March 2015, Vol. 53, No. 3

IEEE Communications MAGAZINE

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Toward the "Space 2.0" Era

- Network Testing
- The Future of Wi-Fi
- Radio Communications
- Mobile Cloud Computing
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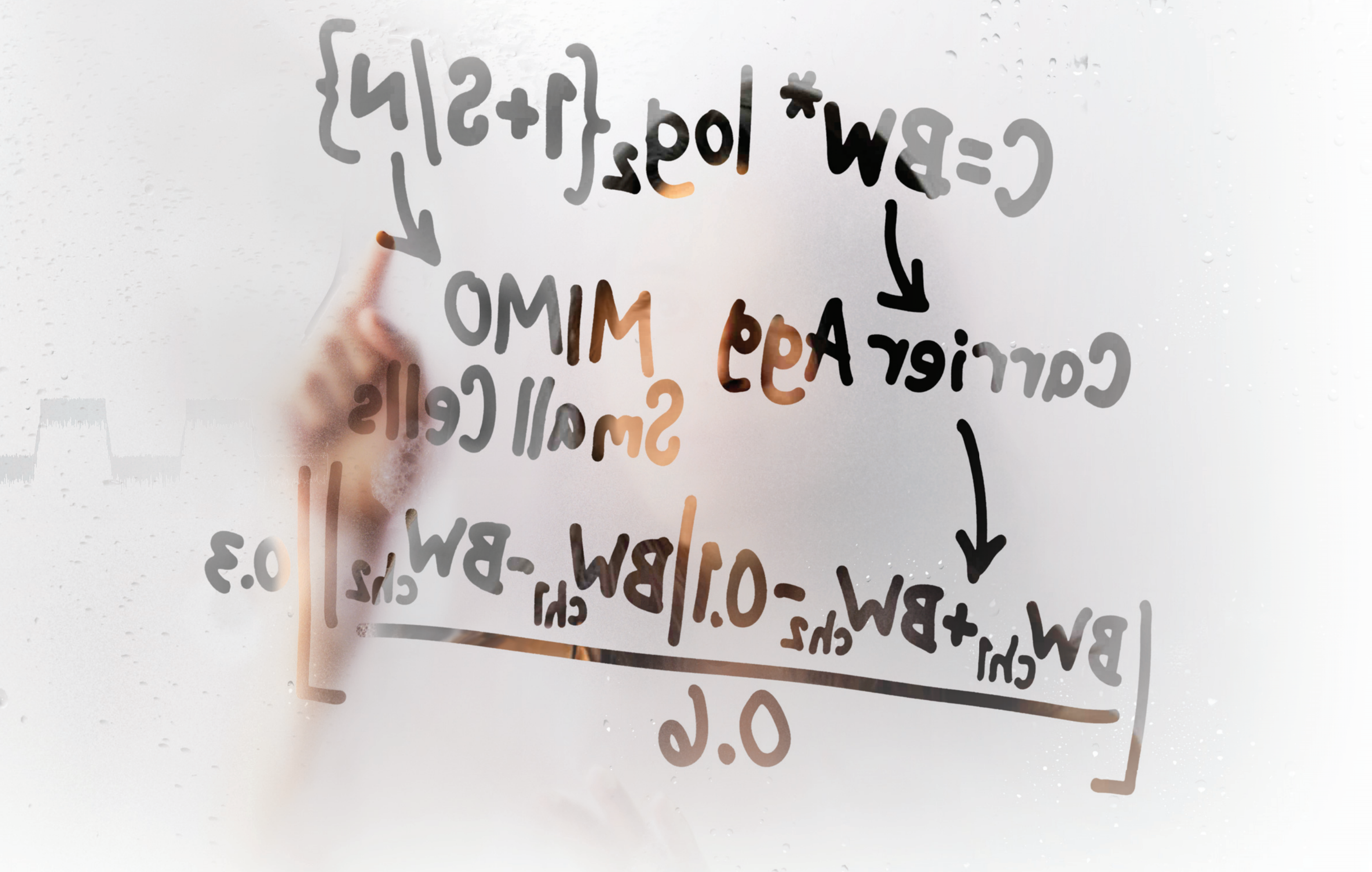

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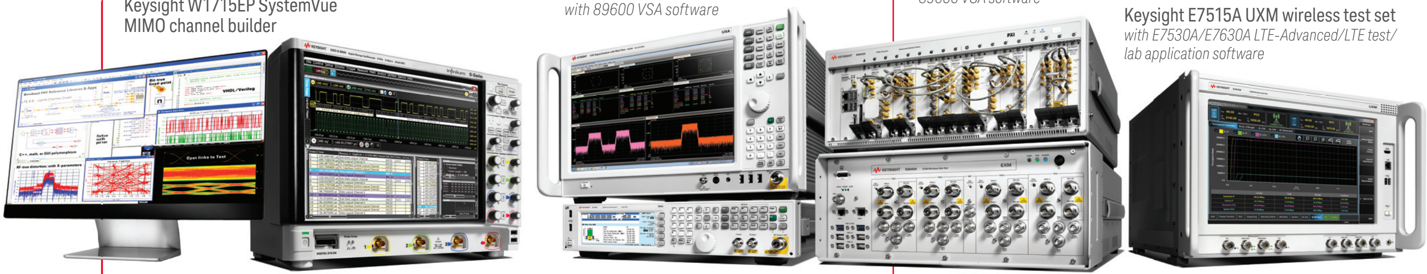

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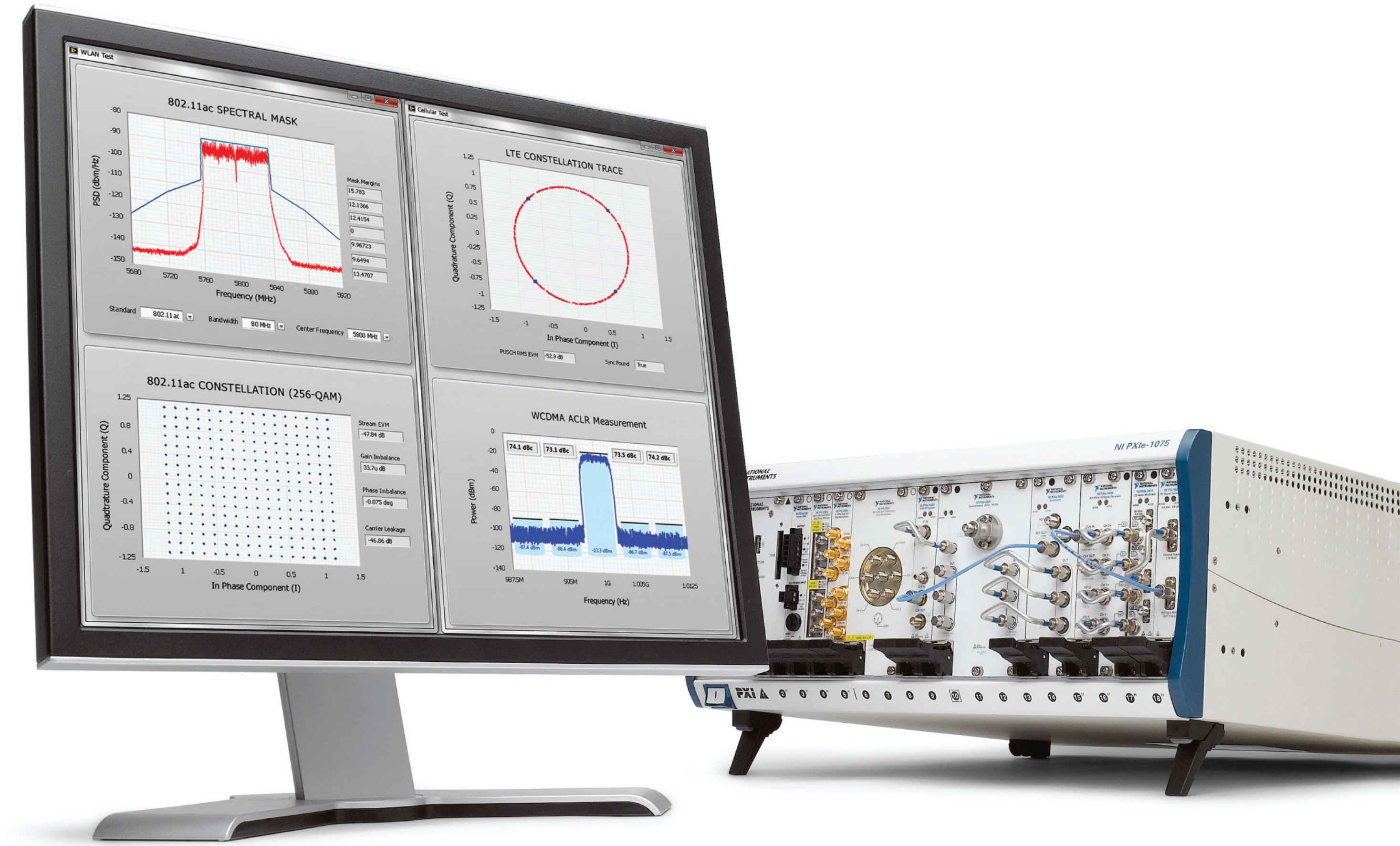

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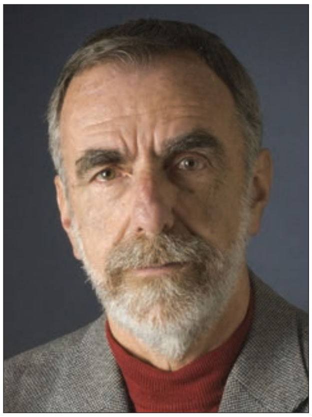
REGAINING INDUSTRY TO COMSOC

Once upon a time, all major telecom industries maintained internal laboratories devoted to medium to long term research, and their engineers were encouraged to publish their results in conferences and journals (think of the impact of an internal industry publication like the *Bell System Technical Journal*). Industries even reimbursed IEEE and Communications Society membership to their employees.

Those golden years have been over for quite some time now, and through the storm of changes in the economy, society, and the profession in the first decade of the new century, ComSoc has witnessed the decline of industry participation in journals (both in terms of submissions and editorial membership), conferences (although in this case less than one would think, approximately three percent less from 2011 to 2014), and membership (simply joining ComSoc as either an active or non-active member).



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For many years, ComSoc's leadership has continued to view this as a serious and growing topic of interest and worry, and indeed, previous Presidents have tried hard to reverse the trend, with mixed results.

There are many good, objective reasons for the decline of industry participation in ComSoc. Quoting from my January 2014 President's Page: "the progressive shift of our core publications toward the academic way of obtaining and presenting results; the strong intellectual property protectionism, favoring the tendency of industry, and in particular of start-ups where innovation begins, to delay publication of results or refrain from publishing at all; and companies focusing their employees on near-term, revenue-producing activities rather than attending conferences or publishing papers that may have longer-term, less quantifiable near-term benefits."

I would add to those the increasing divergence between the scientific and technical roots of ComSoc in the physical layer, and the recent trends of considering telecommunications transmission and switching technologies as commodities, with the shift moving toward upper layers and applications in many different fields, which traditionally our community finds difficult to encompass and serve.

To be faithful to its Constitution, which lists among the society's purposes the "professional, directed toward promotion of high professional standards…," ComSoc needs to persevere in its attempt to regain industry attention and participation. In the last 50 years, our community has been at the core of the most amazing, peaceful revolution in the history of mankind, and the industry has been crucial in this endeavor, reducing the time to market of the innovation generated in research and deploying communication networks and providing communications services to serve people throughout the world.

With the realistic belief that past times will not return, we must continue to work toward finding and providing to industry engineers and telecom professionals valid reasons to become active members of ComSoc.

First, we must partially readdress our conferences and publications to make them more attractive to read and to publish for industry members. Various attempts have already been made. For conferences, our flagship conferences, ICC and Globecom, have offered conference sessions specifically designed for practicing communications engineers, dialogue sessions with industry leaders, executive forums, panels, and interactive sessions, and demonstrations of industrial products. In terms of publications, our magazines are devoting particular attention to inviting and hosting papers from industry, and in December 2014 *IEEE Communications Magazine* published the first *Communications Standard Supplement*, a platform for presenting and discussing standards-related topics in the areas of communications, networking, and related disciplines. If successful, the supplement will transition into a full-fledged new magazine. ComSoc VPs of Conferences and Publications are actively working to increase the industry participation in Technical Committees of conferences and Editorial Boards of journals, so that industry representatives can be directly involved in the process of making both conferences and journals more appealing to industry engineers and telecom practitioners.

Second, we must try to reach those industry employees and professionals who are not directly involved in designing telecom equipment but who deal with the widespread applications of ICT.

Third, we need to widen the spectrum of our activities, in particular by forming new technical communities, to encompass topics that have traditionally been out of ComSoc's DNA. ComSoc has the proper instrument to do that, i.e. the Emerging Technologies Committee, which needs to look at the wider horizon of ICT, perhaps starting collaboration with other IEEE societies.

Fourth, we need to continue increasing our influence and activities in the domain of telecom-related standards. After a long sleeping period, much has been done in recent years to reach this goal, highlighted by the creation of the position of VP-Standards, to joing the four existing VPs. ComSoc is now maintaining a strong and good relation with the IEEE Standards Association, and has been proactively suggesting new fields where IEEE can reinforce its position as a neutral standard facilitator.

Last but not least, ComSoc needs to find ways of involving technological start-ups in its activities, since they are the natural liaison between academic research, where they often come from, and industry. To do so, we need to offer services tailored to their needs and matching their great attention not to disclose their IPR through untimely publications or conference presentations.

This month's President's Page will discuss in detail how ComSoc can improve the appeal of its conferences to industry through the operation of the Industry Content and Exhibition Committee (ICEC). The Committee is chaired by Heinrich (Heiner) Stüttgen, who will describe the structure and scope of the Committee.

Heiner was a Fulbright scholar at the State University of New York at Buffalo (NY), from where he holds a Master of Science degree (1979). He obtained a Doctor of Science degree (1985) in computer science from the University of Dortmund.

In 1985 he joined the IBM Research and Development Laboratory in Germany, developing one of the first mainframe UNIX systems. In 1987 he moved to IBM's European Networking Center in Heidelberg, where he researched protocols for high-speed networks and multimedia communications.

In July 1997 Heinrich joined NEC Europe Ltd. as founding manager of NEC's Network Laboratories in Heidelberg. Since June 2007 he has been Vice President of NEC Laboratories Europe, responsible for NEC's ICT related R&D activities in Europe, now focusing on networking, security, and smart city related technologies, including the Internet of Things, data analytics, and real world optimization problems. As head of an industrial research organization, he has a keen interest in transferring research output into marketable solutions.

Heinrich is an IEEE Fellow. He chaired ComSoc's Switching and Routing Technical Committee (2004–2005); the Globecom and ICC Technical Committee (GITC) (2006–2009); served as Director of Conference Development on the Board of Governors of the IEEE Communications Society (2010–2013); and he has been chairing ComSoc's Industry Content and Exhibition Committee (ICEC) since 2014. From 1998 to 2009 he served as Vice-Chair of the German Information Technology Society's special interest group on communications and distributed systems.

THE INDUSTRY CONTENT AND EXHIBITION COMMITTEE

While ComSoc has been growing and evolving its portfolio of research conferences over the last decades, we realize that the number of industrial and other non-academic participants in our conferences has been on the decline for many years. There are many reasons for this trend. On one hand the number of corporate research functions has been declining, so the number of participants from these organizations has been declining as well. On the other hand, while the practicing engineers in industry seem interested in ComSoc's magazines, it appears that the content offered in the typical research conference is of lesser importance to them. So the question to answer is: How can ComSoc serve this important constituency better?

There could be many approaches to address this challenge. We could try to set aside a section with content primarily geared toward non-academic members at each conference. We could also try to create new conferences focusing specifically at the practicing engineer in the field. Over many years ComSoc has tried to organize exhibitions at some of its conferences, namely IEEE GLOBECOMs, to attract industry participation with highly varying degrees of success. Further, the question needs to be asked whether our topics, our publication formats (e.g. full papers in IEEE Xplore), or review processes are really suitable to address the needs of our non-academic membership.

While these and other questions have been discussed again and again over the years, there has never been a systematic approach to capture what worked well and what did not in this regard. Therefore, ComSoc's President established a new standing committee in 2013 called the Industry Content and Exhibition Committee (ICEC). The ICEC reports directly to ComSoc's President and cooperates with existing ComSoc functions such as the conferences area, the standards area, the technical activities area, as well as the membership development area.

The ICEC is responsible for developing and promoting a strategic vision and a management approach for conference programs that would attract attendees from industry, administration, or other non-academic sectors. This includes processes to assure the quality and value of content in industry oriented conference programs. It is the overall objective and mission of the ICEC to increase industry participation in ComSoc events. This includes, but is not limited to, the individual targets detailed below.

1) Identification of themes, technologies, implementations, and business issues of high relevance to the communications industry at large. In contrast to the more long term academic content supplied by current ComSoc conferences, such issues will generally address near and mid-term issues, relevant for product developers, system integrators, consultants, and business planners.

2) While ComSoc has a strong standing in the academic community, its visibility in the industrial area is much lower. It may therefore be advisable to partner with other societies or governmental organizations when addressing a new technical or business area. The ICEC will identify and if appropriate approach and negotiate with external organizations such cooperative programs and events to reach a critical mass of industry participants quickly.

3) Definition and setup of content delivery processes reflecting the different nature and needs of industry related content, such as publication formats, copyright schemes, review processes, etc, such that industrial engineers are encouraged rather than discouraged to participate in ComSoc events. In addition, such processes shall be clearly documented to allow existing (academic) conferences to integrate an industrial section into their program (e.g. the Globecom IF&E section), while being consistent in terms of processes from time to time and without the need to reinvent the wheel for every event.

4) A long standing question is the issue of exhibitions as stand-alone events or co-located with other major ComSoc events. For instance, Globecom has tried to set up a strong exhibition for more than 10 years with varying but generally limited success. It is one of the objectives of the ICEC to explore the route to successful technology exhibitions, either stand-alone or co-located with existing or new events. If and only if successful exhibitions can be organized by ComSoc, the relevant processes, contacts, and contractual schemes need to be recorded and documented.

Each member of the ICEC covers a certain technical area of ComSoc's technical scope, such as communications circuits, components and devices, physical layer communications technologies, namely wireless, optical, or other transmission technologies, telecommunication networks architecture, control, management, planning, and evolution, communications software and (end-user) services as, well as communications markets and business aspects, including societal impacts and big societal challenges. All of these voting members should be and are currently from industry.

However, in order to extend the reach and impact of the ICEC's activities, the ICEC is forming Special Interest Groups (SIGs) that will each address a specific (industry relevant) technology or business domain, identifying relevant technical topics, high caliber speakers, and potential partner organizations, bridging between the ICEC, specific conference committees, and the relevant experts in industry and ComSoc.

A SIG is an informal group of engineers/researchers with a good view of hot topics relevant to industry members. SIGs are strongly recommended to involve, discuss with, and hence take advantage of ComSoc's existing Technical Committee (TC) structure and their members. TCs by definition have a good view of their respective fields, and although they are often driven by academic members, they can contribute a lot to ICEC's objectives and targets. However, it is important to stress that SIGs are open groups and we very much hope that engineers, so far not so involved with ComSoc, will take the opportunity to involve themselves more and contribute through this new mechanism and organization. You can find more information about the ICEC, its charter and members, as well as the SIG structure, on the ComSoc web page under

http://www.comsoc.org/about/committees/standing

As the current chair of the ICEC, I want to encourage all readers, especially our members from industry, business, or government, to grasp the opportunity this new committee and structure offers them to contribute and to shape ComSoc to server their needs better. If you are interested in contributing, please contact the ICEC members and SIG chairs relevant to your interest. If you have ideas about how ComSoc can serve you better, please contact them. If you are willing to contribute your time, your ideas, and your energy to serve your technical community, this is the perfect time and mechanism to do so. As the ICEC and the SIGs are new organizations with many aspects of their work still under definition and under discussion, your contribution can make the difference. Don't miss this opportunity!

OMBUDSMAN

COMSOC BYLAWS ARTICLE 3.8.10

The Ombudsman shall be the first point of contact for reporting a
dispute or complaint related to Society activities and/or volunteers.
The Ombudsman will investigate, provide direction to the appropriate IEEE resources if necessary,
and/or otherwise help settle these disputes at an appropriate level within the Society…

IEEE Communications Society Ombudsman
c/o Executive Director
3 Park Avenue
17 Floor
New York, NY 10017, USA

ombudsman@comsoc.org
www@comsoc.org "About Us" (bottom of page)

IEEE GLOBECOM 2015 TO HOLD 58TH ANNUAL CONFERENCE DECEMBER 6–10 IN SAN DIEGO, CALIFORNIA

"CALL FOR PAPERS" ENDS APRIL 1 FOR ORIGINAL SUBMISSIONS DEDICATED TO INNOVATIONS AND BREAKTHROUGHS IN NEXT GENERATION COMMUNICATIONS TECHNOLOGIES

IEEE GLOBECOM 2015 (http://globecom2015.ieee-globecom.org/), the premier international event dedicated to driving innovations and technological breakthroughs in nearly every aspect of communications, will hold its 58th annual event from December 6 – 10 at the Hilton San Diego Bayfront Hotel in San Diego, California. The "Call for Papers" will end April 1 for original submissions dedicated to next wave advancements in areas ranging from e-Health, Internet of Things (IoT) and game theory to power-line, satellite, smart grid space, green and 5G cellular networking communications.

"Each year, IEEE GLOBECOM organizers set out to create an experience that will drive global achievements," says Ed Tiedemann, IEEE GLOBECOM 2015 General Chair and Senior Vice President of Engineering at Qualcomm Technologies, Inc. "IEEE GLOBECOM 2014 was a phenomenal success with nearly 3,000 scientists, industry professionals and academics attending more than 2,500 presentations. We are committed to surpassing the excellence achieved by last year's schedule with an industrial and technical program highlighting the many significant achievements, research projects and deployments destined to reshape our lives and the way we do business on a global scale."

Themed Connecting All Through Communications, IEEE GLOBECOM 2015 will start on Sunday, December 6 with the first of two full days of workshops and tutorials exploring topics such as 5G cellular, next generation WiFi, green HetNets, network radio resource and interference management, real-time wireless system prototyping, CloudRAN, M2M/IoT, cloud computing, NFV/SDN, optical communications, and satellite communications.

On the following day, Monday, December 7, the conference will initiate its three-day schedule of learning sessions consisting of keynotes, executive forums and industrial panel discussions highlighting the entire spectrum of broadband, wireless, multimedia, data, image and voice communications. This includes a technical symposia program offering oral and poster presentations with 1,000 scientific papers, grouped into 12 thematic symposia, and more than 15 parallel sessions. IEEE GLOBECOM 2015 will also feature an industry forum and exhibition (IF&E) program and a large exposition showcasing the cutting-edge technologies and interactive demonstrations of leading communications corporations such as Nokia, Cisco, Samsung, AT&T, Qualcomm, Keysight, Huawei and Intel. A wild ideas/lightning talks session is planned for Wednesday evening.

Technology professionals seeking more information on IEEE GLOBECOM 2015 speaking opportunities are urged to visit both the Technical Program (http://globecom2015.ieee-globecom.org/content/technical-program) and the Industrial Program (http://globecom2015.ieee-globecom.org/content/industry-program-submissions) then submit their original proposals by the noted due dates. In the past, specific sessions explored topics such as smart metering, cyber security challenges, next generation 4G/5G cellular networking, Internet governance and programmable cloud networking.

This year, a special emphasis will be placed on access networks and systems, big data networking, cloud computing, data storage, e-health, green communications and computing, IoT, P2P networking, powerline communications, SDN and NFV, satellite and space communications, social networking, smart grid communications, satellite and space communications, social networking and molecular, biological and multi-scale communications. Additional areas of interest also include:

- Ad Hoc and Sensor Networking
- Cognitive Wireless Networks
- Communication and Information System Security
- Communications QoS, Reliability and Modeling
- Communication Software, Services and Multimedia Applications
- Communication Theory
- Next Generation Networking
- Optical Networks and Systems
- Signal Processing for Communications
- Wireless Communications and Networking

In addition, submissions detailing the latest technical and business issues in communications and networking are also welcomed for consideration within the event's industrial program. Organized into six parallel program tracks under the topics of Wireless Access, Wired Access, Networking and Information, Enabling Technologies, Emerging Applications and Business and Government potential topics include, but are not limited to innovations in the field of:

- 5G cellular, LTE/LTE-advanced, WiFi, 802.11ad/NG60, machine-to-machine and vehicular networking communications.
- Optical networking, advances in Ethernet, powerline communications, fiber technologies and cable DSL.
- Big data analytics, Internet technologies, cloud computing, cyber security, software defined networking and network function virtualization.
- CPU/GPU/FPGA/DSP technologies, circuits and systems, RF/Microwave/mmWave devices, design and implementation and test and measurement.
- Internet-of-things, smart grid, social networks, healthcare, public safety and home networking.
- Startups and venture capital, mobile software and applications, spectrum policy and engineering and career management.

Further details on submissions for the IEEE GLOBECOM 2015 industrial program are available at http://globecom2015.ieee-globecom.org/content/industry-program-submissions. All proposals should also be performed through the Easy Chair link located at https://easychair.org/conferences/?conf= gc2015ind.

The annual dinner banquet will be at the San Diego Air and Space Museum, a showplace for San Diego's many firsts in the field of aerospace. Tours are being planned to national attractions like SeaWorld, the San Diego Zoo and the USS Midway Aircraft Carrier.

For more information on IEEE GLOBECOM 2015 speaking opportunities, registration information and conference updates, please feel free to visit http://globecom2015.ieee-globecom.org/. The website's Facebook, LinkedIn and Twitter links are also available for sharing thoughts and comments with peers based worldwide.

IEEE CONFERENCE ON STANDARDS FOR COMMUNICATIONS AND NETWORKING TO HOLD INAUGURAL EVENT 28 – 30 OCTOBER 2015 IN TOKYO, JAPAN

"CALL FOR PAPERS" ENDS 15 MAY 2015 FOR ORIGINAL SUBMISSIONS DEDICATED TO THE LATEST STANDARDS-RELATED DEVELOPMENTS AND RESEARCH IN COMMUNICATIONS AND NETWORKING

The IEEE Conference on Standards for Communications and Networking (CSCN) will hold its first annual event dedicated to the advancement and globalization of the latest industry standards 28 – 30 October 2015 in Tokyo, Japan. The deadline for original paper submissions is 15 May 2015 for high-quality technical and visionary presentations exploring the development, research and implementation of the next wave of standards in communications, networking and all related disciplines.

"Standards are key to the success of the communications industry as well as global systems inter-operability," says Dr. Tarik Taleb, IEEE CSCN General Chair and Professor at Aalto University in Finland. "IEEE CSCN was designed to close the gap, create greater symmetry and forge relationships between scientists and standards experts from academia, industry and differing standardization bodies. The goal is to drive a deeper awareness for the work performed by researchers worldwide, while providing a cooperative environment for sharing ideas and fostering bonds among all key players. It will also offer a meaningful venue for disseminating "pre-standards" technological developments that are likely to drive future and emerging standardization."

Immediately preceding the 94th IETF meeting that will take place in the nearby Yokohama, Japan, IEEE CSCN 2015 will include a three-day, comprehensive agenda focused on the delivery of visionary papers, panels, keynotes and special industry sessions highlighting the broad critical issues impacting telecommunications standards directions. Emphasized will also be the need to foster closer collaborations across various SDOs resulting in the development of more compatible standards and creation of more efficient and effective global communication systems. The specific topic addressed at IEEE CSCN 2015 will include, but not be limited to:

- Enhancements to existing systems and communication protocols developed by standards bodies such as ITU-T, IEEE, IETF, 3GPP, ETSI, OMA, Broadband Forum, Metro Ethernet Forum, oneM2M, ONF, among others
- Visionary papers on hot topics, such as Advanced 5G Radio Access and Network Infrastructure, Converged Access Networks, Optical Networks, Twisted Pair and Coaxial Access Networks, Software Defined Networks and Services, Network Functions Virtualization (NFV)
- The relationship between innovation and standardization, technology governance of standards, the history of standardization, tools and services related to any or all aspects of the standardization lifecycle, and compatibility and interoperability, including testing methodologies and certification to standards
- Current works in progress by global standardization organizations

For more information on IEEE CSCN 2015, including specific submission guidelines, registration, patronage details and ongoing conference updates, please visit

http://www.ieee-cscn.org

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MEMBER RELATIONS

Strengthening the Bridge of ComSoc with Industry

Interview with Ashutosh Dutta, Director of Marketing and Industry Relations

By Stefano Bregni, Vice-President for Member Relations, and Ashutosh Dutta, MIR Director

This is the seventh article in the series started in September and published monthly in the *Global Communications Newsletter*, which covers all areas of IEEE ComSoc Member Relations. In this series of articles I introduce the seven Member Relations Directors (namely: Sister and Related Societies; Membership Programs Development; AP, NA, LA, EAME Regions; Marketing and Industry Relations) and the Chairs of the Women in Communications Engineering (WICE) and IEEE Young Professionals (YP, formerly Graduates Of the Last Decade, GOLD) Committees. In each article, one by one they present their activities and plans.

In this issue I interview Ashutosh Dutta, Director of Marketing and Industry Relations.


Stefano Bregni


Ashutosh Dutta

Ashutosh is a senior member of IEEE and ACM, and served in the past as the chair for IEEE Princeton/Central Jersey Section, Industry Relations Chair for Region 1 and MGA, Pre-University Coordinator for IEEE MGA, and the chair for the Ad Hoc Committee for Public Visibility for ComSoc. Ashutosh co-founded the IEEE STEM conference (ISEC) in 2011, and helped establish STEM clubs and implement EPICS (Engineering Projects in Community Service) in the high schools within the Princeton/Central Jersey Section. In his 25-year career Ashutosh has been Lead Member of Technical Staff at AT&T's Chief Security Office; CTO of Wireless at a Cybersecurity company NIKSUN; Senior Scientist in Telcordia Applied Research; Director of the Central Research Facility at Columbia University; and Computer Engineer with TATA Motors. Ashutosh is co-author of the book *Mobility Protocols and Handover Optimization*, published by Wiley and IEEE. Ashutosh obtained his BS in EE from NIT Rourkela, India, his MS in computer science from NJIT, and he earned his M. Phil. and Ph.D. in electrical engineering from Columbia University, New York, under the supervision of Prof. Henning Schulzrinne.

I am glad to interview Ashutosh and to present the organization and activities of the Marketing and Industry Relations Board (MIRB).

Stefano: Hello Ashutosh. In your opinion, what are the main issues relating to how ComSoc addresses industry needs and areas of interest?

Ashutosh: Industry is one of the important vertices of the ComSoc Member Relations Golden Pentagon. However, as a matter of fact, lack of interest and active participation from Industry has contributed to the sharp decline in industry's share of IEEE Communication Society membership.

IEEE has the following charter for industry engagement:
- Promote and inform the concept of IEEE as a progressive technical information provider to industry and its employees.
- Promote and inform the relevance of support of IEEE in developing/changing technologies.
- Work with industry to establish an understanding of their needs, and to demonstrate how IEEE can help address their needs.
- Encourage the establishment of leadership training within the regions and sections.

However, it is very important to evaluate if IEEE is delivering on Industry's requirements.

Stefano: Might you be more specific?

Ashutosh: A SWOT analysis of IEEE on industry relevance brings forth some of the weaknesses and areas of opportunities that IEEE ComSoc needs to keep in mind while engaging with the Industry. Some points associated with the weaknesses include insufficient industry-relevant publications; lack of industrial content for conferences; and too much academic dominance.

Some threats include: industry people are too busy; industry requires immediate short-term returns; and industry people have little time for conferences. Some strengths include: IEEE's global outreach; its accessibility and affordability to engineers, including the digital library; opportunities to cooperate with standards bodies; a forum and global initiator of discussion and debate on relevant issues; and a professional community to excite, motivate, and energize action.

Some opportunities that could be explored include creation of professional and vocational qualification and recognition; creating opportunities for individuals and companies to gain international recognition; provide and facilitate industry experience for students; IEEE sponsored industry lectures; creating IEEE mentoring for professional development; reaching out to industry leaders in explaining the values of IEEE; creating global job opportunities; recruiting for IEEE actively at industry sites; and demonstrating that IEEE is special and desirable to join.

Stefano: What is ComSoc doing to address such challenges, in particular, the Marketing and Industry Relations Board?

Ashutosh: Within the past few years, both IEEE and the Communication Society have experimented with various ways to re-engage with industry. To re-engage the Communication Society with industry sectors, we have come up with a strategic framework based on the principles that embrace industry's interests and objectives while integrating IEEE and ComSoc's objectives.

With the above strategic framework in place, we have now taken steps to mobilize the team and implement the steps from the framework. We have developed an Industry-relations volunteer support portal (www.ieee-industry.org) where some of the

(Continued on Newsletter page 4)

DISTINGUISHED LECTURER TOUR

Distinguished Lecturer Tour of Rath Vannithamby in Indonesia

By Satriyo Dharmanto, Indonesia Chapter Chair, and Lingga Wardhana CEO of PT Floatway Systems, Indonesia

Indonesia has an estimated population of over 253 million people. Currently there are more than 850 colleges majoring in Information and Communication Technology (ICT), with approximately 500,000 active students and 40,000 engineering graduates per year. This has made IEEE well established in Indonesia for 26 years. This country is also actively participating in international organizations such as ASEAN, OIC, and APEC, and is an active member of ITU. As the largest economy in Southeast Asia, Indonesia also belongs to other economic groups such as the G-20 major economies (G-20) and Developing 8 Countries (D-8), as well as the G-11, G-15, and G-77 groups.

Indonesia is the world's fourth most populous country, with the world's fourth highest percentage of mobile subscribers and the world's third highest growth rate of Internet users with CAGR 58% (source: Internet World Stats). It also has the world's fourth highest number of Facebook users and the world's fifth highest number of twitter users.

The above conditions create unique and heavy demands on the development of an integrated national ICT. Indonesia Digital Network (IDN) 2020 is the main target of the government to develop an ICT network in Indonesia, with three main programs: id-Access, id-Ring, and id-Con.

id-Access has the main programs to achieve 200 Mobile Broadband City, ubiquitous access for 50 big cities and broadband satellite access. id-Ring has the main programs to achieve the Indonesia intelligent transport network, Indonesia's digital hub, and 100% broadband transport to all municipalities. Furthermore, id-Con has the main programs to achieve nationwide ecosystem-based apps, nationwide cloud computing for government, businesses, and individuals, nationwide cyber security systems, and the target to develop a data center facility with total 100,000m2 in 2015 and 500,000m2 by 2020.

The IEEE Indonesia section currently has about 900 active members, with activities in the five different chapters and four different joint chapters, seven IEEE student branches, and two affinity groups, Women in Engineering and SIGHT in Telemedicine.

The Distinguished Lecturer Tour (DLT) Program has been a favorite program hosted yearly by the IEEE ComSoc Indonesia section. The lecture in 2014 was on "5G Evolution and Candidate Technologies & M2M Communications for Internet of Things (IoT)", by Dr. Rath Vannithamby.

Dr. Rath Vannithamby made a great presentation in front of the participants of two different universities, Mercu Buana University, Jakarta, and Telkom University, Bandung. He was also a keynote speaker at the IEEE Asia Pacific Conference on Wireless and Mobile (APWIMOB 2014), held at the Grand Inna Hotel, Kuta Bali.

The lecture at Mercu Buana University was held on Tuesday, 26 August, 2014, and was introduced by Dr. Mudrik Alaydrus, the Dean of the Master Program of Telecommunication Management. The lecture was attended by a total of 34 participants, 15 participants from academia and 19 from industry. Five of the attendees were IEEE members, and 29 were non-members. Most of the attendees were master students from industries that have business activities in the telecommunication area.

Lecture of Dr. Rath Vannithamby in Mercu Buana University, Jakarta.

Prior to Dr. Rath Vannithamby's DLT session, Mr. Lingga Wardhana, CEO of PT Floatway Systems, who was the coordinator of this DLT program, spoke about 4G readiness in Indonesia and promoted their collaborative book writting program in telecommunication technology and business.

At Telkom University, the lecture was held on Wednesday, 27 August, 2014, and was attended by a total of 78 participants, all of whom were from academia. Five of the attendees were IEEE members, and 73 were non-members. Most of the participants were students and lecturers at Telkom University; three people were from Sumatera Utara University, and two were from Lampung University. Most of the non-members expressed interest in joining IEEE. The seminar was introduced by Dr. Rina Pudji Astuti, the Dean of the School of Electrical Engineering, and was also attended by the Vice Dean II, Dr. Basuki Rahmat.

In Bali, on 28 August 2014, Dr. Vannithamby delivered a keynote speech in front of the IEEE APWIMOB 2014 conference, on the topic of 5G Evolution and Candidate Technologies and M2M Communications for IoT. Prior to Dr. Vannithamby's talk, Mr. Rizkan Chandra, CTO of Telkom Indonesia, spoke about Mobile & Wireless System Technology, Business & Industry Highlights in Digital Business Era, moderated by Mr. Satriyo Dharmanto, IEEE COMSOC Indonesia Chapter chair.

The IEEE APWIMOB 2014 conference was attended by 72 people from many countries, of whom 59 were from academia and 13 were from global companies such as Telkom Indonesia, AT&T, Huawei, Samsung, and Intel. The attendees included 31 IEEE members and 41 non-members. The conference was also attended by Mr. Arief Hamdani Gunawan, advisory board and past IEEE Indonesia chapter chair, Mr. Muhammad Ary Murti, and and IEEE Indonesia chapter executive officers.

On the second day in Bali, Dr. Vannithamby discussed 5G Evolution and Candidate Technologies. Of the 18 attendees, 15 were from academia and three were from industry; eight were IEEE members and 10 were non-members.

During the four-day DLT and keynote session, Dr. Vannithamby provided insight on the key development of telecommunications and described the key technologies for the realization of future 5G of telecommunication technology. This lecture was very helpful for researchers, professionals, and students who attended this seminar. Most of the participants said the topic improved their knowledge of telecommunication evolution. The other topic, M2M Communications for IoT, covered challenges, existing M2M technologies, and 5G research directions on M2M/IoT. The Intel collaborative university research program also became a topic of interest for almost all of the participants, so they are excitedly waiting for the next DLT program.

CONFERENCE REPORT

International Conference on Localization and GNSS (ICL-GNSS 2014), Helsinki, June 2014

By J. Nurmi, General Co-Chair, Finland

Walhalla calling…

The best presentation in progress, given by Nyein Aye Maung Maung.

The fourth edition of the International Conference on Localization and GNSS was organized in Helsinki, Finland on June 24-26, 2014, jointly by Tampere University of Technology (TUT) and the Finnish Geodetic Institute (FGI). The conference was chaired by Prof. Jari Nurmi from TUT and Dr. Laura Ruotsalainen from FGI, and the program committee co-chairs were Prof. Elena-Simona Lohan from TUT and Jose Salcedo from Universidad Autonoma de Barcelona, Spain. The conference was technically co-sponsored by IEEE ComSoc, and financially supported by the Federation of Finnish Learned Societies, local positioning industries, and the City of Helsinki. The event brought together 45 positioning professionals for nearly three days to the conference center Wanha Satama (old harbor), which is a brick-built former harbor warehouse with arched windows.

There were four high-caliber keynote speakers. Dr. Lauri Wirola from HERE, Finland talked about mobile positioning and privacy from the service provider point of view; Dr. Marco Lisi from ESA, The Netherlands, shed light on the GALILEO program status and applications of its services; Dr. Valerie Renaudin from IFSTTAR, France, concentrated on pedestrian navigation solutions; and Dr. Vidal Ashkenazi, Nottingham Scientific, UK, looked at the present challenges, potential solutions and future opportunities of Global Navigation Satellite Systems (GNSS).

The session topics were Applications and Services, WLAN and Cellular-based Positioning, GNSS Receivers, Advanced GNSS Processing, Signals of Opportunity, and MULTI-POS Special Session. There were two awards given at the event. The best paper award was given to Enik Shytermeja, ENAC, France who was one of the speakers in the MULTI-POS (Multi-technology Positioning Professionals Marie Curie Innovative Training Network project) special session. The best presentation award went to Nyein Aye Maung Maung, Ritsumeikan University, Japan. There were a total 24 presentations, with an acceptance ratio of exactly 50%.

The social events were also of high quality. On the first evening there was a visit and reception at the Finnish Geodesy Institute, with the house band playing. The banquet took place on the second evening, at the restaurant Walhalla on the old fortress island of Suomenlinna (to which there was the compulsory boat trip as well). The conference was concluded with a sightseeing tour of Helsinki City and barbeque dinner on the courtyard of the brewery restaurant Bryggeri.

In 2015 ICL-GNSS will be organized in Gothenburg, Sweden. For more information, see www.icl-gnss.org

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REGIONAL REPORT

The R. Macedonia ComSoc Chapter wins the 2014 Chapter Achievement Award for the EMEA Region

By Zoran Hadzi-Velkov, Chair, R. Macedonia ComSoc Chapter

The Republic of Macedonia ComSoc Chapter (ComSoc Macedonia) was founded in 1997. Since its founding, many highly motivated and competent volunteers have served the Chapter with great dedication to the benefit of its members. In recent years, ComSoc Macedonia has been particularly successful in providing high-level programs and activities to its members, which resulted in the Chapter being honored by the IEEE Communications Society with the 2014 Chapter Achievement Award (CAA) for Region 8 (Europe, Middle-East and Africa).

According to the CAA citation, the Chapter was awarded for "achieving excellence in Chapter activities and for contributions made in furthering the objectives of the Society". Specifically, the Chapter has seen a rapid rise in its membership (e.g. 59 members in 2013, up from 34 members in 2012), which is one of the most important recent achievements of the Chapter. Some of the Chapter activities that have led to the award are the following.

1) The Chapter has registered and set up its own web page in the comsoc.org domain (http://chapters.comsoc.org/macedonia). The web page has been continuously updated with information about chapter activities in both Macedonian and English languages. The English version has made our Chapter activities visible at the international level, whereas the Macedonian version provides better outreach in the local environment. The Chapter also maintains a Facebook profile. Both the web page and Facebook profile have facilitated the interaction among the Chapter members and have increased their participation in Chapter activities. Information (in Macedonian) that promotes membership in the IEEE and IEEE Communications Society (discounts and benefits, especially for students) is also constantly updated, and is an important contributing factor to the growth in membership.

2) In 2012, the Chapter established the student achievement award, which is given annually to a Bachelor or Master student in telecommunications engineering in Macedonia who has achieved notable research results. The selected student should have had his/her research results published in a scientific article during the previous year, or alternatively should have won an award at an international student competition in telecommunications. Preference is given to students who have published articles as first co-authors in IEEE journals or IEEE conference proceedings. The award consists of an honorarium. The Bachelor student Elena

(Continued on Newsletter page 4)

R. Macedonia Chapter/*Continued from page 3*

Boshkovska is the 2013 award recipient for winning third place at the regional student competition "Elektrijada" in the telecommunications field (photo below).

The 2013 student achievement award ceremony on 14 June 2013 at the Institute of Telecommunications, Ss. Cyril and Methodius University in Skopje (from left: Zoran Hadzi-Velkov, the Chapter Chair, Elena Boshkovska, the Award recipient, and Liljana Gavrilovska, the former Chapter Chair).

3) The Chapter co-sponsored the most important (bi-annual) scientific conference in electrical engineering in the Republic of Macedonia, the 11th International conference ETAI 2013, which was held 26–28 September 2013 in Ohrid, Macedonia. This conference is widely advertised among the research community and ICT industry professionals in Macedonia and Southeastern Europe. The conference had a separate telecommunications track, in which about 30 papers in the telecommunications field were presented. As a conference co-sponsor, the logo of the R. Macedonia ComSoc Chapter was clearly visible on the conference web page.

4) In March 2014 ComSoc Macedonia hosted the Distinguished Lecturer of the IEEE Communications Society, Prof. Norman Beaulieu from Canada, in the framework of his Distinguished Lecturer Tour (DLT) in Southern Europe. During Prof. Beaulieu's three-day visit to Skopje, the Chapter organized a workshop entitled "How to Write an IEEE Paper", during which the Chapter members acquired crucial information about the art of writing high-quality research papers from the former Editor-in-Chief of *IEEE Transactions on Communications*.

5) In April 2014 the Chapter hosted the Distinguished Speaker of the IEEE Communications Society, Prof. Vijay Bhargava from the University of British Columbia, Canada, who gave a lecture on green communications. Since energy efficiency has already become one of the key concerns of modern societies, Prof. Bhargava's lecture sparked many questions by the numerous attendees from academia and the ICT industry in Macedonia.

In conclusion, winning the CAA award is a critical milestone for ComSoc Macedonia, upon which the Chapter will build its future activities. The award is motivation for our Chapter's volunteers to work effortlessly for the promotion of telecommunications and communications technologies in our local environment and beyond, as well as furthering the objectives of the IEEE Communications Society as a whole.

Member Relations/*Continued from page 1*

manuals, industry focused messaging, and promotional fliers/brochures can be found.

Stefano: Would you highlight any specific initiative that can be realized in the short term?

Ashutosh: While the Industry Relations and Marketing Board is working on some long term strategic initiatives to engage industry, a working committee has been set up, which defined some focused key short-term deliverables that include the following items.

•Planned Webinars around Industry Day Tools with Chapter Chairs and Section Chairs.

•The India Council Communication Society and Bangalore Section have started taking many initiatives to engage with the industry. They have organized a series of industry related activities in the area of Software Define Networking, Big Data, and Smart Devices. They organized TechTalks at academic institutions by industry experts. There are also efforts underway to include industry professionals as part of the section executive committee. The India council is also increasingly focusing on ComSoc sponsored conferences across India. In order to further strengthen the industry relationship, Bruce Worthman recently visited India, and with the help from Munir Mohammed made contacts with many key industry leaders at Hyderabad and Mumbai in addition to attending the Intelect conference in Mumbai.

•In order to engage industry members with high value technology, the IEEE Communication Society plans to a have high impact one day summit in emerging technology areas (e.g. SDN/NFV, 5G, IoT, and Cybersecurity). Currently, a high impact 5G Summit is being planned at Princeton University on May 26, 2015.

•The IEEE Communication Society plans to hold "Future Skillset Workshops" at a number of universities. This is an opportunity for collaboration among industry, the academic community, and IEEE.

•Industry initiative with the startup community. According to the 2014 member survey results compiled by ComSoc's Director of Marketing, John Pape, 30.8 percent of ComSoc members are/have been engaged in a start-up, and 30.5 percent plan to hire within the next 12 months. Hence, it is very important that the IEEE Communication Society reach out to the start-up community. There are a few proposals that are being considered such as to name a start-up of the month, and publishing industry news on the startup.

•The industry relation committee plans to liaise with ICEC (Industry Content and Exhibition Committee) and collect feedback for implementing industry content appropriately into some of the conferences.

•The IEEE Communication Society is also exporing potential ways to increase membership, for example by instituting a Member-Get-a-Member (MGM) Program.

•ComSoc has introduced new courses in emerging areas such as 5G, Big Data, and Mobility.

•MIRB plans to hold virtual job fairs at many IEEE sponsored conferences. MIRB also plans to sign an MOU with an interested Industry partner to create an award sponsored by a specific industry.

GCN GLOBAL COMMUNICATIONS NEWSLETTER

Stefano Bregni
Editor
Politecnico di Milano — Dept. of Electronics and Information
Piazza Leonardo da Vinci 32, 20133 MILANO MI, Italy
Tel: +39-02-2399.3503 — Fax: +39-02-2399.3413
Email: bregni@elet.polimi.it, s.bregni@ieee.org

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Regional Correspondents Who Contributed to This Issue

Simona Lohan (elena-simona.lohan@tut.fi)
Ewell Tan, Singapore (ewell.tan@ieee.org)

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TOWARD THE "SPACE 2.0" ERA

Claudio Sacchi

Kul Bhasin

Naoto Kadowaki

Fred Vong

The word "satellite" has historically meant global radio coverage, long-distance inter-continental telephony, global TV broadcast, precise localization available in the sky and across the oceans. This was consequential to the early studies about geo-stationary orbits carried out since the visionary paper of A.C. Clarke published in "Wireless World" in 1945 [1].

The fact that Dr. Clarke has been appreciated worldwide as science fiction writer clearly indicates that the research on satellite communications has been characterized, at least during the early decades, by a strongly visionary attitude. With the passage of time, satellites consolidated their role as "long-distance" communications enablers. However, 70 years after Clarke's article, the role of the satellite has apparently been fixed by the title of that visionary document: "extra-terrestrial relay." Indeed, the most remunerative applications of satellite communications have so far been digital TV broadcasting (standard DVB-S), where the satellite is simply a relay node, and radio localization (GPS, GNSS), where satellites work as "radio-beacons." Just coincidentally with the mass-market explosion of modern Information and Communications Technology (ICT) services during the 1990s, the role of satellites has become marginal and ancillary. The change of philosophy in wireless networking design during the 1980s privileged the local "cellular" connectivity rather than the global interconnection. For almost 25 years now, cellular networks have demonstrated their superior capabilities of supporting mobile connectivity in the presence of high user density. Satellite networks, in the old vision of "extra-terrestrial relays," are no longer competitive. Geostationary (GEO) satellites can provide global coverage, but very long propagation path and high delays and latencies severely limit the link efficiency. On the other hand, Low-Earth-Orbit (LEO) satellites are more suitable to support bidirectional networking, but their coverage area is relatively small. Considering this panorama, a question duly arises: "Apart from the well-consolidated DVB-S and GPS/GNSS market niches, is there still room for space communications in the future ICT market?"

Considering the weaknesses of the existing terrestrial networking technologies, the answer is affirmative. The distribution of cellular networks is not everywhere uniform. In a big European country like Italy, Long Term Evolution (LTE) is available over less than 30 percent of the territory. On the other hand, the recent data published by the FTTH Council Europe evidenced a great difficulty of fiber penetration in the EU countries. Only nine of the 21 nations individually analyzed should achieve "FTTH maturity" (20 percent penetration) by 2016 [2]. Another critical weakness of terrestrial networking is the vulnerability with respect to natural disasters and terrorist attacks, clearly evidenced by the dramatic connectivity disruptions encountered during the aftermath of September 11th and the Katrina floods [3].

The potential of space networking is enormous in terms of broadband, low-cost ubiquitous coverage and disaster resilience. From a merely theoretical viewpoint, satellite link capacity is clearly superior to the terrestrial capacity. From a more practical viewpoint, the monetary costs and the environmental impact of satellite connections are greatly lower than those of the corresponding terrestrial connections.

In order to fully exploit such potential, a radical change of the current vision of space networking is required. Nowadays, space networks are regarded as information broadcasters and/or coverage enhancers. In order to promote a relaunch of space technology for ICT, a complete redefinition of the role and paradigms of space communications and networking should be conceived in the perspective of Future Internet. As stated in [4], Future Internet must be globally "anywhere-anytime," must be capable of assisting society in emergencies, and must be trustworthy. Terrestrial cellular communications cannot effectively fulfil these basic requirements due to their intrinsic "local" nature. For this reason, Future Internet can be thought of as a "building" supported by two basic pillars: the "local" pillar made by terrestrial networking, and the "global" pillar made by space networking. This target is very ambitious: in such a futuristic vision, the space segment would become the main actor in the provision of the global ubiquitous connectivity, instead of a merely ancillary infrastructure broadcasting information and/or patching coverage holes. In some sense, we can highlight this expected revolution with the term “Space 2.0,” which marks a clear discontinuity with the “Space 1.0” era, begun in 1945 with A.C. Clarke’s article.

The potential outcomes are very attractive, but the potential risks are also very high. Indeed, space communications and networking will have to rethink themselves in terms of augmented interactivity, dynamic and context-aware reconfigurability, resilience, and disaster immunity. The Feature Topic, “Satellite Communications and Networking: Emerging Techniques and New Applications,” the first part of which is published in the current issue of *IEEE Communications Magazine*, is intended to show the current status of satellite communications and networking technologies with a look at the change of perspective that will characterize the coming decades.

The first article, “Waveform Design Solutions for EHF Broadband Satellite Communications,” by M. De Sanctis, E. Cianca, T. Rossi, C. Sacchi, L. Mucchi and R. Prasad, illustrates basic design criteria and advanced practical solutions for waveform generation in broadband satellite communications operating in the millimeter wave bandwidths, i.e. Extremely High Frequencies (EHF).

The second article, “Cognitive Spectrum Utilization in Ka-band Multibeam Satellite Communications,” by S. Maleki, S. Chatzinotas, B. Evans, K. Liolis, J. Grotz, A. Vanelli-Coralli, and N. Chuberre, deals with the application of advanced cognitive concepts, already in use in terrestrial wireless communications, to multi-beam satellite systems operating in the Ka-band.

The third article, “IP Mobile Multicast over Next Generation Satellite Networks,” by E. Jaff, P. Pillai, and Y.F. Hu, proposes a novel PMIPv6-based solution for global satellite-based IP multi-cast receiver mobility.

The fourth article, “Contact Graph Routing in DTN Space Networks: Overview, Enhancements and Performance,” by I. Bisio, G. Araniti, N. Bezirgiannidis, E. Birrane, S. Burleigh, C. Caini, M. Feldmann, M. Marchese, J. Segui, and K. Suzuki, discusses the application of Contact Graph Routing (CGR) in different Space Delay Tolerant Networks (DTNs).

The fifth article, “Integration of Satellite and LTE for Disaster Recovery,” by M. Klapez, M. Casoni, C. A. Grazia, N. Patriciello, A. Amditis, and E. Sdongos, considers a very significant topic from a societal viewpoint: the efficient and resilient integration of satellite and terrestrial 4G networking to support emergency communications in the aftermath of natural and/or man-made disasters.

The last article, “Software Defined Networking and Virtualization for Broadband Satellite Networks,” by S. Medjiah, L. Bertaux, P. Berthou, S. Abdellatif, A. Hakiri, P. Gelard, F. Planchou, and M. Bruyère, analyzes the potential advantages that satellite networking could take from SDN and network virtualization in four different technically-relevant user cases.

Approaching the 70th anniversary of A. C. Clarke’s article, we think the papers published in our Feature Topic can contribute to innovate concepts and methodologies of space communications and networking in a renewed perspective where “Sky” will become the true “global” infrastructure available to connect all of humanity.

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BIOGRAPHIES

CLAUDIO SACCHI was born in Genoa (Italy) in 1965. He obtained the “Laurea” degree in electronic engineering and the Ph.D. in space science and engineering at the University of Genoa (Italy) in 1992 and 2003, respectively. Since August 2002 Dr. Sacchi has held a permanent position as assistant professor on the Faculty of Engineering at the University of Trento. He is currently affiliated with the Department of Information Engineering and Computer Science at the University of Trento. He is the author and co-author of more than 80 contributions published in international journals, edited books, international conferences, etc. Dr. Sacchi is Senior Member of the IEEE.

KUL BHASIN is senior space communications architect at NASA Glenn Research Center in Cleveland, Ohio, USA. He has led a number of NASA-wide architecture teams who design space communication systems and ground networks. Currently he is leading and designing NASA’s next-generation space-based communications satellite architecture, as well as designing a network service portal system. Prior to that he held the position system engineering and formulation manager for space communications projects at NASA GRC. He is an associate fellow of AIAA, a Senior Member of IEEE, and a Fellow of the Society of International Optical Engineers (SPIE).

NAOTO KADOWAKI received a B.S. in communications engineering, a master degree in information engineering, and Ph. D. from the University of Tohoku, Sendai, Japan in 1982, 1984, and 2010, respectively. From July 2004 to December 2006 he was the managing director of the Strategic Planning Department at NICT from January 2007 to June 2008, and director general of Wireless Network Research Institute at NICT from July 2008 to June 2013. He is currently senior executive director and executive director of the Strategic Planning Department at NICT. He is a member of the IEEE, AIAA, and IEICE of Japan.

FRED VONG is director of engineering at Asia Satellite Telecommunications Co. Ltd. in Hong Kong. He has over 19 years’ experience in the satellite telecommunication industry. He is managing teams of engineers to support satellite procurement, engineering support to the satellite control center, customer satellite network design and qualification, service product development, frequency spectrum coordination and ITU activities related to satellite communications. He holds a bachelor’s degree and a master’s degree in engineering from the University of Hong Kong.

Waveform Design Solutions for EHF Broadband Satellite Communications

Mauro De Sanctis, Ernestina Cianca, Tommaso Rossi, Claudio Sacchi, Lorenzo Mucchi, and Ramjee Prasad

ABSTRACT

The problematic RF environment experienced by broadband satellite communications at EHF frequency bands, in particular Q/W bands, call for the use of novel waveforms. This paper presents a detailed comparison of several waveforms in presence of nonlinear distortions and typical values of phase noise introduced at Q/W band. Two main types of waveforms have been compared: Constant Envelope multicarrier waveforms (CE-OFDM and CE-SCFDMA) and single carrier impulse-based waveforms (TH-UWB, DS-UWB and PSWF-based PSM). This comparison will allow to draw some practical guidelines for the waveforms design of EHF broadband satellite communications.

INTRODUCTION

Satellites for broadband communications in Ka-band with throughput capabilities around 100 Gbit/s have been recently launched. Examples include KA-SAT, Echostar 17, Viasat-1, Astra 2E and Hylas-2. However, they are still not able to cope with the user broadband demands that are predicted to considerably increase in the next years [1]. Consequently, next-generation High Throughput Satellites (HTSs) will push for the increase of the system capacity and optimization of system resources. The visionary target of "terabit satellite capacity" is considered in [1]. Terabit capacity can be approached in Ka-band by means of intensive frequency reuse through multi-beam satellite systems. However, the use of additional spectrum is essential to boost system capacity to the limit of 1 Tbit/s and such bandwidth resources can be found only in the Extremely High Frequency (EHF) domain.

In Table 1 the slices of available EHF spectrum beyond Ka-band frequencies are listed.

The use of higher bands for the feeder links, like Q/V bands or W-band, would allow a maximization of both the terminal and the gateway spectrum, with a consequential increase of system capacity and minimization of the number of gateways and related costs.

In this framework, the ALPHASAT satellite launched on July 25, 2013 together with its "Aldo Paraboni"[1] payload is an important step towards HTSs allowing to carry out, for the first time, a communication experiment over a Q/V band satellite link using the second generation of the DVB-S2 standard [2]. DVB-S2 presents some new and innovative elements with respect to the older version of DVB-S: introduction of Low Density Parity Check (LDPC) codes and the possibility of using Adaptive Coding and Modulation (ACM) techniques. Nevertheless, the communication technology did not experience a radical change: we are still talking about traditional single carrier modulated signals, with Nyquist shaping and time-domain equalization.

As outlined in [3] the challenges posed by multi-gigabit communications at frequency bands beyond Ka-band call for the use of novel waveforms and in some cases rethinking the traditional baseband and RF design of the air interface. The EHF satellite links are characterized by significant power losses due to large path loss, atmospheric attenuations, and rain fading. Therefore, the link budget is usually constrained at these frequencies. For this reason, the available power resources should be efficiently exploited, taking into account the presence of significant link impairments, nonlinear distortions and phase noise. The tradeoffs to be tackled are not trivial. The designed radio interface should be robust against link impairments, provide high spectral efficiency and – considering the very high data rates involved by broadband applications – characterized by low-complexity in the waveform generation and detection process.

Besides conventional single-carrier modulation formats, like raised-cosine-filtered QPSK, QAM, Minimum Shift Keying (MSK) and Gaussian Minimum Shift Keying (GMSK), other state-of-the-art techniques that have recently gained some attention for satellite communications are multicarrier modulations, namely: Orthogonal Frequency Division Multiplexing (OFDM) and Single-Carrier Frequency Division Multiple Access (SC-FDMA). Multicarrier modulations have been originally conceived for terrestrial communications, in particular, for highly frequency selective channels. The motivation for their applicability to satellite broadband links is not straightforward and stems mainly from the need of an effective integration of terrestrial and satellite broadband systems [4]. Nevertheless, for the applicability to a "dirty RF" environment, such as the EHF channel, the modification to Constant Envelope (CE) multicarrier waveforms, which are natively more robust to nonlin-

Mauro De Sanctis, Ernestina Cianca, and Tommaso Rossi is with the University of Rome "Tor Vergata."

Claudio Sacchi is with the University of Trento.

Lorenzo Mucchi is with the Univesity of Florence.

Ramjee Prasad is with the University of Aalborg.

[1] In memory of Prof. Aldo Paraboni (1940-2011), pioneer of EHF exploitation for satellite communications.

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ear distortions, should be considered. Other typologies of waveforms that have been proposed for satellite communication and, in particular, for EHF satellite communications are impulse-based ultra-wideband (UWB) [5, 6].

In this article, we shall focus our attention on practical solutions for waveform design in future EHF multi-gigabit satellite communications. The analysis of the most significant impairments characterizing EHF satellite links, presented in section II, will drive the waveform design. In particular, we shall consider the following transmission techniques:

- Single-carrier impulse-based UWB transmission techniques, namely Time-Hopping UWB (TH-UWB), Direct Sequence UWB (DS-UWB) and binary Pulse-Shaped Modulation (PSM) using Prolate Spheroidal Wave Functions (PSWF) of order 1 [6]
- Constant-envelope (CE) multicarrier waveforms, namely CE-OFDM [7] and CE-SC-FDMA [8]

We will show the performance comparison among the different proposed waveform solutions by discussing some selected simulation results. After that, some practical guidelines for waveform design in EHF satellite links will be proposed. Finally, article conclusion will be drawn.

EHF Satellite Link Impairments

One of the main characteristics of satellite communications is the need to transmit signals at high power so that the signal received on the ground has enough power for a correct reception. Furthermore, the space and the power onboard the satellite is limited and these resources should be used very efficiently. As demonstrated in [1], the EHF satellite transmission is clearly in the power-limited capacity region. Shannon capacity of the order of 600 Gb/s can be theoretically reached in Q/V band with 200 spots, frequency reuse factor equal to 4 and spectral efficiency of 1.2 bit/s/Hz [1]. Such a low spectral efficiency indicates that the available signal-to-noise ratio (SNR) is substantially reduced by atmospheric attenuations, scintillations and rain fading [3]. Very significant power attenuations at frequencies beyond 30 GHz are also yielded by antenna de-pointing losses, as shown in [3]. For instance, at 80 GHz, a pointing error of 0.3° involves a power loss of 18 dB.

Moreover, to increase the amount of available power resources, the most important role is played by the RF power amplifier. Despite recent advances in microwave solid-state power amplifiers, tube amplifiers such as Traveling-Wave Tubes (TWTs) and klystrons still provide the best combination of power output, power efficiency and bandwidth. The biggest issue of high-power tube amplifiers is related to nonlinear behavior of the power gain. The maximum power gain – of the order of 50-60 dB – is reached at the saturation point of the amplitude-to-amplitude (AM/AM) characteristic. Moreover, a noisy time-varying phase drift is produced by the amplitude-to-phase (AM/PM) conversion typical of TWTs and klystrons. Nonlinear amplification automatically involves nonlinear distortion that might significantly alter both the amplitude and the phase of the modulated waveform. Nonlinear amplification is usually accounted by the Output Back-Off (OBO), which is the difference (in dB) between the output saturation power and the output average power of the amplifier.

	Uplink	Downlink
Q/V-band	42.5-43.5 GHz 47.2–50.2 GHz 50.4–51.4 GHz	37.5–42.5 GHz
W-band	81-86 GHz	71–76 GHz

Table 2. EHF frequencies allocation for satellite communication services.

One of the main characteristics of satellite communications is the need to transmit signals at high power so that the signal received on the ground has enough power for a correct reception. Furthermore, the space and the power onboard the satellite is limited and these resources should be used very efficiently.

Phase noise is an unwanted phase modulation of an RF signal and can be viewed as spurious sidebands on a wanted carrier. Higher-frequency oscillators generally have higher phase noise associated with them, but nonlinear amplifiers may also increase the phase noise of a processed signal. Phase noise is able at compromising the efficiency of carrier synchronization in coherent demodulation systems by introducing significant phase jitters. On the other hand, in OFDM systems, phase noise can produce inter-carrier interference with a consequential error-floor [9]. Some works about W-band satellite communications evidenced in clear manner the detrimental impact of phase noise on the detection of single-carrier trellis-coded [10] and turbo-coded [11] modulated signals. Both [10] and [11] concluded that phase noise should be conveniently limited in order to obtain acceptable performance.

The analysis of EHF link impairments shown in this section may suggest to PHY-layer designers the following basic hints:

- Power resources are very scarce and (therefore) precious in EHF bands and cannot be wasted. This implies that modulation formats highly resilient against nonlinear distortion should be considered in order to exploit the power amplifiers at their maximal efficiency.
- The robustness against thermal noise and phase noise is a primary requirement. It has higher priority than mere spectral efficiency (higher order modulations can be considered only if they cope with robustness requirements).

The basic Nyquist-shaped pulse waveforms, like root-raised cosine, do not match up well with such requirements, because they are vulnerable to nonlinear distortions. Unfiltered binary or quaternary modulations using rectangular pulse shaping (BPSK, QPSK), characterized by constant amplitude, are not acceptable for broadband transmission applications due to their high sidelobe power levels, not complying with the standard recommendation about spectrum usage. Frequency modulation schemes like Minimum Shift Keying (MSK) and Gaussian Minimum Shift Keying (GMSK) are resilient against nonlinear distortion and characterized by negligible sidelobe power level but, as shown in [6], are vulnerable to phase noise. We think that the effi-

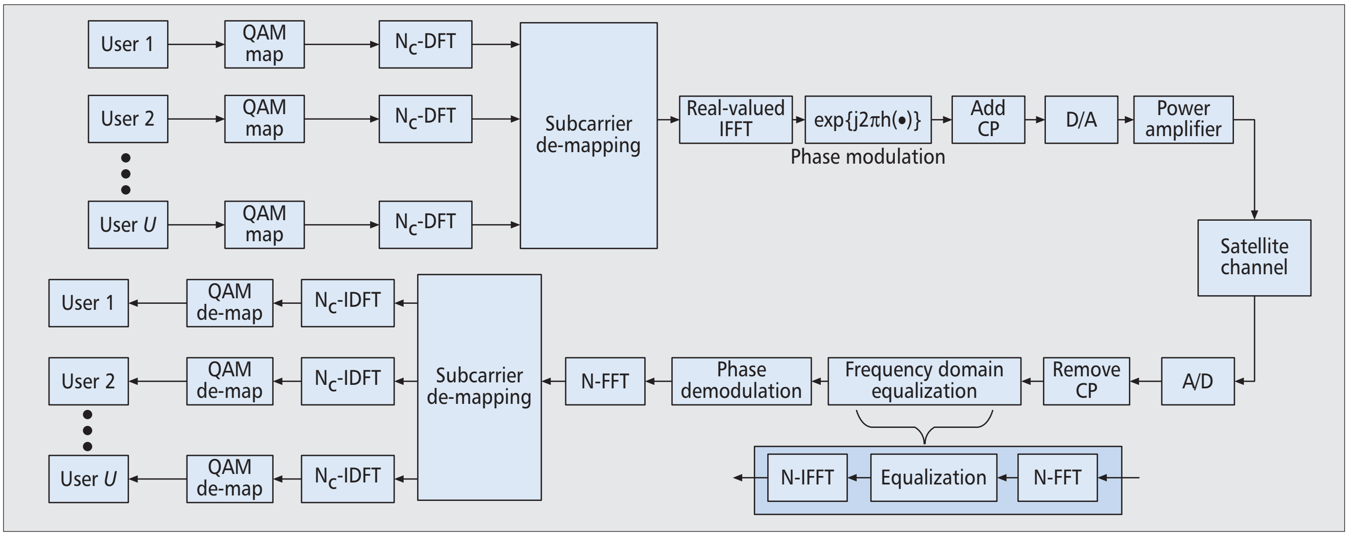

Figure 1. CE-SC-FDMA modulation/demodulation scheme.

cient exploitation of EHF bandwidth portion will require a step-ahead in waveform design with respect to conventional state-of-the-art solutions. This will be dealt in the rest of the article.

Impulse-Based UWB Waveforms

In impulse-based UWB waveforms, the information bits are encoded in various characteristics of the transmitted pulse, such as the pulse presence, position and shape. UWB is mostly applied to short-range wireless applications as a consequence of the limitations to the radiated power required for unlicensed use of the spectrum. Nevertheless, the definition of UWB signals given by the FCC on the 1998, is concerned only with the bandwidth and not with the power emission. The FCC defines UWB as a signal with either a fractional bandwidth of 20% of the center frequency or 500 MHz. Therefore, UWB communications technologies can also be used in licensed systems at the condition that a large bandwidth is available for a single signal and this condition could be fulfilled at Q/V/W bands. Their use in the satellite context is undoubtedly an innovative approach. This type of waveforms could bring some substantial advantages in the direction of relaxing HW requirements and reducing systems costs [12], with the disadvantage of a reduced spectral efficiency.

Several works have demonstrated that IR-UWB at EHF, namely at 60 GHz band, can provide transceiver simplicity, i.e. no need of stable sources, oscillators or mixers [13]. Moreover, as reported in [14] in case of impulse-based communications, it is possible to avoid the use of oscillator and mixers; the idea consists in generating extremely narrow pulses (picoseconds), which have a bandwidth of 100 GHz and then filter them with a bandpass filter centered at 78 GHz. The involved hardware technology is quite complex and far from being space-qualified. Nevertheless, this result encourages in looking for novel RF architectures for pulsed-based communications. Finally, UWB-waveforms have been proposed in all the contexts where power efficiency and robustness to channel non-idealities is preferred to high spectral efficiency (wireless sensor networks, WBAN). In general, a modulation in which the information is encoded in the pulse position such as the Pulse Position Modulation (PPM) is expected to be more robust to nonlinear distortions introduced by the HPA with respect to modulation format where the information is encoded in the polarity of the pulse. This higher robustness could be used to relax the requirements on the HPA.

Different motivations that have led to propose the use of other innovative UWB pulse shaping, such as Prolate Spheroidal Wave Functions (PSWF) waveforms. PSWFs were discovered by Slepian and Pollack in 1961, but their use in practical communication systems has been proposed quite recently. These waveforms, based on the solution of specific differential equations, are characterized by optimal energy concentration in time and frequency domain. In particular, PSWF of order 1 and order 2 can provide RF modulated signals characterized by a near-optimal compromise between spectral compactness (sidelobe power level 31.5 dB below the main lobe [6]) and envelope compactness (their Peak-to-Average-Power-Ratio (PAPR) is 1 dB [6]). Therefore, the effect of nonlinear distortions can be drastically reduced, while maintaining a spectral efficiency rather similar to bandwidth-limited raised cosine waveforms. In Section V, a detailed comparison of PSWFs with other classical IR-UWB waveforms is shown.

Constant-Envelope Multicarrier Modulation Schemes

OFDM and its multi-user extension (OFDMA) found a lot of applications in recent standards for terrestrial wireless networking (IEEE 802.11n, WiMAX, LTE) for three main reasons:

- It offers a lower complexity solutions than current single carrier systems to the problem of performance degradation over frequency selective channels

- It potentially offers good spectral efficiency;
- It efficiently supports variable data rates and provides high flexibility in radio resource management.

On the other hand, the application of this transmission technique has been considered for many years unfit for satellite communications. An OFDM signal is characterized by high amplitude fluctuations that produce large PAPRs. Moreover, high-frequency satellite channels are usually not frequency selective and, hence, the main motivation to use OFDM does no longer hold. Several studies can be found on its use in both mobile and fixed satellite communication systems transmitting in the lower bands of the satellite spectrum (L, S and C bands), where multipath propagation still involves frequency selectivity. Moreover, OFDM has been adopted by the standard for Digital Video Broadcasting Satellite services to Handhelds (DVB-SH) [14], which is the same signal format defined in DVB-H for terrestrial systems. The main reason for adopting OFDM in this context stems from the fact that satellite and terrestrial transmitters form a Single Frequency Network (SFN). Moreover, the use of the same technique for the satellite component would reduce the complexity of the terminal. Less straightforward is the motivation to use OFDM in satellite fixed broadband systems. In this context, some studies have proved that SC-FDMA is less sensitive than OFDM to nonlinear distortions thanks to the reduced PAPR. However, PAPR of SC-FDMA may be still significant, in particular if Localized-FDMA subcarrier allocation is done. Therefore, OBO is still required to make SC-FDMA safely working on satellite links.

A class of FFT-based modulation schemes that are robust to nonlinear distortions and that is worth considering for transmission at Q/V/W bands, is the class of Constant Envelope (CE) multicarrier modulations, namely CE-OFDM [7] and CE-SC-FDMA [8]. The main idea of CE multicarrier systems is to add a nonlinear phase modulation to a real-valued multicarrier signal. The block diagram of the CE-SC-FDMA modulator/demodulator is shown in Fig. 1.

The PAPR of the CE-modulated signal is constrained to 0 dB. The price to be paid is an increase of the signal bandwidth due to the real-valued I-FFT operation [7, 8]. The spectral efficiency S of CE-OFDM and CE-SC-FDMA transmission using a M-QAM constellation ($M = 2^k$, k even) equals to: $S = \log_2(\sqrt{M})/\max(2\pi h, 1)$ b/s/Hz. The tradeoff performance/spectral efficiency is accounted in CE multicarrier modulations by two parameters: M and $2\pi h$, as shown in [7]. Theoretically, fixing M, it is possible to decrease the symbol-error-probability simply by increasing $2\pi h$. Practically, this is not always true because of the threshold effect typical of nonlinear modulations. Anyway, as pointed out in [7] and [8], such tradeoff is very interesting and may deserve a formal analysis.

Both [7] and in [8] confirmed the very good performance yielded by CE-OFDM and CE-SC-FDMA in the presence of nonlinear amplification and frequency-selective multipath fading, which is a typical situation related to low-frequency mobile satellite communications. We think that CE multicarrier modulations may represent an interesting alternative solution for EHF broadband satellite communications, because they would allow benefiting of the favorable features of multicarrier modulations, while avoiding the detrimental effects of nonlinear distortions and power backoffs. A significant open issue is related to the effect of phase noise on such kind of waveforms that have not yet been appreciated. It is demonstrated in [7] that a constant phase offset has no impact on the CE receiver performance. On the other hand, phase noise becomes additive noise after the arctangent detector performing phase demodulation. The arctangent detector is followed by a phase unwrapper aimed at minimizing the effects of ambiguities due to phase jumps crossing the ? boundaries [7]. The introduction of significant amounts of phase noise would increase phase ambiguities and the phase unwrapper might make mistakes. An error-floor would result from such mistakes. In the next section, we shall verify the occurrence of this potential drawback by means of selected simulations.

Parameter	Q band	W band
Center Frequency	40 GHz	85 GHz
Spectral Efficiency	1 bit/s/Hz	1 bit/s/Hz
HPA Output Power Backoff	5 dB	5 dB
Phase Noise at 1 MHz	–140 dBc/Hz	–100 dBc/Hz
Phase Noise at 10 MHz	–160 dBc/Hz	–120 dBc/Hz

Table 2. Main simulation parameters.

Comparison of Different Waveform Solutions

With the final objective to provide guidelines for the design of novel waveforms for satellite transmissions at Q/W bands, and according to the literature review shown in previous Sections on the most related studies, in this section the following waveforms are compared:

- *UWB-based techniques*: TH-UWB, DS-UWB, binary antipodal PSM using order 1 PSWF;
- *CE multicarrier modulations*: CE-OFDM and CE-SC-FDMA.

The main simulation parameters are shown in Table 2.

In order to have a coherent comparison, a fixed spectral efficiency of 1bit/s/Hz has been considered for all the modulation formats. In case of impulse-based UWB, some kind of multiplexing is needed to fully exploit the available bandwidth. For this purpose, two different approaches are considered:

- TH-UWB utilizes low-duty-cycle pulses of duration T_p, which are transmitted with a repetition period T_f. The position of the transmitted pulse within each repetition period is determined by a pseudorandom code c_n which selects one of the N_s slots, each having duration of T_m. The pseudorandom code takes integer value between 0 ~ N_s-1. Assuming that each bit has a length of NTH pulses, the bit duration is $T_b = N_{TH}T_f$.

- DS-UWB utilizes a pseudorandom code c_n that takes value $\{\pm 1\}$ to modulate the amplitude of the DS-UWB pulse train. Each bit has a length of NDS pulses and has duration of $T_b = N_{DS}T_p$.

In the simulations, the transmitted data stream is divided in 8 streams, which are sent in parallel through 8 orthogonal codes of length 8. As far as CE multicarrier modulations are concerned, a 4-QAM constellation has been adopted with angular modulation index of 0.7 rad. According to (4), the resulting spectral efficiency is 1b/s/Hz. The multiplexing is considered also for CE-OFDM and CE-SC-FDMA, in order to exploit all the available subcarriers. In particular, 128 subcarriers have been allocated to 4 streams in block of 32 each.

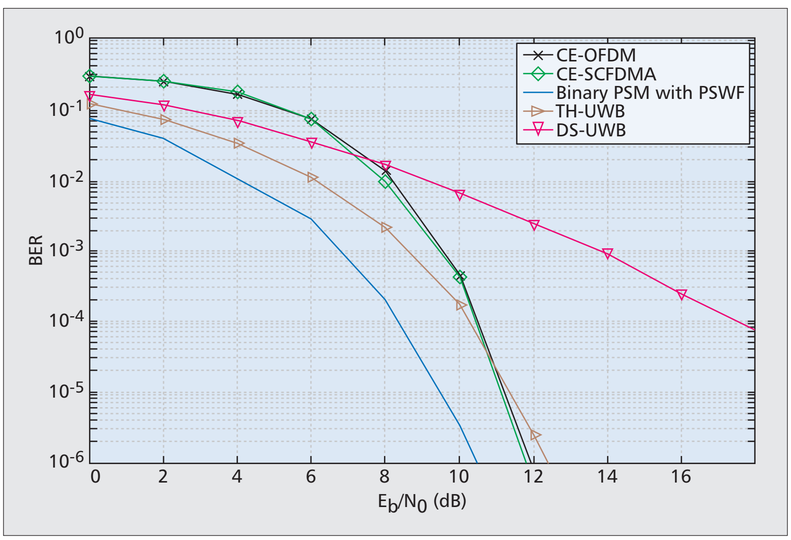

Figure 2. Performance of UWB and CE transmission schemes in the presence of nonlinear distortions.

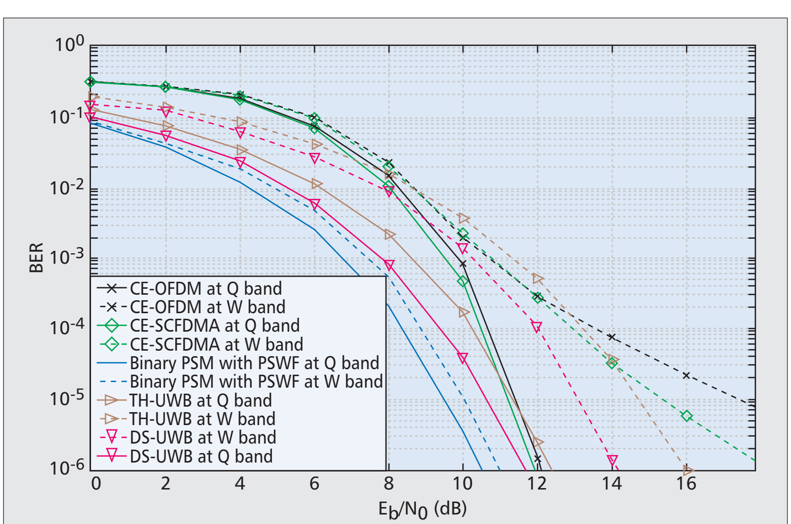

Figure 3. Performance of UWB and CE transmission schemes in the presence of phase noise.

Figure 2, Fig. 3 and Fig. 4 show the comparison in presence of nonlinear distortion only, phase noise only and with both phase noise and nonlinear distortions, respectively. By comparing Fig. 2, Fig 3 and 4, it is possible to infer which RF channel impairment is dominant for each waveform.

Curves in Fig. 2 confirm that CE waveforms are not very sensitive to the nonlinear distortions. On the other hand, an error-floor arises at W-band for CE waveforms, evidently due to the phase noise (Fig. 3). As a matter of fact, the severe phase noise introduced at W-band (at Q-band phase noise is much less severe) it is able to produce noticeable phase discontinuities that the phase unwrapping used in the CE modulations are not able to compensate. Moreover, Fig. 2, Fig. 3 and Fig. 4 show that the considered UWB-based waveforms have very different behavior with respect to these two impairments. As expected, TH-UWB, where the information is on the pulse position, is not much sensitive to the nonlinear distortions introduced by the power amplifier (curves in Fig. 3). Nevertheless, from Fig. 3 TH-UWB appears to be sensitive to the phase noise as the higher phase noise introduced at W-band noticeably degrades the performance with respect to the performance at Q-band. As a matter of fact, phase noise in the time-domain can be seen as fluctuations in the times of zero-crossings of the waveform (time jitters). Even if the phase noise is not so strong to change the polarity of the pulse (in case of bi-phase modulation, as the one considered for DS-UWB), it seems to be able to cause a relevant time-jitter, which has an impact on modulation where the information is encoded in the pulse position. On the other hand, for DS-UWB, nonlinear distortions have a dominant effect. In fact, DS-UWB, which uses a bi-phase modulation is not much sensitive to the phase noise but it is shown to be the most sensitive to the high nonlinear distortions introduced at such high frequency bands, which might cause inversion of the pulse polarity.

The more robust solution with respect to both nonlinear distortions and phase noise is the binary antipodal PSM using order 1 PSWF pulse. Nevertheless, it is worth mentioning that the generation of PSWFs is still unaffordable by state-of-the-art signal processing architectures, at least at the high baud-rate considered in our simulations. Indeed, as shown in [6], 64 samples/pulse are required at minimum in order to maintain the optimal properties of these waveforms. This would result in a sampling rate of 320 GHz, clearly unreachable by state-of-the-art devices.

At W-band, TH-UWB shows performance close to the one of CE waveforms. On the other hand, even if phase noise has a noticeable impact, it does not cause error-floor as it does for the CE waveforms. Furthermore, TH-UWB is practically realizable with current electronic technologies, which is not the case for PSM-PSWF.

Conclusion

In this article, we compared different waveform solutions for future broadband EHF satellite communications. Two main issues should be addressed in such a context: the scarcity of power resources that requires the exploitation of saturating high-power amplifiers at their maximum efficiency and the frequency instability due to phase noise. Some solutions derived by UWB standards, like: TH-UWB, DS-UWB and binary PSM using PSWF have been compared with constant-envelope multicarrier modulations (CE-OFDM and CE-SC-FDMA), whose usage in satellite communications has been very recently proposed. Binary PSM using PSWF theoretically exhibits the best tradeoff between envelope compactness, spectral compactness and robustness against EHF link impairments, but the related waveform generation is complicated, computationally intensive and not affordable by current state-of-the-art signal processing architectures. On the other hand, the generation and the detection of TH-IR signals are very simple and cheap. Such a solution is robust to nonlinear distortion, but it is vulnerable to the effects of phase noise. At the opposite, DS-UWB modulation format is more affected by nonlinear distortions. As far as CE-OFDM and CE-SC-FDMA are concerned, their use may be suggested when the phase noise level is moderate (e.g. in Q band). But, if phase noise increases (e.g. in W band), CE multicarrier techniques may suffer from phase instability. At the present time, TH-UWB looks the best candidate waveform at such high frequency bands, but PSWF-based PSM and CE multicarrier modulations may be considered valuable future alternative, provided that the open issues still impairing them will be solved.

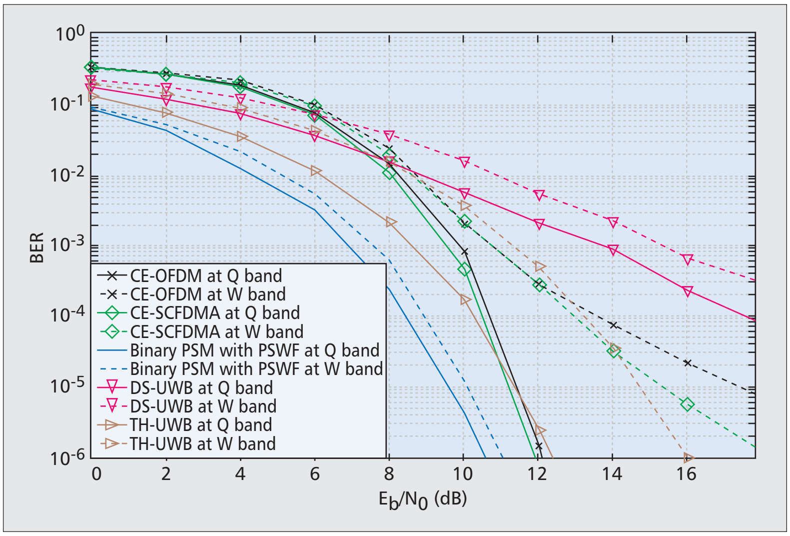

Figure 4. Performance of UWB and CE transmission schemes at Q-band and W-band in the presence of nonlinear distortions and phase noise.

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Biographies

Mauro De Sanctis, Ph.D., is Assistant Professor at the Department of Electronics Engineering, University of Roma "Tor Vergata" (Italy). He is currently involved in the coordination of the Alphasat TDP5 scientific experiments for broadband satellite communications in Q/V band.

Ernestina Cianca is Assistant Professor in Telecommunications at the University of Rome Tor Vergata, Dpt. of Electronics Engineering. She has been the principal investigator of the WAVE-A2 mission, on the design of experimental payloads in W-band. Her research interests are mainly on the wireless access technologies and EHF satellite communications.

Tommaso Rossi is Assistant Professor at the University of Rome Tor Vergata, Department of Electronics Engineering. He is performing experiments on Q/V band satellite communication; in particular he is working on fade mitigation techniques optimization through the "Aldo Paraboni Payload". Moreover he works on radio frequency ultra-wideband radar techniques for medical imaging applications.

Claudio Sacchi [M'01, SM'07] is Assistant Professor at University of Trento, Dept. of Information Engineering and Computer Science (DISI), Trento (Italy). He received the Laurea degree in Electronic Engineering and the PhD in Space Science and Engineering from University of Genoa, Italy. His main research interests include multi-user communications, satellite communications, software-defined radios, etc.

Lorenzo Mucchi [M'98, SM'12] is Assistant Professor at the University of Florence, Dept. of Information Engineering, Firenze (Italy). He received the Ph.D. in Telematics and Information Science by the Univ. of Florence in 2001. His main research areas are: UWB, cognitive radio, physical-layer security and multi-satellite communications.

Ramjee Prasad [M'88, SM'90, F'09] is currently the Director of the Center for Teleinfrastruktur (CTIF), Aalborg University, Aalborg, Denmark, and holds the chair of wireless information and multimedia communications. He is the Founding Chairman of the Global ICT Standardisation Forum for India, established in 2009.

Cognitive Spectrum Utilization in Ka Band Multibeam Satellite Communications

Sina Maleki, Symeon Chatzinotas, Barry Evans, Konstantinos Liolis, Joel Grotz, Alessandro Vanelli-Coralli, and Nicolas Chuberre

Sina Maleki and Symeon Chatzinotas are with the University of Luxembourg.

Barry Evans is with the University of Surrey.

Konstantinos Liolis is with SES TechCom SA.

Joel Grotz is with Newtec Cy.

Alessandro Vanelli-Coralli is with the University of Bologna.

Nicolas Chuberre is with Thales Alenia Space.

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ABSTRACT

Multibeam satellite networks in Ka band have been designed to accommodate the increasing traffic demands expected in the future. However, these systems are spectrum limited due to the current spectrum allocation policies. This paper investigates the potential of applying cognitive radio techniques in satellite communications (SatCom) in order to increase the spectrum oppor- tunities for future generations of satellite networks without interfering with the operation of incumbent services. These extra spectrum opportunities can potentially amount to 2.4 GHz of bandwidth in the downlink and 2 GHz of bandwidth in the uplink for high density fixed satellite services (HDFSS).

INTRODUCTION

The Ka band is mainly considered by the SatCom industry for deployment of satellite high speed broadband networks in un-served and under-served areas. To determine the market demand for Ka band, recent R&D studies in Europe [1, 2] show the potential demand for satellite broadband services in rural areas in order to meet the objectives of the Digital Agenda for Europe, i.e. universal availability of broadband speeds of at least 30 Mb/s throughout Europe, with at least 50 percent of households having access to data rates above 100 Mb/s. Moreover, some studies conclude that the average number of total European households that will choose a satellite broadband connection in 2020 is expected to be between five and 10 million [2]. This represents a market potential for several satellite systems and creates the need to access extra spectrum, including the frequency bands shared with other services, in order to accommodate the increasing bandwidth and data rate demands [3].

It is foreseeable that spectrum congestions can make future Ka band deployments more difficult. High throughput satellites (HTS) already suffer from spectrum scarcity in Ka band in order to deliver meaningful performance. Only 500 MHz of exclusive geostationary satellite orbit (GSO) incumbent spectrum is available in all ITU regions for downlink. The access to a further 2.4 GHz of spectrum in Ka band as discussed in this article potentially allows satellite operators to meet the demand for HDFSS without prior individual frequency coordination. This in turn justifies the need to investigate cognitive radio (CR) techniques in the SatCom context, in order to allow exploitation of the shared frequency bands while guaranteeing acceptable interference to the incumbent users [3].

Another important factor to be taken into account is the long-term and persistent interference from the terrestrial services that affects the core business of satellite operators. In this context, CR based solutions can provide relief as well as a measurable utilization and revenue increase to the SatCom industry.

In this article three scenarios, i.e. A, B, and C, are considered as appropriate opportunities for cognitive SatCom in Ka band with the special focus on the European region. As we show later, these scenarios are in line with the current ITU-R regulations, and are based on the European Conference of Postal and Telecommunications Administrations (CEPT) decisions on dynamic spectrum utilization. Scenario A in the band 17.3–17.7 GHz investigates the spectral coexistence of fixed satellite service (FSS) terminals working in the downlink with broadcasting satellite service (BSS) feeder links in the uplink. This scenario is depicted in Fig. 1a. In this scenario, the cognitive link is from the GSO satellite to the earth FSS terminal, and the incumbent is from the BSS feeder link to a different GSO satellite employed for broadcasting. Scenario B considers a cognitive FSS downlink scheme in the 17.7–19.7 GHz band where the incumbent users are fixed service (FS) links. As shown in Fig. 1b, the cognitive link in this scenario is the same as in Scenario A, but the incumbent link is from one FS terminal to another FS terminal. In Scenario C, in the band 27.5–29.5 GHz where FS links are the incumbent users, the FSS terminal provides cognitive uplink communication. This scenario is illustrated in Fig. 1c. Unlike previous scenarios, here the cognitive link is from the FSS Earth terminal to the GSO satellite, and the

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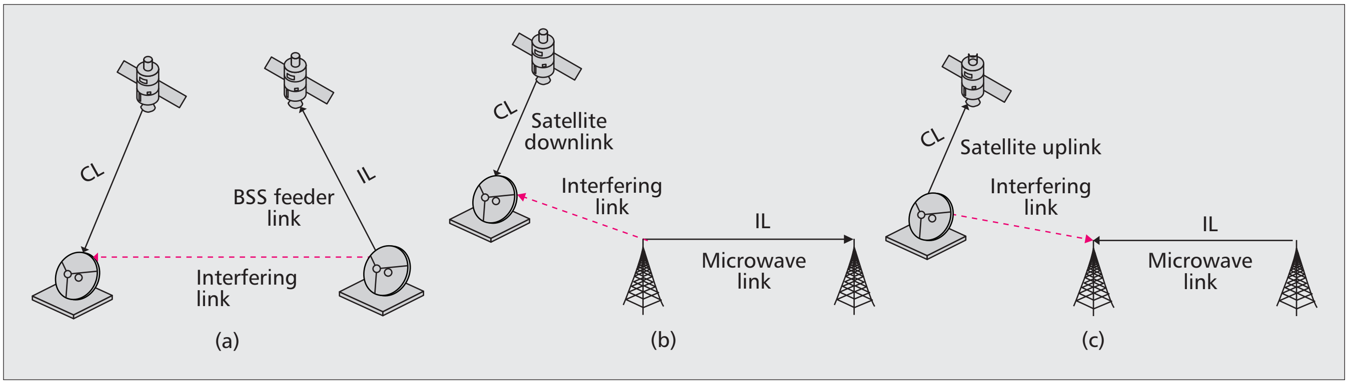

Figure 1. a) Cognitive spectrum utilization in 17.3–17.7 GHz band (Scenario A). CL denotes the cognitive link, and IL denotes the incumbent link; b) Cognitive spectrum utilization in 17.7–19.7 GHz band (Scenario B); c) Cognitive spectrum utilization in 27.5–29.5 GHz band (Scenario C).

incumbent link is the same as in Scenario B. It is important to note that for all three scenarios, the incumbent links are assumed to be fixed with no change in their infrastructure possible due to the coexistence. Furthermore, it is assumed that there is no feedback from the incumbent systems to the cognitive links. Also, it is important to note that all these non-exclusive frequency bands in Ka-band under investigation have shared allocation for many years and are actually shared today. Cognitive radio techniques could allow the use of these shared frequency bands by mass deployed satellite terminals without prior individual frequency coordination, which is needed to satisfy the future market demands.

While the potential gains of cognitive spectrum utilization in Ka band are clear, the required enabling technologies to ensure coexistence within the current regulatory regime need to be developed. These mechanisms include spectrum awareness and spectrum exploitation. Spectrum awareness in turn can be obtained through databases and spectrum sensing. When the spectrum opportunities are known from databases or spectrum sensing, the remaining problem is how to allocate the available carriers in order to optimize the system performance [3].

Dynamic Spectrum Utilization for Ka Band Multibeam Systems

As mentioned earlier, to satisfy the future traffic increase, a wider range of frequency bands is required to provide a high service availability with much higher data rates. In this section we review the spectral regulation in Ka band, and particularly we focus on the potential of cognitive spectrum utilization in the three frequency bands mentioned in the previous section: 17.3–17.7 GHz, 17.7–19.7 GHz, and 27.5–29.5 GHz.

Table 1 provides the ITU-R table of allocations in the aforementioned frequency bands. Considering ITU-R allocation policies as described in Table 1, CEPT has adopted related decisions that give guidance on the use of these bands by FSS. These CEPT decisions are related to the three Ka band scenarios under investigation here and are outlined below [5].

- CEPT has adopted a decision, ECC/DEC/(05)08, which gives guidance on the use of the 17.3–17.7 GHz frequency band by the FSS terminals. The decision stipulates that the incumbent users in this band are BSS feeder uplinks. The deployment of uncoordinated FSS Earth stations is also authorized in these bands. This scenario is depicted in Fig. 1a. As we can see, in the downlink mode, the cognitive transmitter (GSO satellite) does not interfere with the incumbent receiver (GSO satellite). This is controlled by the orbital separation between the GSO satellites. However, the FSS terminal may receive interference from the BSS feeder link earth stations. As shall be shown later, this interference can be managed by employing cognitive techniques.
- CEPT has adopted a decision, ERC/DEC/(00)07, which gives guidance on the use of the 17.7–19.7 GHz frequency band by FSS and FS. The decision stipulates that FSS terminals can be deployed anywhere, but without the right of protection from interference generated by FS radio stations. As shown in Fig. 1b, again the cognitive transmitter does not interfere with the incumbent FS receiver due to power flux density limitations. However, it may receive interference from the FS transmitter. Again, the FSS terminal needs to employ cognitive techniques in order to ensure the minimum system performance.
- CEPT decision ECC/DEC/(05)01 provides a segmentation between FS and FSS stations in the 27.5–29.5 GHz frequency band. The FS segment is lightly used throughout Europe in these frequencies. We envisage an uplink cognitive FSS service in this band where the incumbent users are FS links. As we can see in Fig. 1c, here the cognitive transmitter, which is a FSS terminal, may interfere with the incumbent FS links. Therefore, employing cognitive techniques to avoid harmful interference to the incumbent users is the main challenge in this scenario.

The reference frequency plan, which is considered in this article and is in line with ECC decisions, is depicted in Fig. 2. To sum up the regulation situation, in the downlink mode, the cognitive terminals can freely access the spectrum but need to manage the interference received from the incumbent users; in the cognitive uplink mode, they need to make sure that their transmission does not interfere with the incumbent receivers. To solve these issues, we

The spectrum awareness unit is responsible for gaining knowledge about the incumbent users, and it is based on a joint interaction among the database, spectrum sensing, and beamforming functions, with the aid of a cognitive zone block.

Frequency bands	ITU-R Region 1	ITU-R Region 2	ITU-R Region 3
17.3–17.7 GHz	FSS (space-Earth) BSS (feeder links) Radiolocation	FSS (space-Earth) BSS (feeder links) Radiolocation	FSS (space-Earth) BSS (feeder links) Radiolocation
17.7–19.7 GHz	FSS (space-Earth) BSS (feeder links up 18.1 GHz) FS	FSS (space-Earth) FS	FSS (space-Earth) BSS (feeder links up 18.1GHz) FS
27.5–29.5 GHz	FSS (Earth to space) FS MS (mobile services)	FSS (Earth to space) FS MS	FSS (Earth to space) FS MS

Table 1. ITU-R table of allocations in Ka band [4].

may need to employ CR mechanisms. These techniques are described in the following section. Note that the frequency plan of Fig. 2 is merely illustrative as the techniques can be applied to any Ka band satellite system.

Cognitive Mechanisms for Opportunistic Spectrum Utilization

To enable cognitive spectrum utilization in Ka band, a system architecture as shown in Fig. 3 is employed. This system consists of a spectrum awareness unit that provides information regarding the incumbent users, and a network management or spectrum exploitation unit that allocates the carriers and the transmission power to the users. Each of these units are explained in detail in this section.

Spectrum Awareness

The spectrum awareness unit is responsible for gaining knowledge about the incumbent users, and it is based on a joint interaction among the database, spectrum sensing, and beamforming functions, with the aid of a cognitive zone block as described in this section.

Databases — The aim of the database shown in Fig. 3 is to incorporate BSS feeder links and FS characteristics in order to determine the interference levels at any proposed FSS terminal location for any carrier frequency. In this way, the FSS terminals can operate within the carrier frequencies where the regulatory interference thresholds are respected. If the received interference is deemed to be high, then CR mechanisms should be employed in order to mitigate the interference. In addition, as shown in Fig. 3, the database can be connected to a network management unit to optimize the overall resource allocation in the network.

Cognitive Zones — A cognitive zone is defined as the geographical area around an incumbent user where cognitive techniques need to be employed in order to limit the interference to an acceptable level. Interference modelling connected to the databases (as in Fig. 3) is performed using the ITU-R P.452-15 terrestrial propagation model. The zone boundaries are determined by the respective regulatory interference to noise ratio (I/N) thresholds. The total interference is obtained by summing all the received interferences at a particular location for the FSS terminal. In the case of Scenarios A and B, the zones are calculated around the FSS cognitive receiver using the databases for BSS and FS, respectively. In the case of scenario C, they are calculated with respect to the FS links, as the incumbent, being potentially interfered from the FSS uplink. Scenario C is different from the two downlink cases as in this case, the FSS terminal is the potential interferer to the FS. Cognitive zones can still be used for the FS incumbents. The main challenge here is to gain accurate knowledge about the FS links, and to model the interference accurately, so as to avoid any harmful interference to the incumbent users. A typical cognitive zone plot for scenario A is depicted in Fig. 4. This figure was calculated using the full ITU-R P.452-15 model with terrain effects included. The terrain resolution was 500 m and the interference was calculated for 20 percent of the year. More detailed analysis regarding the cognitive zone, particularly for Scenarios A and B with several databases obtained from regulatory bodies, can be found in [6, 7]. It is shown that a vast geographical area can be used freely for cognitive downlink satellite communications. In the few remaining areas, cognitive mechanisms such as spectrum sensing and beamforming, as described in the following subsections, can be employed to limit the interference.

Spectrum Sensing — Spectrum sensing is a technique to acquire spectral knowledge about incumbent user activities within a specific geographical location [8]. Several techniques have been proposed in order to perform spectrum sensing. Among these, energy detection, cyclostationary detection, and matched filtering are the most common. When it comes to the satellite communication in Ka band, spectrum sensing can be applied either to detect the incoming interference from the BSS feeder links or FS links for downlink scenarios. In the uplink scenario C, since sensing the passive receivers is practically not viable, spectrum sensing has to be done in the FS link and, therefore, it is not considered as an enabling technique for this specific scenario. As shown in Fig. 3, in both cases, the spectrum sensing unit shares the obtained information with the database and network manage-

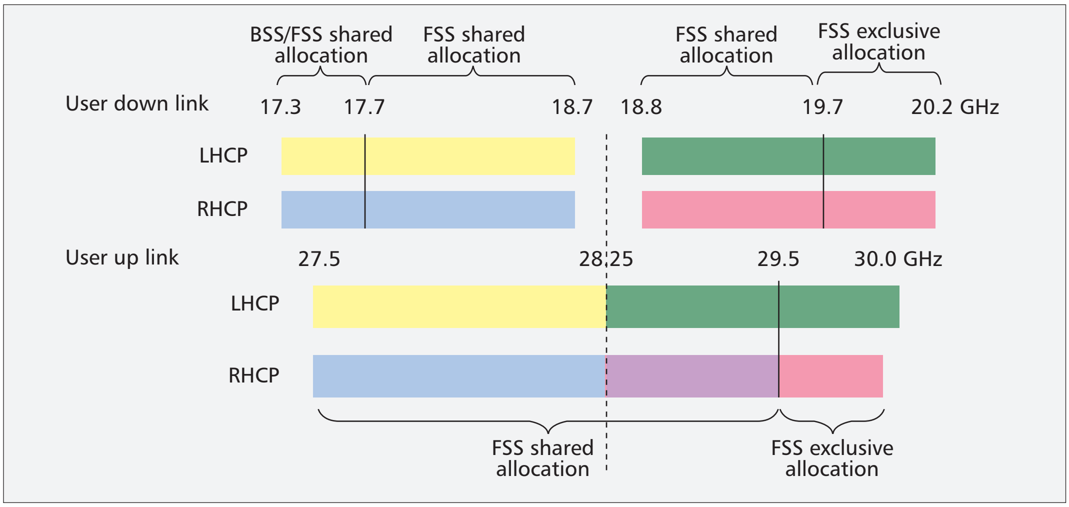

Figure 2. The four color frequency reuse scheme for cognitive multibeam satellite networks.

A beamformer is a spatial filter that operates on the outputs of an antenna array in order to form a desired beam pattern. The signals induced at the different elements of the array are combined to form a single output of the array.

ment units. Here again, traditional sensing techniques such as energy detection and cyclostationary detection can be applied [9, 10].

The main constraint in the downlink scenarios is how much interference a FSS terminal can tolerate. The FSS terminal producers can determine this value which may amount to I/N = –10dB. Moreover, if the FSS link budget side knowledge is available, a higher interference level can also become acceptable if the quality of the received signal is still satisfactory. This encourages spectrum sensing through Signal to Interference and Noise Ratio (SINR) estimation as described in [6, 11]. The limiting factor here is the presence of the satellite beam. Usually, for reliable sensing results, the cognitive transmitter is quiet. This condition cannot be applied to satellite networks, as the satellite beam cannot be turned off for individual users. To solve this issue, as explained in the following subsections, beamforming techniques can be applied to cancel out the satellite signal.

Beamforming — A beamformer is a spatial filter that operates on the outputs of an antenna array in order to form a desired beam pattern. The signals induced at the different elements of the array are combined to form a single output of the array.

Beamforming techniques can be implemented either on the terminal-side or on the satellite-side in order to improve the received SINR at the cognitive receiver or produce a null at the incumbent receivers. It should be noted that implementation of a beamforming technique requires a significant upgrade in the existing FSS system.A terminal equipped with multiple antennas is required to create a desired beam pattern; this can be achieved by either adding omnidirectional antennas or multiple-LNBs. In this context, different approaches for applying existing beamforming and precoding techniques in Ka band cognitive communications can be considered.

In a nutshell, as we can see in Fig. 3, the beamforming techniques can be applied in the cognitive satellite communications in two ways:
- Beamforming for spectrum sensing/interference detection.
- Beamforming for improving the SINR of the satellite terminals.

The latter leads to a smaller size cognitive zone and is investigated with promising results in [6, 12].

In Scenarios A and B, a receiver beamformer can be designed to detect the harmful FS signals so that the satellite terminal can avoid using the harmful FS carriers, or to enhance the SINR by mitigating the interference using side-lobe cancellation [12].

Spectrum Exploitation

Dynamic Carrier Allocation — So far we have determined the available cognitive resources in specific times, frequency carriers, and geographical locations. In this subsection we outline the methods that are necessary to allocate these resources (particularly the carriers) to the users in the network management unit as shown in Fig. 3.

Two major approaches can be employed by the network manager in order to perform carrier allocation. In the first approach, the goal is to assign the carriers so as to maximize the sum of the throughput and thus the overall system throughput. In the second approach, the goal is to maximize the availability through assigning the available carriers to as many users as possible according to their requested rate. Under rate constraints, a carrier can only be assigned to a user if it satisfies a specific rate request, which is in turn directly related to the received SINR at a specific carrier. Cognitive carriers can be a potential help in this case by increasing the system availability.

While these approaches are efficient in terms of network management, several other factors can be considered for carrier allocation, e.g. carrier assignment priorities, carrier aggregation, and shared carrier assignment.

Power Control — As power control in the downlink is well established in current satellite networks, from a cognitive resource allocation perspective, we focus on power allocation related to the uplink scenarios such as the one in the band 27.5–29.5 GHz. Here the transmission power should be determined so as to first increase the system throughput, and second to

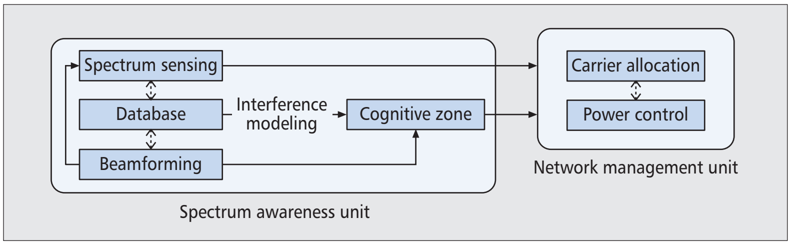

Figure 3. Cognitive system architecture.

keep the interference to the incumbent receivers below a specific value.

Implementation and Technological Challenges

In the previous section, while describing each enabling technique, we outlined some of the challenges that the techniques may face in practice. In this section we evaluate these challenges in detail. Note that the described techniques have been thoroughly examined in a number of practical scenarios in [6]. The obtained results show that the adopted cognitive mechanisms are promising, particularly in the downlink scenarios. The described challenges in this section are provided in order to be considered in the implementation of cognitive terminals to achieve better performances with respect to what is achieved in [6].

As mentioned before, databases leverage a network-level implementation of cognitive satellite communication. Applicability of the database to the considered scenarios, particularly Scenarios A and B, is thoroughly investigated in [6, 7]. It is shown that the database in combination with interference modelling, and cognitive carrier allocation can provide a significant gain in terms of total achievable throughput and geographical availability. However, the database techniques face a number of challenges. The database may become outdated by time, and thus the entity that operates the database needs to update it frequently. Further, some of the frequency bands may be assigned to confidential users, and public databases do not include the information about these links. In these cases, dynamic database techniques in combination with spectrum sensing such as spectrum cartography based on received observation from field sensors can be employed to localize the incumbent users [13].

One approach to obtain a dynamic database is to apply SINR estimation at the satellite terminals. In [6] and [11], a SINR estimation algorithm is introduced that is shown to be effective in producing a dynamic database of the received SINR in Scenarios A and B. However, the specific satellite link conditions create unique challenges for spectrum sensing. In fact, sensing of the interference level has the particular challenge that the signal strength variations can be very large, due to highly directional antennas used at the terminals. Furthermore, the FSS signal received from the satellite has significantly lower power spectral density than the terrestrial FS link. Under worst case power imbalance conditions, this may result in a saturation of the front-end low noise amplifier (LNA). To solve this issue, a special adaptation of LNA is required to prevent the amplifier saturation condition.

Another characteristic that represents a challenge is the heterogeneous feature of the two systems and their signal parameters. The signal parameters of the interferer may therefore be known only to some. This limits applying feature based spectrum sensing techniques. However, this is not a critical issue in Scenarios A and B, as in these scenarios, the most important required information is the received SINR.

In addition to the interference detection challenge, as mentioned earlier, the interference could be also mitigated by using a multiple antenna system. The interference mitigation may be possible if the antenna directivity of the FSS satellite terminal has reconfigurable elements, such as beamforming approaches [14]. These methods require a calibration and configuration of beamforming at the antenna installation and a direction of arrival (DoA) estimation to the interferer. For the FSS downlink scenarios, we can use the reception quality indicators, e.g. SNR estimates and error rates, as calibration and configuration feedback. This is, however, not feasible for the uplink scenario, for which we require a well calibrated and configured setup to ensure that the transmit beamforming direction corresponds to the FS receiver link direction. Because no cooperation between the FS and FSS systems is foreseen so far, this context requires special consideration and possibly a new enabling technology to make the beamforming context work. Successful application of beamforming in interference mitigation for Scenarios A and B is studied in [6, 12].

Another challenge that needs to be taken into account is the effect of atmospheric impairments, especially rain fading, on Ka band communications. This leads to dynamic change of the SINR, which needs to be addressed in the design of dynamic carrier allocation algorithms. However, these are usually short-term effects that can be compensated by adaptive modulation and coding (ACM). In our system and interference modeling (e.g. Fig. 4 and the detailed results in [6]), we focus on long term values, which are more useful in the regulatory context.

Conclusions

The potential of CR applications to SatCom was identified in this article. The regulatory issues were discussed, and three major scenarios were analyzed as the immediate opportunities for extra spectrum. Enabling techniques to facilitate dynamic spectrum utilization were investigated. It was shown that database techniques can enable a network level implementation of cognitive satellite networks. Further, we concluded that spectrum sensing can be used to update the database knowledge about the incumbent users. Considering the challenges therein, novel spectrum sensing could be developed in order to obtain more accurate results. Further, the database of current incumbent users in Ka band needs to be acquired and managed by an entity to enable efficient network level resource allocation.

With respect to CR applicability in the

respective frequency ranges within Ka band, the following points can be noted:

- 17.3–17.7 GHz band: Cognitive utilization of this band in addition to the FSS exclusive band of 19.7-20.2 GHz opens up an additional 400 MHz bandwidth, which can potentially increase the total throughput by 90 percent [6].
- 17.7–19.7 GHz band: CR techniques can significantly increase the spectrum utilization allocated to FSS by enabling access to additional frequency spectrum. CR techniques could act as a dynamic protection of FSS downlink from FS interference.
- 27.5–29.5 GHz band: FSS stations can maximize frequency exploitation by dynamic utilization of the FS segment through the adoption of CR techniques in the satellite uplink able to dynamically control the interference generated to the FS stations.

Last but not least, note that the European Telecommunications Standards Institute (ETSI) recently published the System Reference Document (SRDoc) TR 103-263 v1.1.1 "Cognitive Radio Techniques operating in Ka-band" [15] which is considered to be a major milestone in standardization of relevant CR SatCom activities.

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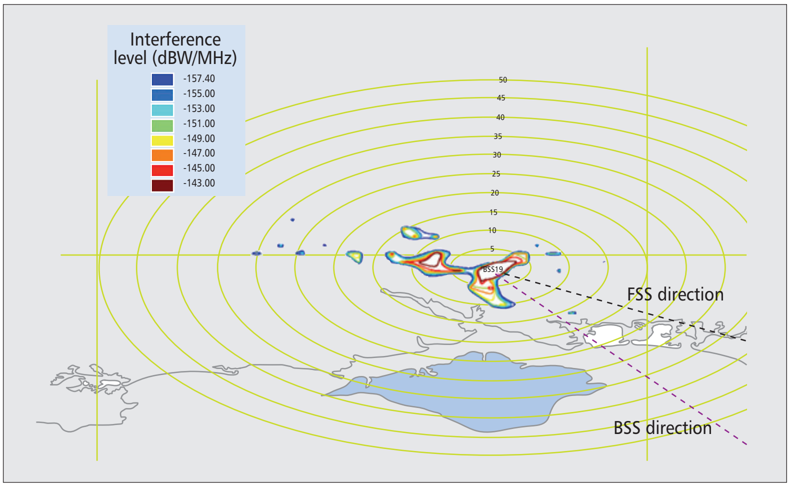

Figure 4. Cognitive zone example based on UK OFCOM BSS database.

Biographies

SINA MALEKI (sina.maleki@uni.lu) received his Ph.D. from Delft University of Technology, Netherlands in 2013. He is currently a research associate in SnT, University of Luxembourg, working on several national and international projects related to interference processing and cognitive radio for satellite communications.

SYMEON CHATZINOTAS is a research scientist in SnT, University of Luxembourg, managing H2020, ESA and FNR projects. He has authored more than 120 technical papers in international journals, conferences, and scientific books. He is the co-recipient of the 2014 Distinguished Contributions to Satellite Communications Award by IEEE ComSoc.

BARRY EVANS was until recently Director of the Centre for Communications Research at Surrey and VP for Research and Enterprise for the University. He has been involved in satellite communications for over 30 years and is editor of the *International Journal of Satellite communications and Networks* and author of the IET Press book on satellite communication systems. He has been involved in many recent EU and ESA funded satellite programs and is on the Steering Board of the European Engineering Technology Platform and the European Network of Satellite Experts.

KONSTANTINOS LIOLIS holds the Ph.D. and Dipl.-Eng. degrees from the National Technical University of Athens, Greece, and the MSc degree from the University of California at San Diego, USA. He is currently with SES TechCom SA, Luxembourg, working as a space systems engineer, mainly on R&D projects. He has more than 40 publications in international journals, conferences, and book chapters, mainly in the satellite communications area.

JOEL GROTZ currently works as senior engineer within the Technology Labs at Newtec Cy, Belgium. He graduated from the University of Karlsruhe and the Grenoble Institute of Technology in electrical engineering and holds a Ph.D. from the Royal Institute of Technology (KTH), Stockholm. He has previously worked at SES on projects related to satellite based Internet access and broadcast systems as well as payload specification and testing.

ALESSANDRO VANELLI-CORALLI received the Dr. Ing. Degree and the Ph.D. from the University of Bologna (Italy) in 1991 and 1996, respectively. In 1996 he joined the University of Bologna, where he is currently an associate professor. He is an elected member of the Steering Board of the Networld European Technology Platforms, representing the SatCom R&D domain. He has been involved in several EU and ESA projects.

NICOLAS CHUBERRE holds an engineering degree from "Ecole Supérieure d'Ingénieur en Electronique et Electrotechnique." He currently manages R&D projects to design advanced GSO and NGSO satellite communication solutions at Thales Alenia Space, France. He is an elected member of the Steering Board of the Networld2020 European Technology Platform and chairs the Satellite Communication and Navigation Working Group at ETSI.

IP Multicast Receiver Mobility Support Using PMIPv6 in a Global Satellite Network

Esua Kinyuy Jaff, Prashant Pillai, and Yim Fun Hu

The authors are with the University of Bradford.

ABSTRACT

A new generation of satellite systems that support regenerative on-board processors (OBPs) and multiple spot beam technology have opened new and efficient possibilities of implementing IP multicast communication over satellites. These new features have widened the scope of satellite-based applications and also enable satellite operators to efficiently utilize their allocated bandwidth resources. This makes it possible to provide cost effective satellite network services. IP multicast is a network layer protocol designed for group communication to save bandwidth resources and reduce processing overhead on the source side. The inherent broadcast nature of satellites, their global coverage (air, land, and sea), and direct access to a large number of subscribers imply satellites have unrivalled advantages in supporting IP multicast applications. IP mobility support in general and IP mobile multicast support in particular on mobile satellite terminals like the ones mounted on long haul flights, maritime vessels, continental trains, etc., still remain big challenges that have received very little attention from the research community. This paper proposes how Proxy Mobile IPv6 (PMIPv6)-based IP multicast mobility support defined for terrestrial networks can be adopted and used to support IP mobile multicast in future satellite networks, taking cognizance of the trend in the evolution of satellite communications.

INTRODUCTION

Recently, the role played by satellites in voice and data communication has witnessed a rapid growth. This is due to the advancement in satellite technologies such as support for regenerative OBP, spot beam technology, and the ability to make use of higher frequency bands, e.g. the Ka-band. The presence of regenerative OBP in today's satellite systems implies that IP multicast packets can be replicated on-board the satellite, and a full-mesh, single-hop communication can be established between two or more satellite terminals/gateways. These features reduce the round trip delay in traditional bent pipe satellite systems by half. Support for multiple spot-beam technology in regenerative satellite systems makes efficient frequency reuse possible within different spot beams. Frequency reuse efficiently utilizes the allocated frequency spectrum and increases the overall satellite capacity. Also, spot-beams make it possible for the satellite to focus its power over a relatively small area using narrow beams resulting in high power density. High power density supports high data rates, reduces the power requirement and size of satellite terminals, and thus reduces the overall cost of satellite communication. Next generation satellite systems such as the Inmarsat Global Xpress operate at the Ka-band. The advantages of operating at this frequency spectrum are: support for higher data rates; offering more available frequency spectrum compared to the Ku-band (five times the availability at the Ku-band); and less competition for spectrum as there are very few operational satellites in the Ka-band [1]. These new features have broadened the scope of satellite-based applications and also made satellite communications more competitive in multimedia, integrated voice and data communications compared with other Internet-based communications technologies.

IP multicast is a network layer protocol that enables a sender to perform a single local transmit operation to deliver the same data simultaneously to a group of interested receivers. This saves processing overhead at the sender associated with sending multiple copies to individual users, and bandwidth overhead in the network since only one copy of the data traverses any network link leading to an interested receiver. The large geographical coverage area and broadcast nature of satellite networks are well suited for multicast applications. Unlike in broadcast, where the traffic is flooded in the whole satellite footprint, in IP multicast, traffic is only sent to spot beams that have at least one interested receiver, thus saving bandwidth in those spot beams that have no receivers. IP multicast applications that could be applicable to mobile satellite scenarios (MSSs) as in long haul flights, global maritime vessels, continental trains, etc., include: on-demand multimedia content delivery (e.g. IPTV), real-time financial data, software distribution and upgrade, important service

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information such as weather conditions, ongoing disaster zones and information, route updates, etc. With all these new applications, next generation satellite networks with their support for fast Internet broadband have a unique opportunity to attract new customers and generate new revenues by deploying these new IP-based services. The aeronautical industry, which is one of the key customers for mobile satellite services, has recently adopted IP as the future network protocol for the Aeronautical Telecommunication Network (ATN) [2]. This opens up new opportunities for satellite-based IP multicast applications on mobile platforms, as real-time important service information could be cost-effectively disseminated using IP multicasting, to a group of airlines in mid-air operating in a particular region or route or from an airline to a group of ground offices/emergency services around the world. Thus, IP multicast support of customers (airliners, trains, ships, etc.) could bring significant financial savings due to the efficient utilization of the allocated bandwidth resources. For satellite operators, the bandwidth resources saved in each satellite footprint could be made available to satisfy the existing customers' demands or sell to new customers.

IP Multicast Mobility Issues in Satellite Networks

In dynamic multicast group membership, when a receiver joins a multicast group, a multicast delivery tree is established linking the receiver to the multicast source. When the source or receiver moves from one satellite beam to another, the delivery tree is broken because its identity (IP address) or location have changed, so multicast traffic from the source cannot reach the receiver. Assuming that Mobile IP (MIP) is supported within the satellite network, the following two problems arise in such a scenario.

- **Mobile Receiver Problems**: For a mobile receiver to re-establish the delivery tree, it has to signal its current location to its home agent (HA) or re-subscribe to the multicast group after handover (HO) as a new member using the newly acquired care-of-address (CoA). Considering the long latency in satellite networks, link-switching delay, MIP protocol operations, member- ship protocol implementation, multicast tree reconstruction, etc., multicast traffic would face a large delay during the HO process, even leading to a break in a real-time multicast application.
- **Mobile Source Problems**: Unlike the HO of a mobile multicast receiver that has just a local and single impact on that particular receiver only, that of a mobile source is a critical issue as it may affect the entire multicast group. During HO procedure, the mobile source cannot send traffic when switching from an old set of satellite resources in the old beam to the new set in the target beam. For an ongoing multicast session, this could result in long multicast latency to the entire multicast group, causing serious problems, especially with real-time applications. If the HO is between beams belonging to different GWs i.e. gateway handover (GWH), then the IP address of the mobile source will change. This creates a serious problem, particularly in source-specific multicast (SSM), where a receiver subscribes to a multicast channel (S, G) i.e. to receive traffic from a specified source identified by its IP address S. When the new IP address (CoA) of the mobile source in the target beam is used as the source address to send traffic, the OBP nor designated multicast router in the foreign network cannot forward the multicast packets until the receivers explicitly subscribe to this new multicast channel (CoA, G). This is known as the transparency problem. Also in SSM, a multicast source is always at the root of the source-specific tree. The reverse path forwarding (RPF) check compares the packet's source IP address against the interface upon which the packet is received. The change of the source location and consequently its IP address during GWH invalidates the existing source-specific tree, as any traffic sent by the mobile source from the target beam using its home address as the source address will result in a failure of the RPF check test and ingress filtering. Hence, the RPF problem points to the fact that the mobile source away from the home network cannot use its home address as the source address to send packets.

This paper looks at the multicast receiver problem in detail and presents a PMIPv6-based solution to support mobile receivers using multicast applications in a satellite environment. The solution for mobile sources is out of the scope of this paper.

PMIPv6-Based IP Multicast Receiver Mobility in Terrestrial Networks

Mobile nodes (MNs) in network-based IP mobility management protocols such as the PMIPv6 protocol [3] do not participate in network layer HO procedures unlike in host-based mobility protocols (e.g. all variants of MIP, etc.), where the MN is an IP mobility-aware node that is fully involved in layer 3 HO signaling. The IETF working group, Multicast Mobility (multimob), charged with the responsibility of providing multicast support in a mobile environment, sees PMIPv6 as the way forward for IP multicast mobility support in terrestrial IP networks.

The authors in [4] have proposed an IP multicast receiver mobility support schemes based on the PMIPv6 protocol for terrestrial networks. Two operational modes, multicast tree mobility anchor (MTMA) and direct routing (DR), are proposed for IP multicast provisioning within the PMIPv6 domain, with the aim of solving the tunnel convergence problem between the local mobility anchor (LMA) and mobility access gateways (MAGs) that exist in [5]. In this proposal, the IP multicast traffic to or from the domain is separated from the unicast traffic. The unicast traffic passes through the LMA as defined in [3], and the multicast traffic passes through the MTMA in the MTAM mode or the multicast router (MR) in the DR mode. The difference between the two operational modes is that in the MTMA, a bi-directional tunnel is established between the MTMA and the MAGs

IP multicast support for customers could bring significant financial savings due to the efficient utilization of the allocated bandwidth resources. For satellite operators, the bandwidth resources saved in each satellite footprint could be made available to satisfy the existing customers' demands or sell to new customers.

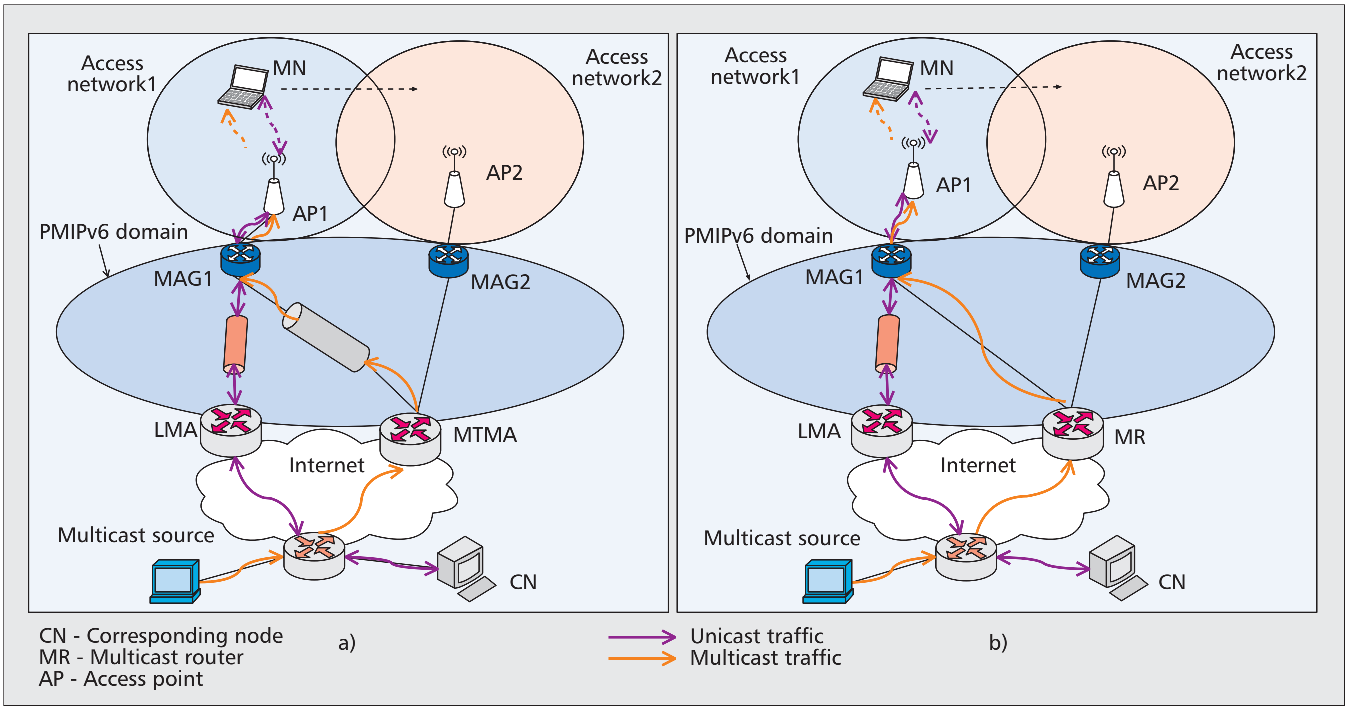

Figure 1. PMIPv6-based architecture for IP multicast receiver mobility support: a) MTMA architecture; b) DR architecture.

that have MNs with multicast group membership, while in the DR mode, native multicast routing takes place between the MR and MAGs. In both modes, the MAGs support the multicast listener discovery (MLD) proxy function where the MNs are connected to the downstream interface, and the upstream interface of the MLD proxy is configured to point toward the internal interface of the MTMA or MR. Figure 1 illustrates the two modes in a PMIPv6 domain.

Existing Solutions for IP Multicast Receiver Mobility in Satellite Networks

Very little has been written about IP mobile multicast support over satellite networks. The authors in [6] proposed the MIP home subscription (HS)-based and remote subscription (RS)-based approaches to support a mobile return channel satellite terminal (RCST) whenever it is away from its home network. In the HS-based approach, the CoA acquired by the mobile RCST in a foreign network is registered with its HA through the foreign GW where the mobile RCST is currently located. A bi-directional tunnel is then established between the mobile RCST's HA at home GW and the foreign GW serving the mobile RCST. The HS-based approach, therefore, relies on the HA at home GW tunnelling multicast traffic to the mobile RCST in a foreign network. On the other hand, in the RS-based approach, the mobile RCST uses its newly acquired CoA to re-subscribe to the groups that it was a member of before handover. While the HS-based approach suffers from triangular routing problems, high HO latency, and signaling overhead, the RS-based approach suffers from multicast delivery tree re-construction, high HO latency, and additional signaling overhead.

IP multicast receiver mobility using multi-homing in a multi-beam satellite network is proposed in [7]. It leverages on the multiple interfaces for seamless HO provisioning whenever a mobile RCST changes its point of attachment from one satellite GW to another. During GWH, it is proposed here that interface 2 should establish connection to the target beam, obtain a CoA, and join all the multicast groups that interface 1 is a member of. Once the mobile RCST starts receiving multicast traffic from all the requested groups through interface 2, then interface 1 can de-register from all the multicast groups and shut down or log off. Despite the advantage of seamless HO, the implementation of this proposed approach requires hardware modification to the standard RCST, which usually would have just one satellite interactive interface. This modification could be very expensive. This approach also suffers from high signaling overhead due to the IP address acquisition process for the second interface.

All these proposed approaches have two common features:
- The mobile RCST must be an IP mobility-aware node.
- They are all host-based IP mobility management protocols that require additional software and complex security configurations on each mobile RCST for IP mobility support since the mobile RCST must participate in IP mobility signaling during GWH.

These two common features are at the center of the drawbacks associated with host-based mobility management protocols. The recent trend in IP mobility is shifting from host-based mobility management protocols to more network-based protocols.

As seen with the previous subsection, PMIPv6

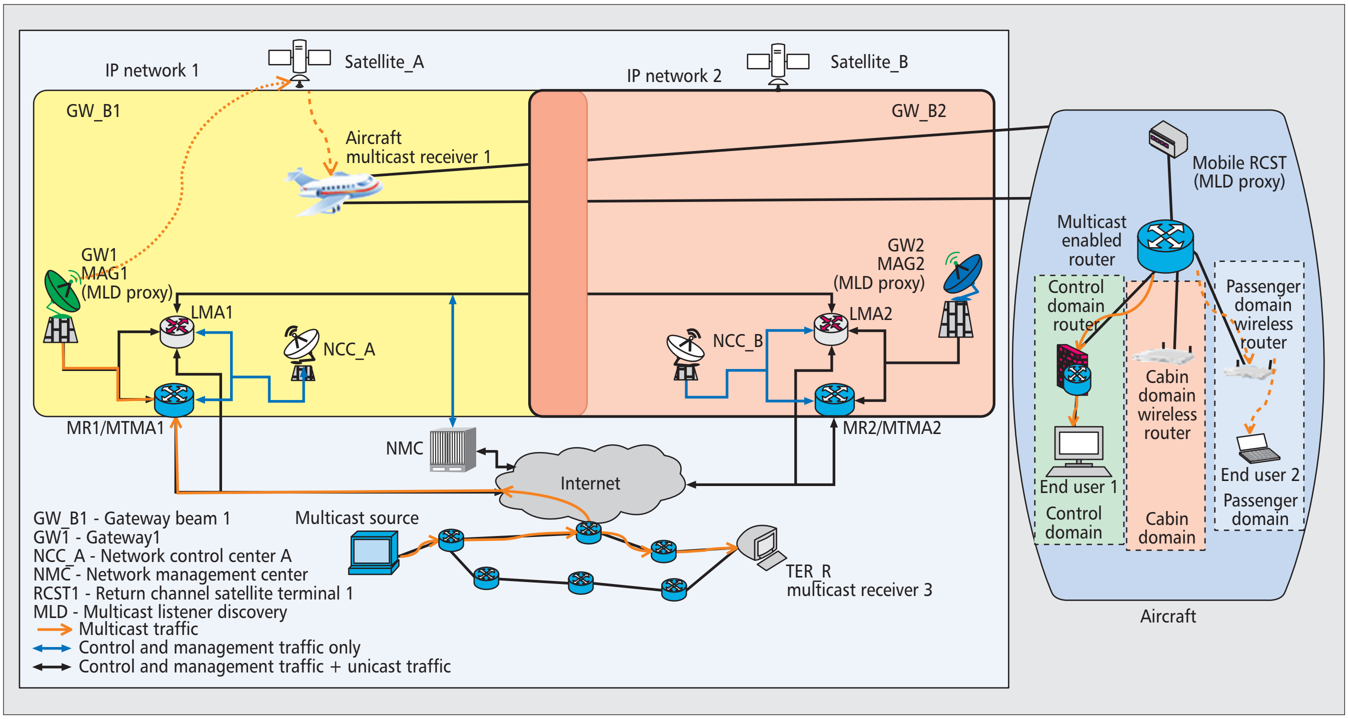

Figure 2. Satellite PMIPv6-based IP multicast communications in a mobile scenario.

provides an elegant solution for supporting receiver mobility in terrestrial networks. However, this has not yet been considered for a satellite network. The following section describes how PMIPv6 can be adopted to support efficient receiver mobility for IP multicast applications within a satellite environment.

Satellite PMIPv6 Network Architecture for IP Multicast Communication

Proposed Network Architecture

Figure 2 shows the proposed PMIPv6 based network architecture for IP multicast receiver mobility in a satellite network. The footprint of each satellite here forms a GW_beam (or global beam) representing a separate IP network and has a GW that interconnects the satellite network to terrestrial networks (e.g. the Internet). There is usually a network control center (NCC) associated with each satellite operator. The NCC provides real-time control and monitoring functions, e.g. session control, connection control, terminal access control to satellite resources, routing, etc. The network management center (NMC) is responsible for the management functions of all system elements in the interactive network (IN). In the proposed network architecture shown in Fig. 2, it is assumed that:

- Three large GW_beams would be used to provide global coverage in order to constitute one global satellite IN under the administration of one satellite network operator.
- The regenerative OBP on each satellite, which decouples the uplink and downlink transponders of each beam, has a data link layer capability (layer 2 switch).
- The regenerative OBP supports on-board replication of multicast packets. Each of the GW_beams is sub-divided into many narrow spot-beams for reasons explained above. The new Inmarsat Global Xpress satellite network has 89 narrow spot beams per satellite.

One of the most difficult tasks in deploying PMIPv6-based mobility management in a global multi-beam satellite-terrestrial hybrid network is to select the most suitable location of the LMA, MR/MTMA, and MAG, taking into consideration their mobility management functions. Each satellite footprint in Fig. 2 is proposed to have one LMA, MR/MTMA, and MAG. The LMA is dedicated to unicast traffic; the MR/MTMA is dedicated to multicast traffic; and the MAG, which is configured at each GW, acts as an MLD proxy. The policy profiles of all mobile RCSTs authorized for global network-based IP mobility management are proposed to be stored at all the LMAs and MAGs. Each mobile RCST's policy profile must contain the mobile RCST's identifier (e.g. MAC address), home network prefix (HNP), link-local address (LLA), and the IPv6 address of its LMA. As shown in Fig. 2, the multicast source is assumed to be a fixed node on the terrestrial network, while the receivers are located both on the satellite and terrestrial networks and are mobile. The aircraft consists of a mobile RCST and acts as a satellite-based mobile multicast receiver. TER-R is a terrestrial-based multicast receiver. Due to the ability to replicate IP multicast traffic on-board the satellite, only one copy of the multicast traffic is sent up to satellite_A regardless of the number of receivers under the satellite's footprint. To efficiently utilize the satellite bandwidth resources, the downlink forwards multicast traffic only to the spot beams that have at least one receiver.

Detailed Working of PMIPv6-based IP Multicast Receiver Mobility Support in MSS

Note should be made here that the role played by the proposed PMIPv6-based support is mainly at the execution phase of the satellite handover (SH) process. As the aircraft (mobile RCST) enters the overlapping area between GW_B1 and GW_B2, it will undergo a SH. A SH takes place when a mobile RCST moves from one beam into another that belongs to a different satellite. A SH within one IN is coordinated by the NMC. The aircraft is assumed to be equipped with a Global Positioning System (GPS)/Galileo receiver and the global satellite network map. These

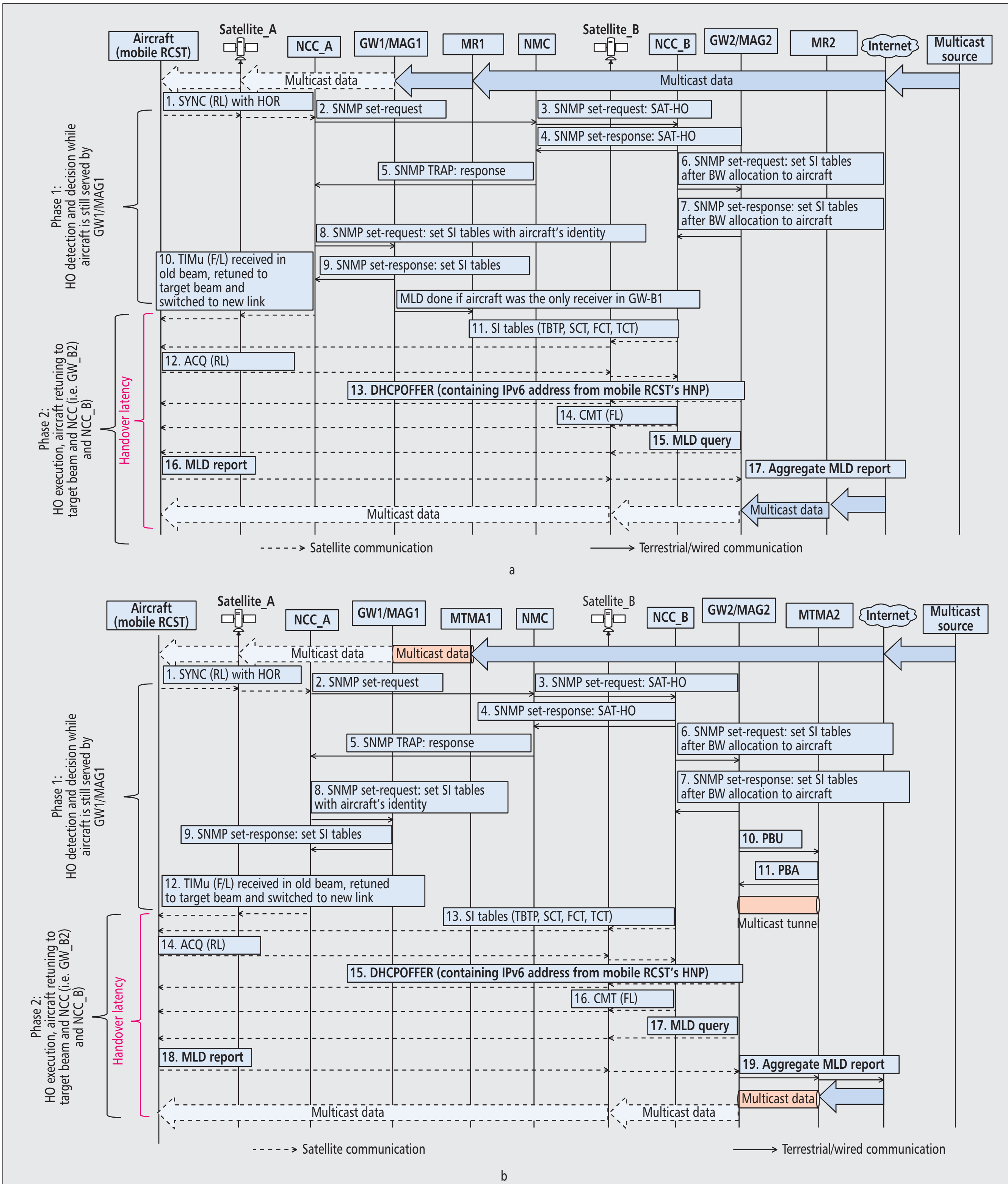

Figure 3. SH signaling sequence for PMIPv6-based IP multicast receiver mobility support.

	Mobile RCST involved in layer 3 signaling at GWH	Efficiency of routing after handover	Layer 3 signaling in satellite network at GWH	Mobile RCST hardware change required	Mobile RCST software change required
HS-based	Yes	No	Yes	No	Yes
RS-based	Yes	Yes	Yes	No	Yes
Multi-homed-based	Yes	Yes	Yes	Yes	Yes
PMIPv6-based: DR	No	Yes	No	No	No
PMIPv6-based: MTMA	No	No	No	No	No

Table 1. Comparison of different IP multicast receiver mobility support schemes.

will enable the aircraft to perform the analysis of its position information and then signal the handover recommendation with a specified target beam to be used in the handover decision process by the NCC. The whole SH process is divided into the following two phases.

Phase 1: HO Detection and Decision: During this phase, the aircraft executes an HO detection algorithm as it enters the overlapping area between GW_B1 and GW_B2, and sends an HO recommendation to NCC_A with a specified target beam identity. Upon reception of the HO request, NCC_A, using its data base, determines that it is an SH. Signaling as shown in Fig. 3, between NCC_A, NMC, NCC_B, GW1, and GW2 then follows, resulting in the aircraft acquiring satellite bandwidth resources in GW_B2 (target beam) [8]. When GW2 receives the resource request for the aircraft, MAG2 configured in GW2 receives the aircraft's identity. Now knowing the identity of the mobile RCST, MAG2 can then extract the mobile RCST's HNP, LLA, and the IPv6 address of the LMA serving the aircraft (LMA1) from the MNs' policy profile store contained in all MAGs within the domain, as proposed above.

Phase 2: HO Execution: This begins when the aircraft receives the HO command in a mobility control descriptor carried in a terminal information message unicast (TIMu) [8]. Once the aircraft receives this command, it retunes to the target beam and switches to a new link provided by GW2/MAG2. Then MAG2, using the mobile RCST's unique LLA extracted from the policy profile, issues the DHCPOFFER message containing an IPv6 address from the mobile RCST's HNP. When the IP mobility-unaware aircraft sees its home network LLA and IPv6 address (from its HNP), it believes that it is in its home network despite the fact that it is now connected to a foreign network.

The mobile RCST on the aircraft receives its IPv6 address immediately after switching to the target beam, which prevents it from issuing router solicitation messages and thus saving satellite bandwidth resources. Following the DHCPOFFER, MAG2 through the mobile RCST's LLA issues the general MLD query to learn about the multicast group membership status of the newly connected aircraft. In response, the aircraft sends back the MLD report containing all multicast groups that it is subscribed to. Once MAG2 receives the MLD report, it checks its multicast membership table to see whether the requested groups already exist. If they do, then MAG2 simply adds the aircraft to the list of downstream receivers and then informs NCC_B to make the necessary signaling with the OBP and aircraft to ensure that the aircraft receives the multicast traffic.

In Fig. 3 it is assumed that the aircraft is the first member of this multicast group in GW_B2. It also shows the difference in signaling for the DR and MTMA modes.

DR mode: The aircraft being the first member of the group in GW_B2 implies that when MAG2 receives the MLD report from the aircraft, an aggregate MLD report will be issued to MR2 for all multicast group subscriptions required to serve all its downstream interfaces, as illustrated in Fig. 3a.

MTMA mode: It is proposed that a MAG should establish only one multicast tunnel to the MTMA located within its satellite footprint for all its multicast needs. This is very important in this satellite scenario to solve the tunnel convergence problem at the MAGs since mobile RCSTs from different GW_beams having different home MTMAs, and subscribing to the same multicast group can coincidently find themselves under the service area of one MAG. This tunnel could be pre-configured or established dynamically when the MAG subscribed to its first multicast group. In such a situation, when MAG2 receives the MLD report from the aircraft, it will issue an aggregate MLD report to MTMA2 (Fig. 3b) for multicast groups that it has not yet subscribed to.

Comparison Between the Proposed PMIPv6 and Other IP Multicast Receiver Mobility Support Schemes in Satellite Networks

Theoretical Comparison

Table 1 summarizes the comparison of some key features of the other existing IP multicast receiver mobility support schemes with respect to the proposed PMIPv6-based approach within a satellite network. From Table 1 it can be seen that in the PMIPv6-based (i.e. network-based) approaches, the mobile subscriber does not require any software/hardware modification in order to join/leave or receive multicast traffic while away from the home network, unlike in the other schemes where this is required. This is because in the PMIPv6 protocol, MNs do not participate in the IP mobility signaling process. Since in the proposed PMIPv6-based approaches layer signaling during handover is done by fixed network entities (LMA, MAG, MTMA, and MR) and not mobile subscribers, complex security configurations required in MNs during layer 3 handover in host-based IP mobility protocols are completely avoidable in the proposed PMIPv6-based approaches. No software or hardware modification in mobile RCST could lead to a cost reduction in mobile RCST equipment.

Also, owing to the non-participation of the mobile RCST in IP mobility signaling for the PMIPv6-based approaches, no layer 3 signaling takes place over the satellite air interface in a satellite system with a layer 2 OBP capability, unlike in the host-based IP multicast receiver mobility support schemes, as shown in Table 1. IP mobility signaling in PMIPv6-based approaches is done by fixed satellite earth stations, which in most cases are linked by wired terrestrial networks. This implies that satellite bandwidth resources that could have been used by the remote mobile RCST in signaling each time it changes its IP point of attachment to the network from one satellite GW to another in host-based approaches will be saved.

From Table 1, multicast receiver mobility support schemes in which routing of multicast traffic after GWH must pass through the home GW is considered inefficient (due to triangular routing problems). The proposed PMIPv6-based DR, RS-based, and multi-homed-based approaches where traffic can be sent straight to the mobile RCST in the foreign network after GWH without passing through its home GW is considered to be efficient.

It is clear from Table 1 that the advantages of deploying the proposed PMIPv6-based approaches to support multicast receiver mobility during GWH in a multi-beam satellite network far outweigh those of the other existing schemes.

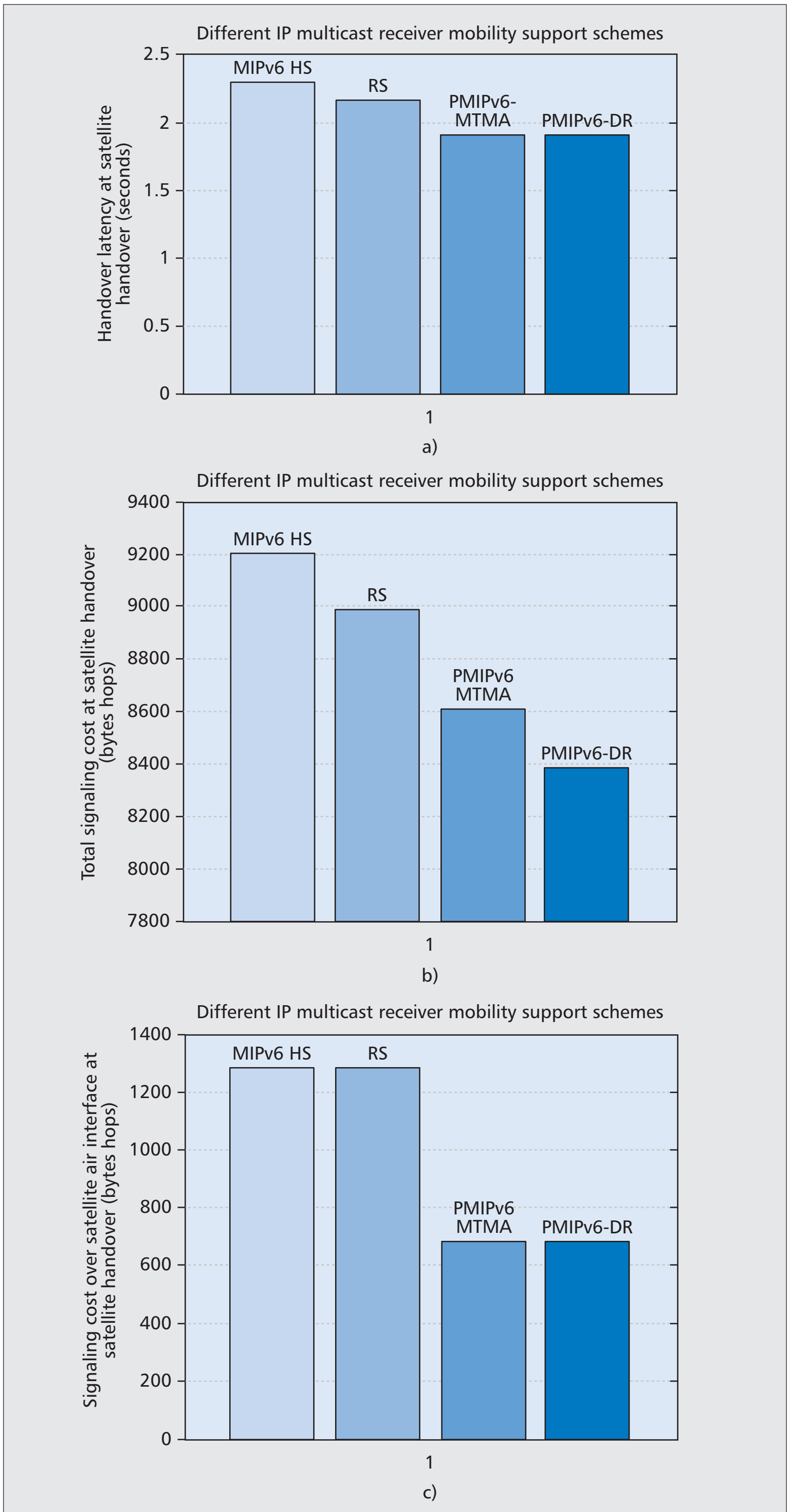

Figure 4. Handover latency and signaling cost during satellite handover: a) Handover latency at SH; b) Total signaling cost at SH; and c) Signaling cost over satellite air interface at SH.

Comparison Using Performance Evaluation During SH

Signaling cost and handover latency are the principal factors in evaluating the performance of any mobility protocol. Signaling cost is defined here as the signaling overhead incurred in supporting a mobile RCST to handover from one GW/satellite to another with minimum disruption of ongoing communications [9]. Handover latency, on the other hand, is the time period during the handover process when the mobile RCST cannot receive or send traffic. This handover latency period for the proposed scheme is highlighted in Fig. 3.

Handover performance analyses of the signaling cost and handover latency using Fig. 3 for the proposed PMIPv6-based approaches are performed. Similarly, implementing the MIPv6 HS-based and RS-based approaches in Fig. 2, the HO signaling cost and latency for these two schemes are also calculated. The results obtained are presented in Fig. 4.

Figure 4a, which compares the handover latency of the four schemes, shows that the handover latencies for the PMIPv6-based approaches are generally lower than those for MIPv6

HS-based and RS-based approcahes. The lower handover latency in the proposed PMIPv6-based approaches occurs because the mobile RCST does not participate in IP mobility signaling during HO. Lower handover latency in the proposed PMIPv6-based approaches implies that fewer multicast packets will be lost during HO.

While Fig. 4b compares the total signaling cost during SH for all four schemes, Fig. 4c compares the signaling cost incurred over the satellite air interface. Figures 4b and 4c show that the PMIPv6-based approaches outperform the MIPv6 HS-based and RS-based approaches in terms of signaling cost because of the efficient and simple HO procedure in the PMIPv6 protocol compared to MIPv6. Less signaling cost over the satellite air interface in PMIPv6-based approaches implies that less satellite bandwidth resources are required to support IP multicast receiver mobility. Considering the cost of satellite bandwidth resources, the implementation of PMIPv6-based approaches will save money.

The drawbacks of the proposed MTMA/DR architecture are:

- The costs and effects of multicast tree reconstruction, assuming the MN is the first member of a multicast group under the service of the new MAG.
- The introduction of a new mobility option in the proxy binding acknowledgement (PBA) from LMA to MAG to support dynamic policies on subscription via MTMA/DR [4].

Conclusion

With the increasing support for IP-based applications over satellite networks and increasing demand for ubiquitous communications, support for IP multicast over mobile satellite terminals is gaining importance. Despite the fact that IP multicast saves satellite bandwidth resources and therefore money for satellite operators and customers, support for global mobile IP multicast communications and dynamic group membership over satellite networks remains a serious problem with no standard solution. This article proposes a PMIPv6-based solution for global satellite-based IP multicast receiver mobility. The paper details the satellite-terrestrial network architecture for the proposed PMIPv6-based support scheme. The proposed solution leverages the advantages of the network-based IP mobility management protocol over the host-based protocols. It was also seen that the proposed PMIPv6-based support schemes outperform the MIPv6 HS-based and RS-based approaches in terms of signaling cost and handover latency for satellite handovers.

Despite the fact that IP multicast saves satellite bandwidth resources and therefore money for satellite operators and customers, support for global mobile IP multicast communications and dynamic group membership over satellite networks still remain a serious problem with no standard solution.

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Biographies

Esua Kinyuy Jaff (E.K.Jaff@student.bradford.ac.uk) received the M.S degree in personal, mobile and satellite communications from the University of Bradford, UK in 2009. He is currently working toward the Ph.D. degree at the School of Engineering and Informatics, University of Bradford. His research interests include IP multicasting, aeronautical/satellite communications, and mobile/wireless communications, especially in the area of designing efficient mobility management support schemes.

Prashant Pillai [M] is a senior lecturer on the faculty of engineering and informatics at the University of Bradford, UK. His main areas of interest are in mobile, wireless, and satellite networks looking into system architecture design, protocol development, RRM, and security. He has worked on projects such as the EU-FP5 WirelessCabin, EU-FP6 SatNEx, EU-FP7 SANDRA, TSB funded SINCBAC and HARNET project. He is a member of IET and a fellow of the HEA.

Yim Fun Hu [SM] is a professor of wireless communications engineering at the University of Bradford. She also holds the Yorkshire Forward Chair in Wireless Communications, established by the Regional Development Agency Yorkshire Forward. She was one of the two UK national delegates to various EU COST Actions, including COST 279, COST253, and COST 256.

Contact Graph Routing in DTN Space Networks: Overview, Enhancements and Performance

Giuseppe Araniti, Nikolaos Bezirgiannidis, Edward Birrane, Igor Bisio, Scott Burleigh, Carlo Caini, Marius Feldmann, Mario Marchese, John Segui, and Kiyohisa Suzuki

Giuseppe Araniti is with the University Mediterranea of Reggio Calabria.

Nikolaos Bezirgiannidis is with the Democritus University of Thrace.

Edward Birrane is with Johns Hopkins University Applied Physics Laboratory.

Igor Bisio and Mario Marchese are with the University of Genoa.

Scott Burleigh is with the California Institute of Technology.

John Segui is with Speedy Packets, Inc. and Pasadena City College.

Carlo Caini is with the University of Bologna.

Marius Feldmann is with Technische Universität Dresden.

Kiyohisa Suzuki is with Japan Aerospace Exploration Agency.

ABSTRACT

Delay- and Disruption Tolerant Networks (DTNs) are based on an overlay protocol and on the store-carry-forward paradigm. In practice, each DTN node can store information for a long time before forwarding it. DTNs are particularly suited to cope with the challenges imposed by the space environment. This paper is focused on routing in space DTNs, and in particular on contact graph routing (CGR) and its most representative enhancements, available in the literature, which are briefly surveyed in this work.

Moreover, the applicability and the obtained performance of the DTN protocol stack and of the CGR have been evaluated by presenting results from real experimental experiences such as the Deep Impact Network experiment (employing the EPOXI space cruise), the JAXA jointly performed space link demonstrations with NASA (where the JAXA's GEO relay satellite called Data Relay Test Satellite has been used), the Space Data Routers European Project, and the pilot operation of a DTN implementation on the International Space Station (ISS).

INTRODUCTION TO SPACE NETWORKS AND DTNS

Every mission into deep space has a communications system to carry commands and other information from Earth to a spacecraft or to a remote planet and to return scientific data to Earth [1]. Communications systems are central to the success of space missions. Large amounts of data need to be transferred (for example, nearly 25 TB in 2013 concerning the Mars Reconnaissance Orbiter (MRO)), and the demand will grow in the future [1] because of the employment of more sophisticated instruments that will generate more data. This will require the availability of high network transfer rates. Satellite systems already have to cope with difficult communication challenges: long round trip times (RTTs); the likelihood of data loss due to errors on the communication link; possible channel disruptions; and coverage issues at high latitudes and in challenging terrain. These problems are magnified in space communications characterized by huge distances among network nodes, which imply extremely long delays and intermittent connectivity. At the same time, a space communications system must be reliable over time due to the long duration of space missions. Moreover, the importance of enabling Internet-like communications with space vehicles is increasing, realizing the concept of extended Future Internet, an IP (Internet Protocol) pervasive network of networks including interplanetary communication [2], where a wide variety of science information values are acquired through sensors and transmitted.

The Delay- and Disruption Tolerant Network (DTN) architecture [3] introduces an overlay protocol that interfaces with either the transport layer or lower layers. Each node of the DTN architecture can store information for a long time before forwarding it. Thanks to these features, a DTN is particularly suited to cope with the challenges imposed by space communication. As summarized in [4], the origin of the DTN concept lies in a generalization of requirements identified for interplanetary networking (IPN), where latencies that may reach the order of tens of minutes, as well as limited and highly asymmetric bandwidth, must be faced.

However, other scenarios in planetary networking, called "challenged networks," such as military tactical networking, sparse sensor networks, and networking in developing or otherwise communications-challenged regions, can also benefit from the DTN solution. Delays and disruptions can be handled at each DTN hop in a path between a sender and a destination. Nodes on the path can provide the storage necessary for data in transit before forwarding it to the next node on the path. In consequence, the contemporaneous end-to-end connectivity that Transmission Control Protocol (TCP) and other standard Internet transport protocols require in order to reliably transfer application data is not required.

In practice, in standard TCP/IP networks,

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which assume continuous connectivity and short delays, routers perform non-persistent (short-term) storage and information is persistently stored only at end nodes. In DTN networks, information is persistently (long-term) stored at intermediate DTN nodes. This makes DTNs much more robust against disruptions, disconnections, and node failures.

The Bundle Protocol (BP) [5] is a key element of the DTN architecture, where the basic unit to transfer data is a bundle, a message that carries application layer protocol data units, sender and destination names, and any additional data required for end-to-end delivery. The BP can interface with different lower layer protocols through convergence layer adapters (CLAs). CLAs for TCP, UDP, Licklider Transmission Protocol (LTP), Bluetooth, and raw Ethernet have been defined. Each DTN node can use the best suited CLA for the forwarding operation.

BP provides useful features such as:

- Custody transfer, where an intermediate node can take custody (i.e. responsibility) of a bundle, relieving the original sender of the bundle which might never have the opportunity to retransmit the application data due to limited resources.
- Proactive and reactive bundle fragmentation, the former to tackle intermittent periodic connectivity when the amount of data that can be transferred is known a priori, the latter, which works a posteriori, when disruptions interrupt an ongoing bundle transfer.
- Late binding, where, for example, when a bundle destination endpoint's identifier includes a dynamic name server (DNS) name, only the CLA for the final DTN hop might have to resolve that DNS name to an IP address, while routing for earlier hops can be purely name based.

Routing is a critical problem in DTN networks. Quoting from [6], "*the routing objective of traditional routing schemes has been to select a path which minimizes some simple metric (e.g. the number of hops). For DTN networks, however, the most desirable objective is not immediately obvious.*" Nodes are not constantly connected. Storage and energy management affect DTN routing. A possible aim may be increasing the probability of bundle delivery, but also reducing the delivery delay may be important. Routing over DTN networks deserves close attention and is the object of the next section. A suitable solution for space networks is represented by Contact Graph Routing (CGR) [7], where each node on the path computes a route from itself to the bundle destination based on a computed graph. We include a brief tutorial on basic CGR, highlight CGR issues and enhancements, and summarize CGR performance over space networks. Finally, the conclusions are drawn.

Routing in DTN Space Networks

Routing and Forwarding in DTN vs the Internet

Given the aforementioned challenges of space communication, it is not surprising that the methods used for computing routes in a space network should be different from those used in Internet routing. To aid in explaining these differences, it may be helpful to return briefly to first principles.

In general, we might say that *routing* is the procedure by which we select the best path for conveying data from source node A to destination node Q in a network. Routing would be trivial if every node could simply transmit directly to every other, but for large networks this is not possible. In recognition of this complexity, a network host plans a route for a data item before issuing it. The network state information on which this planning is based includes the network's "topology" and a list of all known connections between nodes. In a DTN-based network, this list may include additional information such as the speed of each connection and perhaps the storage capacity of each node.

However, network state information may change over time while traffic is traversing the network, and therefore the most efficient route may change while data is en route. For this reason, routing may occur at every branch point to take advantage of newly available information, and consequently it is more accurate to say that *routing* is the procedure by which, at each point in the path from A to Q, we select a neighboring branch point to transmit the data to, believing that branch point to be on the best path for conveying the data to its destination. To make this selection, we may compute a new route based on the network state information currently available at this point or we may simply continue along the path previously computed by another node.

In the Internet this selection can be done with high confidence because information about changes in network state information can be propagated so quickly that each node's current understanding of the state of the network is almost always correct. That understanding may be incomplete, because routing in the network may be compartmentalized: the network state information exposed to any node may be limited to nodes in the local "domain" (including nodes that are on the border between the local domain and adjacent domains that serve as "gateways" between domains). Nonetheless, routing decisions can be made confidently in the expectation that the distribution of network state information within other domains is as rapid and comprehensive as within the local domain. Each node is continuously connected to a small number of neighboring nodes; routing is simply a matter of choosing the neighboring node that's on what seems to be the best path.

In a space network, or in one of the previously mentioned "challenged" networks where DTNs are applied, this is not true: since connectivity is intermittent and/or signal propagation times are long, changes in the network state may occur more rapidly than information about those changes can be propagated. Routing is still a matter of choosing a neighboring node to transmit directly to, but determination of the best path is constrained by lack of knowledge of the current state of the network, and it may not be possible to transmit immediately to the neighboring node that is the nearest branch point on the best path.

Given the aforementioned challenges of space communication, it is not surprising that the methods used for computing routes in a space network should be different from those used in Internet routing. To aid in explaining these differences, it may be helpful to return briefly to first principles.

CGR is a dynamic algorithm that computes routes through a time-varying topology of scheduled communication contacts in a DTN network. It can be successfully applied not only to an Interplanetary Internet, but also to LEO satellite communications, as in both cases link availability is known a priori.

Survey of Current Work

Strategies for dealing with these obstacles have been the focus of most DTN research for longer than a decade. A key discriminator among these strategies is the assumed timeliness and accuracy of the network state information available to every node in the network. Several surveys of DTN routing schemes have been conducted, and a hierarchy of DTN routing approaches, ranging from those with zero configuration information to those with perfect knowledge of the network, has been defined. Approaches that assume minimal accurate network state information have historically been considered "opportunistic" while those that assume complete network state information are regarded as "deterministic."

Significant algorithms belonging to the category of opportunistic approaches include single-hop multi-cast forwarding (Spray and Wait), in-network exchange of link information (DTLSR), and probabilistic analysis of predicted node contact (PRoPHET). All of these rely on the exchange of infrastructure and/or in-network measurements in a timely manner to support on-demand calculations of routes and forwarding hops. Opportunistic approaches often apply a replication-based (alternatively, "flooding-based") strategy. Using this strategy, messages are typically duplicated either a fixed number of times or else a variable number of times based on contact probability. In networks with high node mobility and nearly random contact establishment, the delivery success rate of this class of approaches is higher than approaches that rely on the accuracy of current network state information.

On the other hand, in networks where contacts are predictable, the more deterministic algorithms can achieve high rates of delivery success with less waste of bandwidth and buffer space. Algorithms such as MARVIN and Contact Graph Routing [7] belong to this second category. Accurate contact predictions are distributed to the nodes in the network, enabling network graphs to be built and used to make routing decisions on a hop-by-hop basis. MARVIN encodes information about the operational environment (planetary ephemeris data) and infers contact opportunities from this knowledge. Similarly, the numerous MANET routing approaches also base their operation on evolving graphs. The Contact Graph Routing (CGR) algorithm is a formulation of the perfect knowledge approach, and is currently being extended to work in less-perfect knowledge systems. CGR is discussed in more detail in the next section.

Contact Graph Routing (CGR)

CGR is a dynamic algorithm that computes routes through a time-varying topology of scheduled communication contacts in a DTN network. It can be successfully applied not only to an Interplanetary Internet, but also to LEO satellite communications, as in both cases link availability is known a priori. However, this perfect knowledge does not reduce the complexity of the route computations, as CGR must consider that links among nodes in the network change over time. For an exhaustive explanation we refer the reader to the CGR section of the ION Design Guide (the Interplanetary Overlay Network (ION) implementation of DTN, including the Design Guide, is available at https://sourceforge.net/projects/ion-dtn/) or to the CGR Internet Draft [7]. Here we provide only a few key points of CGR's functionality.

The basic strategy of CGR is to take advantage of the fact that, since space flight communication operations are planned in detail by mission operators, the communication routes between any pair of "bundle agents" in a population of nodes, all of which have been informed of one another's plans, can be inferred from those plans rather than discovered via dialogue.

The foundation of contact graph routing is the "contact plan," a time-ordered list of scheduled, anticipated changes in the topology of the network. The entries in this list are termed "contacts"; each one is an assertion that transmission from node X to node Y at nominal data rate R will begin at time T1 and will end at time T2. Note that this assertion implicitly also defines the "volume" (or "capacity") of the contact, which is the maximum amount of data that can be transferred during the contact, given by the product of contact length (T2 – T1) and nominal transmission rate R.

Each node uses the contacts in the contact plan to build a "routing table" data structure. A routing table is a list of "route lists," one route list for every possible destination node in the network. Each route in the route list for node D identifies a path to destination node D, from the local node, that begins with transmission to one of the local node's neighbors in the network, the initial receiving node for the route, termed the route's "entry node." The route list entry for each neighbor contains the best route that begins with transmission to that neighbor. Also noted for each route are:

- All of the other contacts that constitute the remaining segments of the route's end-to-end path.
- The estimated "cost" of this route, e.g. the end-to-end delivery latency.
- The "forfeit time" for this route, i.e. the latest time by which the bundle must have been forwarded to the route's entry node in order to have any chance of traversing this route.

To compute a new route list for node D:

- We construct an abstract contact graph, a directed acyclic graph whose root is a notional contact from the local node to itself and whose other vertices are all other contacts that can contribute to some end-to-end path to D. A terminal vertex is also included in the graph, constituting a notional contact from node D to itself.
- We perform a series of Dijkstra searches within this graph. On each search we find the lowest-cost route that begins at the root of the graph and ends at the terminal vertex. Each time a route is computed, we add it to the node's list of routes and then remove that route's initial contact from the contact graph before searching for the next best route. The search series is terminated as soon as a search fails to find a route.

Note that the routes in the route list need not be continuous. Each segment of the path is an opportunity to send data from node X to node Y; once a bundle has reached node Y it may well reside in storage at node Y for some length of time, awaiting the start of the opportunity to be forwarded from node Y to node Z, and so on.

So when a bundle must be transmitted from node A to node Q we consult the route list for node Q. Some of the routes in the list may be unusable. For example, a route may be temporarily unavailable (transmission to the entry node is "blocked" due to a detected or asserted loss of connectivity); or the best-case delivery time on a route may be greater than the bundle's time-to-live (the bundle would be purged before delivery); or the "residual capacity" of the initial contact on the route (the capacity that has not been allocated yet to higher-priority bundles) may not be enough to contain the bundle. Note that this latter check is a form of embryonic congestion control: a route is considered unusable if its first contact is already fully subscribed, causing the bundle to be redirected to less congested routes.

Of the usable routes, we choose the one with the lowest cost and queue the bundle for transmission to that route's entry node. If the list of bundles queued for transmission on some route is non-empty at the time that route's forfeit time is reached, new routes must be computed for all of those bundles.

The key advantage of CGR is that, like Internet routing, it can be done with high confidence, as it is based on accurate information about the network's topology. The difference is that:

- The topology on which routing is based is not the currently known current topology but rather an anticipated time-varying topology.
- Since changes in the network's topology are scheduled in the course of mission planning, information about those changes can be propagated long before they occur. Just as in the Internet, each node's understanding of the topology of the network at any moment is almost always correct: while propagation of information about network topology changes is slow, it is still "faster" than the rate at which the changes themselves occur.

So again, routing is a matter of choosing a neighboring node to transmit directly to. Again, it may currently be impossible to transmit to the neighboring node that is the nearest branch point on the best path, but at least determination of the best path is possible because topology knowledge is generally accurate.

CGR Issues and Enhancements

Ever since CGR first appeared, the research community has worked on improving its functionality and usage. For instance, path selection with Dijkstra's algorithm, proposed as "Enhanced CGR," has now become part of the core CGR functionality. Research activity on CGR is still very active and further enhancements have been proposed to cope with residual issues. A short list of the most representative is presented below, divided into short-term modifications to the algorithm and long-term prospects for CGR evolution.

Short Term Evolution

In route computation classical CGR assumes that bundles will be sent at the contact start time or, if the contact is currently in progress, immediately. That is, it does not consider the queueing delay caused by other bundles in the outbound buffer waiting for transmission. For this reason, a modified version of the CGR algorithm, namely CGR-ETO, was introduced in [8] to incorporate the available queue length information. CGR-ETO utilizes the earliest transmission opportunity (ETO) contact parameter, the earliest plausible time that a bundle of a specific priority can be forwarded during this contact, replacing contact start time with ETO during contact graph traversals. Queue length information can be easily obtained at the local node and is updated upon bundle routing. Obtaining useful (i.e. not obsolete) queue length information from other nodes is challenging and requires the transmission of update messages, e.g. using the Contact Plan Update Protocol (CPUP) [8].

A bundle may be assigned to a route that is already fully subscribed, provided that the bundle's priority is higher than that of some of the bundles currently assigned to that route. For this reason CGR does not take into account bundles of lower priority in the "residual volume" computation check. The contact oversubscription that derives from this policy is informally called contact "overbooking." The aim of the overbooking management adaptation is to mitigate as much as possible the consequences of this contact oversubscription.

In an overbooking example of a future contact, some low priority bundles put in the queue to proximate node X will miss their contact, to accommodate higher priority bundles. This situation is tackled by standard CGR a posteriori, by re-forwarding the "bumped" bundles once their forfeit time expires (usually at the overbooked contact's end-time). This handling, although robust, is not efficient. By contrast, overbooking management acts a priori, by re-forwarding as soon as possible any bundles that are destined to miss the contact, i.e. immediately after forwarding the higher priority bundle that has caused the oversubscription. Results presented in [9] show that overbooking management and CGR-ETO are complementary and effective in improving routing decisions.

Long Term Evolution

Path Encoding CGR Extension — The standard CGR model computes a feasible path through the network and uses that to select the most appropriate next step in the routing process. The Path Encoding CGR extension takes that calculated path and attaches it to the message. Downstream nodes may then merely verify the continued feasibility of the encoded path rather than calculate a new path from scratch at every hop in the network [10]. This approach yields four benefits. First, paths are "re-used" as long as they are verified against local knowledge at downstream nodes, thereby avoiding a complex route calculation at every hop in the network. This is a particularly important optimization when implementing routing decisions on resource-

So again routing is a matter of choosing a neighboring node to transmit directly to. Again it may currently be impossible to transmit to the neighboring node that is the nearest branch point on the best path, but at least determination of the best path is possible because topology knowledge is generally accurate.

Due to its flexibility, CGR can be enhanced in order to be applied not only to deterministic but also to opportunistic scenarios. This would allow its application as core routing in large-scale DTN deployments with various, heterogeneous contact types.

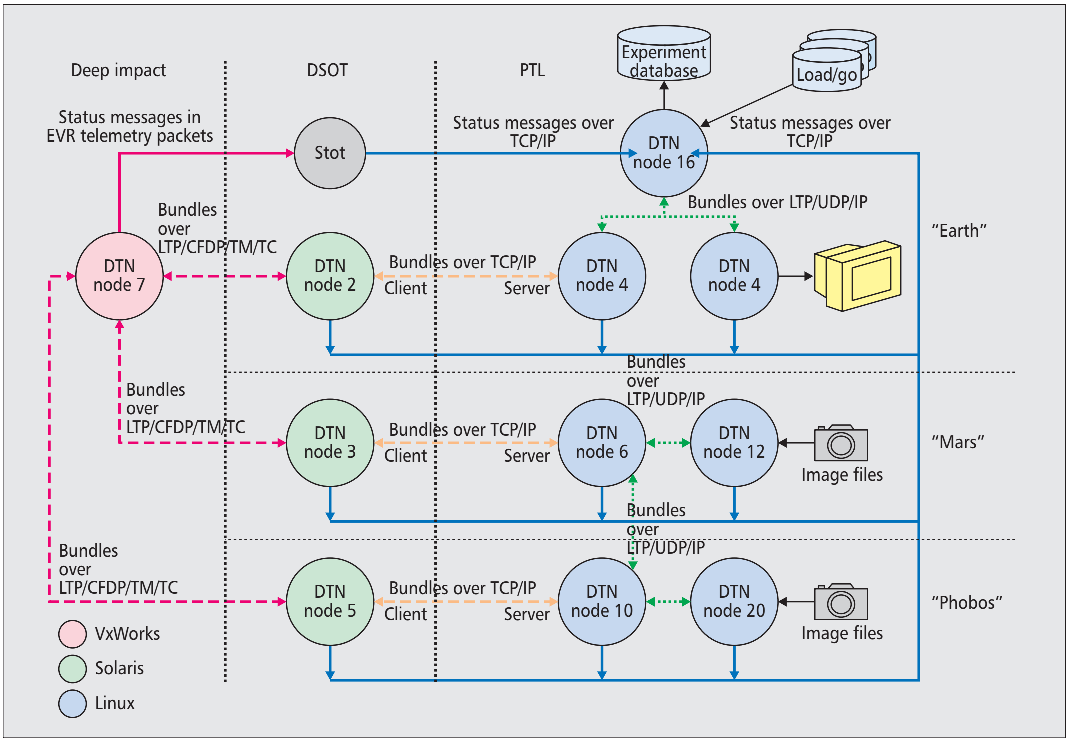

Figure 1. DINET network topology.

constrained flight processors. Second, an encoded path simply needs to remain feasible to be validated, even if a potentially better path could be recalculated. Honoring pre-computed but potentially suboptimal paths provides a natural damping function that resists routing loops in networks undergoing topological changes or congestion. Third, supporting feasible-versus-optimal allows the use of novel cost functions for route selection algorithms that can optimize network utilization rather than individual message delivery cost, which is an important balance in space-based sensing constellations. Fourth, sending path information with a message provides meta-data exploitable for new research such as path-based congestion prediction and topological synchronization.

Opportunistic CGR Extension — Due to its flexibility, CGR can be enhanced in order to be applied not only to deterministic scenarios but also to opportunistic scenarios. This would allow its application as core routing in large-scale DTN deployments with various, heterogeneous contact types, such as scenarios including both terrestrial and space nodes, thus leading to a unified DTN routing approach.

To this end, a "divide and conquer" strategy can be envisaged, where the overarching routing mechanism follows a hybrid (deterministic/opportunistic) approach. The overall network is decomposed into different *regions* where either standard CGR or a modified version able to forward bundles in a probabilistic way is used, depending on the link characteristics (deterministic/opportunistic) of the region. By contrast, regions are always interconnected by standard CGR, as their connectivity is reasonably assumed deterministic.

A CGR evolution proposal to tackle opportunistic forwarding consists of the following steps: first, the contact plan is extended to include contacts with a probability of occurrence lower than 1; second, routes to a destination are calculated as before, thus ignoring the contact probabilities; finally, copies of a message are forwarded to the entry nodes of all opportunistically discovered routes that increase the message's aggregate delivery probability by more than a given threshold. The algorithm is designed to throttle back the number of copies automatically as the aggregate expectation of delivery success on the selected routes increases.

After this short overview of ongoing research, we will address selected experiments that made use of CGR.

Experiments over Space Networks

This section briefly describes four experiences in real space networks carried out by space agencies or within the framework of international projects. These experiences aim at investigating, in general, the effectiveness of the DTN paradigm over operational space networks and, in particular, the effectiveness of the CGR algorithm.

During the DINET experience, NASA performed a first attempt to test their DTN implementation over a real system. The transmissions of photos from remote planets, detailed in the following paragraph, were successfully completed and CGR performed quite satisfactorily. The DRTS DTN project, recently completed by NASA and JAXA, confirmed that it is feasible to use DTN with CGR in real spacecraft operations. The Space-Data Routers project, a European Commission funded initiative, demonstrated that CGR can contribute to efficient data dissemination. Finally, for the sake of completeness, NASA's experience with the ION implementation on the International Space Station (ISS) is described. While this topology was too simple to evaluate the effectiveness of CGR in a large network, the obtained results highlighted important benefits obtained by the application of the DTN paradigm, as listed in detail below.

The Deep Impact Network Experiment

The Deep Impact Network (DINET) project was an experimental validation of "ION" (Interplanetary Overlay Network), JPL's implementation of the DTN protocols. The ION software, including the first implementation of Contact Graph Routing, was uploaded to the backup flight computer of the EPOXI (formerly Deep Impact) spacecraft on 18 October 2008, and was operated continuously from that date until 13 November 2008.

EPOXI was at that time in an inactive cruise period while en route to encounter comet Hartley 2 (in November 2010). The one-way signal propagation time from EPOXI to Earth was initially 81 seconds, dropping to 49 seconds by the end of the four-week exercise. The spacecraft was between 9.1 million and 15.1 million miles from Earth during the experiment.

Uploading the ION software to EPOXI enabled the spacecraft to function as a DTN router in an 11-node network (Fig. 1).

The spacecraft was assigned node number 7 for this exercise and was the only node of the network that was not physically resident at the Jet Propulsion Laboratory (Pasadena, California). Nodes 2, 4, 8, and 16 played the role of "Earth" in the experiment; nodes 3, 6, and 12 functioned as a notional "Mars"; nodes 5, 10, and 20 impersonated "Phobos." The Mars and Phobos nodes simulated the acquisition of images and the transmission of those images back to Earth via the EPOXI spacecraft, which acted as a relay router in space. Each dashed line in the topology diagram represents a sequence of DTN network contacts; the solid blue lines indicate the out-of-band local area network used to instrument the experiment. Note that the topology of the DINET network included cross-links between nodes 6 and 10 (Mars and Phobos), providing alternative paths for data to and from nodes 12 and 20; that is, the DINET network was not a simple tree, so the CGR route selection decisions made at nodes 6 and 10 were non-trivial.

Over the course of the four weeks of flight testing, the DTN software reliably conveyed 292 images (about 14.5 MB) through the network, together with command traffic from the Earth nodes to the Mars and Phobos nodes. No data were lost or corrupted anywhere in the network, and ground station handovers and transient failures in the Deep Space Network uplink service were handled automatically and invisibly. CGR generally performed well, but several bugs in the initial ION implementation resulted in some under-utilization of network capacity. Those bugs were addressed in later versions of ION.

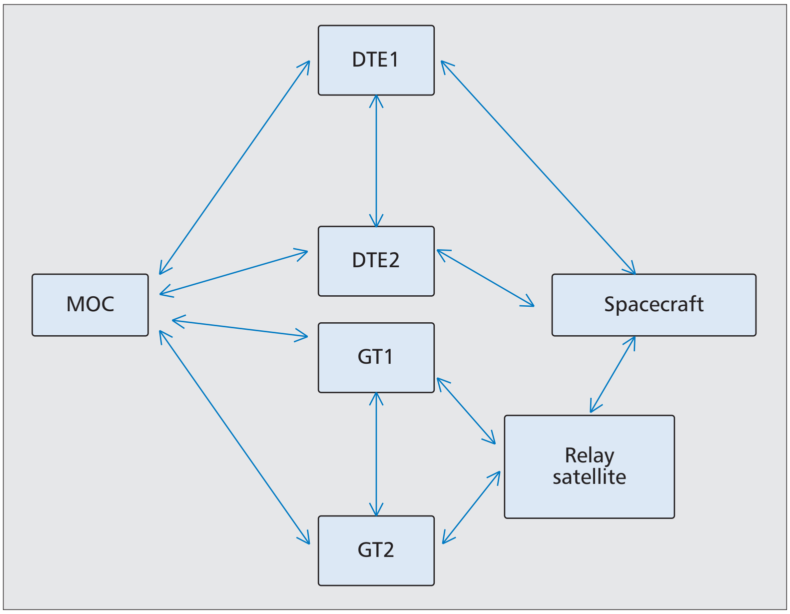

Figure 2. Topology of the JAXA DRTS space experiments.

JAXA DRTS Testing

For the purpose of studying the feasibility of autonomous routing and high-integrity data forwarding in existing and future anticipated space network architectures, JAXA jointly performed a series of experimental tests with NASA in 2012-2013 to evaluate the DTN architecture and CGR [11].

JAXA's GEO relay satellite "Data Relay Test Satellite (DRTS)" and its tracking stations were used in this measurement campaign. The data relay space link is referred to as the "inter-satellite link" in the following discussion. In the tests, ION was used on all nodes to evaluate the performance of BP, LTP, and CGR. Several network topologies were investigated, including direct connectivity between a LEO spacecraft and a ground network and relayed communications between a remote planetary surface and earth's surface connected via a relay spacecraft.

The topology considered here, by contrast, is typical of an earth observation mission. It consists of the following seven DTN nodes shown in Fig. 2: one mission operation center (MOC); one LEO spacecraft; one GEO satellite acting as a relay; two ground terminals in between the GEO satellite and the MOC (GT1 and GT2); and two direct to earth ground stations (DTE1 and DTE2) in between the LEO and the MOC. Data generated on board the LEO satellite can reach the MOC either passing through the GEO relay (and then either via GT1 or GT2) or via DTE1 or DTE2. The task of CGR is to dynamically find the best route, taking into account link intermittency.

The test conditions included the actual signal propagation and processing delay over the DRTS's inter-satellite link and link intermittency, either scheduled (e.g. due to orbital mechanics) or random (e.g. due to space link failures). Contact plans were developed based on the actual resource allocation plan for the inter-satellite link. The obtained results are presented

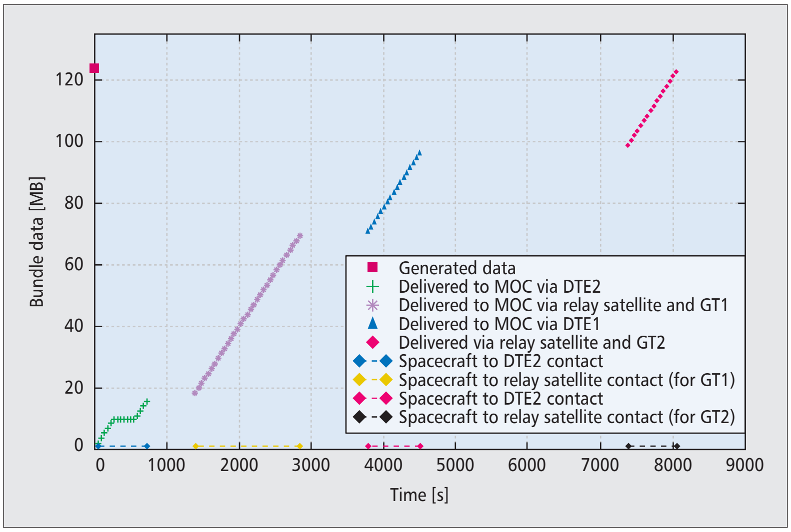

Figure 3. An example of data transfer from a LEO spacecraft and the MOC, via alternative routes dynamically selected by CGR on the basis of the contact plan. All data are routed in accord with contact information, as expected.

in Fig. 3, which shows that all data were automatically transferred to the MOC via four routes, selected by CGR conforming to the contact plan, as expected. These results attest to the feasibility of using DTN with CGR for autonomous routing and reliable data forwarding in real spacecraft operations.

SPACE DATA ROUTERS

The applicability of CGR in the ground segments of space missions was evaluated in "Space-Data Routers" (www.spacedatarouters.eu), a European FP7 project that exploited the DTN architecture to improve the dissemination of space mission data with respect to volume, timeliness, and continuity. More details on that project can be found in [12].

The vision of Space-Data Routers (Figure 4) is to forward data from space missions upon reception, whenever possible, directly to the interested parties (e.g. scientists, research institutes, etc.), utilizing a DTN overlay and applying CGR for routing decisions. The implemented version of CGR applies policy-based forwarding as an alternative to minimizing delivery latency. The contact plan and the forwarding criteria include the level-of-trust of each network node (for confidential data), the storage availability at each node, and a cost rate, enabling data to be forwarded over the lowest-cost route. Evaluation of the Space-Data Routers showed, among other conclusions, that CGR contributes to more efficient space data dissemination and has the potential to administer data confidentiality as well.

OPERATIONS ON THE INTERNATIONAL SPACE STATION.

Pilot operation of the ION implementation of DTN on the International Space Station (ISS) officially began in July 2009: ION was installed in two commercial generic bioprocessing apparatus (CGBA) computers on ISS, where it was used to transmit science experiment telemetry to an experiment center at the University of Colorado (Boulder) continuously, via the Huntsville Operations Support Center at NASA's Marshall Space Flight Center in Alabama, for the next four years. The CGBA deployment on ISS was not topologically complex enough to fully exercise ION's implementation of CGR. However, the success of that pilot deployment convinced the ISS operations team that DTN would be a valuable permanent addition to the networking software infrastructure of the space station. Accordingly, two institutional DTN gateway nodes will be installed on ISS in 2015, serving both ISS operations and payload communications. The topology of the ISS DTN backbone will be as shown in Figure 5, including multiple potential cross-links in end-to-end paths.

The implementation of the DTN stack on the space station provides a variety of benefits, listed below:

- Enables payload developers (PDs) to automate operations and ensure science delivery with little regard for link or facility outages.
- Reduces the need for PD real-time support to access and downlink science data:
  –DTN stores data during loss of signal (LOS) and automatically initiates transfer upon acquisition of signal (AOS).
  –A download transfer can span Ku-Band AOS periods without any special scheduling or scripting.
  –Reduces need for duplicate storage and extra retrieval actions.
- Reliable data transfer for ISS during LOS/AOS cycles:
  –Automatic verification of bundle receipts, retransmissions reduced.
  –When transmission errors occur, only the bundles that have errors are retransmitted, reducing the overall amount of retransmitted data, thus maximizing use of bandwidth.
- Allows PDs to use DTN protocols for their own applications (streaming, telemetry, etc.).
- Efficient use of downlink stream through DTN Quality of Service (QoS)/prioritization.
- Tolerance for high network latency (600ms delay is typical on JSL links).

REMARKS

While the benefits noted in these experiments were provided only to satellite and space networks, the potential advantage in other networks where the DTN paradigm can be applied should be clear. In particular, the reliability guaranteed by DTNs can be very useful in mobile ad hoc networks (MANET) and, similarly in wireless sensor networks (WSN).

CONCLUSIONS AND FUTURE DEVELOPMENTS

Routing in Delay-Tolerant Networks is a challenging problem, but practical solutions that apply principles underlying the design of the Internet are emerging. These solutions seem to offer the potential to support end-to-end data exchanges spanning a very wide range of communication environments.

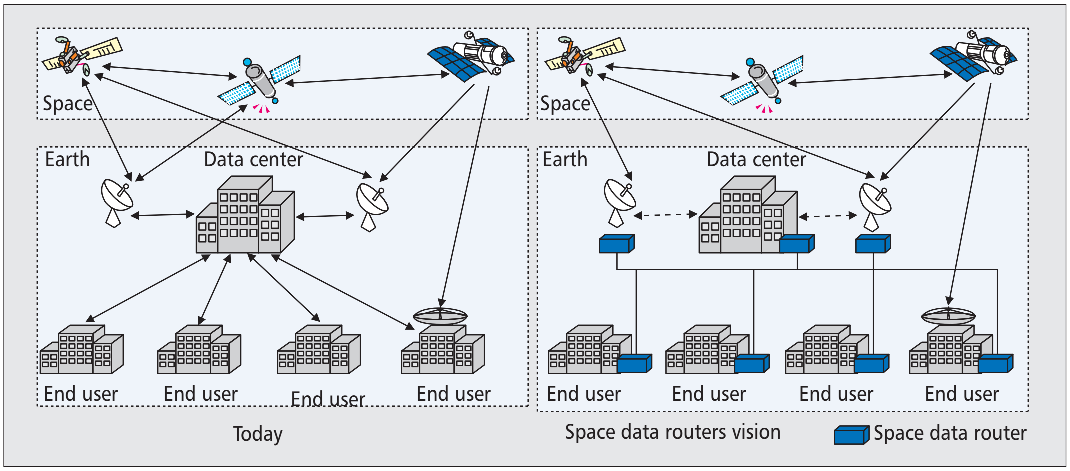

Figure 4. Space data dissemination today and as envisaged by the space data routers project [12].

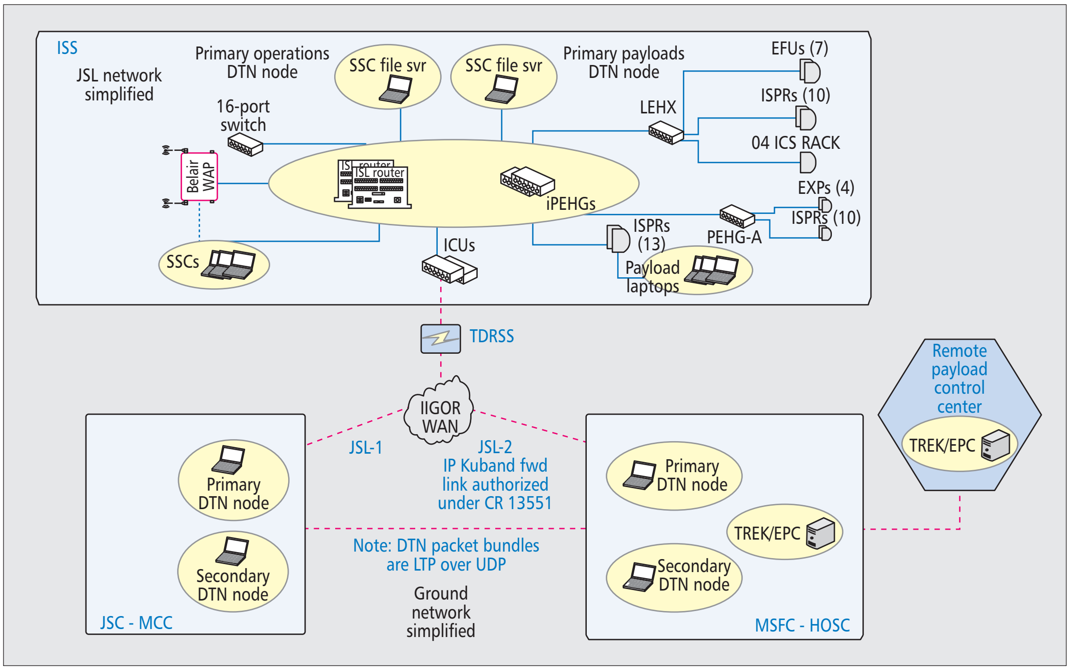

Figure 5. DTN on the international space station.

The applicability of CGR in the Ground segments of space missions was evaluated in "Space-Data Routers" (www.space-datarouters.eu), a European FP7 project that exploited the DTN architecture to improve the dissemination of Space mission data with respect to volume, timeliness, and continuity.

Experience gained in current and future deployments of Contact Graph Routing is vital to the future development of this technology. As noted earlier, future developments of CGR include the "Path Encoding CGR" and "Opportunistic CGR" approaches. In the former case the selected path within the space networks is memorized for validation and re-use by resource-constrained downstream nodes, while in the latter case a decomposition of the network is performed so as to apply a "divide and conquer" strategy.

Another point to be carefully addressed in the future development of CGR as well as other routing/forwarding approaches is processing efficiency. CGR was built for space exploration networks with scheduled communication opportunities, represented as a contact graph. Since CGR uses knowledge of future connectivity, the contact graph can grow rather large. Efficient processing approaches will be required to enable CGR to scale to anticipated future space network complexities.

Acknowledgments

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Routing in delay-tolerant networks is a challenging problem, but practical solutions that apply principles underlying the design of the Internet are emerging. These solutions seem to offer the potential to support end-to-end data exchanges spanning a very wide range of communication environments.

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Biographies

GIUSEPPE ARANITI (araniti@unirc.it) is an assistant professor of telecommunications at the University Mediterranea of Reggio Calabria, Italy. From the same University he received the Laurea (2000) and the Ph.D. degree (2004) in electronic engineering. His major area of research includes enhanced wireless and satellite systems, traffic and radio resource management, multicast and broadcast services, device-to-device and machine type communications over 4G/5G cellular networks.

NIKOLAOS BEZIRGIANNIDIS (nbezirgi@ee.duth.gr) received a diploma in electrical and computer engineering from Aristotle University of Thessaloniki, Greece in 2006. He also received his M.Sc. degree in the scientific area of computer networks in the Department of Electrical and Computer Engineering, Democritus University of Thrace, Greece, in 2010. He is currently a Ph.D. candidate under advisor Prof. Vassilis Tsaoussidis in the same department.

EDWARD BIRRANE (Edward.Birrane@jhuapl.edu) is a senior embedded software engineer, principal investigator, and technical lead in a range of areas including embedded terrestrial telecommunications, space flight software, bioengineering, and national security space at the Johns Hopkins University Applied Physics Laboratory.

IGOR BISIO (igor.bisio@unige.it) received his "Laurea" degree in telecommunication engineering at the University of Genoa, Italy in 2002, and his Ph.D. degree in 2006. He is currently an assistant professor at the University of Genoa. He is the current Chair of the Satellite and Space Communications Technical Committe of the IEEE Communications Society. His main research activities concern resource allocation and management for satellite and space communication systems and signal processing over portable devices.

SCOTT BURLEIGH (Scott.Burleigh@jpl.nasa.gov) has been developing software at JPL since 1986. As a member of the Delay-Tolerant Networking (DTN) Research Group of the Internet Research Task Force, he co-authored the specifications for the DTN Bundle Protocol (RFC 5050) and the DTN Licklider Transmission Protocol for delay-tolerant ARQ (RFC 5326). He has developed implementations of these protocols that are designed to enable deployment of a delay-tolerant Solar System Internet.

CARLO CAINI (carlo.caini@unibo.it) received the master degree in electrical engineering "summa cum laude" from the University of Bologna in 1986. At present he is with the Department of Electrical, Electronic and Information Engineering (DEI) of the same University, as associate professor of telecommunications. His main research interest concerns satellite networking, transport protocols and delay tolerant networks.

MARIUS FELDMANN (marius.feldmann@tu-dresden.de) received his Ph.D. degree from the Technische Universität Dresden in 2011. He has worked for several national and international research projects, including a FP7 project funded by the European Union. His current professional focus lies upon entrepreneurship and he works as a lecturer in the domain of computer networks. His main research interests cover delay-tolerant networking as well as the history of Unix-like operating systems.

MARIO MARCHESE (mario.marchese@unige.it) received the "Laurea" degree (cum laude) in electronic engineering and the Ph.D. degree in telecommunications from the University of Genoa, Genoa, Italy, in 1992 and 1996, respectively. He is currently an associate professor with the DITEN Department, University of Genoa. His main research activity concerns satellite and radio networks, transport layer over satellite and wireless networks, and quality of service.

JOHN SEGUI (John.Segui@jpl.nasa.gov) is a senior technologist at Speedy Packets, Inc., and an instructor at Pasadena City College, CA, USA. He had been with the Jet Propulsion Laboratory and Riverside Community College. He received his bachelor degree in information and computer sciences at the University of California Irvine, and his master degree in computer science at the University of Southern California. His main research interest concerns are satellite networking and delay tolerant networks.

KIYOHISA SUZUKI (suzuki.kiyohisa@jaxa.jp) is currently an engineer in the Department of Consolidated Space Tracking and Data Acquisition of the Japan Aerospace Exploration Agency (JAXA), Japan. He is an active member of the Consultative Committee for Space Data Systems (CCSDS). His main research interest is in satellite networking, delay tolerant networks, and interoperability issues.

Integration of Satellite and LTE for Disaster Recovery

Maurizio Casoni, Carlo Augusto Grazia, Martin Klapez, Natale Patriciello, A. Amditis, and E. Sdongos

ABSTRACT

Wireless communications are critical for public protection and disaster relief (PPDR) professionals during the emergency operations that follow natural or man-made disasters, scenarios in which both commercial and dedicated terrestrial networks often fail to provide the necessary support. The reason is threefold: they simply get destroyed by the disaster, they cannot sustain the sudden surge of network demand or they fail to deliver the necessary bandwidth and/or other QoS guarantees. Because LTE is expected to become the main wireless technology for broadband communication, a lot of studies have been devoted to assess its compliance for PPDR purposes and to find suitable architectural solutions able to meet mission-critical requirements. This approach is surely worthy, but it is based on the assumption that infrastructure-based terrestrial systems are reliable. As a consequence, in worst-case emergency scenarios appropriate guarantees can be provided only in the hypothesis of huge investment costs. Recent developments in satellite technologies are bringing the availability of non-terrestrial high performance channels, with better properties when comparing to LTE for what regards availability and reliability. On this basis, the paper proposes a network architecture based on the integration of satellite and LTE networks for both infrastructure-based and infrastructure-less scenarios. The proposal aims to provide field operators and people in distress with transparent accessibility, coverage guarantees and broadband performance when terrestrial infrastructures are lacking, and to expand their coverage, capacity and resilience otherwise.

Maurizio Casoni, Carlo Augusto Grazia, Martin Klapez, and Natale Patriciello are with University of Modena and Reggio Emilia.

A. Amditis and E. Sdongos are with Institute of Communication and Computer Systems (ICCS).

INTRODUCTION

Current technologies employed for PPDR purposes provide a rich set of voice-centric services that are of paramount importance for field operators, especially in the very early stages of the response; unfortunately, these systems are unable to sustain high-bandwidth data-oriented applications, for which there is an increasing demand from the PPDR community. Furthermore, due to the generalized lack of a common PPDR communication infrastructure between different PPDR entities (e.g. police corps, fire departments, ambulance services), operators often rely on commercial terrestrial networks for coordination and data-oriented communication [1].

To provide field operators with reliable, high performance communication channels, LTE is increasingly being chosen for next-generation public safety networks, for which both dedicated only and hybrid dedicated/commercial solutions have been proposed [2, 3]. These approaches are valuable for day-to-day routine activities and major planned events, because they can offer great improvements to network capacity and in consequence to the range of services and applications that PPDR users can employ. Moreover, LTE is able to exploit spectrum holes through cognitive radio access techniques [4], providing increased transmission quality and coverage. However, in most major incidents and disasters, terrestrial mobile networks are overloaded or their infrastructures are damaged and thus out of service [5]. Even the deployment of ad-hoc mobile networks backhauled by infrastructure-based facilities cannot therefore offer adequate guarantees in the latter, worst-case scenarios.

This article proposes an infrastructure-less approach to provide high-bandwidth connectivity through deployable LTE base stations, backhauled by new-generation satellite systems. This way, it is possible to provide coverage in the majority of the cases in which terrestrial infrastructures are damaged or destroyed by a disaster, without requiring field operators and civilians to employ special equipment. In infrastructure-based scenarios, this proposal provides PPDR operators with extended coverage, higher broadband capacity and greater resilience.

The remainder of the article is organized as follows. First, we briefly present the deficiencies of current PPDR systems. Then, we detail our proposal through the analysis of network architecture, properties and features. Lastly, we outline the conclusions.

CURRENT PPDR SYSTEMS

In the European Union, PPDR communication is currently performed by means of different technologies, depending on the country or even on the specific region: TETRA, TETRA2, TETRAPOL, Analogue/Digital Mobile Radios

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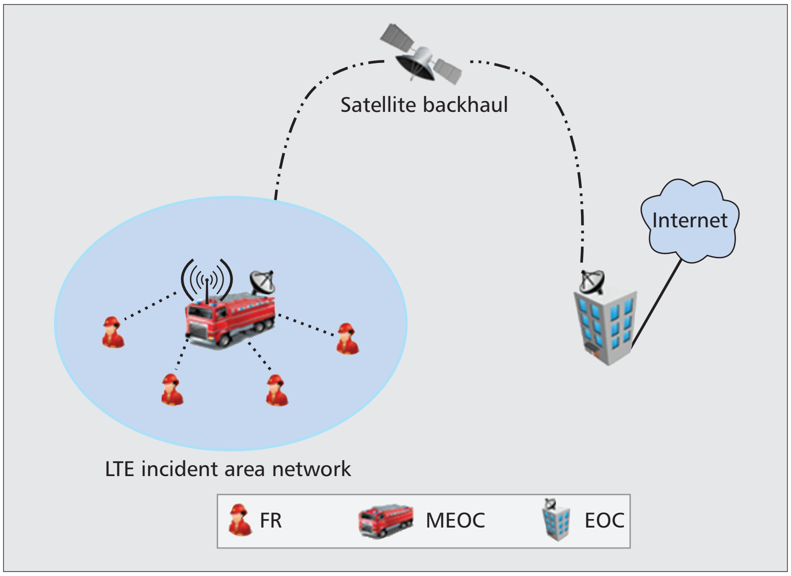

Figure 1. Single reference deployment.

and satellites. Their main purposes are the provision of specific voice capabilities, such as Group Calls and Direct Mode Operation (DMO), and the facilitation of voice calls when terrestrial networks coverage is lacking.

Although these technologies are commonly used in the PPDR domain, they exhibit well-understood deficiencies, especially in relation to the PPDR community demands:

- Data Rate: current PPDR technologies do not offer broadband capabilities; in fact, the achievable data rates are always less than 30 kbit/s, with the exceptions of TETRA2, that allows for data rates up to 400 kbit/s, and satellite communications, for which data rates range from 2.4 kbit/s to an aggregate of tens of Mb/s. It is therefore impossible to exploit performance-demanding applications and data services in general [6].
- Availability: currently deployed networks often fail to guarantee operation in major incidents and disaster scenarios. Even in the absence of damage to terrestrial infrastructures, their coverage is far from being ubiquitous.
- Resilience: current infrastructure-based systems are not redundant and, in general, they are vulnerable to disasters and subsequent incidents (e.g. earthquakes often happen in series, posing a threat to terrestrial infrastructures even if the first event has left them partially or totally operational).
- Spectrum: to date, spectrum allocation is not harmonized between EU countries, affecting the interoperability of PPDR systems, especially in cross-border operations.

Between PPDR stakeholders, there is complete consensus on the paramount importance that voice services play in every disaster relief operations. It is also clear that data and video services have already started to play an important role for PPDR users [1]. PPDR operators need networks that offer high availability, high reliability, high security and faster data transmissions; capabilities that map respectively into coverage, resilience, security and data rate properties.

Integration of Satellite and LTE

In PPDR operations it is often essential to employ satellites or cellular repeater stations to provide wireless network coverage to field operators and people in distress. Traditionally, PPDR agencies have had to choose between the two; instead, we show that they can be employed together, exploiting the benefits of both and at the same time mitigating each technology specific issue. This approach has been chosen because satellites, today, are an almost ubiquitous mean to provide broadband connectivity; however, general-purpose User Equipment (UE) such as mobile phones and smartphones do not embed components to access satellite systems. Furthermore, the propagation delay imposed by satellites would yield unpractical the communication among field personnel. On the contrary, general-purpose UE is built to connect to commercial mobile networks, and it is already clear that LTE and LTE-Advanced are going to be the next standards for mobile devices.

We propose a hybrid architecture based on the integration between LTE and satellite technologies, whose key design requirements have been defined as:

- Accessibility: the deployed network(s) must be easily accessed by general UE.
- Coverage: in absence of adverse external conditions, it must be possible to deploy Incident Area Network(s) where terrestrial coverage is disrupted or absent.
- Performance: it must provide broadband access, at least to PPDR operators.
- Interoperability: it should permit concurrent exploitation of existing terrestrial networks, if operative in the disaster area.

The subsections that follow detail our proposal.

System Architecture

In disaster scenarios, base stations may not be the only damaged network components, also aggregation channels and core infrastructures may be affected. Our proposal aims at being a solution in the worst-case (i.e. when an infrastructure-less network must be deployed and represents the only source of coverage), while adding capacity and functionalities when an infrastructure-based LTE network is still totally or partially operational.

As the general concept, we consider LTE to be the access technology while satellite the backhaul one; i.e. satellites are used as backhaul means to convey coverage through LTE base stations, as depicted in Fig. 1. In this model, the Mobile Emergency Operations-control Center (MEOC) provides First Responders (FRs) with a LTE Incident Area Network (IAN), thus representing a deployable (and mobile) LTE repeater station for field operators. A reliable satellite link serves as the backhaul medium between the MEOC and the Emergency Operations-control Center (EOC), which is non-mobile and represents the operations headquarters, in order for the former to be able to communicate with the latter independently from the geographical position.

We consider the Incident Area Network provided by the MEOC to be an "atomic" element with which to compose the ad-hoc infrastructure-less network topology that will be presented now. The state of network access on the field can be fully described with three cases:

- Persons have no connectivity to MEOCs or terrestrial networks, and are therefore unable to communicate.
- Persons can have connectivity with either a MEOC or a terrestrial base station.
- Persons can have connectivity with both a MEOC and a terrestrial base station.

These conditions are not necessarily fixed for the entire duration of the rescue operations: a person may shift between them at any time, for example if terrestrial congestion increases/decreases, if subsequent disasters damage network infrastructures or if MEOCs move. It is obvious that the first case must be always avoided.

Figure 2 shows the network architecture on a MEOC. By integrating both the E-UTRAN and the EPC subsystems, the MEOC can directly provide IP connectivity to UEs, and in case of necessity it can route data through its satellite link. Furthermore, this way no terrestrial core network is necessary, thus avoiding the presence of a single point of failure when a number of MEOCs are deployed.

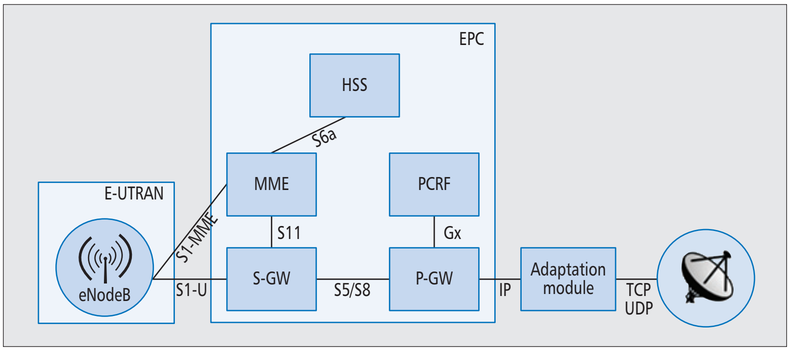

Figure 2. MEOC network architecture.

Service Provisioning

People in distress may use their standard UE to ask for help, communicate their status and position, give a contribution to disaster recovery, and communicate with loved ones. If for whatever reason they are not able to rely on commercial terrestrial networks, they must connect to a MEOC when it provides them with LTE coverage. If, on the other hand, FRs use the terrestrial commercial networks as today often happens [1], they will congest them even more, especially in the presence of prioritization mechanisms; therefore, this option must be used as a last resort. Thus, connectivity (voice and text messaging at least) must be provided to people in distress that may be trapped, injured or dying and additional services (specific voice capabilities, text messaging and broadband data support) must be provided to FRs.

The latest LTE standard releases already contemplate the provision of services such as DMO and Group Calls [7]; in addition, external platforms such as the IP Multimedia Subsystem or service enablers defined by the Open Mobile Alliance can be employed to provide advanced voice services through LTE.

For what regards broadband data support, the optimal solution is for FRs to have a dedicated channel (i.e. high priority) with the MEOC via LTE that provides broadband data capabilities; MEOCs network capacity in excess (i.e. low priority) can be therefore provided for people in distress. MEOCs have thus a twofold function: they provide a dedicated PPDR network for FRs and additional network capacity for people in distress. This prioritization can be realized through LTE bearers. A bearer is an IP packet flow that defines a specific quality of service (QoS) between a gateway and a UE. A user can be associated with multiple bearers; for example, a FR might be engaged in an emergency voice (VoIP) call while at the same time performing a file upload. A VoIP bearer with dedicated resources would provide the necessary QoS for the voice call, while a best-effort bearer would be suitable for the file transfer. The same concept applies for different users, i.e. it is possible to assign by default dedicated bearers to FRs and best-effort bearers to civilians. In fact, when a UE connects to a LTE network, it gets an IP address by the P-GW and at least one default bearer is always established, whose parameters are assigned on the basis of subscription data retrieved from the HSS.

Infrastructure-based and Infrastructure-less Topologies

Our proposal contemplates the possibility to extend infrastructure-based networks if terrestrial LTE infrastructures are still operational, and to deploy completely infrastructure-less networks otherwise, as shown in Fig. 3. Normally, the X2 interface is established between one eNodeB and some of its neighbor eNodeBs in order to exchange signaling information. Its initialization starts with the identification of a suitable neighbor, a process that can be manually performed or automatically carried out by LTE Self-Organizing Network (SON) features. Specifically, the Automatic Neighbor Relation function makes use of UEs to identify neighbor eNodeBs: an eNodeB may ask a UE to read the global cell identity from the broadcast information of another eNodeB for which the UE has identified the Physical Cell Identity (PCI) [8]. Once the IP address of a suitable neighbor has been identified, the initiating eNodeB can establish a transport level connection; then it must trigger the X2 Setup procedure, which enables an automatic exchange of application level configuration data that is core of another SON feature: the automatic self-configuration of the PCIs. Once this procedure has been completed, the X2 interface is operational. This way, it is possible to deploy MEOCs that automatically attach to already existing terrestrial eNodeBs, thus extending the terrestrial infrastructure. In

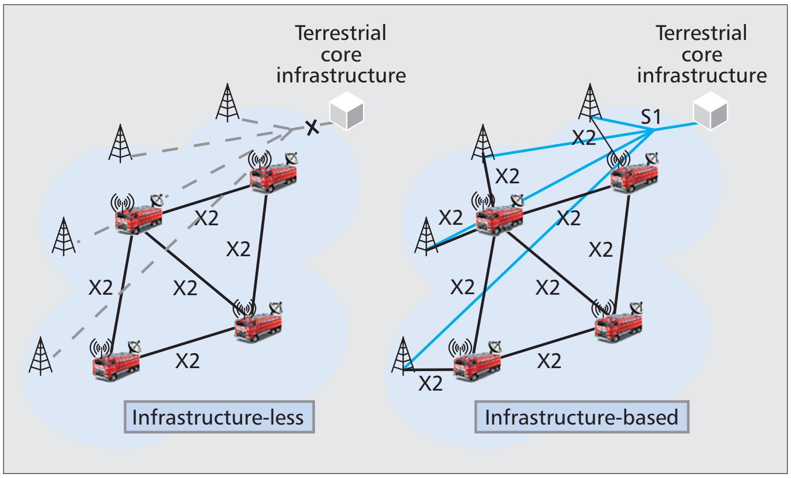

Figure 3. Infrastructure-based vs. infrastructure-less topologies.

the same way, it is also possible to automatically interconnect different MEOCs in an infrastructure-less topology. When a UE has the possibility to attach to different eNodeBs, such network topologies may be very useful to shift traffic from heavily congested MEOCs to less congested ones, thus augmenting the overall system efficiency, and to provide stronger resilience by the presence of different available paths in order to reach a remote host.

An additional LTE eNodeB on each MEOC is an option that may be considered to provide a long-range channel for inter-MEOC communication, when these are not deployed near each other; without providing connectivity and capacity to users on the field, their range may be much more extended comparing to a typical LTE eNodeB used to provide coverage to people in distress and FRs. In absence of these dedicated channels, or if MEOCs are deployed even farther, the satellite backhaul channels may be used among MEOCs, although this solution would add the time for two additional satellite hops to the overall transfers.

Handover

FRs are expected to move in the disaster area, and their specific movements cannot be predicted in advance. When civilians, on the other hand, are not able to rely on commercial terrestrial networks, they must be able to shift to a MEOC when it provides LTE coverage in their area. Therefore, transparent handover provision is an important requirement. With LTE, this can be accomplished through both its S1 and X2 handover procedures. When a UE moves between one cell and another, handover through the X2 interface is triggered by default, unless there is no X2 interface established or the source eNodeB is configured to use S1 handover instead [8]. X2 handover should be performed if the UE is moving between terrestrial eNodeBs. When moving between MEOCs cells or between a MEOC and a terrestrial eNodeB, a S1 handover procedure must be performed instead. In these cases, in fact, the source and target eNodeBs are served by different MME/S-GW nodes, and S1 handover is thus required.

Spectrum Remarks and Interoperability with Legacy Technologies

LTE have been designed to support as many regulatory requirements as possible, and in consequence it is able to operate in a number of different frequency bands. While from a commercial standpoint most European LTE networks are being deployed in Bands 3 (1800 MHz), 7 (2600 MHz) and 20 (800 MHz), EU still lacks a harmonized frequency band dedicated to PPDR purposes. This fact clearly represents an issue for cross-border operations and Pan-European service provision; furthermore, it may happen that equipment from one country is not able to work in others. To help to overcome these issues the proposed architecture may allow for a local spectrum harmonization of the deployed IANs, also through on-demand (i.e. by MEOCs configuration) spectrum alignment with the frequency bands used by terrestrial networks in the specific geographical area of the disaster, feature that may ease accessibility, reduce signal interference and allow for cross-border interoperability. In alternative, under the hypothesis of proper EU regulatory enforcements, the hybrid spectrum management approach proposed in [3] may provide PPDR operators with sufficient guarantees.

Because the legacy PPDR networks listed earlier are already operational in a number of EU countries and are already under development in others, multimode UE may be adopted by FRs in order to allow the exploitation of existing PPDR infrastructures, where those are deployed. LTE also supports interworking and mobility with networks using other technologies, namely GSM, UMTS, CDMA2000 and WiMAX. This allows the interoperability with legacy terrestrial commercial networks, if deployed and operational in the disaster area.

Networks Range, Data Rate and QoS

The range limit of wireless repeaters is primarily determined by cell size, system configuration, signal penetrability and expected number of users. Theoretically, the coverage offered by common LTE base stations may range from 15 km to 100 km [9]. In practice, these values vary. In urban/suburban areas, single cell coverage typically ranges between 0.5 km and 3.5 km, because both the propagation loss and the average user density are high. In rural areas, the nominal single cell coverage typically ranges between 25 km and 50 km; however, the actual coverage is easily affected by environmental conditions that rise the propagation loss (e.g. when obstacles as hills are present between eNodeBs and UE). Considering the peculiarities of a generic emergency situation, the main factors that hinder coverage of a deployed LTE cell are the possibility of high propagation losses (especially when the disaster strikes an urban area) and the certainty of a very high user density, both in the spatial and temporal domains. Therefore, we consider ranges between 0.5 km and 1.5 km in urban areas, between 1 km and 2 km in suburban areas, and between 1 km and 10 km in rural areas to be appropriate estimations for a

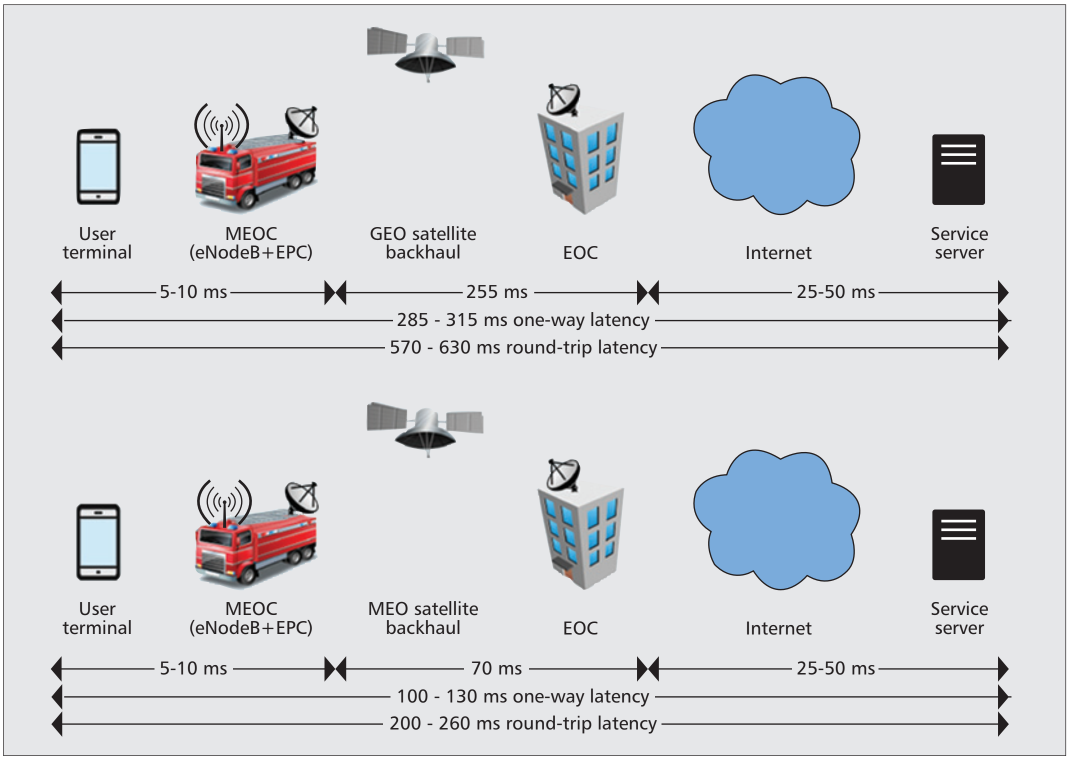

Figure 4. End to end latency comparison between GEO and MEO satellite backhauls.

In PPDR operations, it is not infrequent to meet an indoor or tunnel scenario. In order to provide coverage in those cases, the proposed architecture allows for two different approaches. In general, the FRs UE should embed the possibility to act as repeaters, in order to extend network range and to create a redundant number of network paths.

deployable LTE eNodeB to be used in emergency scenarios. Satellites, on the other hand, are theoretically able to provide global coverage, depending mainly on their orbit type:

- Geosynchronous Orbit (GEO) satellites are deployed at 35.786 km above the Earth's equator. They permanently orbit in the same sky area, allowing ground-based antennas to remain fixed in one direction. These satellites offer the greatest coverage, but being very far from Earth they also exhibit the highest propagation delay.
- Medium Earth Orbit (MEO) satellites orbit at altitudes comprised between 2000 km and 35.768 km. A greater number of satellites is necessary to provide a coverage comparable to GEO solutions. The lower orbit, however, allows better transmission performance.
- Low Earth Orbit (LEO) satellites orbit at altitudes comprised between 160 km and 2000 km. These solutions could provide the least propagation delay. However, an even greater number of satellites must be put in orbit in order to offer a coverage that is comparable to GEO and MEO solutions.

For what regards data rate, LTE can theoretically reach an uplink value of 75 Mb/s, while 326 Mb/s can be reached in downlink with a 4 × 4 MIMO antenna without error control coding. However, simulations and trials have showed [10] that realistic LTE average throughput values stop between 17 Mb/s and 33 Mb/s. The data rate provided by a satellite backhaul depends primarily on its orbit type; because propagation delay plays a key role in the realistically achievable data rates, GEO satellites usually offer the least performance. We consider as the reference for latest MEO systems the O3b solution [11]; it uses the Ka-band frequencies between 17.8 and 19.3 GHz in the downlink and between 27.6 and 29.1 GHz in the uplink. These satellites compose a MEO constellation at a height of 8063 km. Each satellite circles the earth within 4 hours and, for a fixed point on Earth, a new satellite rises every 45 minutes. O3b uses a make-before-break mechanism to ensure seamless handovers, in which a ground terminal temporarily enjoys a simultaneous connection to two satellites. Every satellite possesses 12 spot beams, able to deliver a data rate of up to 1.2 Gb/s each, with a bandwidth of up to 216 Mhz. To achieve such high data rates, the satellite dishes of terminals have to have a diameter that is between 3.5 m and 4.5 m, facilities that cannot be embedded in a UE but can be installed on a MEOC. For future LEO constellations, L-3 Communications estimated that data rates up to 3.75 Gb/s should be achievable between a LEO satellite and a ground station [12], using Ka-band frequencies. However, to date no LEO constellation able to offer broadband capacity is operative.

For QoS purposes, the overall latency plays a key role, especially when considering the usage of satellites as backhaul means. For the LTE access part, thanks to its IP-based architecture, the initial packet data connection typically takes around 50 ms, and between 10–15 ms of roundtrip latency is needed for subsequent transfers. The propagation delay between the eNodeB and the UE does not pose any issue to delay-sensitive applications; in fact, even at a distance of 100 km, the eNodeB needs less than a millisecond to reach the UE. At that distance, however, performance would be significantly

The proposed architecture has the potential to permit interoperability with legacy cellular technologies, to ease cross-border operations and to provide communication in disasters that include both outdoor and indoor scenarios.

impaired by the propagation loss. On the contrary, a GEO satellite backhaul takes *at least* 500–550 ms for a Round Trip Time (RTT), whereas with a MEO solution this value can be reduced to an average of 200 ms [13], depending on the orbit height. LEO satellites typically need less than 100 ms for a RTT, also depending on the orbit height; however, because of the data rate deficiency stated above, current LEO constellations may only be considered as part of the legacy PPDR systems presented earlier. An estimation of the total average latency needed for a complete, remote connection made through the LTE subsystem backhauled by satellite is represented in Fig. 4. In it, the GEO propagation delay is optimistic on purpose, resulting in a mean RTT of 600 ms with GEO satellites and of 230 ms with MEO satellites. Especially in the PPDR context, where stringent QoS guarantees are often needed, the total latency experienced by the user is a very important parameter.

The usage of satellites as backhaul mediums implies that QoS may be strongly affected by TCP, known to perform badly when employed on links with high propagation delay. While also in PPDR operations UDP is preferred for general video streaming, a good number of applications require reliable data delivery [1]. If GEO satellites are employed, specific TCP solutions [14] should be used; instead, with MEO and LEO systems it is possible to maintain the usage of common TCPs (e.g. TCP Cubic) with a limited QoS degradation [13].

In the proposed architecture, the identified bottlenecks are therefore:
- The LTE access subsystem for what regards coverage.
- The LTE access subsystem for what regards data rate if it is backhauled by MEO satellites, while the situation may invert if GEO satellites are employed.
- The satellite backhaul subsystem for what regards latency QoS.

Overall, new-generation Ka-Band MEO satellite constellations offer a high-performance and valuable medium to backhaul LTE connections for PPDR purposes.

Indoor and Tunnel Network Provision

In PPDR operations, it is not infrequent to meet an indoor or tunnel scenario. In order to provide coverage in those cases, the proposed architecture allows for two different approaches. In general, the FRs UE should embed the possibility to act as repeaters, in order to extend network range and to create a redundant number of network paths.

The first option is to deploy picocells and femtocells inside buildings and tunnels, respectively; these may be directly connected to a MEOC, to other eNodeBs that in turn reach the MEOC, or to some FRs UE acting as a bridge. The second option is to create a network chain by deploying the necessary number of FRs, forming a network path from the MEOC into the building or tunnel; this way, their UE may provide connectivity indoor and traffic may be routed outside through multiple hops.

Conclusions

To address the vulnerabilities of terrestrial infrastructure-based networks in major incidents and disaster scenarios, we proposed a hybrid network architecture that integrates LTE and satellite technologies for PPDR purposes. It is based on deployable mobile units that bring LTE coverage to the disaster area through a satellite backhaul. The architecture is designed in order to provide easy connectivity, extended coverage and high performance guarantees. Furthermore, it allows for both infrastructure-less and infrastructure-based service provision, without requiring extensive configuration. This way, existing dedicated and commercial terrestrial infrastructures can be leveraged, and at the same time sufficient reliability from unexpected events can be provided. We conclude by noting that the proposed architecture has the potential to permit interoperability with legacy cellular technologies, to ease cross-border operations and to provide communication in disasters that include both outdoor and indoor scenarios.

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Biographies

Maurizio Casoni (maurizio.casoni@unimore.it) is Associate Professor of Telecommunications in the Department of Engineering "Enzo Ferrari" (DIEF) at University of Modena and Reggio Emilia (UNIMORE), Italy. He received his M.S. with honors and his Ph.D. in electrical engineering from University of Bologna, Italy, in 1991 and 1995, respectively. In 1995 he was with the Computer Science Department at Washington University in St. Louis, MO, as a Research Fellow. He is now responsible at UNIMORE for the EU FP7 Projects E-SPONDER and PPDR-TC.

Carlo Augusto Grazia (carloaugusto.grazia@unimore.it) received his M.S. in Computer Science from University of Bologna in 2012. He is currently a Ph.D. candidate in Information and Communication Technologies at DIEF of UNIMORE. He is involved in the EU FP7 Project PPDR-TC. His research interests include Theoretical Computer Science and Computer Networking.

Martin Klapez (martin.klapez@unimore.it) received his M.S. in Computer Science from University of Bologna in 2013. He is currently a Ph.D. candidate in Information and Communication Technologies at DIEF of UNIMORE. He is involved in the EU FP7 Project PPDR-TC. His research interests include Philosophy of Mind and Computer Networking.

Natale Patriciello (natale.patriciello@unimore.it) received his M.S. in Computer Science from University of Bologna in 2013. He is currently a Ph.D. candidate in Information and Communication Technologies at DIEF of UNIMORE. He is involved in the EU FP7 Project PPDR-TC. His research interests include Distributed Systems and Computer Networking.

Angelos Amditis (a.amditis@iccs.gr) is Research Director at the Institute of Communication and Computer Systems (ICCS). He is founder and Head of I-SENSE Group, President and founding member of ITS Hellas, member of ERTICO Supervisory Board and president of EuroVR Association. He received his M.Sc. and his Ph.D. in Electrical and Computer Engineering from the National Technical University of Athens, Greece, in 1992 and 1997, respectively.

Evangelos Sdongos (esdongos@iccs.gr) is Project Manager and Researcher at ICCS. He holds a M.Sc. in Telecommunications and IT from Patras University, Greece, and has extensive experience in design, implementation and integration of mobile communication networks, embedded systems and emergency communication systems.

Software Defined Networking and Virtualization for Broadband Satellite Networks

Lionel Bertaux, Samir Medjiah, Pascal Berthou, Slim Abdellatif, Akram Hakiri, Patrick Gelard, Fabrice Planchou, and Marc Bruyere

Lionel Bertaux Akram Hakiri, Samir Medjiah, Pascal Berthou, and Slim Abdellatif are with CNRS and Univ de Toulouse.

Patrick Gelard is with CNES.

Fabrice Planchou is with EADS Astrium.

Marc Bruyere CNRS and Dell Inc.

ABSTRACT

Satellite networks have traditionally been considered for specific purposes. Recently, new satellite technologies have been pushed to the market enabling high-performance satellite access networks. On the other hand, network architectures are taking advantage of emerging technologies such as software-defined networking (SDN), network virtualization and network functions virtualization (NFV). Therefore, benefiting communications services over satellite networks from these new technologies at first, and their seamless integration with terrestrial networks at second, are of great interest and importance. In this paper, and through comprehensive use cases, the advantages of introducing network programmability and virtualization using SDN and/or NFV in satellite networks are investigated. The requirements to be fulfilled in each use case are also discussed.

INTRODUCTION

Software-defined networking (SDN) is a new paradigm shift in communication networks, receiving increasing attention from industry and academia. Similarly, network virtualization and network functions virtualization (NFV) are gaining adhesion in the telecom industry. Coupled together, they pave the way to new opportunities in network design, operation, and management.

Terrestrial and satellite networks have always evolved differently. On one hand, terrestrial networks are in constant evolution and are already moving to embrace SDN and NFV [1]. On the other hand, satellite networks have traditionally been considered for specific purposes or as a backup technology (in established markets such as air and sea coverage) and thus rely on technologies locked down by major actors.

Network programmability, openness, and virtualization are the key words of today's networking architectures. Adopting these principles in satellite communications can help reduce CAPEX and OPEX, enhancing the performance and the QoS delivered to satellite communication end-users, extending the range of applications of satellite communications, and achieving seamless integration with terrestrial networks.

In this article we investigate how SDN, network virtualization, and NFV can enhance satellite architecture to achieve the aforementioned objectives. Through practical use-cases, we demonstrate benefits resulting from the integration of these emerging paradigms into communication satellite networks. We also highlight necessary requirements that have to be fulfilled.

This article is organized as follows. Background knowledge on software-defined networking, network virtualization, and network function virtualization is presented. We give a brief overview of the satellite network architecture considered in this study. We detail use-cases where SDN, network virtualization, and NFV can be beneficial to satellite networks. Finally, we conclude with a discussion on their global contributions to communication satellite networks.

BACKGROUND

INTRODUCTION TO SOFTWARE-DEFINED NETWORKS

Recently, SDN [2] has emerged as a new approach for network programmability and management, where the centralized control plane logic is decoupled from the data-forwarding plane. SDN architectures define a new entity (called the controller) that centralizes control intelligence of one or more network elements (switches) as shown in Fig. 1. Various open interfaces [3] have been defined to communicate between the control plane and data plane (south bound interface), and OpenFlow [4] is a de-facto standard. On the north interface of the controller, applications can be deployed with a network-wide view of data path elements.

SDN opens up new opportunities. Most notably, it simplifies network management [5] and allows automated customized on-demand networking with optimal network resource utilization.

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NETWORK VIRTUALIZATION

Network virtualization enables the creation and coexistence of multiple isolated and independent virtual networks over a shared network infrastructure [6]. A virtual network is a logical network with some of its elements (network devices (or nodes) and links) virtual. A virtual node is an abstraction of a network device that is often hosted on a single physical node. It executes network functions such as routing, forwarding, etc. by consuming part of the resources of the hosting node. The resources allocated to a virtual network device are as diverse as CPU, volatile memory, network interfaces, storage, switching, etc. Similarly, a virtual link is an abstraction of a network link that is established on one or multiple physical links or physical paths. It consumes transmission resources (i.e. physical links' bandwidth) as well as switching resources at the traversed physical nodes.

NETWORK FUNCTION VIRTUALIZATION

Historically, the telecom industry has always preferred the use of dedicated equipment to provide network functions. However, this model inevitably leads to long time to market delays and high costs. This model is being challenged by the NFV concept [7]. Indeed, NFV advocates the virtualization of network functions as software modules running on standardized IT infrastructure (like commercial off-the-shelf servers), which can be assembled and/or chained to create services. This approach makes use of the experience on server virtualization learned from the cloud computing industry since a virtual network function may be implemented on one or more virtual machines. The main benefits of NFV are reduction of CAPEX and OPEX and improved network agility.

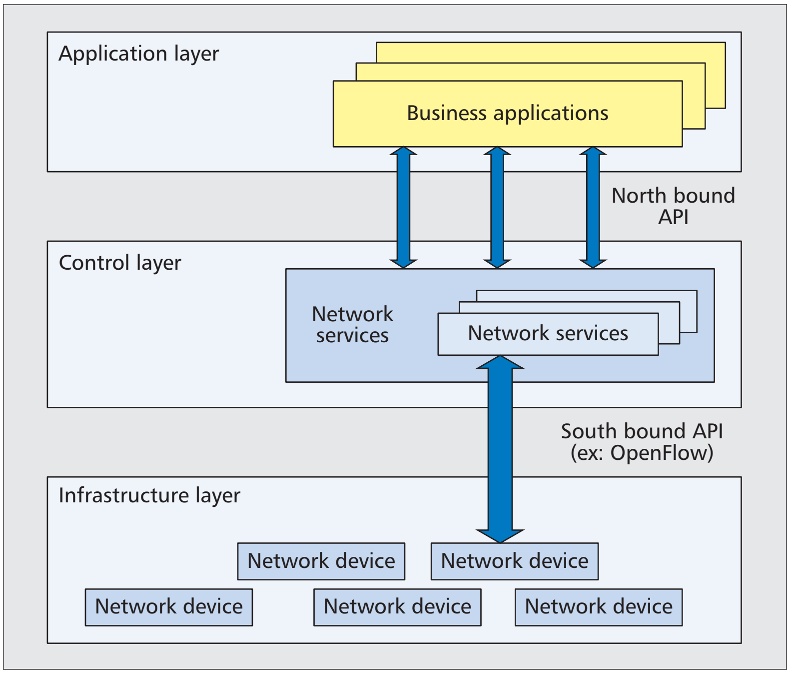

Figure 1. Network vision with the software-defined network (SDN) approach.

SATELLITE NETWORK ARCHITECTURE

This work considers a typical broadband satellite network (BSN) that provides a multi-beam coverage with forward and return links. The ground segment of the BSN gathers multiple hubs that are interconnected via a dedicated backbone network with some PoPs (point of presence) or gateways to external networks, typically the Internet (Fig. 2).

Generally, a hub supports bidirectional traffic on one or many beams. It combines a forward link transmission unit (FL-TU) and a return link reception unit (RL-RU) with a gateway (GW) to terrestrial networks, a network control center (NCC) and a network management center (NMC). The FL-TU performs baseband related functions such as DVB-S2 coding and modulation with adaptive coding and modulation (ACM). The gateway is typically a full-featured IP router with a strong set of functions and protocols (e.g. support for various routing protocols, network address translation, access control lists (ACLs) and firewall services, SNMP, QoS, etc.). The NCC provides control functions; it typically performs satellite terminals (ST) admission control and resources control/allocation on the forward and return links. The NMC performs all management functions, i.e. network element's (ST, hub) configuration, as well as fault, performance, accounting, and security management. Performance enhancing proxy (PEP), designed to improve TCP performance over satellite links, may also be co-located at the hub (or deported at the PoPs or closer to end-users).

The successful delivery of satellite communication services to end-users involves one or many real-life business actors, each playing one or many roles (with the set of functional responsibilities they assume). Referring to [8], three major roles are distinguished:

Satellite Operator (SO): Owns the satellite and assumes its operation. It leases satellite capacity at the transponder level (physical layer) to one or several SNOs.

Satellite Network Operator (SNO): Operates a broadband satellite network with one or more satellite transponders and one or more satellite hubs. It provides satellite forward and return links to second-tier operators by dividing transponder level bandwidth. The NCC controls this bandwidth sharing. Via the NMC, the SNO provides a management interface to the purchased resources.

Satellite Virtual Network Operator (SVNO): Based on the satellite links contracted from one or multiple SNOs, it builds and provides end-to-end higher-level added-value services that are made available via a satellite access.

USES CASES

USE CASE 1: INTER-HUB HANDOVER WITH SITE DIVERSITY

Description and Current Practices — In satellite communications, the use of high frequency bands such as Ka or Q/V makes adaptive coding and modulation (ACM) mechanisms mandatory to counteract signal degradation due to meteorological events such as clouds or rain. In case attenuation is extreme, throughput reduction due to ACM and induced network congestion may not be in agreement with QoS constraints of some flows (VoIP, video conference). If meteorological degradations are due to weather disturbances over ST localization, using robust coding mechanisms cannot be avoided. If degradations concern Hub localization, using another distant site should be considered. The concept of connecting (successively or not) a single ST to several hubs is called site diversity.

Site diversity deployment can follow two different approaches as described in [9]. The N+P approach relies on P redundant hubs that can replace failing sites, resulting in a full handover (HO) of users. The N+0 approach uses frequency multiplexing to serve ST with carriers from different hubs, a failing site inducing losing the corresponding portion of the frequency band. Both cases raise challenges if network service continuity with performance impairments must be guaranteed. Indeed, a hub change (case N+P) or a carrier modification (N+0) has to be signaled to STs and executed. Simultaneously, routing tables have to be updated in the terrestrial network. Also, the handover decision problem is complex since it may concern hundreds of STs and account for several criteria such as flow observation, network knowledge, changing channel quality, etc. Current satellite networks follow the N+P approach with handover performed for a whole beam at a time.

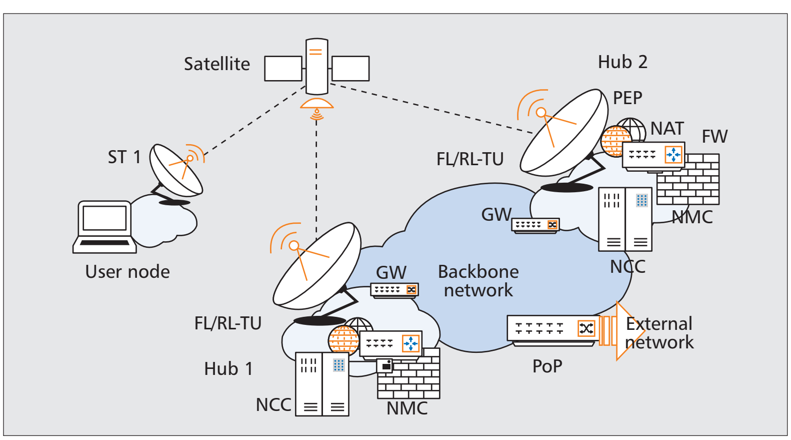

Figure 2. Satellite communication architecture.

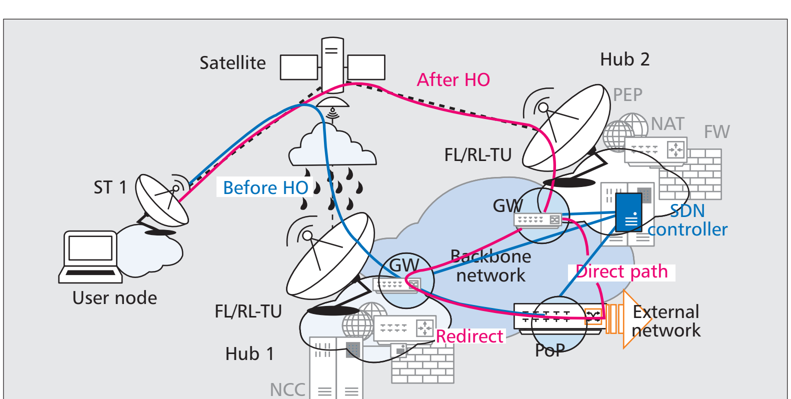

Figure 3. SDN architecture of terrestrial network in site diversity context.

SDN Opportunities for Site Diversity — Obviously, applying SDN principles in the context of site diversity can help devise an effective handover decision algorithm as well as easing the execution of the handover. Basically, this can be achieved with the following enhancements shown in Fig. 3:
- SDN-enabled switches to replace GWs in Hubs.
- One SDN (OpenFlow) master controller located in a hub site running the network application in charge of inter-hub handover management. For clarity one SDN controller is depicted in Fig. 3; however, the use of several controller entities should be considered for scalability and reliability purposes.
- Interfaces to NCC and NMC exposed to the handover application that gather monitoring information and allow triggering some ST configurations.
- Optional: a SDN enabled backbone network.

The handover management application decides when a handover is needed (and the flows or STs that are concerned by the handover) based on:
- Flow constraints: QoS requirements, specific user SLA.
- Flow monitoring: to identify active services, the performance and resources (satellite and backbone) that they are receiving.
- Satellite network as well as terrestrial backbone network performance indicators.
- SNO/SVNO policy: emergency cases or super-user demands.

Once handover of certain flows or STs is decided, the application automatically:
- Informs concerned ST and FL/RL-TU to change their frequencies if needed.
- Updates forwarding rules in the GWs and backbone network.

Two options can be considered:
- “Direct path routing”: Flows are routed directly from their new hub to the nearest PoP.
- “Traffic redirection”: Flows are still passing through their home hub site after having been redirected, at their new hub, via the backbone network,

This is achieved thanks to SDN related programmable functionalities that are added to the packet-processing pipe. For example, OpenFlow can dynamically deploy forwarding rules matching packets based on:
- Incoming network interface.
- IP/MAC addresses.
- Classes of services or protocols used.
- Rate of identified flow or group.
- Deep packet inspection (DPI) using legacy functions.

In conclusion, SDN doubtless contributes to present and future satellite networks by easing the management of inter-hub handover enabled by site diversity, and by extending its capabilities.

Use Case 2: Enhancing VNO Services

Description and State of the Art — The need for virtual network operator (VNO) services is clear and not new. VNO services allow SNOs to partition their satellite resources between multiple SVNOs efficiently by delivering dedicated satellite capacities with different levels of QoS guarantees. Typically, SVNOs in turn repackage these services to provide their customers with added-value end-to-end services. This business model has been in the satellite market landscape for a while. However, the level of control that SVNOs have on their purchased services (and underlying resources) is limited mainly because of the closed nature of satellite devices and the management interface between SNOs and SVNOs.

Figure 4 presents on the one hand the network management system (NMS) used by a SVNO, and on the other hand the SNO's NMC that manages the NCC, GW as well as all STs. Even if some management functionalities can be done directly from the SVNO's NMS to its STs (e.g. routing, etc.), most must go through the NMC (e.g. to get STs' status and statistics). To this end, a management interface (I.SNO-SVNO) is provided by the SNO as part of the VNO service to let the NMS manage the SVNO's satellite terminals. This interface is usually SNMP based complemented with some vendor specific solutions.

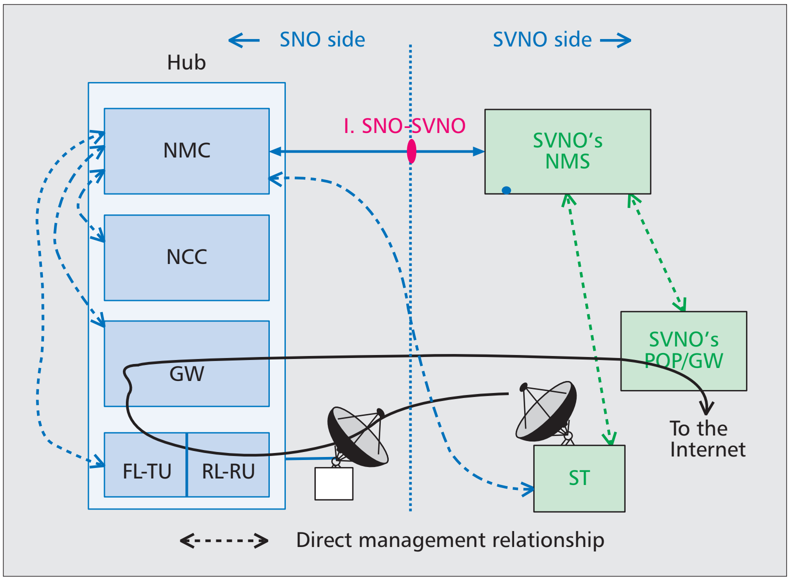

Figure 4. SNO and SVNO management relationships.

The VNO services provided to SVNOs are dependent on the level of visibility and control capabilities exposed by the I.SNO-SVNO management interface that is far from being comprehensive. Moreover, some control capabilities require human intervention from the SNO to validate or perform the required configuration. From the SVNOs perspective, this requires the development of novel services and complicates the provision process of the services they offer.

Network Virtualization and Network Programmability Opportunities for VNO Services — SVNOs are asking for more control of their resources with reduced (or no) intervention from the SNOs. The issues are:

1 Quicker automated service provisioning processes.
2 Enriching their service catalog.
3 Enabling the Satellite Communication as a Service consumption model.

Opening satellite devices via programmatic interfaces (with a rich set of instructions that goes beyond SNMP capabilities) exposed to second tier operators, coupled with network virtualization, is the way to fulfill these goals. By applying device virtualization (i.e. server virtualization applied and adapted to network devices) to SNO satellite hubs, a virtual hub can be assigned on a per SVNO basis (Fig. 5). With the guarantees brought by isolation, which is a key feature of network virtualization, and applies to the data, control, and management planes as well as to performance and security, a SNO can delegate the full control and management of virtual hubs to their customer SVNOs. Therefore, SVNOs can independently enforce their own policies on their satellite virtual networks. Having the control on the NMC to NMS management interface (the range of its capabilities), SVNOs can fully automate the provisioning process of the services delivered to their customers. Indeed, a provisioning engine can be used to orchestrate and perform all the required configurations by accessing the above-cited interface. Moreover, dynamic SLA can be easily supported. Indeed, a SVNO subscriber may ask via a secured portal to dynamically change his bandwidth requirement. The provisioning engine can then autonomously take the right configuration actions to provide and enforce the newly requested SLA within a time frame of few minutes. (In fact, such service exists but with higher response times, often with humans in the loop). New services, such as paying for what you use, can also be considered.

A further step can be achieved by introducing programmability, thus enabling a programmable virtual hub assigned to SVNOs. Programmability may concern the control plane (routing, forwarding, and monitoring, as stated in SDN) allowing SVNOs to devise their own customized traffic control schemes, as well as the data plane, allowing SVNOs to devise customized packet processing algorithms (e.g. PEP, encryption). It paves the way for the diversification and enrichment of the services that SVNOs provide.

The road to programmable virtual satellite networks is in its beginning stages. Network device virtualization with the stringent level of isolation that is required by satellite operators needs to be defined in order to extend the scope of the SVNO to SNO management interface (i.e. I.NMC-NMS). Some vendors already provide some form of hub virtualization; it mainly concerns the gateway (GW) and the management center (NMC) with dedicated I.NMC-NMS to SVNOs. Extending the application of some form of virtualization to the other hub's elements is technically tractable within a reasonable time frame. However, devising a hub virtualization technique that allows a comprehensive management interface between SNOs and SVNOs is still a research topic. Similarly, programming capabilities need to be identified and defined both at the control plane where SDN/OpenFlow capabilities need to be extended or completed with accompanying protocols, and possibly at the data plane. This is another research topic with outstanding issues.

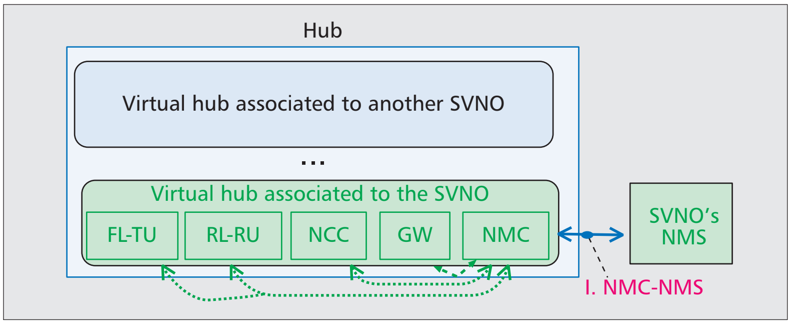

Figure 5. Hub virtualization.

USE CASE 3 : SATELLITE AND TERRESTRIAL NETWORKS INTEGRATION

Description and State of the Art — Whether to provide data backhauling (mobile, military, marine, etc.), or to efficiently deliver communications services along terrestrial access networks, in some deployment environments known as "gray areas" (areas with limited Internet connectivity, i.e. less than 512Kb/s), hybridation of different access networks with satellite would help provide an efficient service offer. Some of its advantages include:

Capacity aggregation: Some applications may require more bandwidth than what a single link provides. In this case, multi-link transmission will help achieve the total required bandwidth. To improve service quality, the additional links may be used to serve a specific purpose, e.g. error correction data.

Load balancing: Data flows from different applications may be forwarded through different links in order to keep the link utilization at its lower level. Similarly, in order to enhance service functionality, the choice of link can be application driven.

To make such a solution a reality, the system architecture should provide a fine-grained control over the carried data flows. Indeed, the ability to dispatch any data stream or any portion of it over the best link is essential. This routing should be done in a seamless fashion to the deployed applications. Nowadays, such control can be sought through a complex combination of various techniques such as policy-based routing (PBR), multi-link protocols (MLPPP, SCTP, etc.), and traffic identification mechanisms (packet marking, DPI, Layer-7 filters, etc.). However, it is worth noting that all these techniques fail to provide the necessary control level for data flows dispatching over different links. Moreover, their behavior lacks dynamicity since forwarding rules are static and do not take into account the evolving link conditions and applications flows.

SDN-Enabled Satellite/ADSL Integration — The SDN paradigm can play an important role in such a solution. Indeed, SDN-based implementations of the hybrid architecture can bring the appropriate control level that current protocols and mechanisms cannot efficiently achieve. Moreover, since packet forwarding decisions are made upon matching rules on packet headers, convergence between the different involved networks, running different communication technologies, can be either achieved at level 3 or at a lower level (level 2).

Figure 6 gives an overview of a network architecture that makes use of both an ADSL access network and a bidirectional satellite network. In this architecture, the global network provider (NP) operates both access networks.

In this scenario, the network operator deploys SDN-enabled devices within its network infrastructure, but also at the customer/user premises. Indeed, in this case the home gateway becomes SDN-enabled and operates under the supervision of an SDN controller hosted at the network operator. Thanks to a network application running on top of the SDN controller, data flow dispatching can be achieved at either the forward or the return links.

In the context of triple-play services over hybrid satellite/ADSL architecture, the freedom in packet forwarding brought by SDN (i.e. OpenFlow packet forwarding rules) enables various scenarios. For example, when starting a phone call and in order to meet the QoS requirement of VoIP, a low-latency link (e.g. ADSL) can be temporarily and dynamically reserved to voice packets, while all other data packets being transmitted over this link are redirected to the satellite link.

However, an SDN-based solution has the following requirements:

Data flow identification: For efficient flow dispatching, the control application needs to identify the service data flows based on parameters such as: IP addresses, port numbers, TOS, or any byte pattern in packet headers or payloads. Thanks to OpenFlow rules expression, such patterns can be easily implemented.

Link monitoring: The control application needs to constantly monitor the links in terms of latency, available bandwidth, etc. in order to optimize the data flow dispatching. OpenFlow in its version 1.3 introduces metering tables, a powerful tool to gather per switch port or even per data flow statistics.

Dynamic forwarding rules generation and update: The control application needs to react to any changes in the link conditions and generate/deploy the appropriate forwarding rules or update the already established rules.

Finally, SDN can make a hybrid architecture more efficient and ease its deployment. Moreover, it will enable novel and innovative services and applications. This use case is already a reality since SDN-enabled switches (e.g. OpenFlow compatible switches) are already on the market; however, the hybridization applications and strategies have to be developed. SDN enabled hybrid and integrated setup boxes must also be proposed.

USE CASE 4 : MIDDLEBOXES VIRTUALIZATION

Middleboxes are prevalent in the Internet architecture, and especially within specific networks such as satellite communication networks. These smart entities are used for various purposes such as performance optimization, security, and address translation. This section analyzes how NFV can improve the classical PEP functions in satellite networks.

TCP Performance Optimization — The TCP/IP model was shown to be not optimum, in terms of performance, in certain WANs and particularly in constrained environments, such as satellite networks. Various TCP protocol versions targeting satellite networks were proposed with the objective of improving the performance of TCP. They were, however, confronted with issues related to their deployment on user terminals. The solution that was found and which is still in use today is to insert devices, at the boundaries of the satellite network, to transform the operation of TCP into a satellite compatible version. These devices, called performance enhancing proxy (PEP) [10], were distributed in satellite networks, offering at the same time advanced services such as Web caching.

Protocol optimizations provided by PEPs are not compatible with several scenarios, in particular in military or aeronautical deployments where security and mobility constraints are present. For instance, implementing a mobile architecture, such as mobile IP, poses complex problems to solve for PEPs. The most problematic case happens during a hybrid handover, i.e. from a satellite network requiring PEP optimization to a network where it is no longer necessary (and potentially counter-productive). In this scenario, TCP connections managed and accelerated by a PEP should survive to a deactivation of the PEP (or more generally to a change of PEP). However, PEPs are physically locked to the infrastructure and cannot move to follow the end-user. Solutions to this problem are proposed in [11] for hybrid satellite/terrestrial; they require context exchanges between PEPs. Other middleboxes that provide advanced services in satellite networks (NAT, firewall, security, etc.) are also subject to the same issues.

PEPs and Network Function Virtualization — The network function virtualization paradigm aims at implementing data-plane processing or control-plane functions in high volume data centers or network elements. This opens a new era in thinking about middleboxes, as they could be easily deployed on demand and under the control of an operator to provide advanced services. Moreover, these middleboxes could be mobile, as they are only relying on software that can be migrated from one standard server to another.

Considering use case #1 presented above (on site diversity), PEPs are typically implemented in the satellite hubs. When a satellite terminal hands over to a new hub, its TCP connections that cross the PEP will be broken as the new PEP will not be aware of the connections' contexts.

With the NFV paradigm, PEPs will no longer be implemented as a dedicated middlebox but rather in software that can be run on different devices. Moreover, the PEP function can be dedicated to a communication context (e.g. dedicated to an ST) and can be tuned according to the application requirements (security, mobility, performances, etc.) If an ST makes a handover from one satellite hub to another, its "dedicated virtual PEP" will migrate to the new hub and will continue to perform the appropriate TCP optimization.

Several cloud computing platforms support NFV and already offer a solution to deploy virtual network functions (VNF). Virtual functions implementing TCP optimization and acceleration for web application servers are proposed by some vendors. From a technical perspective, PEP virtualization can soon become a reality.

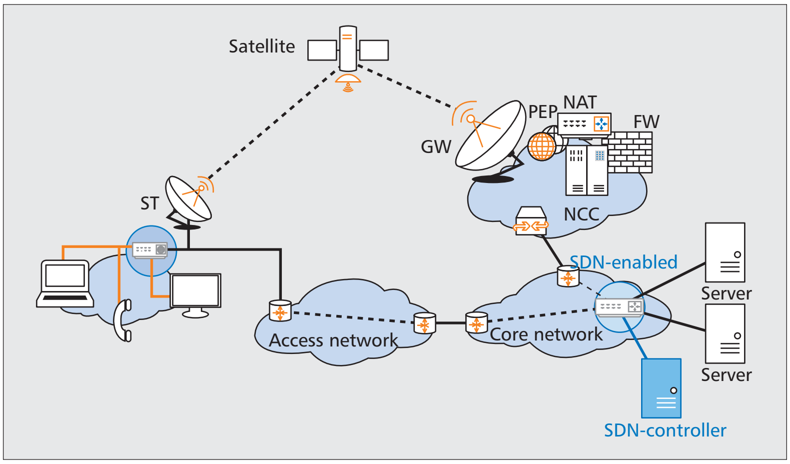

Figure 6. SDN-enabled satellite/ADSL hybrid architecture.

Conclusion

Through the description of four real world use cases, this article has shown some opportunities brought by these emerging paradigms to broadband satellite networks as well as their impacts on a typical satellite system architecture. SDN and NFV are complementary solutions. SDN brings flexibility, automation, and customization to the network. NFV brings agility in the delivery of services and reduces time-to-market development of new services. There is no doubt that they will take a central place in future satellite communication systems.

Acknowledgements

This work is partially funded by the French National Research Agency (ANR), the French Defense Agency (DGA) under the project ANR DGA ADN (ANR-13-ASTR-0024), and the French Space Agency (CNES).

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BIOGRAPHIES

LIONEL BERTAUX (Lionel.Bertaux@laas.fr) received a Ph.D. (2013) in computer science from the University of Toulouse, France. He is a postdoctoral researcher at the Laboratory for Analysis and Architecture of Systems of the French National Centre for Scientific Research (LAAS-CNRS).

SAMIR MEDJIAH (Samir.Medjiah@laas.fr) received a Ph.D. (2012) in computer science from the University of Bordeaux, France. He is an associate professor at Paul Sabatier University in Toulouse (France) and a research scientist at the Laboratory for Analysis and Architecture of Systems (LAAS-CNRS). His main research interests include overlay networks optimization, network virtualization, and software defined networking. He has worked on various R&D projects related to terrestrial/satellite networks convergence, application-driven networking, and network-application co-optimization.

PASCAL BERTHOU (Pascal.Berthou@laas.fr) received a Ph.D. in computer science and telecommunication from National Polytechnic Institute of Toulouse in 2001. He is an associate professor at Université Paul Sabatier. He joined the Laboratory for Analysis and Architecture of Systems of the French National Centre for Scientific Research (LAAS-CNRS) in 1998 as a research staff member, where he works in the area of high-speed networks and protocols and multimedia communications. Since then he has covered two major areas of activity. The first area deals with satellite communication systems. The second research area is sensor networks, particularly the communication system and their application, which in recent years has been directed toward the design of WSN for instrumentation networks. Within this domain he has focused on hardware/software network interface issues and cross layering interactions to reduce the energy consumption in WSN. He recently started research contributions in the area of SDN.

SLIM ABDELLATIF (Slim.Abdellatif@laas.fr) received the M.S. (1998) and the Ph.D. (2002) degrees in computer science from the University of Toulouse, France. He is currently an assistant professor at the Institut National des Sciences Appliquées of Toulouse, and a research scientist at the Laboratoire d’Analyse et d’Architecture des Systèmes (LAAS) from the French National Center for Scientific Research (CNRS). His current research interests include network virtualization, software defined networking, QoS and resource management in wireless networks.

AKRAM HAKIRI (Akram.Hakiri@laas.fr) is an associate professor of electrical engineering and computer science and a research scientist at ISSAT Mateur, Tunisia, and a senior scientist and researcher at LAAS-CNRS, Toulouse, France. His current research focuses on developing novel solutions to emerging challenges in wireless mobile networks, cloud computing, SDN, and QoS.

PATRICK GELARD (Patrick.Gelard@cnes.fr) began his professional experience at ABS (Alcatel Bussiness System) on PABX during 4 years before joining the CNES in 1990 as protocol engineering expert at information systems division. For over eight years now, he works in CNES at Network and telecommunication unit as a terrestrial and satellite networking expert. It has now been five years that he has in charge of research axis “terrestrial and satellite infrastructure convergence.” This led him to work on subject R&T like “Software Defined Networking and Virtualization for Broadband Satellite Networks.” He is working on H2020 Vital project (VIrtualized hybrid satellite-TerrestriAl systems for resilient and fLexible future networks) as network expert.

FABRICE PLANCHOU (Fabrice.Planchou@astrium.eads.net) graduated from ENSEEIHT (Toulouse, 1990), belongs to Airbus Defence & Space (ex-Astrium) since 1990, dealing with telecom space systems. He heads an engineering team devoted to R&T and satellite telecom system validation. His main interest is the promotion of space systems within the telecom service market. Main researches and developments are based around terrestrial/satellite hybrid systems, mobiles systems, for both broadband and broadcast services, and also the M2M domain.

MARC BRUYERE (Marc.Bruyere@laas.fr) is a Technical Consultant at Dell-Force10 and a Phd student at the LAAS/CNRS at Toulouse. In 1996 he worked for the ISP Club-Internet.fr and he left it to pass a Master in Computer Networking at the University Paul Sabatier in Toulouse France. And he worked for Cisco, Vivendi Universal, Credit Suisse First Boston, Airbus/DimensionData, Force10 Networks and Dell. He is a Cisco Certified Internetwork Expert #16651. He is involved in the NetFPGA project for few years and his thesis is about an new architecture for IXP OpenFlow/SDN environment.

MOBILE CLOUD COMPUTING

Xiaoming Fu

Stefano Secci

Dijiang Huang

Rittwik Jana

Mobile Cloud Computing refers to an infrastructure where both the data storage and the data processing occur outside of the mobile device. Mobile cloud applications move the computing power and data storage away from mobile devices and into the cloud, bringing applications and mobile computing not only to smartphone users but also to a much broader range of mobile subscribers.

Cloud computing is an emerging paradigm for cost efficient and reliable service provisioning. Today, the rich availability of energy-harvesting and resource-constrained mobile computing devices is beginning to converge with the great opportunity offered by the powerful cloud computing services hosted by virtualized data center resources. Although mobile applications, cloud computing, and data center networking techniques have been intensively investigated in the past couple of years, mobile cloud computing and mobile cloud networking have not raised the attention of the research community until recently.

In this feature topic we are pleased to introduce a series of state-of-the-art papers on this specific area. These articles cover the subject from a variety of perspectives, offering the readers an understanding of mobile cloud computing. A total of five papers were finally selected for this feature topic out of 33 submissions after a strict peer review process. They cover a broad range of the field of mobile cloud computing. While some articles focus on understanding the challenges and potential ways for improving the mobile cloud computing architectures and mechanisms like mobile code offloading, cloudlets and mobility, others suggest complimenting or applying the state-of-the-art mobile cloud computing technique, e.g. leveraging mobile devices' crowdsourcing and computation capabilities in accessing mobile cloud services.

An important trend occuring in the area of mobile cloud computing is the geographical distribution of cloud facilities, with the deployment of small-size data-centers to form local-area clouds. In the first article, "An Open Ecosystem for Mobile-Cloud Convergence," Satyanarayanan *et al.* discuss how the emergence of Cloudlets, local-area cloud solutions at or close to the premises of cloud users, can improve the user experience and argue how their deployment practices shall follow openness and neutrality principles.

The citizens' locations and movement patterns in daily life are of high relevance to many applications such as city planning, traffic control and personalized recommendations. In the second article of this feature topic, "Estimating Users' Home and Work Locations Leveraging Large-scale Crowd-sourced Smartphone Data," Liu *et al.* utilize crowd-sourcing techniques to collect mobile users' location data, and leverage the cloud to estimate large scale users' home and work location data with a good estimation accuracy.

In the third article, "Mobile Code Offloading: From Concept to Practice and Beyond," Flores *et al.* present issues and challenges in mobile code offloading, that is, when mobile devices can execute part of or the entire application using external cloud computing facilities, and propose improvements in the state of the art code offloading approaches.

As mobility is one of the main sources of energy harvesting and limited connectivity for mobile devices, mobility-augmented cloud service provisioning becomes an emerging issue in mobile cloud computing. In the fourth article, "Mechanisms and Challenges on Mobility-Augmented Service Provisioning for Mobile Cloud Computing," Li *et al.* provide an architecture and taxonomy of research challenges related to mobility augmentation, heterogeneous network convergence, and mobile service provisioning.

Besides data collection, mobile crowdsourcing can offer various other pervasive services, such as data processing

(Continued on next page)

and computing. In the last article of this feature topic, "Exploiting Mobile Crowdsourcing for Pervasive Cloud Services: Challenges and Solutions" Ren *et al.* adopt the mobile cloud computing paradigm for mobile crowdsourcing, leveraging the sensing and computational capabilities of mobile devices as well as user intelligence.

We hope that these articles will help readers understand the state-of-the-art advances in mobile cloud computing, providing current visions of how the mobile cloud computing architectures and applications may be developed, improved and evolving. In preparing this feature topic, we wish to thank all the peer reviewers for their efforts in carefully reviewing the manuscripts to meet the tight deadlines. We are grateful to the editor-in-chief Sean Moore for his timely and constructive suggestions.

BIOGRAPHIES

XIAOMING FU (fu@cs.uni-goettingen.de) received his Ph.D. from Tsinghua University, China in 2000. After nearly two years of postdoc work at TU Berlin, he joined the University of Göttingen, where he is a professor of computer science and head of the Computer Networks Group since 2007. His research interests lie in network architectures, protocols and applications, including content distribution, mobile and cloud computing and social networks. He is an IEEE Distinguished Lecturer, and was IEEE ComSoc TCCC vice chair and Internet TC chair.

STEFANO SECCI received his Ph.D. from Politecnico di Milano, Italy, and Telecom ParisTech, France, in 2009. After postdoctoral experiences at NTNU, Norway, and George Mason University, USA, he joined the University Pierre and Marie Curie, France, where he is an associate professor of telecommunication networks. His research interests lie in network architectures, protocols and applications, including network routing and mobile and cloud networking. He is vice chair of the IEEE ComSoc/ISOC Internet Technical Committee.

DIJIANG HUANG is currently an associate professor at the School of Computing Informatics and Decision Systems Engineering at Arizona State University. His current research interests are in computer and network security, mobile ad hoc networks, network virtualization, and mobile cloud computing. His research is supported by federal agencies (NSF, ONR, ARO, and NATO) and industrial organizations. He is a recipient of the ONR Young Investigator Award and HP Innovation Research Program (IRP) Award.

RITTWIK JANA is a lead inventive scientist at AT&T Labs Research. He received his B.E. degree from the University of Adelaide, Australia, in 1994, and a Ph.D. from the Australian National University in 1999. He worked as an engineer at the Defense Science and Technology Organization (DSTO), Australia from 1996 to 1999, and as a member of technical staff at AT&T, New Jersey since 1999. His research expertise falls in the areas of IPTV, P2P, mobile middleware, and wireless channel modeling.

An Open Ecosystem for Mobile-Cloud Convergence

Mahadev Satyanarayanan, Rolf Schuster, Maria Ebling, Gerhard Fettweis, Hannu Flinck, Kaustubh Joshi, and Krishan Sabnani

ABSTRACT

We show how a disruptive force in mobile computing can be created by extending today's unmodified cloud to a second level consisting of self-managed data centers with no hard state called *cloudlets*. These are located at the edge of the Internet, just one wireless hop away from associated mobile devices. By leveraging low-latency offload, cloudlets enable a new class of real-time cognitive assistive applications on wearable devices. By processing high data rate sensor inputs such as video close to the point of capture, cloudlets can reduce ingress bandwidth demand into the cloud. By serving as proxies for distant cloud services that are unavailable due to failures or cyberattacks, cloudlets can improve robustness and availability. We caution that proprietary software ecosytems surrounding cloudlets will lead to a fragmented marketplace that fails to realize the full business potential of mobile-cloud convergence. Instead, we urge that the software ecosystem surrounding cloudlets be based on the same principles of openness and end-to-end design that have made the Internet so successful.

PRESERVING THE END-TO-END PRINCIPLE

The Internet economy of the G-20 countries is estimated at roughly $4 trillion today [4]. How does an ecosystem that is built on top of a handful of simple protocols (such as IP, TCP, DNS, DHCP, HTTP and TLS) manage to deliver such high value? Key to this success is the openness of the Internet, based on an end-to-end design philosophy for applications and services. Any Internet application works the same way anywhere in the world, and can be used over any low-level networking technology. The network may affect performance and usability, but not functional compatibility. This emphasis on end-to-end design ensures universal interoperability, and leads to an expanding (rather than fragmented) marketplace for applications and services as the Internet grows. Most importantly, applications do not have to be modified to benefit from networking improvements.

This article puts forth the view that it is crucial to preserve this end-to-end design philosophy even as the tectonic forces of mobile computing and cloud computing converge upon each other. Their convergence exposes many short-term business opportunities that are tempting, but would lead to violations of the Internet's end-to-end design philosophy. Instead, we advocate an open ecosystem based on the concept of *cloudlets* that run open-source derivatives of the *OpenStack* cloud computing software. We show how such an ecosystem can support many exciting new classes of mobile applications that leverage both centralized and distributed cloud resources and services.

THE LIMITS OF CLOUD CONSOLIDATION

Why is the cloud relevant to mobile computing? One reason is access to cloud services such as Google Maps, YouTube, Netflix, Facebook and Twitter. A second reason, unique to mobile devices, is for offloading operations to improve performance and to extend battery life. The rich sensing capabilities of a mobile device (such as accelerometer, microphone, and camera) can then be combined with compute-intensive or data-intensive cloud processing. The Apple Siri voice recognition system, the Google Goggles augmented reality system, and the Amazon Silk browser are examples of systems that use this approach.

Improvements in mobile devices will not eliminate the need for offloading some computations. Table 1 illustrates the consistent large gap in the processing power of typical server and mobile device hardware over a 16-year period. This stubborn gap reflects a fundamental reality of user preferences. The most sought-after features of a mobile device are light weight, small size, and long battery life. By using the cloud, such a device can overcome its computational limitations.

The term "cloud" evokes centralization. Today, a cloud service is typically consolidated into a few large data centers. Unfortunately, global consolidation implies large average separation between a mobile device and its cloud. End-to-end communication then involves many network hops and results in high latencies. However, low end-to-end latency is crucial for many

Mahadev Satyanarayanan is with Carnegie Mellon University.

Rolf Schuster is with Vodafone Group.

Maria Ebling is with IBM Research.

Gerhard Fettweis is with TU Dresden.

Hannu Flinck is with Nokia Research.

Kaustubh Joshi is with AT&T Research.

Krishan Sabnani is with Alcatel-Lucent Bell Labs.

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Year	Typical Server		Typical Handheld or Wearable	
	Processor	Speed	Device	Speed
1997	Pentium® II	266 MHz	Palm Pilot	16 MHz
2002	Itanium®	1 GHz	Blackberry 5810	133 MHz
2007	Intel® Core™ 2	9.6 GHz (4 cores)	Apple iPhone	412 MHz
2011	Intel® Xeon X5	32 GHz (2×6 cores)	Samsung Galaxy S2	2.4 GHz (2 cores)
2013	Intel® Xeon® E5	64 GHz (2×12 cores)	Samsung Galaxy S4	6.4 GHz (4 cores)
			Google Glass OMAP 4430	2.4 GHz (2 cores)

Table 1. Evolution of hardware performance (Source: adapted from Flinn [7]).

of the applications sketched later. Can the benefits of cloud computing be preserved without excessive end-to-end latency?

Figure 1 shows how this can be accomplished using a two-level architecture. The first level is today's unmodified cloud infrastructure. The second level consists of dispersed elements with no hard state called cloudlets [12]. A cloudlet is effectively a "second-class data center" with soft state generated locally or cached from the first level. By using persistent caching instead of hard state, the management of cloudlets is kept simple in spite of their physical dispersal at the edge of the Internet. Replacing a cloudlet is conceptually similar to replacing a networking element such as router. The central message of this article is that cloudlets should be viewed as Internet infrastructure, subject to the same principles of openness, transparency and end-to-end design mentioned earlier. This viewpoint is consistent with Cisco's concept of *fog computing* [2] and Wang *et al.*'s concept of *micro clouds* [15].

In the two-level model of Fig. 1, data center proximity to mobile devices is achieved by cloudlets without limiting the consolidation achievable in the cloud. Communication between the cloud and a cloudlet is outside the critical path of interactive mobile applications. In addition to the latency benefit, cloudlets also provide a bandwidth benefit. The one-hop wireless bandwidth between a mobile device and cloudlet can be much higher than the end-to-end bandwidth into the cloud. This can help bandwidth-intensive applications such as real-time computer vision analytics for a large number of video cameras monitoring a city.

The architecture shown in Fig. 1 is agnostic with respect to the type of wireless communication between mobile device and cloudlet. How "low" is low enough for latency, and how "high" is high enough for bandwidth, will depend on the specific application in question. For the demanding class of latency-sensitive applications discussed later, an end-to-end latency of less than 10 ms (including cloud/cloudlet processing) is a good target. For the less demanding class of interactive applications discussed later, a target of 60–70 ms is acceptable. For scalable deployment of the bandwidth-hungry applications discussed later, a data rate of 100 Mb/s is a good target; however, proof of concept deployments can live with data rates in the tens of Mb/s. These targets are most easily achieved today using Wi-Fi technology, cloudlet research has tended to focus on Wi-Fi in the past. However, the growing interest in cloudlets among cellular network providers is likely to lead to enhanced activity in this space. Nokia recently unveiled an LTE base station that includes cloudlet-like service hosting infrastructure [10], and Agarwal *et al.* [1] make the case for cloudlets at the level of femto-cells.

The importance of cloudlets can be seen in the results shown in Fig. 2 for augmented reality and face recognition on a mobile device. Full details of these experiments and many others can be found in the paper by Ha *et al.* [9]. An image from the mobile device (located in Pittsburgh, PA) is transmitted over a Wi-Fi first hop to a cloudlet or to an Amazon data center. The image is processed at the destination by computer vision code executing within a virtual machine (VM). For augmented reality, buildings in the image are recognized and labels corresponding to their identities are transmitted back to the mobile device. For face recognition, the identity of the person is returned. Each curve in Fig. 2 corresponds to the CDF of the observed response time distribution. The ideal curve is a step function that jumps to 1.0 at the origin. Figure 2 shows that this ideal is best approximated by a cloudlet. End-to-end latency plays a dominant role, as shown by the worsening response time curves corresponding to more distant AWS locations. Increasing response time also increases the per-operation energy consumption on the mobile device. This value is shown beside the corresponding label in the middle of the figure. For example, the mobile device consumes 1.1 J on average to perform an augmented reality operation on the cloudlet, but 3.1 J, 5.1 J, and so on when performing it on AWS-East, AWS-West, etc. Although these results were obtained on AWS, similar results can be expected with any offload service that is concentrated in a few large data centers.

The label "mobile-only" in Fig. 2 corresponds to a case where no offloading is performed, and the computer vison code is run on the mobile device. In spite of avoiding the energy and performance cost of Wi-Fi communication, the data shows that mobile-only does worse than using the cloudlet. Offloading is clearly important for these applications.

There is a tradeoff between the density of cloudlet deployment and total infrastructure cost. Higher density tends to increase the likelihood that a mobile device will find a cloudlet just one network hop away. However, dense infrastructure is also more expensive. Striking the right balance will require real-world deployment and business experience. The use of persistent caching, leading to the "second-class" nature of cloudlets, reduces the marginal cost of deploying a cloudlet relative to deploying that hardware in the cloud.

MOBILE TOMORROW: A NEW WORLD BECKONS

Cloudlets are a disruptive force in mobile computing. First, they bring energy-unlimited high-end computing within one wireless hop of mobile devices, thereby enabling new applications that are both compute-intensive and latency-sensitive. Second, they can process high data rate sensor inputs (such as video) close to the edge of Internet, thereby reducing demand for ingress bandwidth into the cloud. Third, they can serve as proxies for cloud services when the cloud is unavailable due to failures or cyberattacks, thereby improving the robustness and availability of cloud services. In the sections below, we briefly sketch some of the new genres of mobile applications made possible by cloudlets.

WEARABLE COGNITIVE ASSISTANCE

Using wearable devices for deep cognitive assistance was first suggested nearly a decade ago, but is only now within reach of practical implementation. Such a system could be created by combining context-aware real-time scene interpretation (including recognition of objects, faces, activities, signage text, and sounds) with deep reasoning (using a cognitive engine such as IBM's Watson [5]). Ha *et al.* [8] describe the architecture of a system called *Gabriel* (Fig. 3) that uses Google Glass devices with cloudlets to provide mobile, real-time cognitive assistance. Gabriel is layered on top of OpenStack++, which is described later. Such a system could offer helpful guidance, much as a GPS navigation system guides a driver. Imagine scenarios like this:

After his traumatic brain injury in the Iraq war, Ron often forgets the names of friends and relatives. He also forgets how to do simple tasks. His new cognitive assistance system offers hope. Now, when Ron looks at a person for a few seconds, that person's name is whispered in his ear; when he is cooking, he is guided step by step through the recipe; if he is outdoors, he is reminded of where he is going and how to get there. The quality of Ron's life has improved greatly.

The low end-to-end latency of cloudlets is crucial in this application because humans are acutely sensitive to delays in the critical path of interaction. Delays longer than tens of milliseconds will distract and annoy a mobile user who is already attention challenged. At the same time, cognitive engines such as those shown in Fig. 3 require the memory and processing resources of a small data center. The massively parallel nature of systems like Watson are simply not compatible with running them entirely on a mobile device.

Human cognition involves the synthesis of outputs from real-time analytics on multiple sensor stream inputs. A human conversation, for example, involves many diverse inputs: the language content and deep semantics of the words, the tone in which they are spoken, the facial expressions and eye movements with which they are spoken, and the body language and gestures that accompany them. All of these distinct channels of information have to be processed and combined in real time for full situational awareness. There is substantial evidence that human brains achieve this impressive feat of real-time processing by employing completely different neural circuits in parallel and then combining their outputs [11]. Each neural circuit is effectively a processing engine that operates in isolation of others, a viewpoint shared by work on Hierarchical Temporal Memory [3]. Coarse-grain parallelism is thus at the heart of human cognition, and this is reflected in the Gabriel architecture shown in Fig. 3.

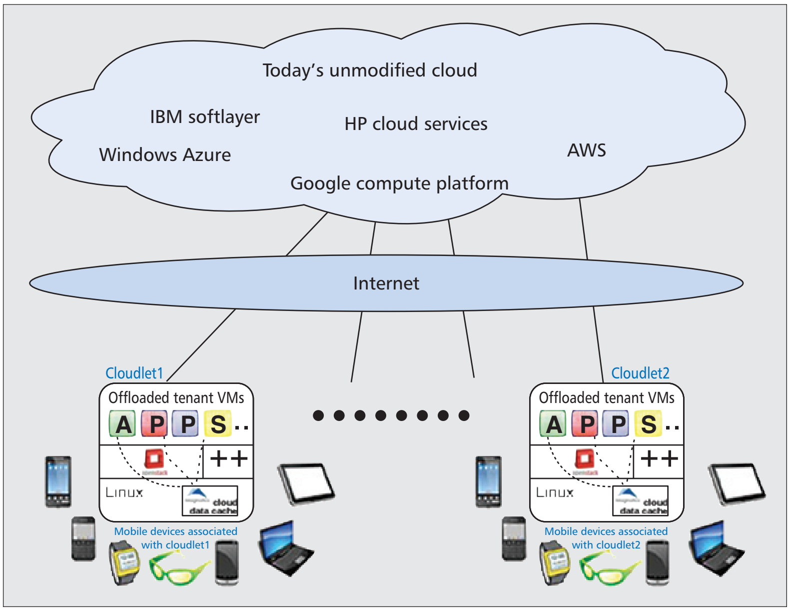

Figure 1. Two-level cloud-cloudlet architecture.

EDGE ANALYTICS IN THE INTERNET OF THINGS

Cloudlets can also reduce ingress bandwidth into the cloud. For example, consider an application in which many of the mobile devices shown in Fig. 1 are continuously transmitting live video to the cloud for content analysis. The cumulative data rate from even a small fraction of users in a modest-sized city would saturate its metropolitan area network: 12,000 users transmitting 1080p video would require a 100 Gbps link; a million users would require 8.5 Tbps!

GigaSight [14] shows how cloudlets can solve this problem. Video from a mobile device only travels as far as its currently-associated cloudlet. Computer vision analytics are run on the cloudlet in near real-time, and only the results (e.g. content tags, recognized faces, etc.) along with meta-data (e.g., owner, capture location, timestamp, etc.) are sent to the cloud. This can reduce ingress bandwidth into the cloud by three to six orders of magnitude. GigaSight also shows how tags and meta-data in the cloud can guide deeper and more customized searches on the content of a video segment during its (finite) retention period on a cloudlet.

A video camera is only one example of a high data rate sensor in the future Internet of Things. Another example is a modern aircraft, which can generate nearly half a terabyte of sensor data on

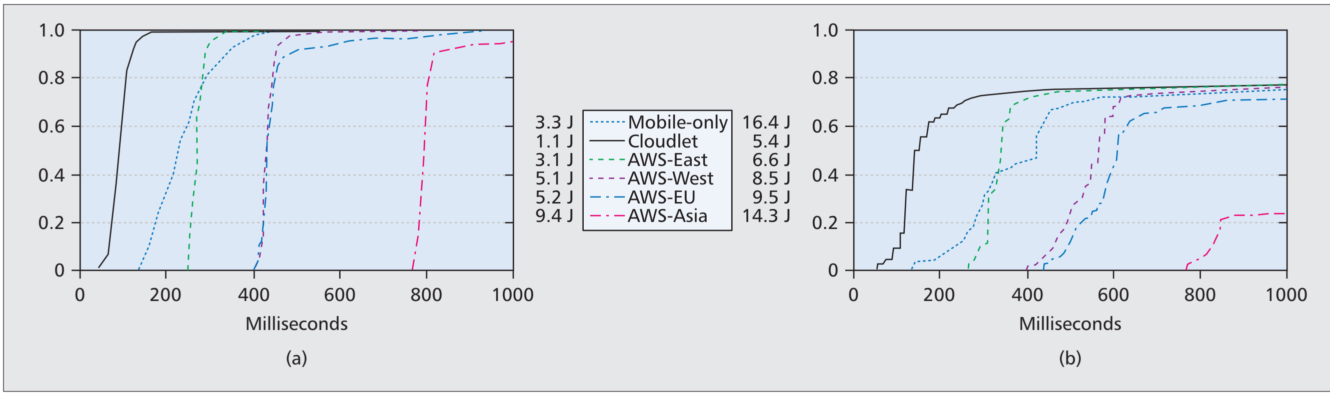

Figure 2. Response time distribution and per-operation energy cost: a) augmented reality and b) face recognition (source: Ha *et al.* [9]).

a flight (www.computerworlduk.com/news/infrastructure/3433595). Real-time analysis of this data on a cloudlet in the aircraft could generate timely guidance for preventive maintenance, fuel economy, and other benefits.

Automotive Environments

The latency and bandwidth benefits of cloudlets are especially relevant in the context of automobiles. For the foreseeable future, cloud connectivity from a moving automobile will be 3G, 4G or (eventually) 5G. An important question is whether cloudlets should be placed within automobiles or at cell towers. We see value in both alternatives, as shown in Fig. 4. This architecture can be viewed as a mapping of Fig. 1 to the automotive context.

An application such as a multi-player video game for the passengers of an automobile is best hosted on the automobile's cloudlet. Only for automobiles without cloudlets (such as old cars on the road, or for the lowest-end cars of the future) does it make sense to use the cell tower cloudlet for such an application. For other applications such as collaborative real-time avoidance of road hazards, the cell tower cloudlet is the optimal hosting site. For example, if one automobile in a cohort hits a pothole or swerves to avoid a fallen tree branch, the coordinates of the hazard can be rapidly shared within the cell tower cloudlet and then used for avoidance by the other automobiles in that cohort. This kind of transient local knowledge can also be provided when an automobile arrives in the coverage region of a cell tower.

An automobile cloudlet could also perform real-time analytics of high data rate sensor streams from the engine and other sources. They can alert the driver to imminent failure or to the need for preventive maintenance. In addition, such information can also be transmitted to the cloud for integration into a database maintained by the vehicle manufacturer. Fine-grain analysis of such anomaly data may reveal model-specific defects that can be corrected in a timely manner.

Mobile Access to the Legacy PC World

In addition to the futuristic applications discussed earlier, cloudlets can also enable mobile access to the huge legacy world of Windows-based desktop applications. These are likely to remain important for many years to come. Using these applications on a mobile device such as an Android tablet is a challenge today. Running a local VM with a Windows guest environment is rarely feasible since these mobile devices are typically ARM-based and therefore not instruction-set compatible with Intel x86 desktops. Modifying individual applications to become cloud-based (e.g., Microsoft Office 365) is feasible, but time-consuming and expensive. Enabling seamless access from mobile devices to the unmodified legacy Windows world is a less disruptive and more cost-effective solution.

Figure 5 shows how cloudlets can help. A VM encapsulating the personal desktop environment of a user is run on a cloudlet, and the user connects to it through a remote desktop protocol. No legacy code executes on the user's mobile device, since it is merely being used for input and output. The usability of a remote desktop protocol is critically dependent on low latency and good bandwidth. These attributes are exactly what cloudlets offer. To achieve the state shown in Fig. 5, an instance of the user's desktop VM image has to be rapidly created on a cloudlet nearby. As the user moves, it may become necessary to suspend the VM and resume it on a different cloudlet that is closer to the user. Our work in the Internet Suspend/Resume (ISR) system has demonstrated the feasibility of these steps.

Hostile Environments

Implicit in the convergence of mobile and cloud computing is the assumption that the cloud is easily accessible at all times. In other words, there is good end-to-end network quality and few network or cloud failures. However, there are usage contexts in which cloud access has to be viewed as an occasional luxury rather than a basic necessity. This viewpoint applies to several important contexts that we refer to as *hostile environments*.

The prime example of a hostile environment is a theater of military operations. Another example is a geographical region where recovery is under way after a natural disaster or terrorist attack. A third example is a developing country with weak networking infrastructure. More generally, even a well-connected region of the public Internet may become a hostile environment under conditions of cyber attack.

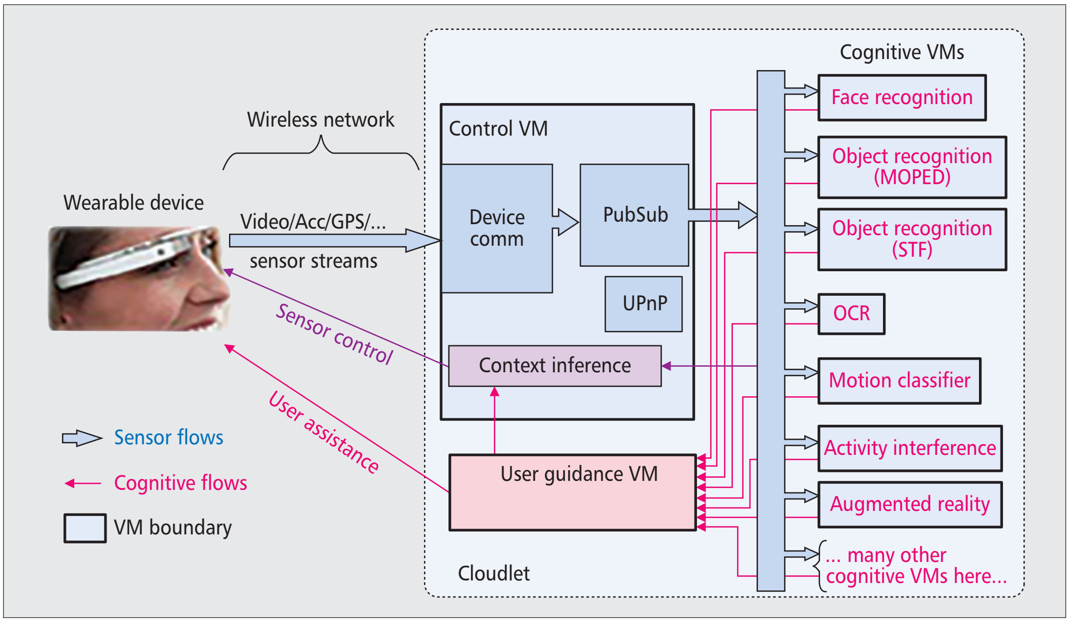

Figure 3. Gabriel architecture for wearable cognitive assistance (Source: adapted from Ha *et al.* [8]).

Many new applications will arise as we move from today's world, where the Internet is used to deliver content, to the next level where applications will control real and virtual objects as laid out in the concept of the Tactile Internet.

We have discussed elsewhere [13] that cloudlets can play a foundational role for service availability in hostile environments. Because of physical proximity, the survivability characteristics of a cloudlet are closer to its associated mobile devices than to the distant cloud. During failures, a cloudlet can serve as a proxy for the cloud and perform its critical services. Upon repair of the failure, actions that were tentatively committed to the cloudlet may need to be propagated to the cloud for reconciliation. These issues are explored in more detail in the paper cited earlier.

Tactile Internet

Many new applications will arise as we move from today's world, where the Internet is used to deliver content, to the next level where applications will control real and virtual objects as laid out in the concept of the Tactile Internet [6]. These new applications will drive the requirements for end-to-end latency towards one millisecond. Only the concept of locally-available cloudlets will enable us to realize this vision. Even at the speed of light, 1 ms of total propagation delay requires a cloudlet within 300 km. In practice, cloudlets will have to be much closer in order to ensure end-to-end delays that are less than 1 ms. The emergence of 5G cellular networks and the deployment of cloudlets will together create the technology platform for the Tactile Internet.

Industry-Wide Business Opportunities Drive an Open Ecosystem

We have sketched a tantalizing new world. However, without viable business models for deploying cloudlets, this vision will merely be a mirage. We face a classic bootstrapping problem. Without unique applications that can benefit, there is no incentive for deploying cloudlets. Yet, without large-enough deployments, it is too risky to create new applications that critically depend on cloudlets. How do we break this deadlock?

This state of affairs is similar to what existed at the dawn of the Internet (early 1980s). An open ecosystem attracted investment in infrastructure and applications, without any single entity bearing large risk or dominating the market. Over time, this lead to the emergence of a critical mass of Internet infrastructure and applications (such as email) that could uniquely benefit from that infrastructure. By the time the World Wide Web emerged as a "killer application" circa 1992, sufficient Internet infrastructure had already been deployed for growth to explode.

We can follow a similar path to success with cloudlets by nurturing the creation of an open cloudlet ecosystem. Emergence of a broad range of cloudlet-based mobile applications (E2E Services) require the involvement and support of a complex set of industries, communities and technology standards. Creating successful business opportunities from these services will require partnerships that jointly drive the business model innovation and agreement on core technology platforms. We examine below how all parts of the value chain can benefit from the new E2E Services:

• **Infrastructure and Device Manufacturers**: These are industry players that provide telecoms infrastructure, cloud infrastructures or mobile devices. The common products in these areas are commoditized already and the players are under significant price pressure. Cloudlets offer opportunities to differentiate and become an innovation leader.

• **Cloud software providers and application developers**: Cloud software providers are eager to enable their cloud software stacks for new cloud environments. The mobile edge cloud will

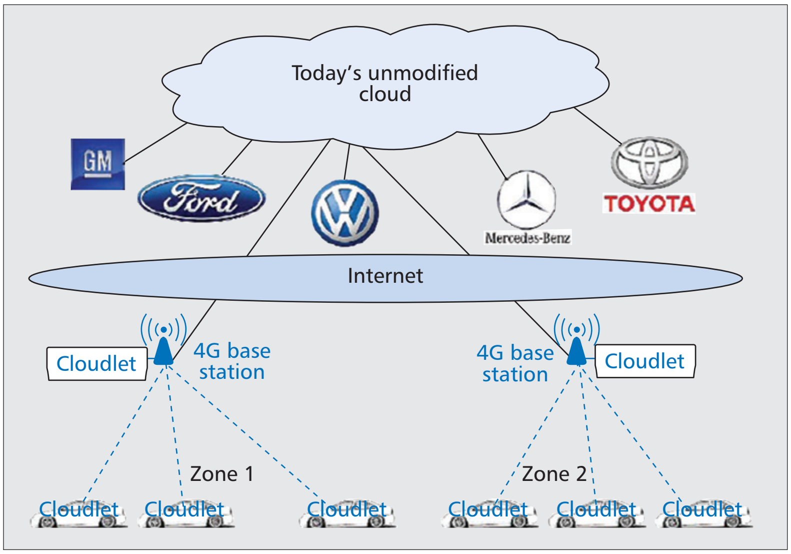

Figure 4. Cloudlets in automotive environments.

be an excellent opportunity for them. Eventually, the cloudlet concept has to become an integral part of mobile application development. For this to happen, cloudlets have to be simple to integrate into mobile application software. In addition, the relevant developer communities have to embrace the cloudlet concept.

• **Telecoms Operators**: Cellular network operators and Wi-Fi hotspot providers are under significant price pressure and are very interested in new revenue opportunities. Since they own the network edge, they are eager to offer cloudlet-based services. Furthermore, cellular network operators have already started to transition their network components into the cloud (network function virtualisation). Hence, the cloudlet components at their network edge are just an extension of the distributed cloud system they are already starting to build. Last but not least, mobile operators are starting to roll out small-cell components (e.g., pico-cells and femto-cells), C-RAN technologies and indoor coverage services. These technologies can be easily combined with cloudlet components to enable new business.

• **E2E service providers**: These are providers of services that utilize cloudlet infrastructure. Each of the new application classes discussed earlier is a source of E2E services. This is a broad set of very different services and business drivers that directly benefit from the cloudlet concept.

• **Service users**: This is a mobile Internet user (consumer or enterprise user) who derives value from cloudlet-enabled E2E services. The key advantage for this part of the value chain is a paradigm shift in user experience driven by a very responsive user interface. The phenomenal market success of Apple products shows that users are willing to pay significant premiums for an improved mobile experience. Cloudlets enable new services (such as wearable cognitive assistance, described earlier) that are transformative. Crisp, low-latency access to data center resources is essential for these services. The concept of a Tactile Internet is well aligned with this viewpoint.

As mentioned earlier, mainstream cloud computing today is driven by the cost efficiencies and economies of scale achievable through consolidation. Obviously, dispersed cloudlet infrastructure at the network edge will not offer these cost efficiencies. However, their unique and specific value to E2E services complements and enhances the highly consolidated and cost-effective cloud systems of today. Premium pricing for these services and infrastructure will therefore be possible. These benefits to all parts of the value chain are only achievable if fragmentation of the marketplace is avoided. The goal can be simply stated as follows:

As long as authentication, authorization and billing criteria are met, any E2E service that is running on any mobile device should be able to leverage any cloudlet in the world. The decision to use a particular cloudlet at a specific point in time should be based solely on current pricing of cloudlet resources and service-relevant performance criteria (such as end-to-end latency, multi-tenancy load, storage cache state, and availability of hardware accelerators) rather than software compatibility.

The open cloudlet eco-system has already started to evolve towards this goal. For example, telecoms infrastructure provider NOKIA and cloud software provider IBM started their joint development of mobile edge technology in 2013 (http://www.ibm.com/press/us/en/pressrelease/40490.wss). In February 2014, the first mobile edge technologies were launched in the market. In October 2014, four major cloud and infrastructure companies (Huawei Technologies, IBM, Intel, NOKIA) and two global operators (NTT Docom and Vodafone) announced a new industry specification group called "Mobile Edge Computing (MEC)" (http://portal.etsi.org/tb.aspx?tbid=826&SubTB=826). The first meeting of this group was held in December 2014 in Munich, and broader participation by other companies is now being explored. This initiative is being held under the auspices of the European Telecommunications Standards Institute (ETSI), and it is well aligned with the cloudlet vision put forth in this article. The MEC white paper referenced above states:

"Mobile-edge Computing transforms base stations into intelligent service hubs that are capable of delivering highly personalized services directly from the very edge of the network while providing the best possible performance in mobile networks. Proximity, context, agility and speed can be translated into unique value and revenue generation, and can be exploited by operators and application service providers to create a new value chain."

Furthermore, Carnegie Mellon University is creating an open source platform called *OpenStack++* (http://elijah.cs.cmu.edu) that is a derivative of the widely used OpenStack platform for cloud computing (http://openstack.org). The "++" refers to the unique extensions necessary for use of OpenStack in cloudlet environments. Some key components of OpenStack++ such as cloudlet discovery and just-in-time provisioning have already been developed and are available as open source. Work is in progress on other components such as optimal cloudlet selection and seamless cloudlet handoff. OpenStack++ aims to be a universally deployable platform for cloudlet-enabled mobile computing, above and below which many proprietary hardware, software and service innovations can emerge.

Conclusion

The early stages of the convergence of mobile computing and cloud computing are already under way. There is a small window of opportunity to shape this convergence so that it preserves openness and cohesiveness of software interfaces and network protocols in the new infrastructure that will emerge. This path can lead to the kind of explosive growth seen in the Internet itself. The alternative path of multiple proprietary ecosystems will lead to a fragmented marketplace that fails to ignite exponential growth. Fortunately, there is growing awareness among industry players of the benefits of openness and end-to-end thinking in this space. We are therefore optimistic that a single open ecosystem will prevail in the converged new world. Although simple in concept, cloudlets are a disruptive force in mobile computing. Their ability to provide low latency, high bandwidth access to energy-unlimited high-end computing within one wireless hop of mobile devices is transformative. As this article has shown, many valuable applications can be created using cloudlets. The potential marketplace for these services and their infrastructure is enormous.

Acknowledgements

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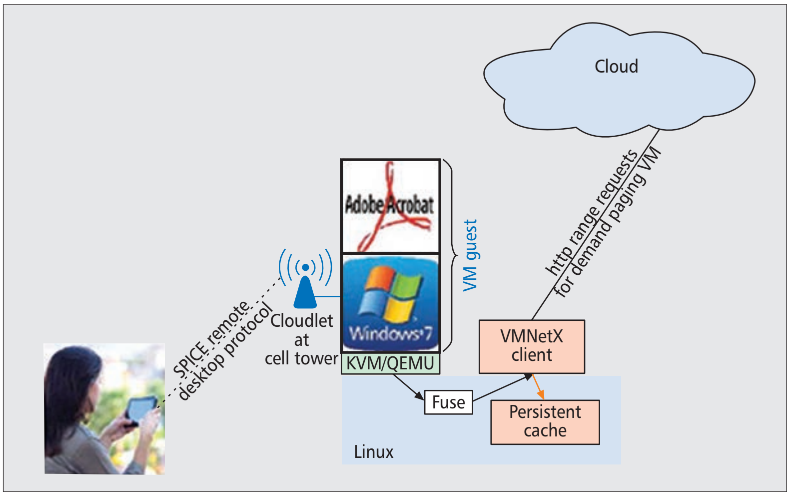

Figure 5. WAN virtual desktop infrastructure for the legacy PC world.

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Biographies

Mahadev Satyanarayanan [F] (satya@cs.cmu.edu) is the Carnegie Group Professor of Computer Science at Carnegie Mellon University. He is an experimental computer scientist whose many pioneering contributions to distributed systems and mobile computing are in widespread use today. He received the Ph.D. in Computer Science from Carnegie Mellon University, after Bachelor's and Master's degrees from the Indian Institute of Technology, Madras. He is a Fellow of the ACM

Rolf Schuster received his BSc in Telecommunications from Fachhochschule Mannheim (Germany), his M.Sc. in Electrical Engineering from the New Jersey Institute of Technology (USA), and a Ph.D. in Computer Science from Technical University Munich (Germany). He also holds a Business Administration degree from Fernuniversitaet Hagen (Germany). He held a number of senior executive roles in large international telecommunications companies and the CEO role in two startup companies. Currently, he is with Vodafone Group Services (Germany) and has the global responsibility for the Partner Innovation area as well as all Research Cooperations of Vodafone.

Maria R. Ebling [SM] is a research staff member and director at the IBM T.J. Watson Research Center. She manages a team building systems capable of supporting a Smarter Planet while not forgetting about the people who use such

systems. Ebling received her Ph.D. in computer science from Carnegie Mellon University. She is a member of the IBM Academy of Technology, and a distinguished member of the ACM.

GERHARD P. FETTWEIS [F'09] earned his Ph.D. under H. Meyr at RWTH Aachen. After one year at IBM Research, San Jose, CA, he moved to TCSI Inc., Berkeley. Since 1994 he is Vodafone Chair Professor at TU Dresden, Germany, with 20 companies sponsoring his research on wireless transmission and chip design. He coordinates 2 DFG centers at TU Dresden (cfaed and HAEC), is member of the German academy acatech, and has spun-out eleven start-ups.

HANNU FLINCK is a Research Manager at Nokia Solutions and Networks, Espoo, Finland. He received his M.Sc. degree (1986) and Lic.Tech. degree (1993) in Computer Science and Communication Systems from Helsinki University of Technology, Finland. His current research agenda includes cloud technology, SDN and content delivery in mobile networks.

KAUSTUBH JOSHI is at AT&T Research. His technical interests include dependability, probabilistic models, distributed systems, cloud computing, and data center management. He received his Ph.D. in Computer Science from the University of Illinois at Urbana-Champaign. His research straddles both theory and practice, ranging from analytical techniques to model and reason about system behaviors under conditions such as faults, attacks, and changing workloads, to runtime control systems that use the models to implement adaptive behaviors such as fault recovery and self-reconfiguration.

KRISHAN SABNANI [F] is a VP at Bell Labs responsible for research on NFV and web communications. He was VP of Networking Research from Jan. 2000 to Sept. 2013. He received a B.Tech. from the Indian Institute of Technology, Delhi in 1975 and a Ph.D. from Columbia University in 1981. He is an ACM Fellow and a Bell Labs Fellow. Krishan won the 2005 IEEE Sumner Award and the 2005 IEEE McDowell Award.

Estimating Users' Home and Work Locations Leveraging Large-Scale Crowd-Sourced Smartphone Data

Hao Liu, Yuezhi Zhou, and Yaoxue Zhang

Hao Liu and Yuezhi Zhou are with Tsinghua University

Yaoxue Zhang is with Central South University and Tsinghua University.

ABSTRACT

Estimating the home and work locations of users is important for applications such as city planning and personalized recommendations. Although existing approaches can achieve a reasonable precision, they rely on fine-grained sensor data with high sampling rate. Therefore, these approaches come with a high cost and are only studied in small samples of volunteers and thus cannot benefit large-scale Internet users. In this article we propose a method to use crowd-sourced location data from mobile devices to estimate the home and work locations of large-scale users, leveraging the computation power of the cloud. Experimental results demonstrate our approach achieves a good estimation precision. Moreover, we further study how the estimated home and work locations can be used in two typical applications that are difficult problems using traditional methods but can be elegantly solved by leveraging the proposed crowd-sourced approach.

INTRODUCTION

Home and work locations are two important places for each individual, which are highly related to individuals' moving and living patterns. Accurately estimating people's home and work locations is beneficial to a number of applications such as city planning, traffic prediction, personalized recommendations, and epidemic study. For example, a personalized recommendation system can recommend business interests nearby a user's home and work locations.

Existing approaches to estimate users' home and work locations are typically based on fine-grained sensor data with a high sampling rate. The sensor data includes location (GPS and Wi-Fi), movement (accelerometer), application usage, communication (calls and SMS), and voice (microphone). Since the data is densely sampled, each location corresponds to a large amount of sensor data. Therefore, researchers are able to use the fine-grained sensor data in each location as features, and employ classification methods [1] to identify whether a location corresponds to a user's home or work location [2]. Due to the high cost involved, these studies are typically based on the data from small samples of volunteers.

With the development of the smartphone and the ubiquity of wireless connectivity in recent years, the diverse apps on smartphones provide us an unprecedented opportunity to collect large-scale crowd-sourced location data from users in a *mobile-cloud* manner. Users' behaviors in smartphones (*the mobile side*) communicate with the services in the servers (*the cloud side*). In this process, a huge amount of detailed mobile data is generated and transferred to the cloud in real time, which enables us to study mobile data at a scale leveraging the power of the cloud. For example, in online location-based social networks such as Foursquare and Jiepang, users can share their location records online. Each location record includes a timestamp and the precise geographic coordinates, which enable us to analyze users' movement and thus infer users' home and work locations, as well as an optional message that we can use to infer other information of a user (e.g. mood). However, although we can collect a huge amount of crowd-sourced data from users in these applications, the records we obtained per user are quite sparse. Therefore, the typical methods of employing classification to identify home and work locations do not apply, as here we do not have densely sampled fine-grained locations to be used as features.

In this article we use the crowd-sourced location records from smartphones to estimate users' home and work locations. Based on the results, we then study two applications that are typically difficult problems but can be elegantly solved here. In the first application, leveraging the patterns in which people use their IP addresses, we design a method to accurately estimate the location of any given IP address. In the second application, based on the estimated precise home and work locations of each user, we estimate their commuting distance and then study the impact of the commuting distance on individual mood. As illustrated in Fig. 1, the data generated in mobile devices is transferred to the cloud for large-scale analytics and study.

The contributions of this article can be summarized as follows:

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Based on the information we can leverage, we design the estimation algorithm in two scenarios: the message-independent scenario, in which we only use the time and location information in each location record, and the message-dependent scenario, in which we leverage time, location, and messages to achieve a high estimation precision.

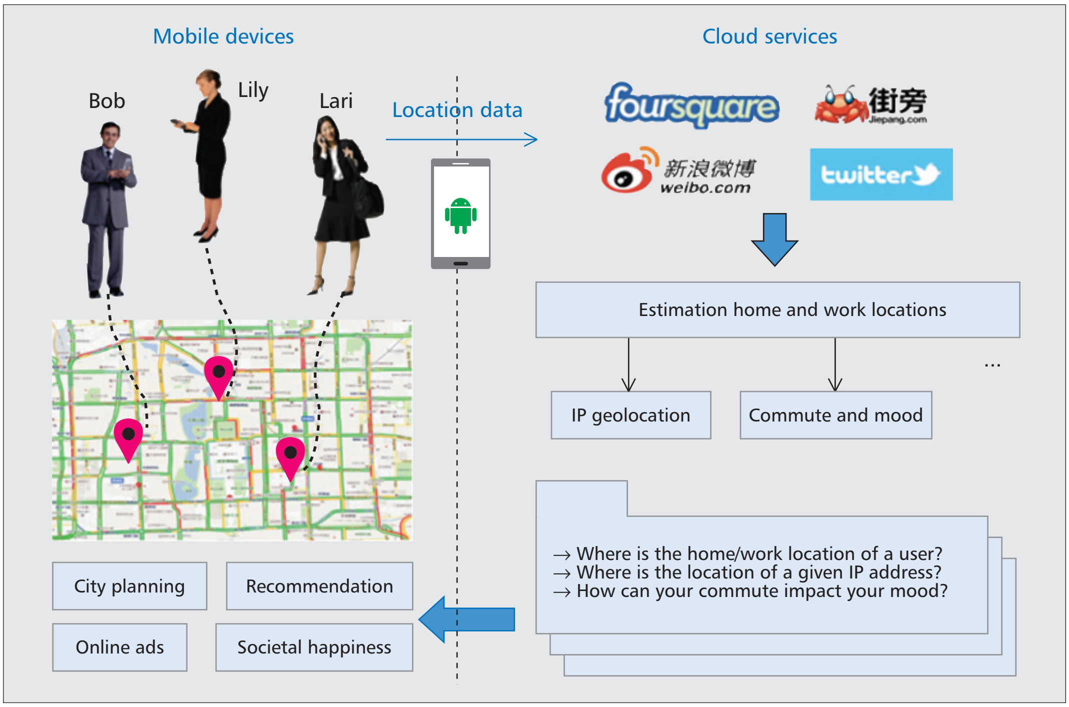

Figure 1. Crowdsourcing location records from smartphones for estimating users' home and work locations, and its applications. The mobile devices generate the data, and the cloud side is used to analyze the data.

- Leveraging the crowd-sourced location records, we propose the algorithms for estimating a user's home and work locations in both a message-independent scenario and a message-dependent scenario. Experimental results show that our algorithms achieve an estimation precision of 2 km for 88 percent of users in the message-independent scenario, and for 98 percent of users in the message-dependent scenario.
- Based on the estimated home and work locations for each user, we propose a novel method to estimate the location of an IP address using geolocation resources fundamentally different from existing approaches. Experimental results show that it achieves a median estimation error of 799 m (an order of magnitude smaller than existing database-driven approaches). Moreover, it has been deployed on a large scale in the Tencent Online Advertising Service[1].
- We leverage the estimated home and work locations to estimate the commuting distance of each user, and correlate it to the mood of users measured by the messages/tweets shared by users. We demonstrate unique findings across different societies regarding the impact of individuals' commuting distances on daily mood, which are difficult to obtain from survey-based research: longer commuting distance has a more negative impact on happiness but only beyond a distance threshold.

Dataset

We collected 4.9 million location records from 31,252 users in Japan between January 2010 and September 2012, and 7.1 million check-ins from 59,660 users in the U.S. between Januarty 2010 and October 2012, both from Foursquare. We collected 7.2 million location records from 70,794 users in China between May 2010 and February 2012 from Jiepang. We conducted the collection in two steps:

1 Collecting users' Twitter or Weibo accounts in Foursquare and Jiepang (many Foursquare/Jiepang users shared their Twitter/Weibo accounts in their profile pages).

2 Collecting the publicly shared location records of these accounts from Twitter.com and Weico.com.

Using APIs of Twitter.com and Weibo.com,[2] we collected only location records *publicly* shared from Foursquare and Jiepang, and did not invade the privacy of users whose location records are restricted to be accessed only by their friends.

Although the location records are generated on the *mobile* side, we collected them using the public APIs from the *cloud* side. The connection between the mobile and cloud side enables us to combine the advantages of the two: the data of the mobile and the power of the cloud. Therefore, we can study users' mobile behaviors on a large scale with low cost.

Estimating Home and Work Locations

We used crowd-sourced location records of a user to estimate the home and work location of that user. Typically, a location record includes the following information: time, location, and message (optional). Based on the information we can leverage, we designed the estimation algorithm in the following two scenarios:

- Message-independent scenario: In this scenario we only use the time and location information in each location record.

[1] http://www.tencent.com/en-us/ps/adservice.shtml

[2] A large portion of Foursquare and Jiepang users sync/share their check-ins *publicly* in Twitter and Weibo.

- Message-dependent scenario: In this scenario we leverage time, location, and messages to achieve a high estimation precision.

In the first scenario we provide a relatively coarse-grained estimation precision, but this approach can be used in all applications as long as we can collect location records. The approach in the second scenario can only be used in limited applications (where we can also collect messages) but it has a higher precision.

MESSAGE-INDEPENDENT SCENARIO

Based on the observation that a user is likely to be near his or her home at night and near the office during working hours, we cluster his location records at night and working hours respectively to see whether clustering is effective for estimating the home/work locations. For each user, we use a hierarchical clustering method [3] to cluster location records at night (from 8:00 pm to 7:59 am every day) and location records during work hours (from 8:00 am to 6:59 pm on weekdays), respectively. For this clustering, the distance between two location records is defined as the geographic distance of the two corresponding locations. We run the hierarchical clustering with a distance threshold of 150 m. In this case, we stop clustering if the distance between each pair of clusters is farther than 150 m; otherwise, the nearest two clusters are aggregated into a new cluster. We experimented with a range of distance thresholds and found that a distance threshold between 100 m and 500 m works well in practice for our dataset (we use a threshold of 150 m only to demonstrate the effectiveness of our technique). The produced clusters are the estimated home locations (referred to as home candidates) and the estimated work locations (referred to as work candidates) for each user. We reorder the candidates for each user by the number of location records in each candidate in descending order.

Next, we use users with known home or work locations to evaluate the effectiveness of our clustering method. We use as ground-truth the exact home or work locations of the small subset of users that explicitly state it to be their locations in their location record messages. For example, say a user Bob has shared a location record at the same place 10 times, and each time with a message such as "I am at home." Then, we can infer this place is the home location of Bob. We explore location records of all users by filtering their location record messages with keywords such as"Home," "At Home," and "My Home." We extract users who satisfy the following criteria:

- The user has shared his/her home location more than 10 times.
- All the home locations the user has shared are within a distance of 100 meters.

In this way, for our dataset, we get 753, 101, and 193 users with known home locations in China, Japan, and the U.S., respectively. Similarly, we get 417, 72, and 112 users with known office locations in China, Japan, and the U.S., respectively.

Figure 2a shows the errors when estimating the home locations of 753 users with known home locations in China, using the home candidates ranking the top three. For 88 percent of users, the home candidate ranking the first is within 2 km of the actual home location. The chance that the home candidates ranking the second and third are near to the actual home location is lower but still promising. The estimation for work locations demonstrates similar results as shown in Fig. 2b. Here we demonstrate the results from China; the results from Japan and the U.S. are similar.

MESSAGE-DEPENDENT SCENARIO

In this subsection, for each user, we select the most probable home/work candidate from all the home/work candidates produced using the clustering method described in the previous subsection. We use the messages in location records as auxiliary information for estimating the home/work location for a user.

- Strong indications. If in his messages a user mentions "off work/duty in one minute," "want/wait to be off work/duty," etc., the user is probably at work. Similarly, if a user mentions "leave home and go to work," etc., the user is probably at home.
- Weak indications. If a user mentions "go to work," "off work," "morning shift," "night shift," etc., we cannot be sure where the user is. However, if the corresponding location record is shared in an office building or a dining hall, the user is likely at work. If the corresponding record is shared in places such as a house, apartment, village or community, the user is likely to be at home.

Therefore, we can leverage the location record messages associated with a home/work candidate to estimate the probability that the candidate is near to the actual home/work location. A simple notion is to rank all the candidates for a user based on the strong and weak indications. However, our problem is different from a typical ranking problem in which an ordering for all the items is needed. Instead, here we only care about whether the candidate ranking the first is nearest to the actual home/work location, and the relative ordering of the other candidates is not important. In other words, our objective is to maximize the probability that the candidate nearest to the actual home/work location is ranked in front of other candidates.

We design a logistic regression to maximize this probability. We use the occurrences of the above mentioned indications and the ranking of the number of check-ins as features for a candidate, based on which we compare each pair of candidates and infer whether one candidate is more likely to be nearer to the actual home/work location than the other. In the training process, for each user, we

1) Calculate the feature differences between the candidate nearest the actual home/work location and any other candidate, and then set the corresponding label to 1, which means the former candidate is more likely to be near the actual home/work location than the latter.

2) Calculate the feature differences between the latter and the former, and set the corresponding label to 0.

We use the produced feature differences and labels as input and output, respectively, to train

Based on the observation that a user is likely to be near his or her home at night and near the office during working hours, we cluster his location records at night and working hours respectively to see whether clustering is effective for estimating the home/work locations.

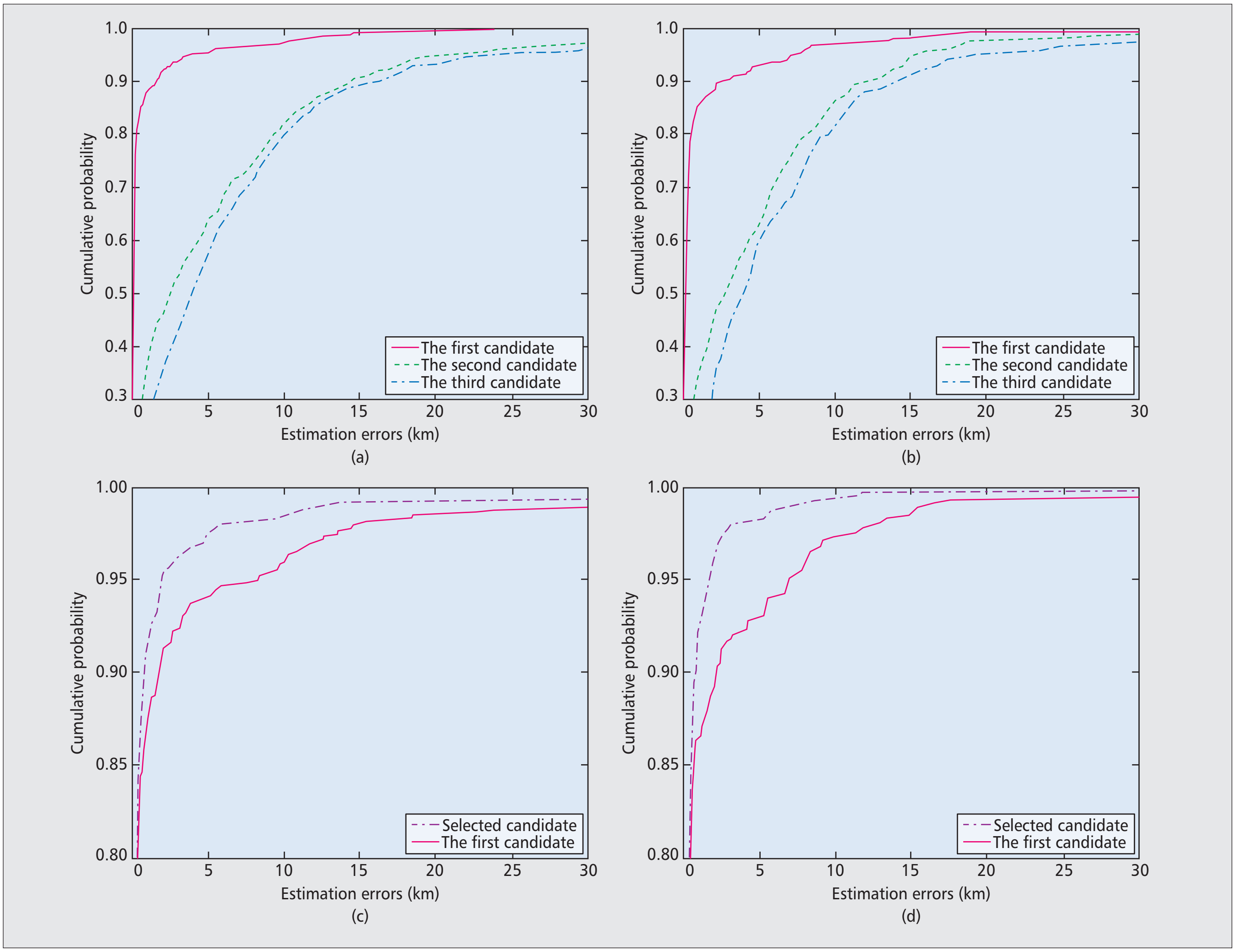

Figure 2. The errors between the estimated home/work locations and actual home/work locations: a) estimation errors to home; b) estimation errors to office; c) estimation errors to home; and d) estimation errors to office.

a logistic regression model. In the testing process, for each user, we use the trained model to compare the likelihood that one candidate is nearer the actual home/work location than the other for any two candidate pairs, and select the candidate that defeats most other candidates as the estimated home/work location.

Figures 2c and 2d show the estimation errors for

1) Simply selecting the candidate that has the most location records (the first candidate).

2) Selecting the candidate using the logistic regression based on supplemental information from location record messages with a five-fold cross validation (selected candidate).

For estimating both home and work locations, the estimation errors are significantly reduced. For 98 percent of users, the selected home/work candidates are within 2 km of the actual home/work locations. Here we demonstrate the results from China; the results from Japan and the U.S. are similar.

Based on the observation that phone call records contain both the time and the associated cell tower ID of each call event, Becker *et al.* [4, 5] leverage call records from service providers to characterize the daily travel, carbon emissions, and traffic volumes of a city. Particularly, they use these data to estimate the home and work locations of a user. Compared with their work, here we use large-scale crowd-sourced location records from mobile devices and leverage their characteristics (e.g. messages) to obtain much higher precision, i.e. 95th percentile error of about 1 km (about 0.6 miles) for both home and work compared to their precision of 3.86 miles (home) and 21.23 miles (work). Also note that they only have 37 users with ground-truth of home/work locations while we have hundreds of users whose locations are known.

Application: IP Geolocation

Accurately determining the geographic location of an Internet host, referred to as IP geolocation [6, 7], is the basis for effective location-aware applications. Existing work toward IP geolocation can be categorized into database-driven geolocation and delay measurement based geolocation approaches. Database-driven approaches typically maintain a database with records of IP/location mappings whose geolocation information comes from the Whois database, DNS, user contributions, etc. Database-driven geolocations are widely used for their simplicity of deployment and fast response time. However, such IP/location mappings are very coarse-grained and usually at most achieve a city-level precision [6]. Based on the idea that the network delay between two hosts has a positive correlation with the geographic distance between the two hosts, a number of delay measurement based approaches [8–11] have been proposed. These approaches estimate the location of the target IP address by measuring network delays from geographically distributed servers. Because of these delay measurement based approaches, the geolocation precision is improved to less than 10 km. However, these delay measurement based approaches have not been widely deployed in reality due to the following reasons [6]:

- High deployment cost: a large number of geographically distributed servers are needed for probing the target.
- Long response time: the response time of localizing a single PC is typically tens of seconds due to large measurement delays, which cannot meet the real-time requirement of many location-aware applications.

In this section we demonstrate our approach, Checkin-Geo [12], which is fundamentally different from existing approaches and leverages the crowed-sourced location records for precision IP geolocation. This crowd-sourced method enables us to achieve good precision and at the same time avoid the deployment/time cost of existing delay-measurement based approaches.

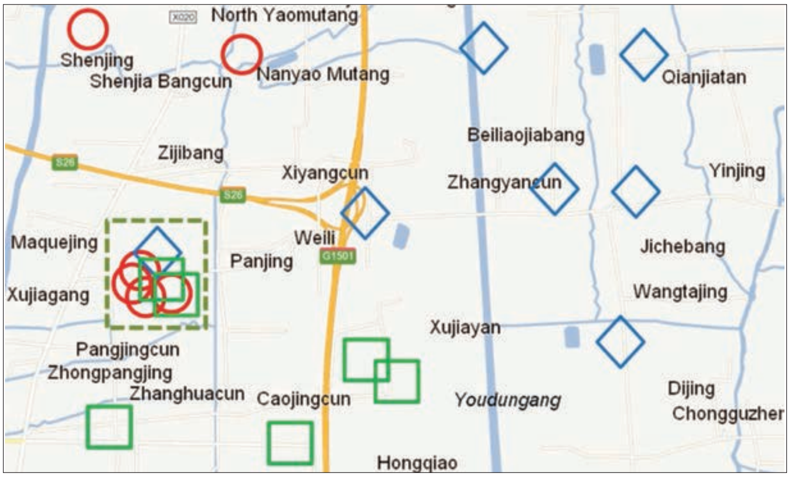

Figure 3. Determining the Geographic Location of an IP Segment. The location of an IP segment is estimated to be at the intersection area of the home/work candidates of users in this IP segment.

Estimation Method

We note that an IP/24 segment is usually assigned to a small area, e.g. a building. Therefore, we can estimate the location of an IP/24 segment leveraging the home/work locations of users using this IP segment. The idea is that this IP segment is likely to be in the intersection area of the home/work locations of users using this IP segment, as shown in Fig. 3. Assuming there are three users A, B, and C who have used an IP in the same IP segment S, we use the square, diamond, and circle to indicate the home/work candidates of the three users, respectively. As shown in Fig. 3, the home/work candidates of the three users are dispersed. However, each of the three users has at least one home/work candidate in the dashed rectangle region. In this case, there is a great chance that S is located at that region. We describe this idea in detail in the following three scenarios.

Colleague Scenario: Users A, B, and C are colleagues. They work in the same place, but live in different places. In this case, it is likely that A, B, and C are using the same IP segment when they are working. Therefore, we can estimate their commonly used IP segment to be at their working place. We can use the intersection area of their home and work places to estimate this place.

Neighbor Scenario: Users A, B, and C are neighbors. They live in the same place, but work in different locations. In this case, it is likely that A, B, and C are using the same IP segment when they are at home. Therefore, we can estimate their commonly used IP segment to be at their home location. Here we can similarly use the intersection area of their home and work locations to estimate this place.

Hybrid Scenario: A user A works in the place where B lives. In this case, the IP segment A uses when he is working is the same as the IP segment B uses when he is at home. Therefore, we can estimate their commonly used IP segment to be at A's work location or B's home location. Again, here we can use the intersection area of their home and work locations to estimate this place.

In all three of these scenarios, we can simply estimate an IP segment to be at the intersection area of the home/work locations of users using that IP segment, and do not need to identify whether a user uses the target IP segment at home or at work. Since the notion of *intersection* is used here, limited estimation errors of home/work locations would not impact the intersection effects, and thus the home/work locations are estimated using the approach in *the message-independent scenario*.

We use clustering [3] to identify the intersection area, with the following two objectives, respectively:

- Find the intersection with the most users (maximizing users).
- Find the intersection with the most location records (maximizing records).

For a given IP segment, we cluster all the home/work candidates in it with a predefined distance threshold, and get a number of clusters. In this case, home/work candidates in the same cluster are within the predefined threshold, and home/work candidates from different clusters have a distance farther than the predefined threshold. We select the cluster with the most users for the objective of maximizing users, and select the cluster with the most records for the objective of maximizing records. The location of an IP segment is then estimated to be at the center of the intersection.

Evaluation

Database-driven geolocations are widely used in location-based applications, while delay measurement based geolocations are not used in real applications due to their high deployment cost and long response time [6]. We propose Checkin-Geo to be a feasible commercial IP geolocation technique with good scalability,

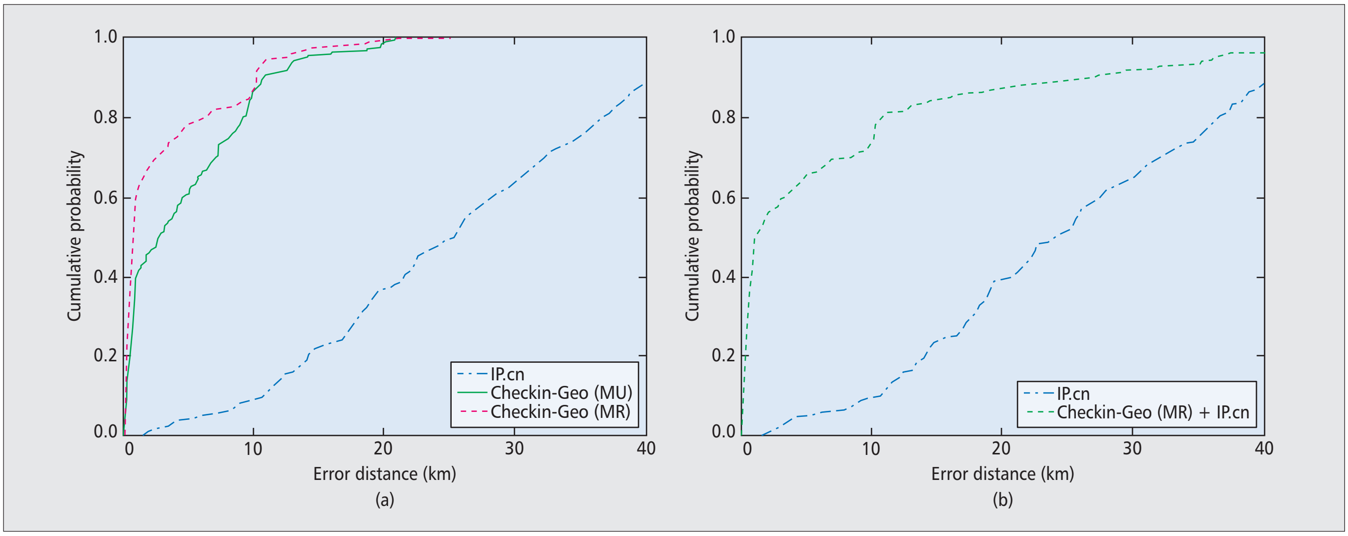

Figure 4. Error distances for (a) the 243 IPs that checkin-geo covers and (b) All the 301 IPs in our test dataset.

which can replace/complement existing database-driven geolocations, and provide fine-grained and fast geolocations for real applications. In this section we compare Checkin-Geo with IP.cn,[3] which is based on database-driven techniques and is also the most popular IP geolocation service in China.

We use IPs with known ground-truth locations to compare Checkin-Geo with IP.cn. In order to collect the ground-truth dataset, we set up a web page with a map to enable users to mark their locations. This web page records the latitude and longitude coordinates of the marked point, as well as the IP address that the user uses. We then request volunteers to access our web page and contribute their IPs and locations. We filter out IPs from mobile devices based on the user-agent field of HTTP request, and get a total of 301 IPs with ground-truth locations.

Checkin-Geo covers 243 of the 301 IPs (80.7 percent). We first compare Checkin-Geo with IP.cn using the 243 IPs that Checkin-Geo covers. We then use all of the 301 IPs as the test data, to evaluate the scenario where Checkin-Geo is combined with IP.cn (use IP.cn for IPs that Checkin-Geo does not cover) to provide a fast and precise IP geolocation, which we expect to be the model for Checkin-Geo to be used for general location-based applications.

We use the error distance, i.e. the distance from the estimated location to the actual location, to quantify the geolocation error. Figure 4a demonstrates the cumulative probability of the error distance for the 243 IPs that Checkin-Geo covers. We use legends Checkin-Geo (MU) and Checkin-Geo (MR) to represent Checkin-Geo with the objectives of maximizing users and maximizing records, respectively. We can see that Checkin-Geo significantly outperforms IP.cn. We use the metric median error to further compare Checkin-Geo and IP.cn. Since the average error can be influenced by abnormal or large errors from a few IPs, the median error is widely used in geolocation systems [8, 9] to indicate the average case for estimation error. The median errors of IP.cn, Checkin-Geo (MU), and Checkin-Geo (MR) are 24,783 m, 2,829 m, and 799 m, respectively, which indicates that for an average case, Checkin-Geo (MR) achieves an estimation error with an order of magnitude smaller than IP.cn.

We have investigated the causes for the performance difference between Checkin-Geo (MR) and Checkin-Geo (MU). Let us consider a case in which several users in the same IP segment S have an intersection area with a small number of location records, but this area is not the home/work location of anyone of these users. In Checkin-Geo (MU), these intersection areas are likely to be estimated to be the location of S. Therefore, Checkin-Geo (MR) is more applicable in real applications.

We next consider the scenario where Checkin-Geo is expected to be used in reality: we reuse existing database-driven geolocations for IPs which Checkin-Geo does not cover. Therefore, Checkin-Geo can complement existing widely used database-driven techniques by improving the precision significantly without increasing response time or deployment cost. Figure 4b shows the results for these cases for all 301 IPs in our dataset. The median errors of IP.cn, Checkin-Geo (MR) + IP.cn are 25,732 m, and 1,079 m, respectively. Therefore, if we combine Checkin-Geo with IP.cn, we achieve a median estimation precision of about 1 km for all IPs, which is promising for a number of location-based applications such as targeted advertising.

We deploy Checkin-Geo in Tencent, a large Internet company in China with about 800 million users. Evaluation results based on 5000 ground-truth samples show that Checkin-Geo achieves an estimation precision within 2.5 km (such precision can be of significant help in practice for location-based applications such as targeted advertising) for 68 percent of these users.

Application: Impact of Commuting on Mood

Individual mood is highly related with health, creativity, decision-making, and immune response. Moreover, individual mood is an important dimension to indicate people's happi-

[3] http://www.ip.cn.

ness, which is a universally recognized goal for human existence and also part of people' working existences to both get compensated to earn a livelihood as well as to be happy. So, does your working pattern have an impact on your mood and how can you improve your daily mood? Validated answers to these questions are beneficial to workers, corporations, and urban planners, who can find ways to improve workers' happiness through elegantly redesigning their working patterns and facilitating easier commutes.

Social and psychological scientists invest large amounts of research to study the impact of working patterns on individual mood [13, 14]. Typically, their research is based on samples or surveys asking participants to record/recall their behavior, thoughts, and feelings. In these approaches, because of the significant manual work involved processing these surveys, only small samples of people are surveyed in a limited region (a class/university/city) and for a limited duration (a day, a few weeks, etc.). In this section we use commuting as an example to demonstrate how we can leverage crowd-sourced location records to study the impact of working patterns on mood, on a large scale but with a low cost.

INFERRING MOOD ASSOCIATED WITH LOCATION RECORDS

We use the message in a location record to infer a user's mood at the specific time associated with the location record. For example, if a message is "I am enjoying the food!" we can infer the user is happy at that time. Similarly, if a message is "Oh, terrible traffic," we can infer the user is unhappy. A message without explicit sentimental polarity is believed to be neutral.

For both Chinese and English, we find three publicly available sentiment tools each:

- Chinese: *ROST*, *NTUSD,* and *Keenage.*
- English: *Sananalytics*, *OpinionFinder* and *General Inquirer*.

We use 3000 Chinese and 3000 English messages with manually tagged sentiments (positive, negative, and neutral) to evaluate the effectiveness of these tools. Each of these messages was labeled by two persons independently, and messages with inconsistent sentiment were already kicked out and not counted in the 3000. We then choose ROST and OpinionFinder to infer the sentiments for Chinese and English, respectively, as they perform best in our evaluation. We could only find one publicly available sentiment analysis tool for Japanese, Jnlp, which we used. All three tools produce a positive, negative, and zero value to indicate a positive, negative, and neutral sentiment, respectively.

IMPACT OF COMMUTING DISTANCE ON INDIVIDUAL MOOD

Since we can estimate the home and work locations for each user, we can then calculate the commuting distance for each user. As we have to use the messages in location records to estimate sentiment, here we try to make full use of the messages and use the method in *the message-dependent scenario* to estimate users' home and work locations. Note that the calculated distances are straight-line distances between home and work, and thus the actual commuting distance associated with each user is likely longer depending on the commuting trajectory.

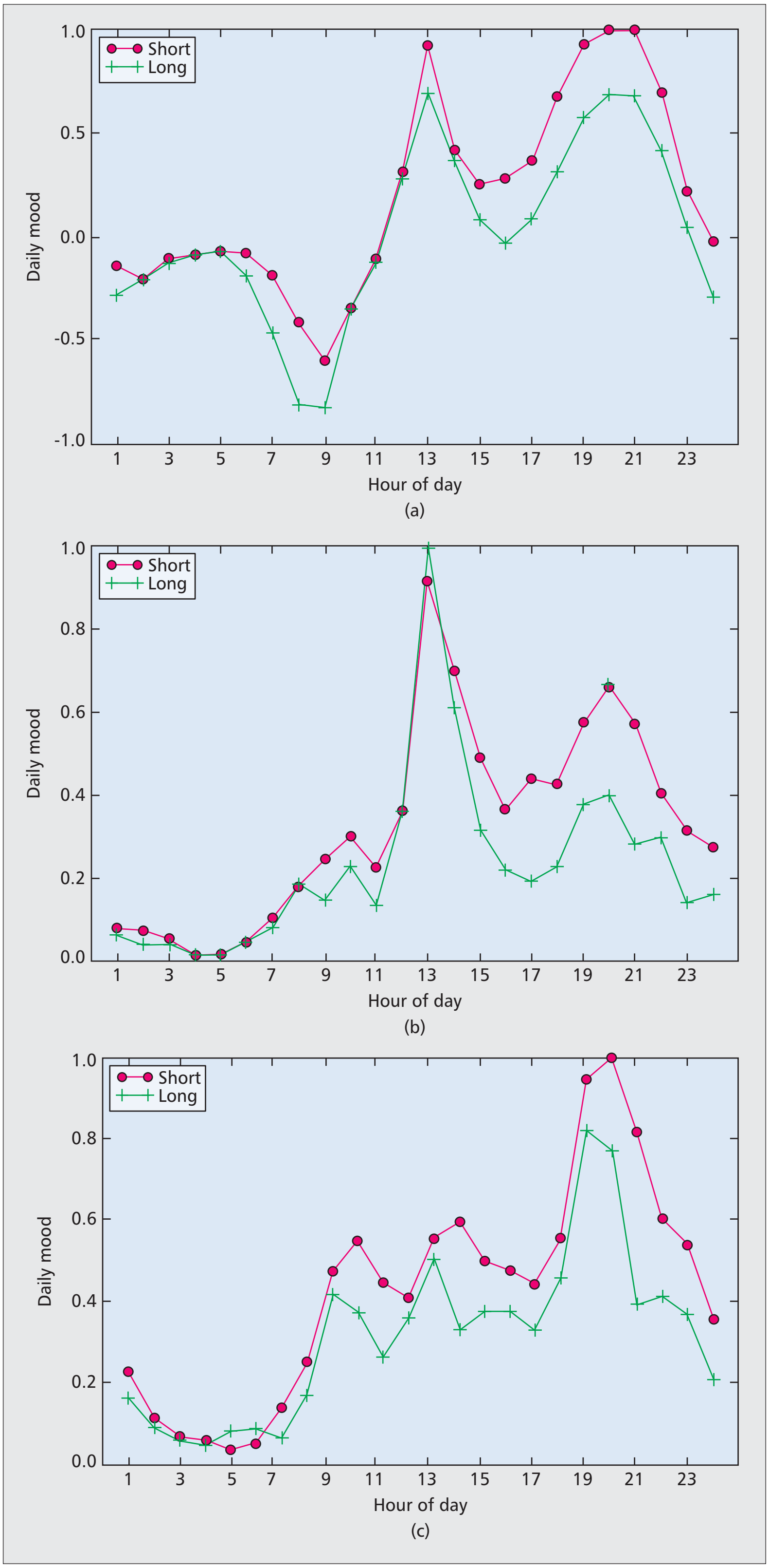

Figure 5. Daily Mood Variations with Hour of Day: a) China; b) Japan; and c) US.

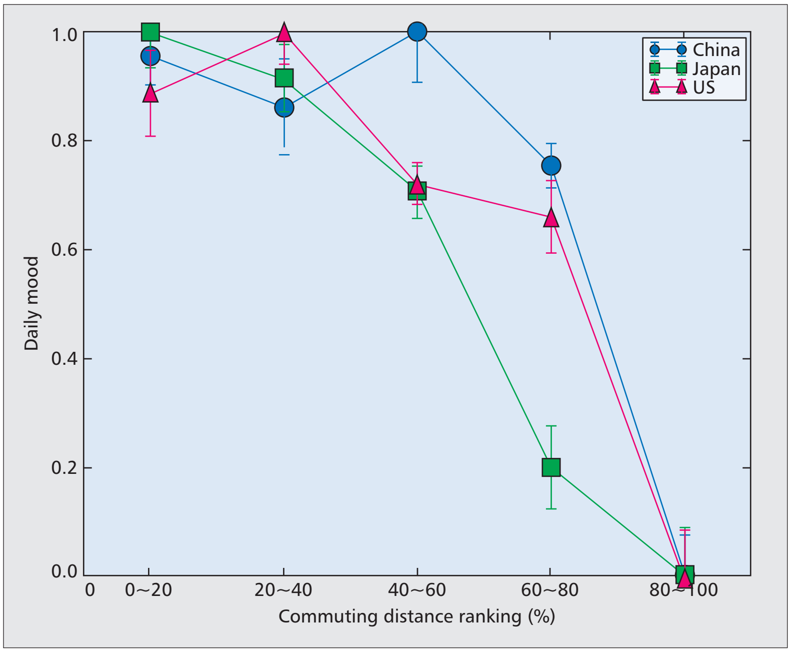

Figure 6. Daily mood variations with commuting distances.

We first compare the normalized hourly mood of users with long and short commuting distances on a detailed hourly basis. As shown in Fig. 5 (the maximum value is scaled to 1 to make figures more clarified), the mood for the group with commuting distances shorter than the median commuting distance (group A) is generally better than the group with longer distances than the median (group B), with a significance level of $P<0.01$ for China and Japan and $P<0.02$ for the U.S. In particular, group A is much happier during the morning and evening commuting times, which might be because

- Commuting is typically associated with a negative mood.
- Group B needs to get up earlier and early rising has a negative effect on mood (also as noted by others regarding their health and well-being).

When we explored our dataset, we found a number of early morning users who frequently complain they are on the train/subway in the early morning with tweets such as "sleepy on the train," " I hate early morning shift," and "a long way to the office." Their complaints are shared in a very regular manner, almost the same time every day. In the very beginning of the research for this article, we were inspired by these tweets and had the idea of studying the impact of commuting on mood using these data. Further, group A is much happier in the afternoon (2 pm to 5 pm), which suggests that people who have a long commute (group B) might be more likely to be tired in the afternoon.

Next we compare the normalized mood of users with different commuting distances. Users in each country are sorted by increasing commuting distances to enable us to compare across different societies. As shown in Figure 6 (the mood of the three countries is scaled to the same range of [0,1] to make the figure more clarified), for people who have relatively short commuting distances, their mood is not impacted significantly by their commuting distances. However, for people who have relatively long commuting distances, their mood drops rapidly with increasing commuting distance. These patterns suggest that a long commuting distance, above a threshold (the thresholds in Japan, the U.S., and China are about 10.4 km, 8.4 km, and 11.5 km, respectively), tends to make people unhappy.

In this example application, our consistency with existing results and reconfirmation demonstrate the reliability of leveraging crowd-sourced location data for social/psychological studies in a fast/automated way, while the differences and new findings show the power of large-scale study across different societies using the crowd-sourced data.

CONCLUSION

In this article we propose to leverage large-scale crowd-sourced location data from smartphones to estimate users' home and work locations. Experimental results demonstrate that our approach achieves a good estimation precision. We further study how to use the estimated home and work locations for two applications. In the first application, we estimate the location of a given IP address, which has an estimation error with an order of magnitude smaller than existing approaches. In the second application, we study the impact of commuting distance on individual mood. We confirm a survey-based result that a longer commuting distance is associated with more negative moods [15]. But unlike their results, we have found critical new insights, i.e. there is a threshold for that distance: happiness does not vary significantly below that threshold, and drops rapidly above it. This demonstrates the power of leveraging crowd-sourced data on the mobile side and the computation performance in the cloud side, which enables us to have unique findings that are difficult to obtain through traditional survey-based research. In both applications, our approaches based on crowdsourcing avoid the high cost involved in traditional methods, and perform well on a large scale.

It should be noted that we focus on home/work locations in this article since they are the places where everyone spends the most time. The home/work locations characterize the life of a person and usually are more important compared with other places such as the supermarket or gym. However, the methods proposed in this article can also be extended for estimating other places by similarly analyzing the relationship between the pattern of users' location records and the new place.

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Biographies

Hao Liu (liuhao.buaa@gmail.com) received his B.S. degree in software engineering from Beihang University, China, in 2009. He is currently a Ph.D. candidate in the Department of Computer Science and Technology, Tsinghua University, China. His research interests include high-speed transport protocols, energy saving in smartphones, and data mining in location-based services.

Yuezhi Zhou (zhouyz@mail.tsinghua.edu.cn) obtained his Ph.D. degree in computer science and technology from Tsinghua University, China in 2004, and is now working as an associate professor in the same department. His research interests include ubiquitous/pervasive computing, distributed systems, mobile devices and systems. He has published over 30 technical papers in international journals or conferences. He received the IEEE Best Paper Award at the 21st IEEE AINA International Conference in 2007. He is a member of IEEE and ACM.

Yaoxue Zhang (zyx@csu.edu.cn) received his B.S. degree from Northwest Institute of Telecommunication Engineering, China, and received his Ph.D. degree in computer networking from Tohoku University, Japan in 1989. He is a fellow of the Chinese Academy of Engineering. He is a professor in the School of Information Science and Engineering, Central South University, and the Department of Computer Science and Technology, Tsinghua National Laboratory for Information Science and Technology (TNList), Tsinghua University. His major research areas include computer networking, operating systems, ubiquitous/pervasive computing, transparent computing, and active services.

Mobile Code Offloading: From Concept to Practice and Beyond

Huber Flores, Pan Hui, Sasu Tarkoma, Yong Li, Satish Srirama, and Rajkumar Buyya

Huber Flores and Satish Srirama are with the University of Tartu, Estonia.

Pan Hui is with the Hong Kong University of Science and Technology. He is also affiliated with Telekom Innovation Laboratories and Aalto University.

Sasu Tarkoma is with the University of Helsinki.

Yong Li is with Tsinghua University.

Rajkumar Buyya is with the University of Melbourne.

ABSTRACT

The emerging mobile cloud has expanded the horizon of application development and deployment with techniques such as code offloading. While offloading has been widely considered for saving energy and increasing responsiveness of mobile devices, the technique still faces many challenges pertaining to practical usage. In this article, we adopt a systemic approach for analyzing the components of a generic code offloading architecture. Based on theoretical and experimental analysis, we identify the key limitations for code offloading in practice and then propose solutions to mitigate these limitations. We develop a generic architecture to evaluate the proposed solutions. The results provide insights regarding the evolution and deployment of code offloading.

INTRODUCTION

Mobile and cloud computing convergence is shifting the way in which telecommunication architectures are designed and implemented [1]. In the presence of network connectivity to bind mobile and cloud resources, the potential of code offloading lies in the ability to sustain power-hungry applications by releasing the energy consuming resources of the smartphone from intensive processing. Multiple research works have proposed different code offloading strategies to empower smartphone apps with cloud-based resources [5–9]. However, the utilization of code offloading is debatable in practice as the approach has been demonstrated to be ineffective in increasing the remaining battery life of mobile devices [2].

The effectiveness of an offloading system is determined by its ability to infer where the execution of code (local or remote) represents less computational effort for the mobile, such that by deciding *what*, *when*, *where*, *and how to offload* correctly, the device obtains a benefit. Code offloading is productive when the device saves energy without degrading the normal response time of the apps, and counterproductive when the device wastes more energy executing a computational task remotely rather than locally. Current works offer partial solutions that ignore the majority of these considerations in the inference process. Most of the proposals demonstrate the utilization of code offloading in a controlled environment by connecting to nearby low-latency servers (e.g., lab setups) and inducing the code to become resource intensive during runtime [3] (e.g., passing an input that requires lot of processing). As a result, in practice, in most cases, computational offloading is counterproductive for the device [2, 7]. Thus, at this point, the main questions about the strategy are *can code offloading be utilized in practice?* and *what are the issues that prevent code offloading from yielding a positive result?*

This article takes a systemic perspective to answer these questions by analyzing in detail the components that form an offloading architecture. The work highlights the drawbacks of code offloading architecture and proposes solutions that exploit intrinsic cloud features (e.g., massive data analysis) in order to overcome the issues. Unlike other offloading architectures that focus on specific issues in the offloading process, such as *what or when*, the aim of our proposal is to generalize an offloading service that can estimate the most effective offloading outcome for the device by focusing on multiple perspectives at once (*what*, *when*, *where*, *how*, etc.) Based on the solutions, we present and evaluate use cases, which give insights about how the computational offloading process can be managed to accelerate the response time of mobile applications without inducing extra overhead in the device, and to reduce the offloading traffic of computational requests. Moreover, the work also emphasizes the importance of the cloud beyond the ordinary provisioning of services.

The rest of the article is structured as follows. We describe the essential elements and functionality of an offloading architecture. We describe an overview of the obstacles that inhibit the adoption of code offloading. We present the solutions we envision to potentiate the applicability of the approach, and finally, we evaluate how much gain can be obtained by leveraging the proposed solutions.

BACKGROUND

Code offloading has evolved considerably from its initial notion of cyber-foraging [1]. Table 1 describes most relevant proposals in code offloading. The table compares the key features of the offloading architectures: the main goal,

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	Code offloading strategies				Mobile perspective	Cloud perspective
Framework	Main goal	Code profiler	Offloading adaptation context	Offloading characterization	Applications effect	Features exploited (Besides server)
MAUI [5]	Energy saving	Manual annotations	Mobile (*what, when*)	None	Low resource consumption, Increased performance	None
CloneCloud [6]	Transparent code migration	Automated process	Mobile (*what, when*)	None	Accelerate responsiveness	None
ThinkAir [8]	Scalability	Manual annotations	Mobile + Cloud (*what, when, how*)	None	Increased performance	Dynamic allocation and destruction of VMs
COMET [9]	Transparent code migration (DSM)	Automated process	Mobile (*what, how*)	None	Average speed gain 2.88×	None
EMCO [10]	Energy saving, Scalability (Multi-tenancy)	Automated process	Mobile + cloud (*what, when, where, how*)	Based on historical crowdsourcing data	Based on context (Low resource consumption, increased responsiveness, etc.)	Dynamic allocation and destruction of VMs, Big data processing, Characterization-based utility computing

Table 1. Code offloading approaches from the mobile and cloud perspectives.

how code is profiled, the adaptation context, the characterization of the offloading process, and how code offloading is exploited from the mobile and cloud perspectives. From the table, the main goal defines the actual benefit of using the associated framework. The mechanism used to profile code provides information about the flexibility and integrability of the system. The adaptation context specifies the considerations taken by the system to offload. The characterization means whether the offloading system has a priori knowledge or not about the effects of code offloading for the components of the system. Finally, the exploitation highlights the mobile benefits obtained from going cloud-aware, and the features of the cloud that are leveraged to achieve those benefits. Moreover, we can also observe that currently, most of the effort has been focused on providing the device with an offloading logic based on its local context. To gain an understanding of the actual benefits and functionality of code offloading, we described the key properties of an offloading architecture.

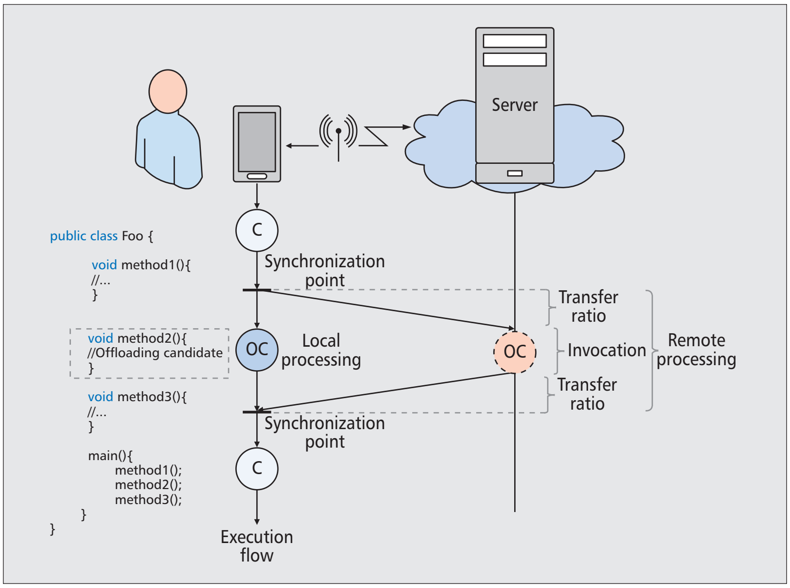

Figure 1. A code offloading architecture: components and functionalities.

Code Offloading

Offloading is the opportunistic process that relies on remote servers to execute code delegated by a mobile device. In this process, the mobile is granted the local decision logic to detect resource-intensive portions of code, such that in the presence of network communication, the mobile can estimate where the execution of code will require less computational effort (remote or local), which leads the device to save energy [9]. The evaluation of the code requires consideration of different aspects, for instance, *what code to offload* (e.g., method name); *when to offload* (e.g., round-trip times thresholds); *where to offload* (e.g., type of cloud server); *how to offload* (e.g., split code into *n* processes); and so on. Most of the proposals in the field do not cover all these aspects; thus, we describe a basic offloading architecture, which is shown in Fig. 1. The architecture consists of two parts, a client and a server. The client is composed of a code profiler, system profilers, and a decision engine. The server contains the surrogate platform to invoke and execute code. Each component is described below.

The *code profiler* is in charge of determining *what to offload*. Thus, portions (*C*) of code — *Method, Thread*, or *Class* — are identified as offloading candidates (OCs). Code partitioning requires the selection of the code to be offloaded. Code can be partitioned through a diversity of strategies; for instance, a software developer can explicitly select the code that should be offloaded using special static annotations [5] (e.g., *@Offloadable*, *@Remote*). Other strategies

Annotations can cause poor flexibility to execute the app in different mobile devices. Similarly, automated strategies are shown to be ineffective and require major low-level modifications in the core system of the mobile platform, which may lead to privacy and security issues.

analyze the code implicitly during runtime by an automated mechanism [6]. Thus, once the application is installed in the device, the mechanism selects the code to be offloaded. In order to estimate whether or not a portion of code is intensive, the mechanism implements strategies such as static analysis and history traces. Automated mechanisms are preferable over static ones as they can adapt the code to be executed in different devices. Thus, automated mechanisms overcome the problem of brute force development in code offloading, which consists of adapting the application every time it is installed in a different device.

System profilers are responsible for monitoring multiple parameters of the smartphone, such as available bandwidth, data size to transmit, and energy to execute the code. These parameters influence *when to offload* to the cloud. Conceptually, the offloading process is optional, and should take place when the effort required by the mobile to execute the OC is lower in the case of remote invocation than local execution. Otherwise, offloading is not encouraged as excessive amount of energy and time is consumed in transmission of data to the cloud.

The *decision engine* is a reasoner that infers *when to offload* to the cloud. The engine retrieves the data obtained by the system and code profilers, and applies certain logic over them (linear programming, fuzzy logic, Markov chains, etc.) such that the engine can measure whether or not the handset obtains a concrete benefit from offloading to the cloud. If the engine infers a positive outcome, the mechanism to offload is activated, and the code is invoked remotely; otherwise, the code is executed locally. A mobile offloads to the cloud in a transfer ratio that depends on the size of the data and the available bandwidth [4]. Usually, when code offloading is counterproductive for the device, it is due to a wrong inference process, which is inaccurate based on the scope of observable parameters that the system profilers can monitor [7].

The *surrogate platform* is the remote server located in proximity of the device or in the cloud, which contains the environment to execute the intermediate code sent by the mobile (e.g., Android-x86). The remote invocation tends to accelerate the execution of code as the processing capabilities of the surrogate are higher than those of most smartphones, which is translated into higher app responsiveness for the mobile user.

Challenges and Technical Problems

Computational offloading for smartphones has not changed drastically from its core principles [3]. However, the effectiveness of its implementation in practice shows it to be mostly unfavorable for the device outside controlled environments. In fact, the utilization of code offloading in real scenarios proves to be mostly negative [7], which means that the device spends more energy in the offloading process than the actual energy that is saved. As a consequence, the technique is far from being adopted in the design of future mobile architectures. Our goal is to highlight the challenges and obstacles in deploying code offloading. The issues are described below.

Inaccurate Code Profiling — Code profiling is one of the most challenging problems in an offloading system, as the code has non-deterministic behavior during runtime, which means that it is difficult to estimate the running cost of a piece of code considered for offloading. A portion of code becomes intensive based on factors [7] such as the user input that triggers the code, type of device, execution environment, available memory, and CPU. Moreover, once code is selected as *OC*, it is also influenced by many other parameters of the system that come from multiple levels of granularity (e.g., communication latency, data size transferred). As a result, code offloading suffers from a sensitive trade-off that is difficult to evaluate; thus, code offloading can be productive or counterproductive for the device. Most of the proposals in the field are unable to capture runtime properties of code, which makes them ineffective in real scenarios.

Integration Complexity — The adaptation of code offloading mechanisms within the mobile development life cycle depends on how easily the mechanisms are integrated and how effective the approach is in releasing the device from intensive processing. However, implementation complexity does not necessarily correlate with effective runtime usage. In fact, some of the drawbacks that make code offloading fail are introduced at development stages; for example, in the case of code partitioning, which relies on the expertise of the software developer, portions of code are annotated statically, which may cause unnecessary code offloading that drains energy. Moreover, annotations can cause poor flexibility to execute the app in different mobile devices. Similarly, automated strategies are shown to be ineffective and require major low-level modifications in the core system of the mobile platform, which may lead to privacy and security issues.

Dynamic Configuration of the System — Next generation mobile devices and the vast computational choices in the cloud ecosystem make the offloading process a complex task, as depicted in Fig. 2a. Although savings in energy can be achieved by releasing the device from intensive processing, a computational offloading request requires meeting the requirements of a user's satisfaction and experience, which is measured in terms of responsiveness of the app. Consequently, in the offloading decision, a smartphone has to consider not just potential savings in energy, but also has to ensure that the acceleration in the response time of the request will not decrease. This is an evident issue as the computational capabilities of the latest smartphones are comparable to some servers running in the cloud. For instance, consider two devices, Samsung Galaxy S (i9000) and Samsung Galaxy S3 (i9300), and two Amazon instances, m1.xlarge and c3.2xlarge. In terms of mobile application performance, offloading intensive code from

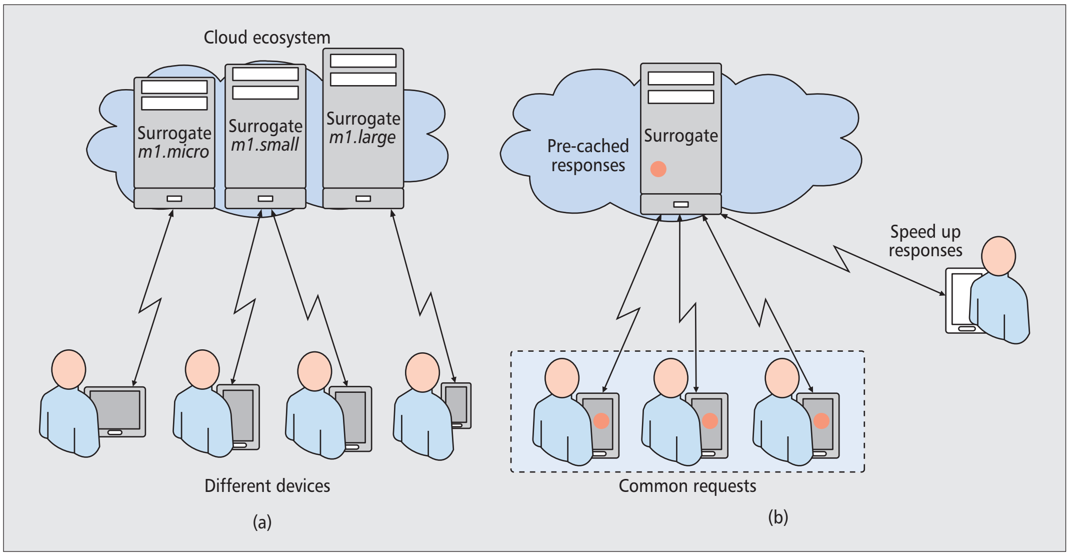

Figure 2. a) Characterization of the offloading process that considers smartphone diversity and the vast cloud ecosystem; b) acceleration of a code offloading request via pre-cached results.

From a cloud point of view, allocation of resources cannot occur indiscriminately based on processing capabilities of the server as the use of computational resources is associated with a cost. Consequently, the need for policies for code offloading systems are necessary considering both the mobile and the cloud.

i9000 to m1.xlarge increases the responsiveness of a mobile application at comparable rates to an i9300. However, offloading from i9300 to m1.xlarge does not provide the same benefit. Thus, to increase responsiveness it is necessary to offload from i9300 to c3.2xlarge (refer to results later). It is important to note, however, that constantly increasing the capabilities of the back-end does not always speed up the execution of code exponentially, as in some cases, the execution of code depends on how the code is written; for example, code is parallelizable for execution into multiple CPU cores (parallel offloading) or distribution into large-scale graphics processing units (GPU offloading).

Offloading Scalability and Offloading as a Service — Typically, in a code offloading system, the code of a smartphone app must be located in both the mobile and server as in a remote invocation, a mobile sends to the server, not the intermediate code, but the data to reconstruct that intermediate representation such that it can be executed. As a result, an offloading system requires the surrogate to have a similar execution environment as the mobile. To counter this problem, most of the offloading systems propose relying on the virtualization of the entire mobile platform in a server (Android-x86, .Net framework, etc.), which tends to constrain the CPU resources and slows down performance. The reason is that a mobile platform is not developed for large-scale service provisioning. As a result, offloading architectures are designed to support one user at a time, in other words, one server for each mobile [8, 11]. This restrains the features of the cloud for multi-tenancy and utility computing. Moreover, while a cloud vendor provides the mechanisms to scale service-oriented architectures (SOAs) [12] on demand, such as Amazon autoscale, it does not provide the means to adapt such strategies to a computational offloading system as the requirements to support code offloading are different. The requirements of a code offloading system are based on the perception that the user has of the response time of the app. The main insight is that a request should increase or maintain a certain quality of responsiveness when the system handles heavy loads of computational requests. Thus, a code offloading request cannot be treated indifferently. The remote invocation of a method has to be monitored under different systems' throughput to determine the limits of the system to not exceed the maximum number of invocations that can be handled simultaneously without losing quality of service. Furthermore, from a cloud point of view, allocation of resources cannot occur indiscriminately based on the processing capabilities of the server as the use of computational resources is associated with a cost. Consequently, the need for policies for code offloading systems are necessary considering both the mobile and the cloud.

Mobile Crowdsourcing for Computational Offloading: Vision

Despite the large amount of related works [5–11], the instrumentalization of apps alone is insufficient to adopt computational offloading in the design of mobile systems that rely on the cloud. Computational offloading in the wild can impose more computational effort on the mobile rather than reduce processing load [2]. In contrast to existing works, we overcome the limitations of computational offloading in practice by analyzing how a particular smartphone app behaves in a community of devices [7]. Computational crowdsourcing strategies are viable solutions to understand potential problems in software applications with high accuracy (bugs, leaks, etc.). The main advantage of relying on a community is to capture the diversity of cases in which an applications works. Our fine-grained framework at code level is inspired by the coarse-grained solution Carat [13], which attempts to

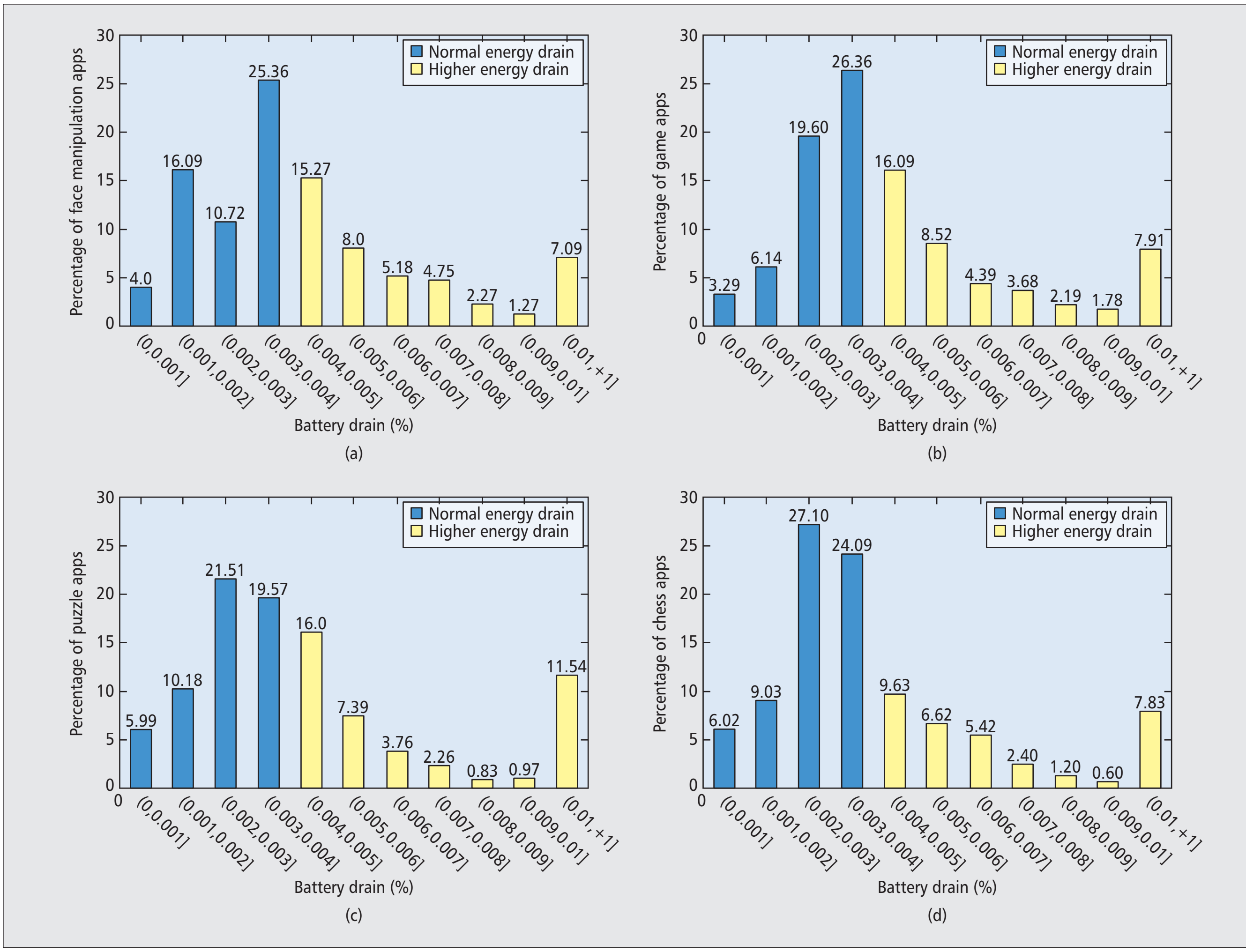

Figure 3. Smartphone apps that depict higher energy drain.

find energy bugs and hogs in mobile apps. Carat analyzes big repositories of data that depict the runtime behavior of a mobile app in order to find anomalies that can turn into customized recommendations for preferable configurations, (e.g., apps to kill) the mobile user can apply to make the battery life of his/her device longer

We believe that in the same way Carat detects energy anomalies, it is possible to determine the conditions or configurations for offloading smartphone applications. For instance, by applying the Carat method [13] over a subset of data (≈328,000 apps), we can get an idea about what is a resource-intensive app. Based on this data, which is collected in a real life deployment, we develop several case studies to motivate the viability and applicability of our approach. The subset contains for each app the expected percentage of energy drain. The Carat method consists of determining the energy drain distribution of a particular app, and then comparing it with the average energy drain distribution of other apps running in the device. The key insight of the method is to determine the possible overlap between application energy distributions in order to detect anomalies in application energy usage.

By analyzing apps' categories and specific purposes, we develop four case studies, which consist of face manipulation, games, puzzles, and chess. Figure 3 shows the results of the case studies. For each case, we extract a group from the subset of apps, where each app in a group is different from the rest. The four groups consist of about 550 face manipulation apps, about 7805 game apps, about 717 puzzle apps, and about 166 chess apps. Each group is compared to the average energy drain of the subset, which excludes its energy drain. From the results, we can observe that around 43.84 percent, 44.56 percent, 42.75 percent, and 33 percent of apps implement computational operations that require higher energy drain than normal, which is a significant number of apps. As consequence, computational offloading is required (e.g., customized alarm) to overcome the extra overhead introduced by the apps when showing resource-intensive behavior.

Generally speaking, higher granularity data is able to provide insights about the right conditions of these apps to offload to the cloud. However, beyond equipping the apps with computational offloading, this requires the mechanisms to be instrumentalized with the ability to record their own local/remote execution, such that we can capture more specific details about what induces the code to become resource-intensive and the runtime behavior of the code in the device/surrogate. Thus, it is reasonable that the characterization of the computational offloading process can be modeled through a community of devices, such that by taking advantage of the huge amount of devices that connect to cloud, it can be possible to foster a more effective offloading strategy for smartphones.

Implicit crowdsourcing that does not need incentives, but rather is extrapolated from application usage, can be used to collect history traces of the computational offloading process across the entire system. Traces can be analyzed using cloud analysis features to extract the characterization. The purpose of the characterization is to define the effect of remote code execution in different conditions and configurations, where a condition depicts the interaction aspects of the user with the mobile (e.g., available memory and CPU, input variability), and a configuration represents the state of the components of the systems (bandwidth size, capacity of the cloud-surrogate, performance metrics of the mobile/back-end, etc.). In this manner, we can find out the most accurate configurations (what to offload) for a specific application in a particular device based on multiple criteria, such as type of surrogate (where to offload) and conditions of the system (when to offload). Additionally, it can be possible to determine offloading plans (how to offload) that enable the device to schedule code offloading operations (e.g., computational parallelization of code).

Furthermore, the characterization can also be utilized to identify reusable results. A reusable result is a portion of code that is commonly offloaded by multiple devices. These results can be cached in the cloud to respond to duplicate offloading requests from other devices. Logically, this accelerates the offloading process as the surrogate avoids the invocation time of the request. We envision that as part of the characterization process, pre-cached functionality from the entire mobile application can be requested on demand, as depicted in Fig. 2b. In this manner, our cloud assistance approach delivers a system in which the cloud is the expert, and mobile devices ask the cloud for its expertise.

Evaluation and Results

The binding between computational cloud services (Amazon EC2, Microsoft Azure, etc.) and smartphones is proven to be feasible with the latest technologies [5, 6, 9] (Android, Windows Phone, etc.), mostly because virtualization technologies and their synchronization primitives enable mobiles' transparent migration and remote invocation of code. In this section, we evaluate our ideas about equipping the offloading architectures with cloud assistance (analytics and stratified features) and support on demand (multi-tenancy). We conduct experiments using our own code offloading framework [10]. Our goal is to demonstrate the most significant insights about how code offloading is enhanced with data captured by a community of devices. In the evaluation, as clients we used a Samsung Galaxy S3 (i9300), Samsung Galaxy S2 (i9100), and Samsung Nexus (i9250), and as a surrogate; we used a nearby local computer (64-bit, 2.3 GHz Intel® Core™, 8 GB of memory), general-purpose servers (instances m1.small, m1.medium, m1.large, m1.xlarge), and a compute optimize instance (c3.2xlarge) from Amazon EC2.[1] To measure the energy consumed by the mobile in our offloading experiments we relied on the Mobile Device Power Monitor.[2]

Code Offloading Framework

Our framework enables the mobile applications to offload code at the method level using Java reflection. The framework follows a client-server model, where the client is located at the mobile and the server is located in the cloud. One mobile subscribes to one server, and a server can handle multiple devices. The framework is equipped with the mechanisms to record the offloading process at different levels of the architecture, such that offloading traces can be gathered from the devices that connect to the cloud. The mobile implements an automated mechanism that profiles code based on the information defined in a JSON schema. The schema is created at the cloud by analyzing the traces of the offloading process and defines the code to be offloaded from different points of view (e.g., *what*, *when*, *where*, and *how*). Each point describes the attributes a smartphone app must meet in order to offload; for instance, *what* defines the name of the candidate methods, *when* describes the informational thresholds the device must detect (e.g., latency), *where* defines the type of server in which the code has to be offloaded (e.g., server of type m1.small), and *how* introduces an execution plan for the code (e.g., parallelize the code into *n* processes). Additionally, other points of view can be introduced (e.g., user's location). However, this requires exploitation of code execution patterns from the community (e.g., pre-cache results based on location).

The characterization is sent to the mobile asynchronously using push notification technologies. Evidently, the use of crowdsourcing also implies solving the scalability problem of the cloud surrogate. Since its only purpose is to execute code, we think that the virtualization of the entire mobile platform in a server is unnecessary as it wastes CPU resources, and is counterproductive as it slows down performance. Thus, in our system, we avoid such virtualization approach, and instead rely on a low-level compiler, which was extracted from the mobile platform (Android) and built straight in the host operating system of the server. The compiler is built by downloading and compiling the source code of Android Open Source Project (AOSP)[3] over the server to target an x86 server architecture. In this manner, the framework takes advantage of the server at full capacity, which means that the surrogate is released from the limitations imposed by the mobile operating system, such as activating multiple instances from the same , and executing multiple applications concurrently, among others.

Furthermore, our framework implements an auto scaling mechanism that allows the server to scale horizontally in the cloud. Figure 4 depicts how the scaling process occurs. The framework

We envision that as a part of the characterization process, pre-cached functionality from the entire mobile application can be requested on demand. In this manner, our cloud assistance approach delivers a system where the cloud is the expert and mobile devices ask the cloud for its expertise.

1 http://aws.amazon.com/ec2/instance-types/

2 http://www.msoon.com/LabEquipment/PowerMonitor/

3 http://source.android.com/

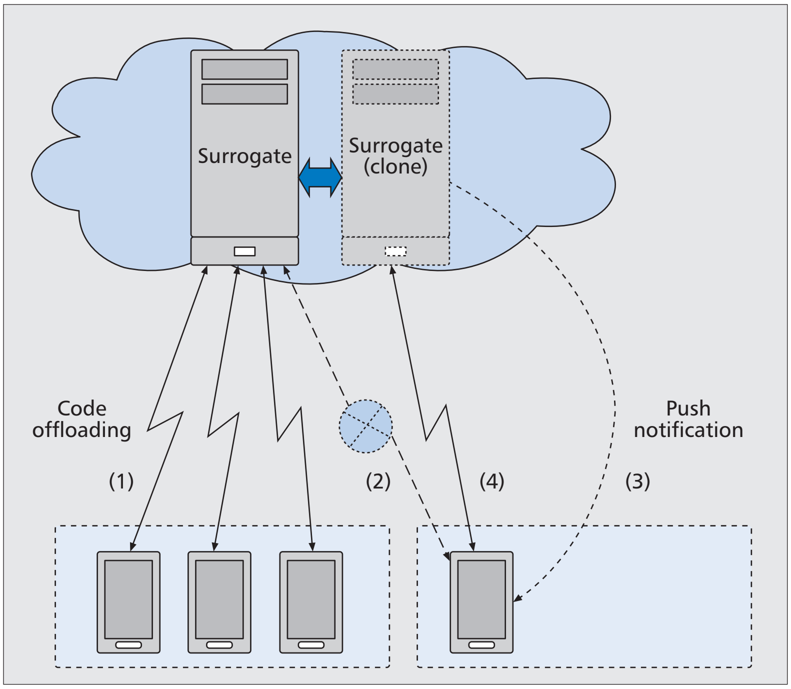

Figure 4. Representation of a cloud-based surrogate supporting multi-tenancy for code offloading.

implements a performance-based mechanism that monitors the utilization of CPU of the server, such that when a server is handling multiple offloading requests (1) and the CPU utilization is too high for method invocation (2), another server is created (3). In this process, the load of the subscribers is split between the available servers (4). The servers are in charge of updating the information of their subscribers via push notification.

Results

To demonstrate that the analytic properties of the cloud beyond service provisioning enhance the offloading process, we develop a use case based on the results of puzzle apps presented in a previous section and analyze the effects of offloading an NQueens algorithm from the smartphone to cloud. The aim of the algorithm is to calculate how to place n queens on an $n \times n$ chessboard. It uses backtracking with pruning to calculate all the solutions. The motivation of using this algorithm lies in its inherent characteristics to turn into a resource-intensive operation based on a normal user's interaction (play time). Definitely, we could rely on other uses cases, such as image or video processing (e.g., face recognition), to strongly emphasize the need for going cloud-aware. However, we believe that the applicability of code offloading beyond obvious use cases is possible, but it requires a priori knowledge about the possible executions of a mobile application. Of course, the knowledge should cover the effect of local and remote execution to assess multiple trade-offs.

Figure 5a shows the execution of NQueens with $n = 13$ in various cases, locally and remotely. Under the assumption that the same code can be offloaded multiple times by a smartphone app running in a particular or different devices, a code offloading request can be generic enough, such that its result can be reused in future invocations by other devices. In other words, the cloud can allocate an extended cache to store reusable results. We assumed that the NQueens is offloaded multiple times by a crowd of i9300, such that our framework identifies the request as a common offloading operation. A computational task offloaded to the cloud is serializable. Thus, it is possible to calculate for each request a MessageDigest key based on a standard checksum (e.g., a SHA-1 checksum) in order to uniquely identify each request. By collecting and clustering all the checksums of the requests using DBSCAN, it is able to find all the generic requests offloaded by a community of devices. By this approach, the result of a previous remote method invocation is pre-cached in the cloud. The diagram also shows the total offloading time of the pre-cached result. Based on the diagram, we can observe that the offloading process obtains further acceleration as the invocation time in the server is avoided. Arguably, the offloading result could be stored temporally in the cache of the device. However, the limited space of the cache in the device is unsuitable for long-term reuse. Clearly, the cache can be increased, but a cache that is too large can cause out-of-memory exceptions and leave the mobile application with little memory to work. In parallel, Fig. 5b shows the energy consumed by the mobile device in each of the previous cases. Since the acceleration of the offloading process influences the effort required by the mobile to execute the code, the faster the time to execute the code, the less computational effort is required to maintain an active communication channel to remote resources. Of course, this is certain when the code invocation requires a long execution time.

Finally, Fig. 5a also compares how the algorithm is processed by different smartphones and multiple servers of the cloud ecosystem in terms of response time. The diagram includes the communication latency, invocation, and synchronization time of the code. From the diagram, we can see that the latest mobile technologies are computationally comparable with some servers in Amazon's cloud such as m1.large and m1.xlarge. Moreover, we can visualize that the utility computing features of the cloud are critical in the decision of *where to offload*. While the latency in communication cannot be controlled, the total time of the invocation can decrease by adjusting the trade-off between utilization price and computational capabilities of the server. As a consequence, the characterization of an offloading operation is associated with multiple levels of enhancement. In this case, i9300 should offload smartly to m1.xlarge or c3.2xlarge to obtain maximum benefits. Clearly, the characterization is not obvious for the devices or the architecture in general. The characterization is a process computationally exhaustive and long-term for a single device, and to an even greater extent if we consider specific mobile applications. Thus, crowdsourcing can be utilized to capture information that can be transformed into knowledge to assist in the offloading process.

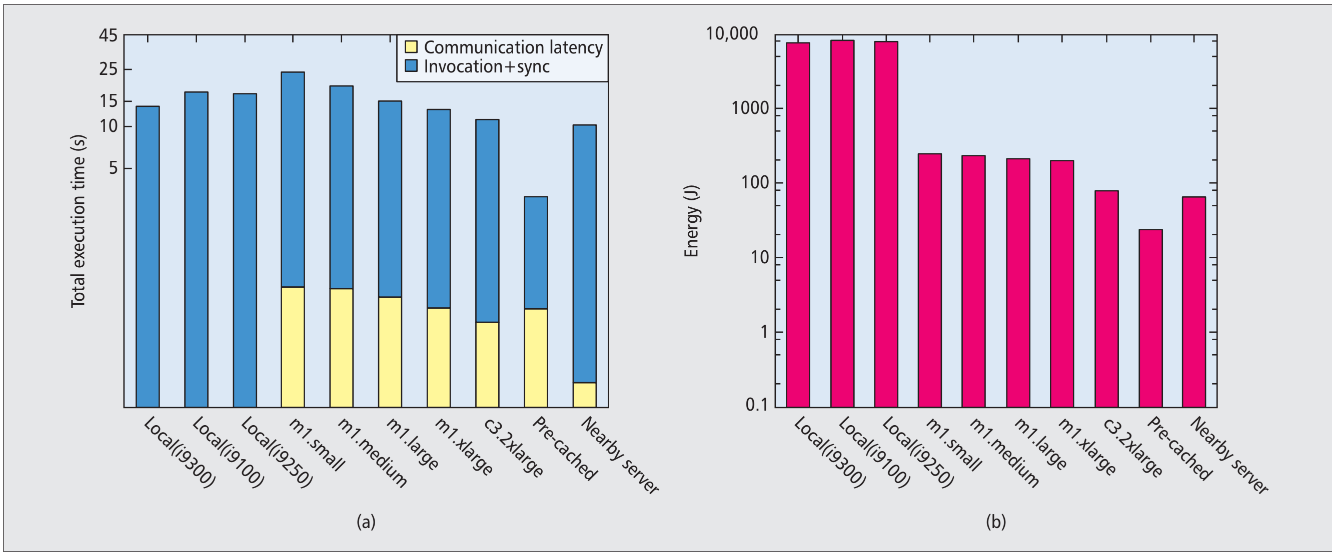

Figure 5. a) Acceleration gains that can be achieved by caching offloading results in terms of response time; b) energy consumed in each offloading case (local, remote, and pre-cached).

Discussion

As with any other crowdsourcing solution, our strategy becomes gradually more effective as data is being collected to understand the offloading process for a particular mobile application. However, initially, the advantages of our framework lie in the ability to consider many parameters from different perspectives in the inference process that determines when to offload. Once cloud assistance is available, the advantages of our framework take place, such that a computational request can speed up the performance of a smartphone app up to 10×, and up to 30× using pre-cached functionality. This acceleration is variable as it is proportional to the actual time required to execute the computational task. We believe that the assistance in the offloading process can be improved further by analyzing different aspects of the architecture at multiple levels of granularity [13]. The benchmarking (support for multi-tenancy) of the framework is left as future work since it cannot be shown with simple insights, but instead requires large-scale detailed analysis.

Conclusions and Future directions

The bridging of the mobile and the cloud has led to rethinking the binding mechanisms that enable the cloud to be exploited for the benefit of the handset. Code offloading is a key technique in augmenting the computational capabilities available for mobile applications with elastic cloud resources. However, the sustainability of the technique in practice is an open issue. In this article, we explore the challenges for code offloading from a systemic point of view and identify the key limitations that prevent the adoption of code offloading. We evaluate a number of proposed design strategies to overcome these limitations by using our own code offloading framework. The source code used in the experiments is available as open source in GitHub.[4] Our study highlights new directions for the design of future mobile architectures supported by cloud computing. Essentially, our work proposes the use of data analytics over offloading history captured by a community of devices in order to enrich the context of the device to offload to the cloud without inducing a counterproductive outcome for the mobile. Moreover, data analytics can also empower the offloading architectures with adaptive features that respond to the behavior of code during runtime, such as quality of experience based on code acceleration in different cloud servers, pre-cache functionality of the apps, auto scaling according to code execution, and so on.

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Biographies

HUBER FLORES [S] (huber@ut.ee) is currently a Ph.D student at the Faculty of Mathematics and Computer Science, University of Tartu. He received his B.Eng. in computer science from the University of San Carlos of Guatemala and his M.Sc. in software engineering from a combined program between the University of Tartu and Tallinn University of Technology, Estonia. He is also a Student Member of ACM (SIGMOBILE). His major research interests include mobile offloading, mobile middleware architectures, and mobile cloud computing.

PAN HUI (panhui@cse.ust.hk) received his Ph.D. degree from the Computer Laboratory, University of Cambridge, and earned both his M.Phil. and B.Eng. from the Department of Electrical and Electronic Engineering, University of Hong Kong. He is currently a faculty member of the Department of Computer Science and Engineering at the Hong Kong University of Science and Technology, where he directs the HKUST-DT System and Media Lab. He also serves as a Distinguished Scientist of Telekom Innovation Laboratories (T-labs) Germany and an adjunct professor of social computing and networking at Aalto University, Finland. Before returning to Hong Kong, he spent several years in T-labs and Intel Research Cambridge. He has published around 150 research papers, and has some granted and pending European patents. He has founded and chaired several IEEE/ACM conferences/workshops, and has served on the Organizing and Technical Program Committees of numerous international conferences and workshops including ACM SIGCOMM, IEEE INFOCOM, ICNP, SECON, MASS, GLOBECOM, WCNC, ITC, ICWSM, and WWW. He is an Associate Editor for *IEEE Transactions on Mobile Computing* and *IEEE Transactions on Cloud Computing*.

SASU TARKOMA [M'06, SM'12] (sasu.tarkoma@helsinki.fi) received an M.Sc. degree in 2001 and a Ph.D. degree in 2006 in computer science from the University of Helsinki. Since 2009 he has been a full professor of computer science at the University of Helsinki. He has led and participated in national and international research projects at the University of Helsinki, Aalto University, and Helsinki Institute for Information Technology (HIIT). He has worked in the IT industry as a consultant and chief system architect as well as principal researcher and laboratory expert at Nokia Research Center. He has over 140 scientific publications, four books, and four U.S. patents.

YONG LI [M'09] (liyong07@tsinghua.edu.cn) received his B.S. degree in electronics and information engineering from Huazhong University of Science and Technology, Wuhan, China, in 2007 and his Ph.D. degree in electronic engineering from Tsinghua University, Beijing, China, in 2012. During July to August 2012 and 2013, he was a visiting research associate with Telekom Innovation Laboratories and the Hong Kong University of Science and Technology, respectively. During December 2013 to March 2014, he was a visiting scientist with the University of Miami, Coral Gables, Florida. He is currently a faculty member in the Department of Electronic Engineering, Tsinghua University. His research interests are in the areas of networking and communications, including mobile opportunistic networks, device-to-device communication, software-defined networks, network virtualization, and future Internet. He is currently an Associate Editor of the *EURASIP Journal on Wireless Communications and Networking*.

SATISH SRIRAMA (srirama@ut.ee) is an associate professor and head of the Mobile & Cloud Lab at the Institute of Computer Science, University of Tartu. He received his Ph.D in computer science and Master's in software systems engineering from RWTH Aachen University, Germany, and his Bachelor's degree (BTech) in computer science and systems engineering from Andhra University, India. His current research focuses on cloud computing, mobile web services, mobile cloud, Internet of Things, and migrating scientific computing and enterprise applications to the cloud.

RAJKUMAR BUYYA (rbuyya@unimelb.edu.au) is a professor of computer science and software engineering and director of the Cloud Computing and Distributed Systems (CLOUDS) Laboratory at the University of Melbourne, Australia. He is the founding CEO of Manjrasoft, a spin-off company of the university, commercializing its innovations in cloud computing. He has authored over 430 publications and four textbooks. He has also edited several books, including *Cloud Computing: Principles and Paradigms* (Wiley, 2011). He is one of the most highly cited authors in computer science and software engineering worldwide (h-index = 66 and 21,300+ citations). Software technologies for grid and cloud computing developed under his leadership have gained rapid acceptance and are in use at several academic institutions and commercial enterprises in 40 countries around the world. He has led the establishment and development of key community activities, including serving as founding Chair of the IEEE Technical Committee on Scalable Computing and five IEEE/ACM conferences. These contributions and his international research leadership have been recognized through the 2009 IEEE Medal for Excellence in Scalable Computing award. Manjrasoft's Aneka Cloud technology developed under his leadership has received the 2010 Asia Pacific Frost & Sullivan New Product Innovation Award and the 2011 Telstra Innovation Challenge, People's Choice Award. He is currently serving as the first Editor-in-Chief of *IEEE Transactions on Cloud Computing*.

Mechanisms and Challenges on Mobility-Augmented Service Provisioning for Mobile Cloud Computing

Wenzhong Li, Yanchao Zhao, Sanglu Lu, and Daoxu Chen

The authors are with State Key Laboratory for Novel Software Technology, Nanjing University.

ABSTRACT

While mobility is the inherent cause of resource poverty and low connectivity in wireless environment, it also enables the opportunity for surrounding sensing, therefor providing mobility-augmented cloud service becomes the key challenge for the emerging paradigm of Mobile Cloud Computing (MCC). This article provides an overview of the mechanisms and open issues for mobility-augmented service provisioning in MCC. We first outline the concept, system architecture, and taxonomy of research issues. Then we introduce three key mechanisms with respect to mobility augmentation, heterogeneous network convergence and mobile service provisioning. Moreover, we discuss the open challenges to reveal the future direction of MCC.

INTRODUCTION

The past years have witnessed a tremendous growth of mobile market. According to the report of Portio Research,[1] the number of worldwide mobile subscribers is forecast to increase at a rate of 7.3 percent per year and will reach nearly 8.5 billion by the end of 2016; the worldwide mobile applications market is forecast to reach 63.5 billion USD with more than 200 billion downloads per year by 2017. With the rapid growing of mobile devices, the demand for ubiquitous media applications increases continuously towards the vision of "information at my fingertips" at anytime and place [1].

Despite the advances in the capacity of smartphones and wireless communication technologies, mobile hardware is still resource-poor compared to desktop and server hardware. Constrained by battery life, computation capacity, storage, wireless bandwidth, and communication delay, the resource-poor mobile devices encounter the difficulty of supporting content-rich or resource-intensive applications such as real-time image processing for video games, augmented reality, and location-based service [2] with guaranteed Quality of Service (QoS). The Mobile Cloud Computing (MCC) paradigm is introduced as a promising solution to overcome the above difficulty, which provides a uniform platform for cloud-based resource sharing and augmentation for mobile computing. MCC allows the computing, data storage and mass information processing be offloaded to the cloud servers for enhancing the reliability and availability of services while minimizing the energy and computational requirements in mobile devices. Many mobile applications have employed cloud computing to provide enhanced services [3]. For example, Dropbox[2] enables mobile applications extending their storage in the data center; Google Glass[3] can enhance wearers' vision with map services provided by a remote cloud.

The convergence of mobile computing and cloud computing technology forms the leading trend of mobile cloud computing research in recent years. Surveys such as [4–6] have provided extensive overview on different aspects of mobile computing which include its definition, architecture, platform, application, etc., and cover a broad range of technologies such as network operation, resource sharing, service management, and distributed application processing. However, as mobility has been known as a key factor of mobile cloud environment, the issues of mobility-augmented service provisioning for MCC has not dragged enough attention in literature and there is lacking comprehensive study to address the mobility augmentation issues and challenges.

While mobility has the inherent problems such as resource poverty, finite energy, and low connectivity, there are strong motivations to enable mobility-augmented service provisioning in MCC environment. First, users have the preference to shift from static computers to portable devices, which forms the booming market of mobile computing. Second, the development of wireless communication technologies such as 3G/4G connectivity and Wi-Fi access have pro-

[1] http://www.portioresearch.com

[2] https://www.dropbox.com/

[3] https://www.google.com/glass/start/

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The evolving new applications such as location-based service, mobile crowdsourcing, healthcare, and environmental monitoring, where mobility is no longer an issue but an opportunity to obtain rich sensing data from surroundings, have high expectation of enabling mobility-augmented cloud services.

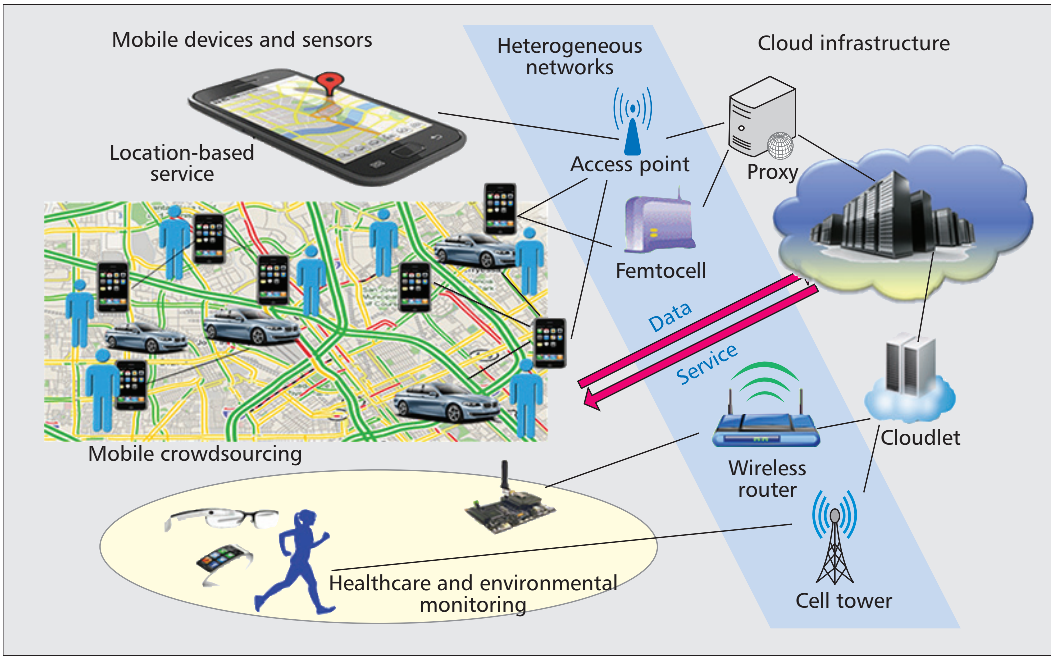

Figure 1. The scenarios and architecture of mobility-augmented cloud service.

vided the infrastructure for ubiquitous mobile applications. Third, the demand of running resource intensive applications in mobile devices requires the augmentation of resource-rich clouds. Last but not least, the evolving new applications such as location-based service, mobile crowdsourcing, healthcare, and environmental monitoring, where mobility is no longer an issue but an opportunity to obtain rich sensing data from surroundings, have high expectation of enabling mobility-augmented cloud services.

The contribution of this article is highlighted as follows. We first provide the definition and framework for the emerging mobility-augmented services in MCC environment. Then, we outline a taxonomy of research issues from the perspective of mobile terminal, network, and cloud infrastructure accordingly. Thereafter, we examine the detailed mechanisms to enable mobility-augmented cloud service, which include mobility augmentation, heterogeneous network convergence and mobile service provisioning. Finally, we discuss a number of open research challenges with respect to overhead reduction, heterogeneity handling, QoS assurance, privacy and security issues, to envision the future direction of MCC.

FRAMEWORK

This section overviews the framework and mechanisms of mobility-augmented service in MCC.

DEFINITION AND SCENARIOS

Mobility-augmented service refers to a type of application relying on mobile devices for data collection, processing vast amount of data in the remote data center, and providing aggregated services to end users via cloud. Traditional mobile computing regards mobility as a technical issue to be handled by communication and coordination mechanisms. In the context of mobility-augmented service, mobile devices are considered as sensors that provide rich source of sensing data from the physical world. Figure 1 depicts several typical scenarios of mobility-augmented services.

Location-based service (LBS) uses information on the geographical position of the mobile device to provide customized services such as searching a route in the city, discovering popular restaurants in local area, recommendation nearby friends in a social network, and advertisement based on locations.

Mobile crowdsourcing is a type of applications that provide information service or carry out a task by using the data collected from smart phones of a large group of people. For example, Google Traffic implemented a feature on Google Maps[4] that displays live traffic condition by collecting the GPS-location of a large number of drivers. It can even detect accidents by analyzing the reports from individuals.

Healthcare and environmental monitoring are the emerging applications that smart devices equipped with sensors can sense the activities of human body and the surroundings. The sensing data is collected and analyzed by a remote server to estimate the human health conditions and to assess environment.

ARCHITECTURE AND TAXONOMY

To provide the supportance of mobility-augmented service over MCC, we propose a system architecture as illustrated in Fig. 1, which contains three perspectives: cloud infrastructure, heterogeneous networks, and mobile devices.

Cloud infrastructure consists a pool of sharing resources, which are virtualized and provided as service. Virtualization technology integrates and encapsulates heterogeneous resources to enable resource sharing in an isolated environment called virtual machine (VM). With virtualization, cloud resources can be accessed via a uniform

[4] maps.google.com.

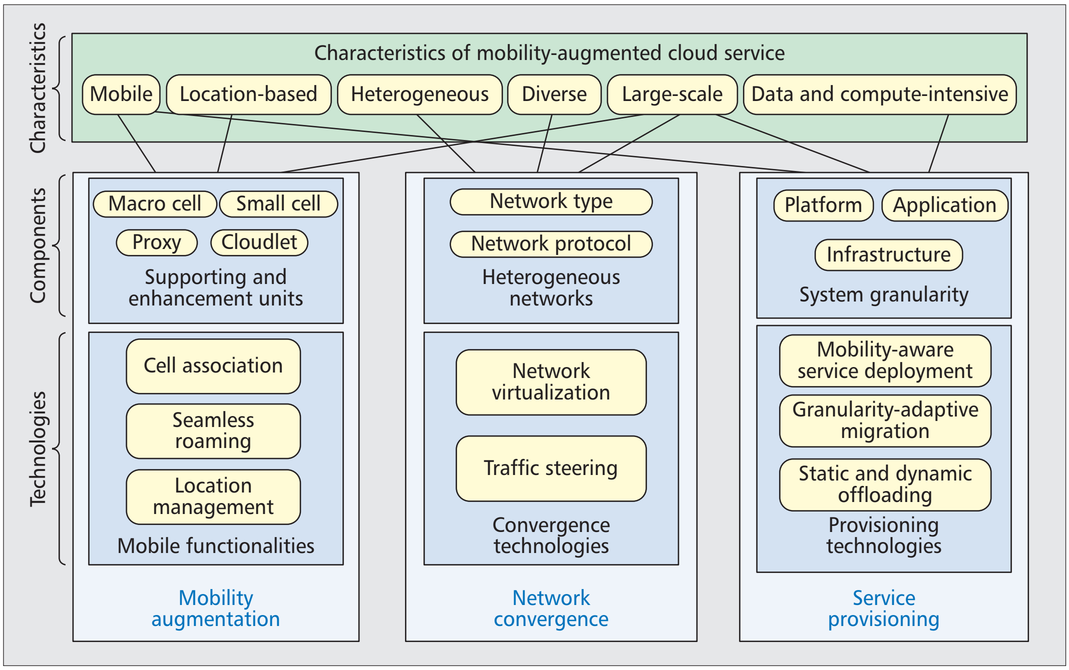

Figure 2. Taxonomy of mobility-augmented cloud service provisioning mechanisms.

Mobility augmentation techniques support constant moving wireless devices accessing to cloud service seamlessly and pervasively. The major components for mobile enhancement consist of a set of mobile supporting and enhancement units to provide wireless access to the cloud, and a number of mobile functionalities to support seamless roaming.

interface where infrastructure, platform and software are offered as service.

Heterogeneous networks consist of wired and wireless connections. High-speed wired backbone connects to cloud infrastructure to assure the capacity and quality of service for cloud service delivery. Wireless networks are diverse networks that support different type of communications (device-to-infrastructure and device-to-device communications) with different network structures (i.e., centralized network, ad hoc network [7]) that operate on different mechanisms and protocols. Supporting units (i.e., radio tower, base station, wireless routers, and access points) and enhancement units (i.e., proxy and cloudlet [1]) are deployed to enable mobile devices connecting to wireless networks and accessing to the remote cloud service.

Mobile devices include smartphones, laptops, tablets, vehicles, etc. Mobile devices are heterogeneous in nature with a vast variety of hardware, operating systems, platforms and wireless network standards. Since nowadays mobile devices are typically equipped with sensors that are capable of sending context information, they are not only cloud service consumer, but also the data source and producer of cloud services.

The scenarios and architecture in Fig. 1 reveal several important characteristics of mobility-augmented service: mobile, location-based, heterogeneous, diverse, large-scale, data-centric and compute-intensive. To tackle such issues, several technologies have been developed to facilitate mobility-augmented service, which can be classified from the perspective of client, network and infrastructure accordingly.

As illustrated in Fig. 2, the three mechanisms named mobility augmentation, network convergence, and service provisioning work together to support mobility-augmented service in the paradigm of MCC. Mobility augmentation addresses mobile, location-based and scalability issues, which focus on the mechanisms of deploying mobile supporting and enhancement units to collect sensing data and service requests, and implementing a set of mobile functionalities to support ubiquitous connectivity. Network convergence studies the heterogeneity and diversity in network types and connection protocols, which enables the transmission of large-scale traffic over heterogenous networks. Service provisioning addresses the data-centric and compute-intensive issues by deploying, migrating and offloading cloud services, which enables big data processing in remote data center and service delivery to end users via infrastructure.

The detailed mechanisms are presented in the following sections.

Mobility Augmentation

Mobility augmentation techniques support constant moving wireless devices accessing to cloud service seamlessly and pervasively. The major components for mobile enhancement consist of a set of mobile supporting and enhancement units to provide wireless access to the cloud, and a number of mobile functionalities to support seamless roaming.

Supporting and Enhancement Units

Mobile supporting units are dedicate wireless communication infrastructure and devices to provide Internet service, which can be cataloged by macro cells and small cells [7].

Macro cells are normally implemented by GSM/3G/4G radio towers, where data service can cover a wide range of area to provide stable and constant network connectivity for highly mobile devices. However, the data rate of macro cells is relatively low due to channel fading in long range communication. Small cells can usu-

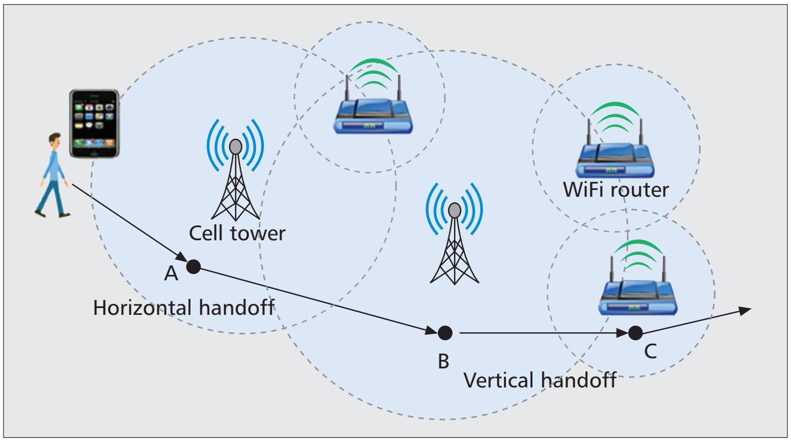

Figure 3. A scenario of seamless roaming.

ally cover a small range less than 100 meters. For example, Femtocells are small, low-power cellular base stations connecting to the service provider's network via broadband, which are typically designed for use in a home or small business. Wi-Fi routers and access points (APs) employ a local area wireless technology that allows a wireless device to exchange data or connect to the Internet using 2.4 GHz or 5 GHz radio waves. Small cells allow service providers to extend service coverage indoors or at the cell edge with high data rate, but they suffer from handoff issues for frequent mobility [8].

Mobile enhancement units are physical or logical components that are deployed to extend the capacity of mobile devices. Typical enhancement units includes proxy and cloudlet.

Proxy is a component being proximately deployed between mobile devices and the cloud. To access cloud service the mobile devices can simply send queries to their nearby proxy. Thus user mobility is supported by monitoring the user's location and directing his queries to the proxy in range. Cloudlet [1] is a resource-rich computer connected to the Internet and available for serving nearby mobile devices via one-hop wireless local area network (WLAN). In this scheme, the mobile device acts as a thin client, outsources its computation to the nearby cloudlet, and accesses to a remote cloud only when the cloudlet has no enough capacity to handle the request. A cloudlet is self-managing and easy to deploy, which functions like a mini data center to leverage mobile device's resource poverty to meet the need for real-time interactive response for mobile applications.

Mobile Functionalities

To support devices roaming among different cells and provide location-based cloud services, a number of functionalities should be implemented for mobile enhancement, which includes cell association, seamless roaming, location management, etc.

Cell Association —Cell association decides which mobile support unit should be associated with the mobile devices for Internet access. Due to the diverse and heterogeneity of supporting units, choosing the best cell to associate with is not trivial. Simple policies will associate the device with the cell tower or access point that has the largest received signal strength (RSS) or least geographical distance. However, since the supporting units, especially small cells, are typically deployed in an unplanned manner with diverse transmit powers and uneven user populations, such heuristic policies will suffer from communication interference and unbalancing workload. In MCC, sophisticated cell association policies are needed to exploit dynamic interference management and inter-cell coordination techniques to optimize link capacity and coverage gains.

Seamless Roaming — Seamless roaming in mobile cloud environment allows mobile devices moving from one cell to another without interrupting network service. Handoff is such a technology that enable smooth inter-system signal switching along heterogeneous wireless networks, which can be classified as either horizontal or vertical [8].

Figure 3 shows a scenario of seamless roaming in wireless environment. Horizontal handoff occurs when a mobile device crosses from one cell into a neighboring cell (supporting the same technology) triggering the signal communication changing from one base station to another. For example, moving from A to B in Fig. 3 triggers horizontal handoff. While horizontal handoff is a changeover among base stations by the same wireless network interface, vertical handoff is a more recent scheme taking place between different network interfaces operating on different technologies (e.g., switching from macro cell to small cells or vice versa). As the example shown in Fig. 3, vertical handoff occurs when the user moves from location B to C causing the change of his connectivity from a cell tower to a wireless router.

The key challenge of handoff is to reduce the time for suspending and resuming network service while keeping the guarantee of quality of service. A measurement-based proactive handoff strategy can make this process smooth: by constantly measuring the wireless signal strength, a device can estimate its distance to the cell tower and its movement direction; when it enters the overlapping area of another cell, the device can initiate handoff proactively without waiting for the connection time-out. Emerging techniques such as multihoming and Multipath TCP can be applied to enhance fault tolerance and fast recovery during vertical handoff.

Location Management — Location management is the mechanism to determine a device's current location and track its potential movement away from or towards the range of a communication cell. Works on wireless localization falls into two categories: infrastructure-based and peer-based [4].

Infrastructure-based localization systems mostly rely on triangulation using physical signals from the fixed-deployed infrastructures such as GPS, GSM, and Wi-Fi. For example, Global Positioning System (GPS) allows receivers to determine their positions by communicating with the GPS satellites with the aid of trilateration;

Wi-Fi based localization collects the radio fingerprints quantified from the Wi-Fi signal strengths at many physical positions and APs, and identifies user location by retrieving and matching the fingerprints.

Peer-based localization systems determine device's position by sensing information from surroundings and other peers. As introduced in [9], mobile device can estimate the relative location using the measurements of accelerometer and compass equipped by smartphones, and calibrated estimation errors by observing the locations of other peers passing by to achieve decentralized cooperative localization in sparse wireless network. Peer-based localization is infrastructure-free or only relies on very few anchor nodes, which is quite flexible for both indoor and outdoor location management.

NETWORK CONVERGENCE

In mobile cloud computing environment, wired and wireless devices operate on different networks with different communication protocols. The convergence of heterogeneous networks is an important aspect to enable pervasive mobile cloud service.

HETEROGENEOUS NETWORKS

In MCC, diverse network types and their communication protocols impact the delivery of cloud services and affect mobility and usability of mobile devices.

Heterogeneity in Network Types — Most mobile clouds operate in a centralized architecture, where cloud resource is deployed in a remote centralized data center. Cloud service is transferred through wired backbone network, and delivered to the mobile devices via wireless connections. Sometimes proxies or cloudlets are deployed closed to the MTs to reduce latency and enhance mobility support.

Wired networks use high capacity physical networks to support a high-quality voice and data services, which form the core infrastructure for resource-rich cloud service provisioning. While wired networks provide a fixed service, wireless networks enable communications while in motion, which provide ubiquitous coverage and flexible service usage.

To extend coverage and adapt to hostile environments, mobile cloud works on ad hoc mode were proposed to enable distributed resource sharing. In ad hoc mobile cloud [3], tasks from mobile devices are processed in a distributed and collaborative fashion on all the neighboring mobile devices pooled together or handled by a particular mobile device acting as a server. Delay tolerant network (DTN) is a sparse mobile ad hoc network supporting device-to-device communication relying on the mobility of nodes and their encounter opportunities [9], which is found a suitable way for mobile crowdsourcing. As an example, vehicular ad hoc network (VANET) forms a type of DTN that allows car-to-car communication with intermittent connections and enables live traffic monitoring and safe driving service via roadside supporting units connecting to a cloud.

Heterogeneity in Network Protocols — The current mobile cloud computing supports a variety of connection protocols for wireless communication.

Macro cell communication protocols for mobile cloud service include 3G and 4G standards. Typical 3G standards include EDGE, W-CDMA, TD-SCDMA, CDMA2000, etc. Two 4G candidate systems are commercially deployed: the first-release Long Term Evolution (LTE) standard which has a theoretical data transfer rate of up to 100 Mb/s in the downlink and 50 Mb/s in the uplink, and the Mobile WiMAX standard which offers peak data rates of 128 Mb/s downlink and 56 Mb/s uplink over 20 MHz wide channels.

Small cell communication protocols provide short distance communication with higher data rate and more reliable network services. The Wi-Fi technology enables mobile devices to connect to Internet via a wireless access points. The Femtocell technique deploys low-power cellular base stations, which is compatible to most 3G and 4G standards, to allow service providers extending service coverage indoors or at the cell edge with better communication quality. For high mobility devices such as vehicles, multiple networking technologies/protocols such as IEEE 802.11p, WAVE IEEE 1609, WiMAX IEEE 802.16, Bluetooth, and ZigBee are applied to enable mobile cloud applications (e.g., Apple CarPlay and Google's Android Auto).

NETWORK CONVERGENCE TECHNOLOGIES

The vision of MCC depicts the convergence of heterogeneous networks with QoS assurance, which concerns several techniques including network virtualization and traffic steering.

Network Virtualization — To deal with the heterogeneities appearing from different level of networks, a coherent paradigm is desirable for allocating network resources and managing network functionalities in a mobile cloud environment. Network virtualization is an emerging technology to create multiple heterogeneous logical/virtual networks that are decoupled from the underlying network hardware and service providers. With network virtualization, customized end-to-end network services can be deployed and managed dynamically by sharing and utilizing the rich network resources leased from multiple infrastructure providers, which achieves the integration of coexisting heterogeneous network architectures in the Internet [10].

The idea of software-defined networking (SDN), which truly decouples the control plane and data plane in a network, has become the focus of industry and academy in the recent years. As a de facto standard to enable SDN, OpenFlow[5] was proposed as the communication interface between the control and data planes of an SDN architecture, which allows effective manipulation and configuration of the functionalities of the forwarding plane of network devices such as switches and routers. With the open interfaces provided between network devices and controllers, OpenFlow-enabled

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[5] http://archive.openflow.org/.

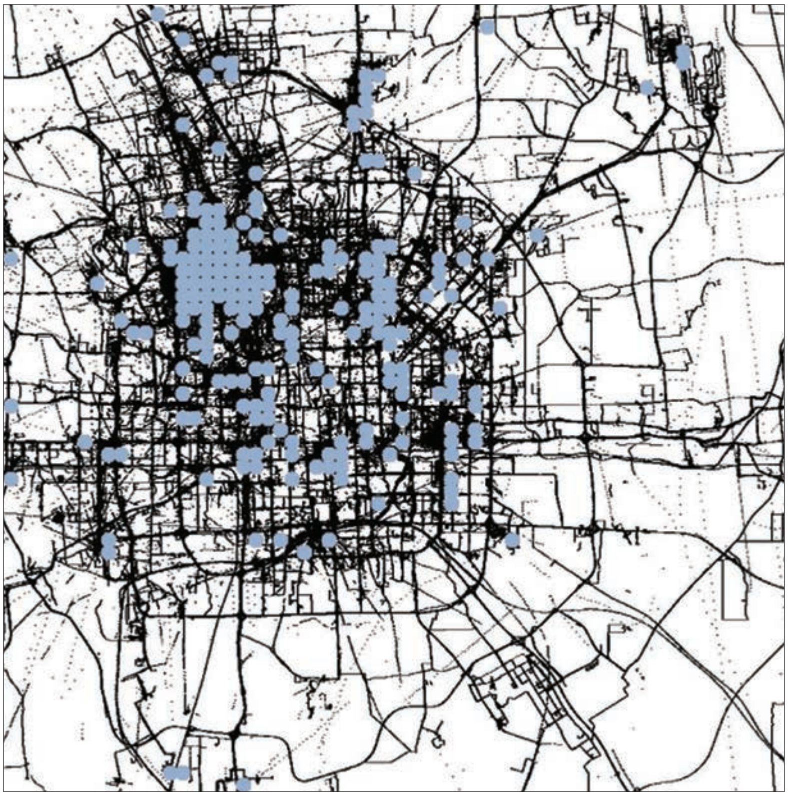

Figure 4. Deployment of small cells to cover mobility trajectories..

SDN has been widely adopted as a way for network virtualization, with which the heterogeneous networks can be managed by a centralized control panel over a logical view of the entire network.

Traffic Steering — Although network virtualization bridges the heterogeneity of network devices, there is the heterogeneity of user traffic in different network levels. In the MCC environment, a heterogeneous network is composed of several layers of subnetworks with different radio access technology and cell size. As the coverage area of such layers overlaps, the mobile devices can select among different network layers for cloud service. The concept of traffic steering includes those decisions made by the operator to modify the user distribution across the layers to meet the ever-increasing traffic demand of mobile applications [11]. To meet this goal, the hybrid networks should be managed as a unified whole for traffic steering. From the client perspective, mobile users are allowed to decide the priorities of the cells to associate with. From the management perspective, the operator has some degree of freedom to adjust user priorities or force handoffs to achieve the optimization of several objectives such as cost, energy consumption, load balancing, quality of service, etc.

SERVICE PROVISIONING

Service provisioning extends the capacity of mobile computing by introducing the mechanisms of service deployment, migration and offloading.

MOBILITY-AWARE SERVICE DEPLOYMENT

Since mobile devices must connect to a mobile supporting unit in order to access to remote cloud service or communicate with nearby proxies or cloudlets, a variety of mobile supporting units should be deployed to enable mobile cloud service. The deployment of mobile supporting units has several objectives, which includes maximizing coverage area, improving network capacity, reducing service delay, etc.

Generally speaking, the deployment of macro cell radio tower is regular and planned by the ISPs, which can provide wide-range coverage for high mobility devices with low data rate. Small cells can provide much higher data rate in short range to extend service coverage indoors or at the network edge. However, since the human mobility is irregular and the small cells such as Wi-Fi and Femtocell are deployed in an unplanned manner, it leaves a wide space to optimize. Optimizing the deployment of cells to cover mobility trajectories can be reduced to the classic facility allocation problem, which is NP-hard. Heuristic strategies can be applied to seek near-optimal solutions for cell deployment. Figure 4 shows a scenario of deploying access points in an urban area in order to provide Wi-Fi access on the street. In the figure, black lines depict the movement trajectories of 178 mobile users in Beijing City from April 2007 to October 2011 (dataset from the Goelife project[6]), and the blue circles represent the most visited locations that are suggested to deploy access points. Approximately, this figure implies that if small cells are greedily deployed to cover a small percentage (≤ 5 percent) of the urban area, a large amount of mobile users (≥ 80 percent) are benefit from enhanced network service.

GRANULARITY-ADAPTIVE MIGRATION

The mobile enhancement units are usually implemented as logical visual components, and they are flexible to be migrated to "follow" the mobile terminals in order to enhance mobility support. Migration concerns several granularity such as service or VM migration. The Service Cloud [12] proposed on demand service deployment, where proxy service is deployed by overlay hosts on the wireless edge to connect to mobile devices, and user mobility is handled by migrating the service to different locations following the user. The Cloudlet [1], implemented by virtualization technology, deploys resource-rich virtual machines (VMs) to provide cloud service for physical proximate mobile devices. When a device moves to a guest environment, the VM migration technology is applied to enhance mobility: it suspends the original executing VM, transfers the whole status of its processor, disk and memory, and resumes the VM execution at the destination from the point of suspension. An granularity-adaptive migration approach is desirable to enhance the efficiency of mobile access.

STATIC AND DYNAMIC OFFLOADING

Offloading is the technique that transfers resource-intensive applications from mobile devices to the cloud to bridge the resource constraints and enhance mobility support, which is a key mechanism of the mobile cloud computing

6 http://research.microsoft.com/en-us/projects/geolife.

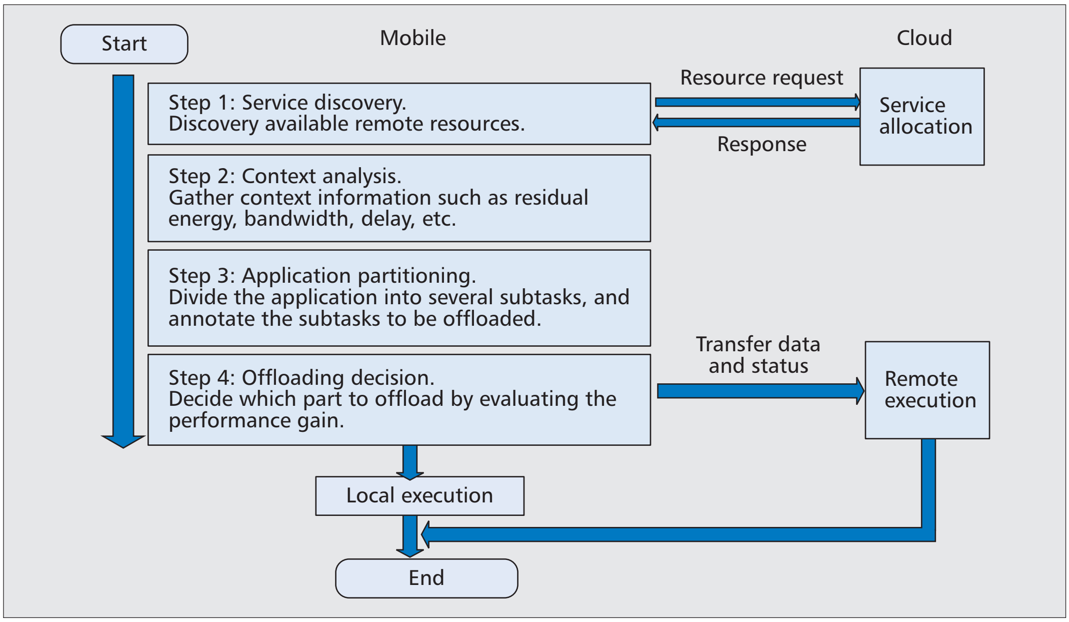

Figure 5. The process of application offloading.

Offloading computation and storage to a remote cloud is an effective mechanism to reduce local overhead, which concerns a number of factors including offloading granularity, device capacity, network delay, local and remote execution overhead. Application offloading to achieve multiple optimization objectives remains an open issue in MCC.

paradigm. The process of application offloading in mobile cloud environment is illustrated in Fig. 5. Offloading technologies can be classified into two categories: static offloading and dynamic offloading [13].

Static offloading decides which parts of codes should be offloaded prior to execution (at design or installation time). In the offloading scheme described in [14], the developer examines the source code of an application and specifies the function prototype of each procedure deemed worthy of remote execution and the combination of these procedures to produce a result. Offloading decision is made by maximizing a static utility function of each application regarding resource usage, latency, etc.

Dynamic offloading occurs at run time and starts to offload when the required resources are insufficient. As an example, Mobile Assistance Using Infrastructure (MAUI) [15] employs fine-grained dynamic offloading to reduce energy consumption of mobile devices. In this scheme, the developer are required to manually annotate the individual methods that cloud be offload. On application execution, a MAUI profiler assesses the energy consumption of the methods according to current situation to make the decision of offloading, which can cope with the mobility of users and adapt to the changing of network condition and user location.

A more complicated yet practical scenario is that multiple mobile devices offloading computation/storage tasks to the cloud dynamically via multiple wireless channels in a WLAN. The offloading performance will be affected by wireless access efficiency: it will cause severe interference if nearby devices decide to offload data in the same channel simultaneously. Due to the dynamic network topology and the shortage of central controller, a promising solution to this problem is to adopt a distributed game theoretic approach, where individual performance is optimized when a Nash equilibrium is achieved.

Challenges

Despite the mechanisms introduced above, there exist several open challenges for mobility-augmented service provisioning in mobile cloud environment, with respect to overhead, heterogeneity, QoS, privacy and security issues.

Overhead Reduction

The implementation of the mobile cloud computing paradigm will cause the overhead concerning a variety of technical and management issues. Overhead reduction is an open challenge toward the mobile cloud computing era.

Hardware overhead concerns the tradeoff between deploying new hardware and upgrading existing infrastructure to support mobile cloud service. The former is limited to scalability (the number of new servers) while the latter is limited to network capacity (the available bandwidth). They will co-exist and evolve in the long term.

Since mobility-augmented service deals with large-scale data sensed by mobile devices, the overhead of data delivery and processing is a key issue. On one hand, mobile device are densely deployed and devices in the same region may generate duplicated data. The in-network processing mechanism is needed for network components to aggregate data from downstream nodes and remove the duplicated and noise data. On the other hand, the sensing data from different source may be correlated. Thus the coding and compression mechanisms can be employed to further reduce the volume of network traffic. Seeking efficient algorithms for in-network processing and compression (e.g., compressive sensing) is a popular topic in the recent years.

Offloading computation and storage to a remote cloud is an effective mechanism to reduce local overhead, which concerns a number of factors including offloading granularity, device capacity, network delay, local and remote execution overhead. Application offloading to achieve

Security issues arise from every aspect of mobile cloud computing: security for mobile devices (eliminating the threat of viruses and worms), security for data transmission over networks (encrypting communication protocols), and security in the cloud datacenter nodes (preventing unauthorized access to the personal data stored in the cloud).

multiple optimization objectives remains an open question in mobile cloud computing.

HETEROGENEITY HANDLING

Heterogeneity is a major concern for ubiquitous mobile cloud service provisioning. There are different levels of heterogeneity in MCC. For example, mobile devices employ different hardware architecture and operating system platforms; heterogeneous networks operate with different technologies and connection protocols; network traffics are uneven and time-varying; cloud services are managed by different providers for different purposes; runtime environments of applications are mobile and dynamic. Bridging the heterogeneity to provide a homogenous distributed application processing solution remains a challenging issue for MCC.

QOS ASSURANCE

Satisfying the QoS for mobile applications is challenging due to the inherent mobility property of wireless devices. As mobile devices travel through different geographical locations with diverse speed, maintaining seamless connectivity and uninterrupted access to cloud datacenters is a non-trivial task.

Mobility-augmented services could be delay-sensitive or delay-tolerant regarding their QoS requirements. Since most smart devices support multiple communication modes, maintaining multiple communication paths via multiple interfaces could be a promising solution to enable always-on connectivity for delay-sensitive applications.

Other aspects affecting QoS assurance in MCC include the unavailability of remote cloud service due to heavy workload, insufficient network bandwidth due to communication congestion, and long latency caused by hostile communication environment.

PRIVACY AND SECURITY

Last but not the least, privacy and security are important aspects for establishing users' trust and loyalty for mobile cloud service.

Since mobile devices leverage their personal data to be stored and processed by the remote cloud, there is a major concern about the leakage of user privacy such as their personal interests, health condition, location and movement trajectories. Simple anonymity mechanisms could not prevent information leakage in mobility-augmented service where personal sensing data are transferred through insecure wireless channels. To achieve privacy protection such as k-anonymity [2] is still an open challenge in MCC.

Security issues arise from every aspect of mobile cloud computing: security for mobile devices (eliminating the threat of viruses and worms), security for data transmission over networks (encrypting communication protocols), and security in the cloud datacenter nodes (preventing unauthorized access to the personal data stored in the cloud).

CONCLUSION

In the paradigm of mobile cloud computing, the capacity of resource-poor mobile devices can be enhanced by exploiting the service provided by a remote cloud. The major challenge of enabling ubiquitous mobile cloud application arises from the mobility nature of devices in wireless communication environment. This article provides an overview of the mechanisms for mobility-augmented service provisioning. The architecture of mobility augmentation for MCC is outlined, and the current technologies are classified into three categories: mobility augmentation, heterogeneous network convergence and mobile service provisioning. Open challenges are discussed to reveal the future research directions.

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BIOGRAPHIES

WENZHONG LI [M] (lwz@nju.edu.cn) receives his B.S. and Ph.D degree from Nanjing University, China, both in computer science. He is now an associate professor in the Department of Computer Science, Nanjing University. Dr. Li's research interests include wireless networks, pervasive computing, mobile cloud computing, and social networks. He has published over 40 peer-review papers at international conferences and journals, which include INFOCOM, ICDCS, IWQoS, ICPP, IEEE Transactions on Parallel and Distributed Systems, IEEE Transactions on Wireless Communications, IEEE Transactions on Vehicular Technology, etc. He served as Program Co-chair of MobiArch 2013 and Registration Chair of ICNP 2013. He was the TPC member of several international conferences and the reviewer of many journals. He is the principle investigator of two funding from NSFC, and the co-principle investigator of a China-Europe international research staff exchange program. He is a member of ACM, and China Computer Federation (CCF). He was also the winner of the Best Paper Award of ICC 2009.

YANCHAO ZHAO received his B.S. degree in Computer Science from Nanjing University in 2007. He is currently a Ph.D. candidate in the Department of Computer Science and Technology, Nanjing University, Nanjing, China. In 2011 he was a visiting student in the Department of Computer and Information Sciences, Temple University, Philadelphia, USA. His research interests include wireless network optimization, mobile computing, and data-center networks.

SANGLU LU [M] received her B.S., M.S., and Ph.D. degrees from Nanjing University in 1992, 1995, and 1997, respectively, all in computer science. She is currently a professor in the Department of Computer Science and Technology and the deputy director of State Key Laboratory for Novel Software Technology. Her research interests include distributed computing, pervasive computing, and wireless networks. She is a member of ACM.

DAOXU CHEN [M] was a visiting professor in Purdue University and City University of Hong Kong. He is now a professor in the Department of Computer Science, Nanjing University. His research interests include distributed computing, parallel processing, and computer networks. He has published over 100 research papers in international conference proceedings and journals. He is a member of ACM and a senior member of the China Computer Federation.

Exploiting Mobile Crowdsourcing for Pervasive Cloud Services: Challenges and Solutions

Ju Ren, Yaoxue Zhang, Kuan Zhang, and Xuemin (Sherman) Shen

Ju Ren and Yaoxue Zhang are with Central South University.

Ju Ren is also a visiting scholar at the University of Waterloo.

Kuan Zhang and Xuemin (Sherman) Shen are with the University of Waterloo.

ABSTRACT

With the proliferation of increasingly powerful mobile devices, mobile users can collaboratively form a mobile cloud to provide pervasive services, such as data collecting, processing, and computing. With this mobile cloud, mobile crowdsourcing, as an emerging service paradigm, can enable mobile users to take over the outsourced tasks. By leveraging the sensing capabilities of mobile devices and integrating human-intelligence and machine-computation, mobile crowdsourcing has the potential to revolutionize the approach of data collecting and processing. In this article we investigate the mobile crowdsourcing architecture and applications, then discuss some research challenges and countermeasures for developing mobile crowdsourcing. Some research orientations are finally envisioned for further studies.

INTRODUCTION

According to the report of *eMarketer* in June 2014, the number of global smartphone users surpassed the one billion mark in 2012, and is estimated to be 1.75 billion in 2014. With the explosion of mobile devices, mobile computing has become an overwhelming trend in the development of IT technology as well as the fields of commerce and industry. However, mobile devices are facing some limitations on various resources, e.g., computation, memory, and energy. To overcome these limitations, mobile cloud computing has become a promising solution to enable mobile devices to consume varied cloud resources via wireless networks. Such a cloud computing service model, i.e. mobile as a service consumer (MaaSC), can improve the computation capability and energy efficiency of mobile devices by offloading computation tasks onto cloud servers [1].

New mobile devices are embedded with a set of versatile sensors, providing a novel paradigm to collect a vast amount of data about individuals, human society, and environments. Meanwhile, since mobile devices are usually associated with human users, human-intelligence can be leveraged for the tasks that are intractable for machine-computation, e.g., entity resolution and image annotation. Empowered by these capabilities, mobile devices shift from service consumers to service providers, offering a new service model for mobile cloud computing, i.e. mobile as a service provider (MaaSP). In this emerging service model, a large number of mobile devices connect with each other via wireless networks, forming an unprecedentedly powerful mobile cloud to provide pervasive data collecting, processing, and computing services. With this powerful mobile cloud, mobile crowdsourcing has been gaining momentum as a feasible solution for solving very large-scale problems. By outsourcing tasks to the mobile cloud, cost-effective and pervasive cloud services can be achieved, using a possibly huge number of mobile users and devices to work together in a distributed way. The ideas behind mobile crowdsourcing involve a wide range of applications and are utilized in different business models [2]. For example, OpenStreetMap [2] is a crowdsourced map of the world, created by worldwide voluntary mobile users using their local knowledge, GPS trajectories, and donated sources. The rapid development of OpenStreetMap indicates that mobile crowdsourcing has the potential to revolutionize traditional data processing and collecting approaches, and in fact already has.

Despite the promising computing paradigm and tremendous advantages, mobile crowdsourcing is still in its infancy and facing many challenges. As mobile users become service providers, social relationships and interactions play a significant role in mobile crowdsourcing. It poses a particular challenge on exploiting the underlying social impacts, such as personal social attributes, preference, selfishness, etc. Meanwhile, in the presence of malicious users, mobile crowdsourcing is vulnerable to various kinds of attacks, e.g., denial-of-service attacks and Sybil attacks. In addition, the private information of both service consumers and mobile users may also be disclosed without sophisticated privacy preservation techniques. Having these security and privacy concerns, mobile users would lose their passion for mobile crowdsourcing. Therefore, it is extremely critical to address these challenges to facilitate the development of mobile crowdsourcing.

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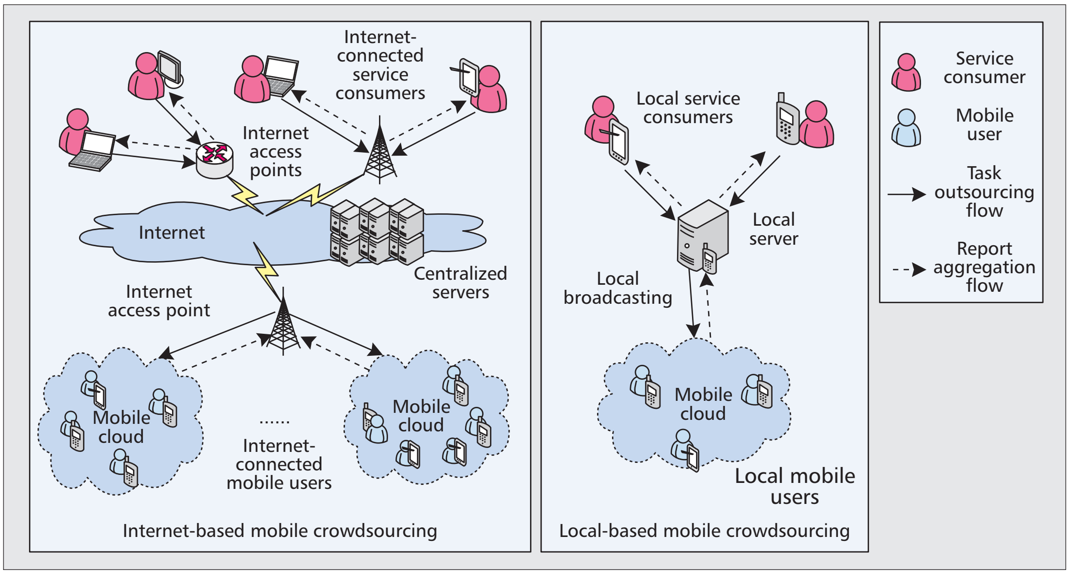

Figure 1. Network architectures of mobile crowdsourcing.

Mobile crowdsourcing is a type of electronic commerce service, where mobile users form a mobile cloud to sell cloud resources and services (e.g., data collecting, computing, and processing) for service consumers.

In this article we investigate the mobile crowdsourcing architecture and existing applications to achieve pervasive cloud services. In addition, we identify some key challenges that impede the implementation of mobile crowdsourcing. Two countermeasures are then presented to address these challenges. Finally, we envision some future research directions and open challenges, and conclude this article.

Mobile Crowdsourcing: Architecture and Applications

Mobile Crowdsourcing Architecture

Mobile crowdsourcing is a type of electronic commerce service, where mobile users form a mobile cloud to sell cloud resources and services (e.g., data collecting, computing, and processing) for service consumers. Different from the traditional cloud computing that depends on Internet connection, mobile crowdsourcing can provide pervasive cloud services for both online and local terminals. Figure 1 shows the architectures of mobile crowdsourcing in an Internet-based scenario and a local-based scenario, respectively. The main difference between the two kinds of mobile crowdsourcing models is that all the Internet-connected mobile users can potentially be a service provider in the Internet-based mobile crowdsourcing, while only the mobile users in the vicinity can provide cloud services in local-based mobile crowdsourcing. We describe the key components of mobile crowdsourcing as follows.

Service Consumers: Service consumers refer to the online and local users that require cloud services through the mobile crowdsourcing system. They utilize the cloud services by outsourcing tasks to mobile users.

Mobile Users: Mobile users with mobile devices can autonomously form a mobile cloud to provide cloud services, for online service consumers via cellular/WiFi networks, or for local service consumers by communicating with local servers or neighboring users using Bluetooth/NFC techniques. When a mobile user participates in an outsourced task, it can adopt local computing or require mobile cloud computing to execute this task.

Centralized Servers: Centralized servers can be seen as a mobile crowdsourcing platform for Internet-based service consumers. They store all the crowdsourcing information (e.g., users' profiles, historical service records) that can be used for task outsourcing and service evaluation. Generally, centralized servers can provide trusted services for task publishing, allocating, report collecting, and feedback processing for the Internet-connected service consumers and mobile users.

Local Servers: Local servers can provide local crowdsourcing services, such as outsourced task broadcasting and task result aggregation, for service consumers and mobile users in the vicinity. Local servers are generally equipped with dedicated mobile local gateways to disseminate the task information to neighboring mobile users and collect user report results. In addition, they can also query or update necessary information from centralized servers to support mobile crowdsourcing. However, local servers are usually deployed for commercial purposes and not trusted by mobile users.

Mobile Crowdsourcing Applications

In the last five years mobile devices have become sensor and information hubs in our daily life. By integrating mobile computing and crowdsourcing, some emerging applications have shown the potential to achieve highly efficient and cost-effective data computation, collection, and processing services. In this section we present two representative mobile crowdsourcing applications, as shown in Fig. 2: mobile crowdcomputing and mobile crowdsensing. Some related works on the two types of mobile crowdsourcing applications are and compared in Table 1.

Mobile Crowdcomputing: Mobile crowdcomputing is used to outsource data computation tasks to mobile users. The mobile users who participate in the outsourced tasks can locally exe-

Generally, mobile users have various capabilities for the outsourced tasks (e.g., personal knowledge, available resources for data collecting and computing), thus, incentive mechanism should stimulate more well-suited mobile users for a specific outsourced task, instead of indiscriminate stimulation.

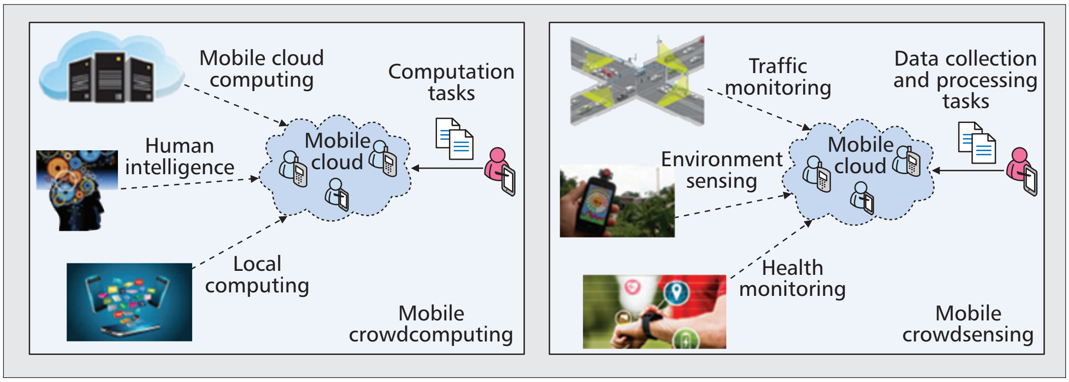

Figure 2. Mobile crowdcomputing and mobile crowdsensing.

cute these tasks or offload them to the cloud severs based on their own data and computation resources. Due to human intervention, mobile crowdcomputing can leverage human-intelligence to deal with the tasks that are more suitable for human evaluation than machine computation (e.g., entity resolution, image annotation, and sentiment analysis). Honeybee [4] is a local-based mobile crowdcomputing application, in which face detection and photography tasks are outsourced to local mobile users. The mobile users use their mobile devices to run face detection algorithms and take specific photos, together with their personal evaluation. CrowdDB [3] crowdsources the computing tasks in the form of querying and answering, based on the Amazon Mechanical Turk platform.

Mobile Crowdsensing: Data collection and processing, such as environment sensing and monitoring, generally require enormous technical efforts and significant economical resources. Mobile crowdsensing is used to outsource data collection and processing tasks to mobile users, who can perform data sensing with sensor-equipped mobile devices, and execute data processing by local computing or mobile cloud computing. By motivating mobile users' participation, mobile crowdsensing can provide cost-efficient mobile cloud services for data collection and processing. SignalGuru [5] is a local-based mobile crowdsensing application, utilizing smartphones to opportunistically detect current traffic signals and collaboratively exchange their detection information via an ad-hoc network. The smartphones can predict the future schedule of traffic signals based on the collection of exchanged information to guide the driving decision-making. Medusa [6] is a mobile crowdsensing application that collects specific sensing data by outsourcing sensing tasks, including video documentation, auditioning, and road monitoring, to Internet-connected mobile users via secure-HTTP based wireless communication.

Key Challenges in Mobile Crowdsourcing

Selfishness and Incentive

Motivating mobile users to participate in mobile crowdsourcing is critical for forming a powerful mobile cloud. The incentives to motivate mobile users could be varied, including financial rewards, personal contribution, social gains, etc. Experiences with micro-task markets, such as Amazon Mechanical Turk, provide positive indications on monetary incentives, while Wikipedia is a good example of human contribution for non-financial gain. However, since both service consumers and mobile users are selfish and aim to benefit from crowdsourcing, incentive mechanisms should economically balance the requirements of the two parties, and create mutual benefits and a win-win situation. Yang *et al.* [7] propose two incentive mechanisms, including reward-sharing and auction-based, to simulate mobile users to participate in mobile crowdsourcing. In reward-sharing incentives, service consumers offer fixed rewards for their outsourced tasks, and each reward is shared by the task participants according to the time they worked on the corresponding task. They also design a truthful auction-based incentive mechanism, where mobile users make offers for different outsourced tasks and service consumers choose appropriate participants to maximize their own utilities. Both the reward-sharing and auction-based incentive mechanisms can economically stimulate the formation of a mobile cloud and also can achieve mutual benefits. However, neither of them shows discrimination for different mobile users. Generally, mobile users have various capabilities for the outsourced tasks (e.g., personal knowledge, available resources for data collecting and computing), thus incentive mechanisms should stimulate more well-suited mobile users for a specific outsourced task, instead of offering incentives indiscriminately.

Task Allocation

In cloud computing it is essential to apply a type of server instance (e.g., high-memory instance, high-CPU instance) for the computation tasks from a larger number of networked cloud servers, according to the task characteristics and requirements. Similarly, in mobile crowdsourcing, task allocation aims to allocate a specific set of outsourced tasks to a set of mobile users who can potentially finish these tasks more accurately and efficiently. Some factors that may impact task allocation have been investigated in existing works. Reddy *et al.* [8] claim geographic and temporal availabilities of mobile users would highly impact the task delay, which should be considered in participant selection. He *et al.* [9]

Application	Service type	Working platform	Task type	Computation resources	Internet-based or local-based
CrowdDB [3]	Mobile crowdcomputing	Based on Amazon Mechanical Turk	Querying and answering	Human intelligence	Internet-based
Honeybee [4]	Mobile crowdcomputing	Android	Face detection	Human intelligence and machine computation	Local-based
SignalGuru [5]	Mobile crowdsensing	iOS	Traffic signal detection	Machine computation	Local-based
Medusa [6]	Mobile crowdsensing	Android	Environment sensing and data processing	Machine computation	Internet-based

Table 1. Representative existing mobile crowdsourcing applications/systems.

design a recruitment algorithm for a mobile crowdsensing application taking the mobility paths of mobile users into consideration. Despite the existing considered factors, the underlying social impacts between the outsourced task and mobile users should be considered in task allocation. For instance, if the outsourced task is "Find an unoccupied basketball court at the University of Waterloo," the mobile users who are interested in "Sport" and study at this university might be preferred to be recruited in the task. Therefore, investigating the social impacts and determining a matching degree for each pair of task and mobile user, and to describe the potential utility of a mobile user participating in an outsourced task, is necessary and crucial for mobile crowdsroucing. Furthermore, based on the determined matching degrees, an efficient task allocation scheme should be developed to maximize the potential utility for both service consumers and mobile users.

Security Threats

Security is one of the primary concerns for cloud service consumers, while the mobile crowdsourcing philosophy originates from the assumption that mobile users would honestly provide accurate results. This is a contradiction and also a persistent problem for mobile crowdsourcing, since there may be malicious mobile users attempting to misbehave in or undermine the mobile crowdsourcing. The malicious users can fabricate computation or sensing results, or maliciously suspend the ongoing tasks, or launch other types of attacks that can directly or indirectly cause negative impacts on the outsourced tasks and service consumers. Some related work has been proposed to mitigate the impacts of malicious task reports and identify the misbehaving users. Zhang *et al.* [10] develop a robust trajectory estimation strategy, to alleviate the negative influence of abnormal crowdsourced user trajectories and identify the normal and abnormal users. Huang *et al.* [11] employ the Gompertz function to compute the device reputation score and evaluate the trustworthiness of the contributed data. Although trust evaluation is an effective solution to measure the credibility of task reports and assist malicious user detection, the users' privacy may be disclosed by linking the trust values associated with multiple task reports [12]. In summary, a major challenge is to design sophisticated security countermeasures to resist malicious attacks and guarantee a reliable mobile crowdsourcing system.

Privacy Leakage

Another obstacle to the widespread deployment and acceptance of mobile crowdsourcing is the privacy concerns of both mobile users and service consumers. The outsourced tasks could reveal the personal interests and objectives of service consumers. Meanwhile, task reports generally tagged with spatio-temporal information disclose abundant personal information of mobile users, such as location, personal activities, and social relationships. Therefore, privacy preservation is of paramount importance in mobile crowdsourcing. For instance, mobile user's location privacy could be exposed when participating in environment sensing of a small space. Generally, we can use cryptography to provide protection for mobile users and service consumers from being eavesdropped by outside attackers when data transmitting and processing. To preserve data privacy from service consumers, Liu *et al.* [13] propose a collaborative learning scheme for classification tasks, e.g., activity or context recognition, in mobile sensing, which can ensure the classification accuracy without disclosing mobile users' privacy, by utilizing the feature perturbation and regression techniques. Anonymity, as an effective solution for privacy preservation, has also been adopted to preserve mobile users' privacy in mobile crowdsensing [12]. In particular, privacy leakage concerns should be given more attention for local-based mobile crowdsourcing, since local servers are generally deployed for commercial purposes and not trusted by mobile users. Therefore, anonymous techniques should be well developed for information transfer between mobile users and local servers.

Solutions for Mobile Crowdsourcing Applications

In this section we discuss two promising solutions for mobile crowdsourcing. Specifically, we present a social-aware task allocation scheme to address the incentive and task allocation challenges for Internet-based mobile crowdcomputing

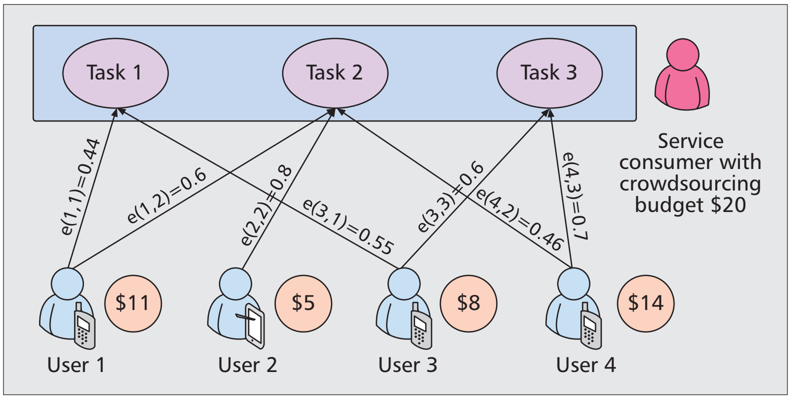

Figure 3. An example of task allocation by SATA.[1]

applications, and an anonymous reputation system to mitigate the security and privacy concerns for local-based mobile crowdsensing applications.

Social-Aware Task Allocation for Internet-based Mobile Crowdcomputing

In Internet-based mobile crowdcomputing, all the Internet-connected mobile users have the potential to provide cloud services for outsourced computation tasks. With such a huge service pool, incentives and task allocation are particularly important for this type of applications to achieve targeted computing services. In addition, since crowdcomputing generally depends on human-intelligence, the social attributes of mobile users (e.g., specialties, social activities) may highly impact the quality of computation tasks. In this section we present a social-aware task allocation (SATA) scheme for a typical Internet-based mobile crowdcomputing application [14], where the crowdsourcing procedures can be detailed as follows. A service consumer first publishes a set of tasks to outsource, which generally contain specific requirements (e.g., interested specialties, delay, and task budget). The mobile users, interested in these tasks, then estimate the task execution information (e.g., processing cost, delay), and apply for participation. Afterward, the service consumer allocates these tasks to a subset of applicants based on their application information. After task allocation, the selected participants execute the outsourced tasks with their own data and computation resources. After the tasks are finished, the task reports will be submitted to the service consumers. Finally, the service consumer evaluates the task reports and gives feedback (e.g., report evaluation, rewards), to the participants.

To motivate mobile users' participation and achieve mutual benefits, SATA adopts an auction-based incentive mechanism, where mobile users announce their bid prices for the interested outsourced tasks, according to their costs and capabilities. Service consumers then choose a subset of mobile users to take the outsourced computation tasks under a fixed budget. In order to identify the well-suited mobile users for a specific computation task, SATA introduces a matching degree, determined by three social factors, for each pair of mobile user and task. Social attribute overlap degree is the primary factor considered in the matching degree calculation. Each mobile user can be characterized by a set of social attributes based on their specialties, social interactions, and personal features, to identify the advantages in addressing some specific types of tasks. If we can specify the interested social attributes for an outsourced task, a higher social attribute overlap degree generally indicates a potential matching between a mobile user and this task. Moreover, each user can estimate a task delay to finish the task based on his available data resources and capabilities, which directly impacts the matching degree, especially for the delay-sensitive tasks. In addition, the reputation of the mobile user is a crucial factor in task allocation, since it indicates the trustworthiness of the mobile user according to his historical task evaluation. For each pair of mobile user and task, if we use three functions f, g, and h, to denote the impacts of social attribute overlap, estimated task delay, and reputation, respectively, the total matching degree can be calculated as a function $e(f, g, h)$ related to f, g, and h.

After the determination of matching degree, task allocation changes to a knapsack problem to maximize the potential utility of the service consumer. More specifically, if a service consumer has a set of tasks to crowdsource under a fixed budget, and there are a set of interested mobile users announcing their bid prices for participating in a subset of tasks, task allocation can be formulated as choosing a subset of interested mobile users to maximize the total matching degree $\Sigma\{e(f, g, h)\}$, subject to the constraint that the sum of the bid prices of chosen mobile users should not be larger than the task budget. It is known that a knapsack problem is NP-hard, but a fully polynomial time approximation solution can be achieved to address this problem [14]. Figure 3 shows an example of task allocation by SATA.

By employing SATA, service consumers can recruit the well-suited participants that can potentially optimize the quality of outsourced computing tasks. Meanwhile, SATA can lead mobile users to participate in the tasks with higher matching degrees, for which they can flexibly adjust their bidding strategies to maximize their own profits. Figures 4a and 4b show the task quality and average user profit comparison, respectively, between SATA and greedy allocation scheme (GAS). Here, GAS refers to allocating tasks to mobile users only according to their bidding prices, without considering the underlying matching degrees [7]. The experiment results show that SATA can notably improve the outsourced task quality and the benefits of mobile users.

Anonymous Reputation Management for Local-based Mobile Crowdsensing

Mobile crowdsensing has been widely adopted as an efficient and economical solution for environment sensing. However, since sensing reports generally contain abundant sensitive personal information, privacy leakage is a significant challenge for mobile crowdsensing applications. Moreover, due to the untrusted local servers, privacy preservation becomes particularly important in local crowdsensing scenarios [15]. In addition, in the presence of malicious users, the

[1] The matching degree of each pair of task and mobile user is denoted by the weight of edge. By employing SATA, User 1 and User 3 will be chosen to potentially optimize the quality of outsourced tasks.

sensing reports are easily fabricated or tampered with to pollute the final sensing results. Therefore, sophisticated security and privacy preservation techniques should be elaborately developed to mitigate these severe challenges for local-based mobile crowdsensing.

In [12], Wang *et al.* propose an anonymous reputation system based on blind signatures to simultaneously achieve trust evaluation and privacy preservation. By adopting trust evaluation and reputation management, malicious users can be detected accurately after repeated misbehaviors [15]. The reputation values of mobile users can also be used for determining the trustworthiness of sensing reports and generating the final sensing results. Meanwhile, blind signatures ensure the authenticity of signed messages without disclosing their content to the signing entity, and also can prevent the signing entity from linking the message with the identity of its generator. Therefore, mobile users' privacy can be preserved from both local servers and service consumers. The anonymous reputation system is illustrated in Fig. 5, which consists of a mobile user, a service consumer, local servers, and a reputation and pseudonym manager (RPM). We describe the procedures of mobile crowdsensing with anonymous reputation management as follows.

First, a mobile user should register with RPM for participating in the task t. RPM obtains the reputation level based on his actual reputation, and creates two certificates, C_u and C_0, for u, where C_0 is the anonymous certificate provided for the service consumer, while C_u contains the actual user ID u and is used by RPM to update u's reputation. Both the certificates are signed by RPM and contain the reputation level of u and task ID. If the mobile user wants to participate in a new task, it has to require two refreshed certificates with a different task ID, which can also partly prevent Sybil attacks [15]. After receiving certificates, the mobile user generates his blind ID as B_u based on the certificate C_u. Every time the user submits a sensing report, it should generate a different blind ID. The service consumer will assess the quality of the submitted sensing reports with the user's reputation level, and create a reputation feedback (RF) as $RF = B_u \mid \{f_{SR}\}_{K_{sp}} \mid C_0$, where f_{SR} is the reputation feedback encrypted by RPM's public key. Therefore, both mobile users and local server cannot decrypt this feedback. After receiving the RF, the mobile user can obtain an unblinded RF (URF) by retrieving C_u from B_u as $URF = C_u \mid \{f_{SR}\}_{K_{sp}} \mid C_0$. Note that both RF and URF are signed by the service consumer and hence cannot be forged by the user. When RPM receives a URF, it processes a security check to validate the URF. Only if the validation is passed will RPM extract the user's ID and reputation feedback to update his current reputation value.

Throughout the whole process, neither service consumers nor local servers can link the sensing reports and reputation values to the real identities of mobile users. Furthermore, by changing the blinded ID for each report submission, it becomes impossible to de-anonymize mobile users by multiple sensing reports.

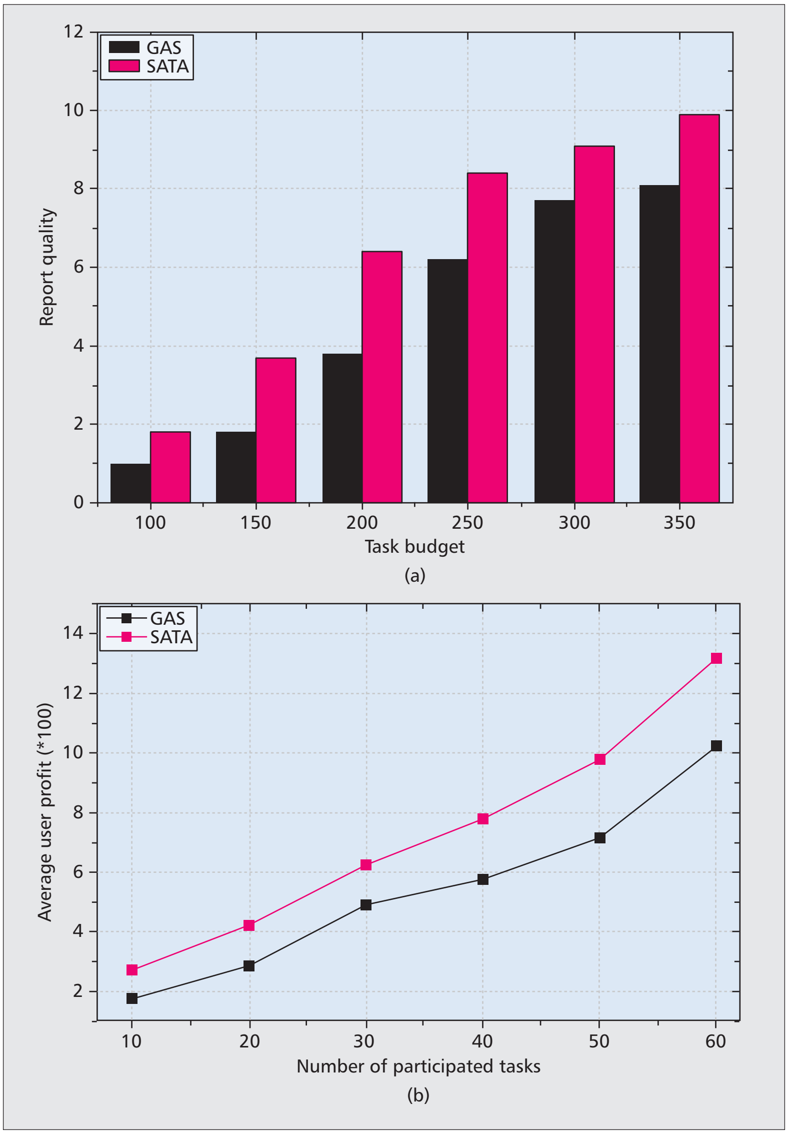

Figure 4. Performance comparison between SATA and GAS[2]: a) task report quality comparison; b) average user profit comparison.

Future Research Directions and Challenges

Recent research has provided some feasible solutions for the key challenges in mobile crowdsourcing, triggering an explosion of mobile crowdsourcing applications. However, there is still a long way to go before researchers will witness the flourishing of mobile crowdsourcing. In this section we present future research directions and challenges to foster continued advancement in this emerging and evolving field of study.

Community Oriented Mobile Crowdsourcing

In mobile crowdsourcing, mobile users themselves form a social network, where the underlying social relationships and interactions cause a significant impact on task crowdsourcing. Social community, as a social structure consisting of individuals with common social interests or attributes, can be introduced into mobile crowdsourcing to improve

[2] The task report quality is determined by the accuracy and actual delay of task reports, while the user's profit is calculated by received task rewards minus task costs.

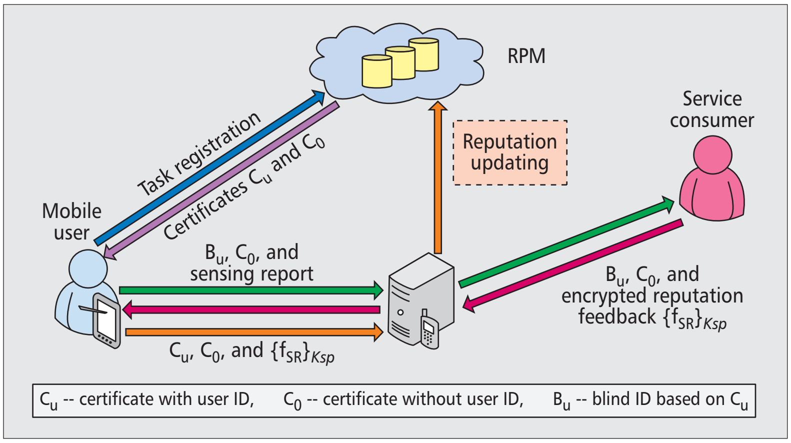

Figure 5. Illustration of anonymous reputation system for local-based mobile crowdsensing (K_{sp} is the public key of RPM).

the social aware task crowdsourcing. By introducing social community, service consumers can focus on outsourcing tasks to different social communities without considering which mobile user should be involved. Meanwhile, communities can recruit well-suited mobile users with required specialties and capabilities for service consumers to accomplish the specific tasks. By leveraging the hierarchical crowdsourcing, service consumers can enjoy more reliable services and achieve targeted task crowdsourcing, especially for tasks requiring specific data resources and processing background. However, exploring community oriented mobile crowdsourcing still faces some challenging issues. Mobile users of a community may contribute their efforts to maximize the profit or reputation of the community, rather than always being selfish. Therefore, task crowdsourcing would be impacted by stimulating users' contribution, and meet new challenges in balancing the benefits of mobile users and their community. In addition, since mobile users can simultaneously belong to different communities, task allocation of a community should consider many social factors (e.g., benefits for mobile users and communities, strength of the social ties between mobile users and different communities) to address the potential conflict of parallel task crowdsourcing in different social communities.

Exploring Big Data Applications by Mobile Crowdsourcing

Digging from previously inaccessible data sets is allowing companies and governments to improve operations and to discover some hidden regularities and new solutions to problems, making big data a hot topic in recent years. However, sorting and analyzing the mountains of information is a challenge even for the largest enterprises and institutions. This embarrassment could be eliminated by mobile crowdsourcing, which is able to exploit the spare processing power of millions of mobile devices and human brains via wireless communication networks. Furthermore, mobile crowdsourcing also provides a faster and more efficient way to access and collect a huge amount of data information from mobile users. The rapid evolution of wearable mobile devices and healthcare applications has proved that the vast potential market for personalized services is attracting industry attention based on mobile crowdsourcing assisted big data applications.

We can foresee the future of integrating mobile crowdsourcing and big data analytics from a successful example in Paris: Tranquilien. This is a smartphone app to help commuters pick a train where they are able to find a seat, based on urban mobility pattern modeling and mobile users' data contribution. By combining user contribution with search queries and location signals, it can accurately observe and predict origin-destination patterns, as well as where people change trains. The unpredicted success of Tranquilien indicates that mobile crowdsourcing and big data are converging, and providing unprecedented opportunities for this big data era. Unfortunately, the integration of mobile crowdsourcing and big data analytics brings not only opportunities but also challenges from both techniques. Since data collection is crowdsourced to a huge amount of various "always-on" mobile devices, the original challenges of data volume, velocity, and variety in big data analytics are amplified by mobile crowdsourcing. Furthermore, both big data and mobile crowdsourcing are still facing the challenge of privacy disclosure. Due to increasingly powerful data mining, personal privacy can be fully exposed even with advanced anonymity techniques. If we aim to outsource data collection to personal mobile devices, privacy concerns would be the biggest obstacle impeding the development of mobile crowdsourcing based big data applications.

Robust Evaluation of Multimedia Reports

In mobile crowdsourcing, fabricated or inaccurate task reports generally lead to useless and even misleading task results and cause significantly negative impacts on users' experiences. Therefore, report evaluation is critical for mobile crowdsourcing to evaluate the quality of aggregated task reports and identify false task reports, which is also the foundation of malicious attack detection. Existing works dealt with this challenge by recruiting multiple participants for a single outsourced task, and detected the false task reports by similarity analysis of these participants' reports [9, 12]. However, since task reports usually contain multimedia content, such as voice, pictures, and videos, especially in mobile crowdsensing, similarity analysis cannot apply to the evaluation of multimedia reports. For instance, if Bob outsources a real-time traffic-monitoring task for a specific area, the task reports could be in the form of pictures or videos. Regardless of which form the report is in, Bob will face the challenge of multimedia report evaluation. Although human evaluation can partially mitigate this challenge, it is not an efficient or even feasible solution for scalable report evaluation. Moreover, report evaluation is also vulnerable to collusion attacks and Sybil attacks, wherein malicious users collaboratively submit false task reports to subvert report evaluation. Therefore, future research can include exploiting an intelligent and robust evaluation scheme for multimedia reports to guarantee the quality and credibility of aggregated task reports, as well as malicious attack detection.

CONCLUSION

In this article we have investigated the mobile crowdsourcing architecture, and presented technical challenges with possible solutions to facilitate the implementation and development of mobile crowdsourcing. By outsourcing tasks to mobile users, mobile crowdsourcing can provide a highly efficient and cost-effective way to achieve pervasive cloud services. We have also discussed future research directions and challenges to nurture continuous improvements for mobile crowdsourcing. It is envisioned that mobile crowdsourcing will accelerate the pervasiveness and evolution of data collecting, processing, and computing.

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BIOGRAPHIES

JU REN [S'13] (ren_ju@csu.edu.cn) received his B.Sc. and M.Sc. degrees in computer science from Central South University, China, in 2009 and 2012, respectively. He is currently a Ph.D. candidate in the Department of Computer Science at Central South University, China. Since August 2013 he has also been a visiting Ph.D. student in the Department of Electrical and Computer Engineering, University of Waterloo, Canada. His research interests include wireless sensor network, mobile sensing/computing, and cloud computing.

YAOXUE ZHANG (zyx@csu.edu.cn) received the B.S. degree from Northwest Institute of Telecommunication Engineering, China, and received the Ph.D. degree in computer networking from Tohoku University, Japan, in 1989. Currently he is a professor in the Department of Computer Science at Central South University, China, and also a professor in the Department of Computer Science and Technology at Tsinghua University, China. His current research interests include computer networking, operating systems, ubiquitous/pervasive computing, transparent computing, and active services. He has published over 200 technical papers in international journals and conferences, as well as nine monographs and textbooks. He is a fellow of the Chinese Academy of Engineering and the President of the Central South University, China.

KUAN ZHANG [S'13] (k52zhang@bbcr.uwaterloo.ca) received his B.Sc. degree in electrical and computer engineering and M.Sc. degree in computer science from Northeastern University, China, in 2009 and 2011, respectively. He is currently working toward a Ph.D. degree in the Department of Electrical and Computer Engineering, University of Waterloo. His research interests include packet forwarding, and security and privacy for mobile social networks.

XUEMIN SHEN [M'97, SM'02, F'09] (xshen@bbcr.uwaterloo.ca) received his B.Sc.(1982) degree from Dalian Maritime University, China, and his M.Sc. (1987) and Ph.D. (1990) degrees from Rutgers University, New Jersey, all in electrical engineering. He is a professor and university research chair, Department of Electrical and Computer Engineering, University of Waterloo. He was the associate chair for graduate studies from 2004 to 2008. His research focuses on resource management in interconnected wireless/wired networks, wireless network security, wireless body area networks, and vehicular ad hoc and sensor networks. He is a co-author/editor of six books, and has published more than 600 papers and book chapters in wireless communications and networks, control and filtering. He has served as the Technical Program Committee Chair for IEEE VTC '10 Fall, Symposia Chair for IEEE ICC '10, Tutorial Chair for IEEE VTC '11 Spring and IEEE ICC '08, Technical Program Committee Chair for IEEE GLOBECOM '07, IEEE INFOCOM '14, General Co-Chair for Chinacom '07, QShine '06 and ACM MobiHoc '15, Chair for IEEE Communications Society's Technical Committee on Wireless Communications, and P2P Communications and Networking. He also serves/served as editor-in-chief for *IEEE Network*, *Peer-to-Peer Networking and Applications*, and *IET Communications*. He is a founding area editor for *IEEE Transactions on Wireless Communications*; an associate editor for *IEEE Transactions on Vehicular Technology*, *Computer Networks*, and *ACM/Wireless Networks*. He has served as a guest editor for *IEEE JSAC*, *IEEE Wireless Communications*, *IEEE Communications Magazine*, and *ACM Mobile Networks and Applications*. He received the Excellent Graduate Supervision Award in 2006, and the Outstanding Performance Award in 2004, 2007, and 2010 from the University of Waterloo, the Premier's Research Excellence Award (PREA) in 2003 from the Province of Ontario, and the Distinguished Performance Award in 2002 and 2007 from the Faculty of Engineering, University of Waterloo. He is a registered Professional Engineer of Ontario, Canada, an Engineering Institute of Canada Fellow, a Canadian Academy of Engineering Fellow, and a Distinguished Lecturer of the IEEE Vehicular Technology and Communications Societies. He has been a guest professor of Tsinghua University, Shanghai Jiao Tong University, Zhejiang University, Beijing Jiao Tong University, Northeast University, and others.

Report evaluation is also vulnerable to collusion attacks and Sybil attacks. Therefore, future research can include exploiting an intelligent and robust evaluation scheme for multimedia reports to guarantee the quality and credibility of aggregated task reports, as well as malicious attack detection.

GUEST EDITORIAL

NEXT GENERATION CABLE NETWORKS WITH DOCSIS® 3.1 TECHNOLOGY

Mehmet Toy

Jim Martin

Matthew Schmitt

Victor Blake

Modern cable networks are continuing to evolve at a rapid pace, driven by consumer demand for new interactive services. This is driving a convergence between traditional broadcast video and Internet video, as well as the need for higher capacity broadband networks. At the same time, Radio Frequency (RF) transmission technology has continued to evolve, offering the opportunity to utilize the resources of the cable network more efficiently and cost effectively. The needs of network evolution, combined with the opportunity of new technology, recently resulted in the development of a new version of DOCSIS® technology, the DOCSIS 3.1 specifications.

DOCSIS 3.1 technology was developed based on the requirements of MSOs (multiple system operators) to ensure the technology would meet their needs for years to come, utilizing the expertise of equipment manufacturers to ensure it was practical and could be built, and with the leadership of CableLabs to help pull it all together. The DOCSIS 3.1 specifications introduce a number of advancements, including: orthogonal frequency division multiplexing (OFDM) to improve network flexibility and utilization; large channel sizes (up to 192 MHz) to improve cost effectiveness at higher bandwidths; low density parity check (LDPC) FEC to improve performance; and adaptive modulation to adjust to different network conditions. Collectively these advances enable a roadmap to downstream data rates of 10 Gb/s and upstream data rates of 1 Gb/s.

The DOCSIS 3.1 specifications were initially released to the public in October, 2013. Since that time equipment manufacturers have been steadily working to turn those specifications into reality. This past December, CableLabs hosted the first DOCSIS 3.1 interoperability event, during which several equipment manufacturers demonstrated DOCSIS 3.1 technology with their products. There have been demonstrations at trade shows as well, and by the time this article is published, more interoperability events will have taken place.

In light of the rapid development of DOCSIS 3.1 devices, the Cable Networks and Services Subcommittee of IEEE ComSoc has organized this Feature Topic to provide more insight into this emerging technology, consisting of four papers.

The first article, titled "DOCSIS 3.1: Scaling Broadband Cable to Gigabit Speeds," describes the building blocks of DOCSIS 3.1 technology and highlights the specific capabilities that will help broadband cable systems scale to support gigabit per second network speeds. This tutorial article also identifies a set of open issues that represent challenges and opportunities for academic researchers to explore.

The second article, titled "Bit-loading Profiles for High-Speed Data in DOCSIS 3.1," describes a key new feature in the DOCSIS 3.1 specifications: multi-tone modulation with different bit-loading per subcarrier, to adapt the transmission to specific conditions as the Signal-to-Noise ratio (SNR) conditions vary for different cable modems (CMs).

The third article, titled "An Experimental RF Noise Cancellation Analysis for Cable Access Systems," describes the design of an experimental RF noise cancellation system to reduce interference signals from Long Term Evolution (LTE) or other RF devices that overlap with the DOCSIS or cable TV spectrum. The designed system is able to achieve an optimal noise cancellation result by adjusting the amplitude and phase of the reference signal to minimize the interference signal transmitted in the coax cable.

The fourth article, titled "Active Queue Management in DOCSIS® 3.1," describes an Active Queue Management (AQM) algorithm providing good application layer quality of experience when multiple applications share a network connection. A variant of the Proportional Integral controller Enhanced (PIE) algorithm, called DOCSIS-PIE, is required for DOCSIS 3.1 cable modems.

We hope the readers find the articles informative, and that this feature topic will contribute to better understanding of the current issues and challenges with DOCSIS 3.1 technology. We would like to thank the authors of all the articles submitted to this special issue, and the reviewers who have given their time generously, providing valuable feedback and comments on the papers to make this feature topic a reality.

BIOGRAPHIES

MEHMET TOY (SM) (Mehmet_Toy@cable.comcast.com) is a distinguished engineer at Comcast, involved in network architectures and standards. He received his B.S. and M.S. degrees from Istanbul Technical University, and a Ph.D. from Stevens Institute of Technology. He has held management and technical positions at well-known companies, and tenure-track and adjunct faculty positions at universities. He has contributed to research, development, and standardization of various technologies, authored five books, a video tutorial, and numerous articles, and he has four patent applications.

JIM MARTIN (jim.martin@cs.clemson.edu) is an associate professor at the School of Computing at Clemson University. His research interests include broadband access, wireless networks, Internet protocols, and network performance analysis. Current research projects include heterogeneous wireless systems and DOCSIS 3.x cable access networks. He has received funding from NSF, NASA, and various corporations. He received his Ph.D. from North Carolina State University and worked for Gartner and IBM, prior to joining Clemson.

MATTHEW SCHMITT (m.schmitt@cablelabs.com) is the vice president, lab services at CableLabs. In this role he is responsible for the labs and testing activities related to the technologies developed at CableLabs, including the certification programs for DOCSIS® and PacketCable™ technology. Prior to that he led the development of the DOCSIS 3.1 specifications at CableLabs. He has worked in the cable industry on DOCSIS technology since 1997 with CableLabs and several vendor companies.

VICTOR BLAKE (victorblake@victorblake.org) is a cable industry consultant. He is the chair of SCTE IPS WG5 and has chaired the initial development of DOCSIS Provisioning of EPON (DPoE). He has taught undergraduate and graduate courses at RIT and Syracuse University. He is a member of the IEEE, SCTE, OSA, SPIE, and CTAM. He has filed two patent applications and is a co-inventor and one issued patent.

DOCSIS 3.1: Scaling Broadband Cable to Gigabit Speeds

Belal Hamzeh, Mehmet Toy, Yunhui Fu, and James Martin

ABSTRACT

The cable industry has recently released the DOCSIS® 3.1 specifications which are the fifth generation of the DOCSIS family of specifications. In this article, we introduce the building blocks of DOCSIS 3.1 and highlight the specific capabilities that will help broadband cable systems scale to support gigabit-per- second network speeds. While this article is a tutorial on DOCSIS 3.1, an important objective is to engage and motivate the academic community to participate in research related to emerging DOCSIS-based cable systems. We identify a set of open issues that represent challenges and opportunities for academic researchers to explore.

INTRODUCTION

In the United States, cable companies serve about 50 million broadband access subscribers, 40 percent more than DSL and fiber subscribers.[1] A major reason for this trend is due to the continued evolution of cable technology. Current cable networks are based on the Data-Over-Cable System Interface Specification (DOCSIS) version 3.0 (DOCSIS 3.0) [1]. Cable Television Laboratories, Inc. (CableLabs), the research and development organization for multiple system operators (we refer to this simply as network operators), has recently released an update to DOCSIS, referred to as the DOCSIS 3.1 specifications [2, 3].

Figure 1 illustrates a modern broadband cable network that applies to both DOCSIS 3.0 and 3.1 systems. Hybrid fiber coaxial (HFC) systems are designed to push content over fiber with the fiber terminating as close to end users as possible. A cable modem termination system (CMTS) interacts with cable modems (CMs) over the coaxial cable. The CMTS resides in the head-end that houses network operator facilities while the CM resides at the subscriber's home residence or business. A subscriber is likely to have multiple devices that consume traditional broadcast video such as televisions and digital video recorders and that use broadband data service such as IP delivered video content and Internet applications. The DOCSIS standard specifies the media access control (MAC) and the physical (PHY) layer procedures.

The network operator's Operational Support System (OSS) contains, among other capabilities, various necessary services including Dynamic Host Configuration Protocol (DHCP), a configuration file and software download server for CMs, a Network Time Protocol server, and a certificate revocation server. The combined functionality of these servers allow for the proper connection and operation of the CMs with the DOCSIS network.

DOCSIS 3.1 was developed to provide subscribers with service speeds up to 10 Gb/s downstream and up to 1 Gb/s upstream. The most significant area of change provided by DOCSIS 3.1 involves spectrum access at the physical layer. Motivating application directions for DOCSIS 3.1 might include:
- Higher video quality such as 4K
- Higher levels of on-demand video
- Network operator provided digital video recorder services;
- Backhaul network service to support cellular operator WiFi offloading;
- Emerging applications enabled by advances in virtual environments, cloud computing, and machine-to-machine communications.

The central focus of this tutorial article is to explore how future cable systems based on DOCSIS 3.1 specifications will scale to support gigabit-per-second network speeds. We introduce the building blocks of modern cable systems provided by the current DOCSIS 3.0 specifications. We highlight DOCSIS 3.1 extensions and present open issues and challenges. An important objective of this article is to identify research issues related to emerging DOCSIS-based cable systems.

This tutorial article is organized as follows. The next section provides a brief overview of the DOCSIS 3.0. Next, we DOCSIS 3.1 enhancements, paying particular attention to modulation and coding profile management and to active queue management (AQM). We highlight specific challenges and identify several research questions. We conclude the article with a brief summary.

DOCSIS BACKGROUND

The DOCSIS 3.1 specifications define the fifth generation of high-speed data-over-cable systems and is part of the DOCSIS family of specifications developed by CableLabs. It builds upon the previous generations of DOCSIS specifications commonly referred to as the DOCSIS 3.0 and earlier specifica-

Belal Hamzeh is with CableLabs.

Mehmet Toy is with Comcast USA.

Yunhui Fu is with Central South University at Changsha.

James Martin is with Clemson University.

[1] http://www.leichtman-research.com/press/081514release.html.

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tions. In this section, we provide an overview of the DOCSIS 3.0 and earlier specifications.

The coaxial medium provides in excess of 3 GHz of usable bandwidth (in ideal signal-to-noise conditions). Frequency division multiplexing is used to isolate bandwidth for downstream (CMTS to CM) and upstream (CM to CMTS) communications. In current DOCSIS 3.0 systems, downstream communications assumes 6 MHz (8 MHz in Europe) channels which are located in the frequency range between 108 MHz and 870 MHz (and optionally 1002 MHz). Most of the spectrum is allocated for traditional broadcast video. Some downstream channels are allocated to support Internet access. Downstream channels can achieve, assuming 256 quadrature amplitude modulation (QAM), data rates of 42 Mb/s per 6 MHz channel. The CMTS uses asynchronous time division multiplexing to share downstream bandwidth between multiple end-stations. In DOCSIS 3.0 systems, packets sent over the downstream channel are segmented based on a 188 byte MPEG frame format that includes 4 bytes of header and a 184 byte payload. The use of MPEG frames facilitates multiplexing the coaxial medium for use with both data and broadcast video. The CM rebuilds IP packets that arrive over the downstream channel.

Upstream channels are 3.2 or 6.4 MHz wide and are located within the frequency range between 5 MHz and 42 MHz (65 MHz in Europe, and optionally 85 MHz in North America). Upstream communications, assuming 64 QAM and a 6.4 MHz channel, can achieve data rates of 30 Mb/s. Additional upstream physical layer operating modes were added in the DOCSIS 2.0 standard. Asynchronous Time Division Multiple Access (ATDMA) and Synchronous Code Division Multiple Access (S-CDMA) to provide communications options that can improve robustness and increase data rates over impaired channels.

In a DOCSIS network, downstream and upstream traffic is classified and managed by service flows. A service flow provides a unidirectional transport of packets over the cable network. Traffic is shaped, policed, and managed based on service quality parameters that were defined for the service flow. A service flow typically includes multiple IP flows. By default, a subscriber is provisioned with two service flows: a downstream service flow that represents all downstream user traffic and an upstream service flow that represents all upstream user traffic. A further example involves a subscriber configuration that supports toll quality IP telephone service and best effort data. This could be provisioned with four service flows: one each for the upstream and downstream VoIP signaling and one each for upstream and downstream aggregate consisting of all the best effort data traffic.

To support higher data rates, DOCSIS 3.0 specifications introduce channel bonding which requires the CMTS to schedule downstream traffic from services flows over potentially more than one channel. A bonding group is an abstraction that represents the group of channels that are available to a service flow. CMs, equipped with multiple tuners and transmitters, are assigned a set of downstream and upstream channels. A common configuration involves 8 bonded downstream channels and 4 bonded upstream channels which can support data rates

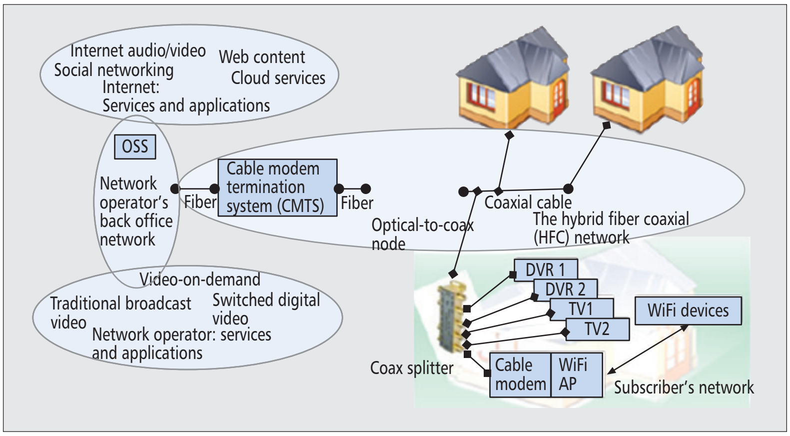

Figure 1. Modern broadband cable network.

up to 320 Mbit/s downstream and 120 Mbit/s upstream. The downstream scheduler, which operates at the CMTS, can allocate bandwidth to service flows from any channel that is in the bonding group assigned to the service flow.

Support for upstream bonding groups is more challenging than downstream due to the request-grant mechanism. Prior to DOCSIS 3.0, a CM was limited to a single outstanding request for bandwidth. This stop-and-wait behavior does not scale to higher speeds. Therefore, DOCSIS 3.0 defined a new request-grant mechanism referred to as continuous concatenation and fragmentation (CCF). CCF allows a CM to send multiple requests for bandwidth concurrently. The CMTS stripes data grants over the set of upstream channels in the service flow's bonding group. A process referred to as segmentation allows the CM to stream data over the bonding group. The CM organizes user data (i.e., packets) into segments which include a sequence number allowing the CMTS to reorder the segments if necessary.

Since every CM tuned to an upstream channel is a potential sender, controlling access to the upstream channels is considerably more complex than downstream operation. The upstream channel is time division multiplexed with transmission slots referred to as minislots. Permission to transmit data in a block of one or more minislots must be granted to a CM by the upstream schedule which operates at the CMTS. The CMTS grants mini-slot ownership by periodically transmitting Upstream Bandwidth Allocation Map messages, which we refer to as MAP messages, on the downstream channel. In addition to ownership grants, the MAP also typically identifies some minislots as contention slots in which CMs may bid for quantities of future minislots. To minimize collisions in the contention slots, a non-greedy back-off procedure is employed. When a CM has a backlog of upstream packets it may also "piggyback" a request for minislots for the next packet in the current packet.

Quality of Service Mechanisms

Many applications have distinctive network service requirements which in turn requires broadband networks to provide quality of service (QoS) mechanisms. VoIP, online gaming, and video streaming are just a few examples of such applications. Additionally, the migration to an all IP video broadcast

Parameter	Value
Lower Band Edge	Mandatory: 254 MHz Optional: 108 MHz
Upper Band Edge	Mandatory: 1218 MHz Optional: 1794 MHz
Signal Type	OFDM
Subcarrier Spacing	50 kHz, 25 kHz
FFT Time Duration	20 µs (50 kHz subcarriers) 40 µs (25 kHz subcarriers)
FFT Size	50 kHz: 4096 FFT; 3800 Maximum active subcarriers 25 kHz: 8192 FFT; 7600 Maximum active subcarriers
Minimum OFDM Channel Bandwidth	24 MHz
Maximum OFDM Channel Bandwidth	192 MHz
Number of Independently configurable OFDM channels	Minimum of 2
Modulation Type	Mandatory: BPSK, QPSK, 8-QAM, 16-QAM, 32-QAM, 64-QAM, 128-QAM, 256-QAM, 512-QAM, 1024-QAM, 2048-QAM, 4096-QAM Optional: 8192-QAM, 16384-QAM

Table 1. Downstream channel parameters.

network makes it critical for the network to provide service guarantees. Each service flow is assigned a scheduling service which is associated with a set of service attributes. For downstream service flows, attributes include traffic priority, traffic shaping parameters (maximum sustained traffic rate and maximum traffic burst), and minimum traffic rate. Because upstream data rates are typically lower than downstream, a set of specific network services have been defined to support application service requirements. As an example, the Unsolicited Grant Service (UGS) is intended to support upstream latency sensitive applications that generate fixed size data packets on a periodic basis, such as Voice over IP. UGS offers fixed size grants (in bytes) on a real-time periodic basis. The default upstream service flow provisioned for each subscriber is treated as a best effort service. The details of the upstream and downstream schedulers are not specified in the DOCSIS specifications and consequently are left for vendor differentiation. Please refer to [1] for further information on DOCSIS services and configuration parameters.

DOCSIS 3.1 System Enhancements

In this section, we provide an overview of the main enhancements provided by the DOCSIS 3.1 specifications. We focus in particular on profile management and AQM.

DOCSIS 3.1 Physical Layer

The DOCSIS 3.1 specifications adopt the use of Orthogonal Frequency Division Multiplexing (OFDM) in the downstream channel and Orthogonal Frequency Division Multiple Access (OFDMA) in the upstream channel. The DOCSIS 3.1 specifications supports 25 kHz and 50 kHz subcarrier spacing options on both channels; the downstream supports channel bandwidths from 24 MHz up to 192 MHz while the upstream channels' bandwidths range from 6.4 MHz to 96 MHz.

The DOCSIS 3.1 specifications extend the range of frequencies that can be used over the coaxial medium. Prior to DOCSIS 3.1 specifications, upstream channels operate in specific band ranges such as 5–42 MHz, 5–65 MHz or 5–85 MHz. The DOCSIS 3.1 specifications introduce two additional bands at 5–117 MHz and 5–204 MHz. Although the use of specific band ranges is typical, for operational reasons a network operator might support other upstream band ranges such as 5-80 MHz by taking advantage of the highly granular channel bandwidths enabled by OFDM signaling. On the downstream, the lower band edge can be as low as 108 MHz (assuming the upstream upper band edge is less than 108 MHz), while the upper band edge can extend up to 1794 MHz.

To maximize the overall spectral efficiency of the system (i.e., to operate as close to the Shannon limit as conditions allow) the DOCSIS 3.1 specifications introduce higher order modulation schemes, with mandatory support up to 4096-QAM on the downstream and 1024-QAM on the upstream. The DOCSIS specifications 3.1 also allow for optional support of 8192-QAM and 16384-QAM on the downstream and 2048-QAM and 4096-QAM on the upstream; this allows the network operator to maximize network capacity and take advantage of signal quality improvements.

Table 1 and Table 2 summarize the downstream and upstream channel parameters. The tables indicates that 4096 or 8192 point Fast Fourier Transform (FFT) is used for downstream and that 2048 or 4096 point FFT is used for upstream. To further enhance spectral efficiency, forward error correction based on low density parity check (LDPC) coding is applied along with interleaving of code words in time and in frequency prior to transmission in both downstream and upstream.

Proactive Network Maintenance — The DOCSIS 3.1 specifications provide enhanced diagnostic tools and statistics monitoring and reporting for proactive network maintenance (PNM). The statistics and measurements are collected at the CMs and the CMTSs distributed within the network. The goal for PNM is to rapidly and accurately characterize, maintain and troubleshoot the upstream and downstream cable plant. Information such as mean error rates per subcarrier, noise power, and forward error correction statistics analyzed over large timescales can be used by the network operator to proactively identify potential performance issues and respond accordingly.

The measurements for PNM are in-service measurements, which mean they are transparent to the end-user without disruption of service.

The PNM functionality provides the network operator with a centralized view of the complete network health and can help predict network faults before they cause a service disruption. The collected measurements and statistics enable the network operator to have embedded functionalities including network analyzers, spectrum analyzers, vector signal analyzers in addition to various functionalities provided by the collected metrics. A detailed description of the collected measurements and statistics is provided in [3].

DOCSIS 3.1 MAC Layer

In this section we summarize the main MAC layer functional extensions introduced with DOCSIS 3.1 specifications.

Profile Management — By using OFDM/OFDMA, DOCSIS 3.1 systems can vary the modulation details over all subcarriers, effectively optimizing the transmission to account for different channel conditions observed by different CMs. In its simplest form, a profile defines a set of subcarriers and associated modulation orders that are assigned to a CM. On the downstream, a CM is required to support 4 profiles for live traffic and 1 test profile for each OFDM channel. For upstream, a CM is must support at least 2 profiles for each OFDMA channel. Profiles are assigned on a per CM basis, thus CMs will be able to operate using the highest modulation order possible according to the signal-to-noise ratio (SNR) of the channel (or profile-assigned subcarriers). Profiles are channel dependent but independent of service groups. For example, if a CM is operating with two downstream channels, the profiles for each channel can be set independently. This can potentially increase system efficiency. This is in contrast to previous versions of the DOCSIS specifications where CMs were required to use the same profile (or modulation order) within the same service group.

Figure 2 illustrates the use of five downstream channels along with four profiles (referred to as Profiles A, B, C, and D). The downstream scheduler must be "channel aware" in that it determines which profile should be used for each transmission. Channel quality information is periodically sent by the CMs to the CMTS. The CMTS selects the best burst profile for a CM based on current channel conditions and on the set of profiles available to the CM. Over longer time scales, the CMTS can update the set of profiles available at a CM. The specification indicates that Profile A will offer a robust configuration through the use of relatively low bit loading to ensure all CMs can receive it. This profile will be used by the CMs for initialization and to carry broadcast MAC management messages. A test profile will be used to check whether a CM can receive a certain profile before the profile is assigned to the CM and live traffic sent on it. The methodology to select the profile configuration, which profiles to assign to a CM, and when to switch between profiles is not specified in the DOCSIS 3.1 specification. As with scheduling, this functionality is meant to be vendor specific.

For upstream, the CMTS specifies the specific channel as well as the specific burst profile that a CM should use for a particular grant.

Parameters	Value
Frequency ranges	5–42 MHz 5–65 MHz 5–85 MHz 5–117 MHz 5–204 MHz
Signal Type	OFDMA
Subcarrier Spacing	50 kHz, 25 kHz
FFT Time Duration	20 ms (50 kHz subcarriers) 40 ms (25 kHz subcarriers)
FFT Size	50 kHz: 2048 FFT; 1900 Maximum active subcarriers 25 kHz: 4096 FFT; 3800 Maximum active subcarriers
Minimum OFDMA Occupied Bandwidth	6 MHz for 25 kHz subcarrier spacing 10 MHz for 50 kHz subcarrier spacing
Maximum OFDMA Channel Bandwidth	96 MHz
Number of Independently configurable OFDMA channels	Minimum of 2
Modulation Type	Mandatory: BPSK, QPSK, 8-QAM, 16-QAM, 32-QAM, 64-QAM, 128-QAM, 256-QAM, 512-QAM, 1024-QAM Optional: 2048-QAM, 4096-QAM

Table 2. Upstream channel parameters.

CMs use a profile denoted as IUC 13 for initialization and then can be assigned to a different profile with higher bit loading for live traffic. A second upstream profile may be assigned to the CM for test purposes or for live traffic.

Downstream multicast supports the use of multiple profiles. The CMTS should attempt to conserve bandwidth by selecting the highest bandwidth profile common to the CMs that are members of the multicast group. If this is not possible, or if a CM requests to join a session but does not support the current session profile, the CMTS can either move all CMs in the session to a lower bandwidth profile or it can replicate the multicast on multiple profiles.

Active Queue Management (AQM) — Bufferbloat in access networks has received significant attention recently [4]. Bufferbloat is caused by network equipment that is provisioned and configured with large unmanaged buffer queues. Its effects are persistently large queue levels that in turn lead to large and sustained packet latency. Active Queue Management (AQM) has long been considered a solution to bufferbloat. Controlled Delay (CoDel) and Proportional Integral controller Enhanced (PIE) are two recently proposed AQM algorithms that address Random Early Detection (RED) issues [5, 6]. Both are delay-based as they proactively drop packets to maintain an average packet queue delay that is less than a configured latency

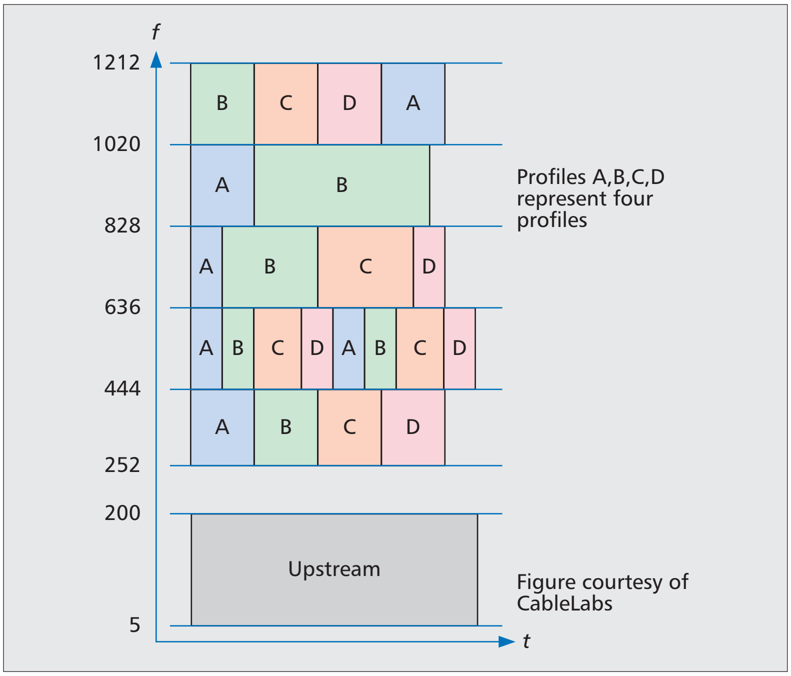

Figure 2. DOCSIS spectrum management.

target. The DOCSIS 3.1 specification requires AQM turned on by default for both downstream and upstream service flows. Further, the CM must support PIE AQM on a per-upstream service flow basis. For downstream, the CMTS is required to support a published AQM. Presumably this will be PIE or CoDel.

The DOCSIS 3.1 specifications introduce other MAC enhancements which we briefly describe below. A complete description of the enhancements is available in [2]. Hierarchical QoS (HQoS) adds an aggregate service flow abstraction allowing an intermediary level of scheduling located between aggregates of service flows and bonding groups. The concept is similar to class-based queueing where different groups of flows (perhaps organized by organization or type of traffic) is allocated a percentage of available bandwidth [7]. Scheduling within one class has no effect on flows that are scheduled within a different class. An energy management mode specifically targeting CMs uses OFDM downstream channels. The CM supports multiple levels of CM functionality and operation, with each level offering reduced capabilities but with lower power requirements. A new DOCSIS Timing Protocol allows CMs to maintain time synchronization with the CMTS within 3000 nanoseconds.

Deployment Discussion

The DOCSIS 3.1 specification is backwards compatible with DOCSIS 3.0 (or earlier) specifications, where a DOCSIS 3.1 CMTS must interoperate seamlessly with DOCSIS 3.0, DOCSIS 2.0 and DOCSIS 1.1 CMs and a DOCSIS 3.1 CM must interoperate seamlessly with a DOCSIS 3.0 CMTS. The CMTS is required to support time and frequency multiplexing of OFDMA and Single Carrier QAM signals (DOCSIS 3.0 signaling) on the upstream. Depending on the scheduling methodology used by the CMTS, DOCSIS 3.1 and DOCSIS 3.0 signaling can be frequency division, time division or time and frequency division multiplexed which provides the highest efficiency in spectrum utilization.

The various generations of video technology deployed in operators' networks define the required video transmission formats over the network. In the extreme case an operator may have to broadcast traditional video channel in multiple forms such as analog, digitally using MPEG2 in both standard and high definition formats, and in IP format for DOCSIS capable set-top boxes. As network operators migrate their access network to all IP (including video), the amount of duplicate broadcasts that has to be maintained is greatly reduced. For example, once an operator decides to no longer support analog, there is no longer the need to have commonly viewed content broadcast in both analog and digital formats. Further, network operators might need to broadcast the same content over a 6 MHz channel as well as over an OFDM channel. The transition from analog video, to digital video to IP video is happening but at different rates depending on the number of legacy devices deployed in the network. Once beyond this migration period, emerging DOCSIS systems will greatly increase the spectral efficiency of the coaxial medium.

Emerging Issues

There are many areas related to the DOCSIS 3.1 specifications that are fertile for innovative research. We briefly identify several examples.

DOCSIS 3.1 Scheduling — The DOCSIS specifications purposely do not address specific scheduling techniques for either the downstream or the upstream. We expect that some form of channel aware scheduling will be supported by a CMTS. The following simple downstream example is meant to illustrate the concept. We consider two subscribers that have the same service plan in a DOCSIS 3.1 network. Assume that one subscriber is poorly connected and is assigned a profile that is optimized to mitigate low signal-to-noise level. The second subscriber is well connected and is assigned a profile that can achieve high data rates over the channel. If these are the only two flows in the network and all packets are the same size, simple round robin downstream scheduling at the CMTS will allocate bandwidth such that the throughput achieved by both flows will be the same (this is referred to as max-min fair allocation [8]). Unfortunately, this underutilizes spectrum capacity. An alternative allocation strategy referred to as 'max throughput' would favor the better connected flow but might starve the poorly connected subscriber. This scheduling complexity has been well studied in wireless networks. A widely accepted compromise is proportional fair scheduling that applies a slight bias towards the better connected subscriber but ensures the poorly connected subscriber also receives a share of available bandwidth [9].

High Bandwidth Applications and Issues — It is well known that TCP has issues in certain network environments. Well studied examples include TCP over wireless and TCP over high speed networks. TCP over cable also has been studied. In fact much of the early research in cable networks focused on TCP issues [10, 11]. Better performing DOCSIS upstream support along with more robust TCP implementations have addressed these early concerns. However, DOCSIS 3.1 systems that offer gigabit-per-second data rates over networks that involve new delay-based AQM mechanisms will require further study to better understand and engineer future deployments.

Wireless Last Hop — Broadband access is likely to involve wireless as the last hop network. A natural evolution for cable networks is to extend services over wireless networks.

Ethernet over DOCSIS — Extending carrier grade Ethernet over DOCSIS is an open issue. The challenge is matching the precise timing and quality required by Ethernet to available DOCSIS profiles and service qualities.

Autonomic Capabilities — Network fault characterization based on PNM measurements and statistics. This can be viewed as a "big data" problem where software needs to analyze large amounts of information searching for early signs of errors, faults, network performance issues, and subscriber quality of experience issues.

Innovative Economic Models — If current tiered pricing models are scaled to support gigabit per second data rate services, the top tiers (i.e. tiers that offer the highest service rates) might not be widely used due to the high cost. New economic models are required.

Virtualization and Software Defined Networking — In the future, cable systems involving virtualizations of CM and CMTS functionality are likely as this can simplify equipment and reduce equipment cost. Further simplification of equipment and automation of provisioning are also expected with applications of Software Defined Network (SDN) concept to DOCSIS systems.

Conclusions

The DOCSIS 3.1 specifications define a flexible and highly extensible technology that is capable of meeting the growing capacity requirements of broadband networks. The main enhancements include more flexible and efficient use of coaxial spectrum. These capabilities will allow broadband cable to scale to gigabit speeds. The more challenging issue might be related to economics. Innovative applications and network services must synergistically evolve to ensure appropriate economic factors are in place so that gigabit-per-second broadband access services see wide usage.

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Biographies

Belal Hamzeh is Director of Broadband Evolution with CableLabs driving technology development activities for wired and wireless technologies. Belal has extensive experience in research and development activities for wired and wireless technologies and currently leads the DOCSIS 3.1 specification development efforts and is the Principal Architect for the RF and Physical layers. Prior to that, Belal led research and development efforts for 3G/4G systems, including standardization, product development and network deployment. He holds an M.Sc. and Ph.D. degrees in Electrical Engineering from Penn State.

Mehmet Toy [SM] is a Distinguished Engineer at Comcast, USA, involved in network architectures and standards. He received his B.S and M.S from Istanbul Technical University, Istanbul, Turkey, and Ph.D from Stevens Institute of Technology, Hoboken, NJ. He held management and technical positions at ADVA Optical Networking, Intergenix, Intel Corporation, Verizon Wireless, Axiowave Networks, Fujitsu Network Communications, AT&T Bell Labs and Lucent Technologies. In addition, he served as a tenure-track and adjunct faculty at universities in the US and Turkey; and served in IEEE Network Magazine Editorial Board, IEEE, and IEEE-USA. He has contributed to research, development and standardization of Cloud Services Architectures, Data Center Network Virtualization Overlays, Self-Managed Networks, Metro Ethernet, DOCSIS Provisioning of EPON (DPoE), IP Multimedia Systems (IMS), Asynchronous Transfer Mode (ATM), optical, IP/MPLS, and wireless. He has four patent applications; authored five books, a video tutorial, and numerous articles and standards contributions in these areas and signal processing. He received various awards from IEEE and the companies, including multiple Inventor Awards from Comcast and Exceptional Contribution Awards from AT&T Bell Labs. He is a member of Cloud Ethernet Forum (CEF) Board, co-chair of CEF Architecture Committee, and founder and chair of IEEE ComSoc Cable Networks and Services (CNS) sub-committee.

Yunhui Fu received a B.S. degree in electrical engineering from the Central South University at Changsha, China, in 2000, and an M.S. degree in Computer Science from the University of Texas-Pan American in 2010. He is currently a Ph.D. student in the School of Computing at Clemson University. His research focuses on improving the reliability and scalability of multimedia communication over wired and wireless packet networks.

Jim Martin is an associate professor in the School of Computing at Clemson University. His research interests include broadband access, wireless networks, Internet protocols, and network performance analysis. Current research projects include heterogeneous wireless systems and DOCSIS 3.x cable access networks. He has received funding from NSF, NASA, the Department of Justice, BMW, CableLabs, Cisco, Comcast, Cox, Huawei, and IBM. He received his Ph.D. from North Carolina State University. Prior to joining Clemson, he was a consultant for Gartner, and prior to that, a software engineer for IBM.

The more challenging issue might be related to economics. Innovative applications and network services must synergistically evolve to ensure appropriate economic factors are in place so that gigabit-per-second broadband access services see wide usage.

Bit Loading Profiles for High-Speed Data in DOCSIS 3.1

Haleema Mehmood, Syed Rahman, and John M. Cioffi

ABSTRACT

DOCSIS 3.1 aims to provide higher speeds over coaxial cable systems than existing cable standards. It uses multicarrier modulation and allows different bit loading per subcarrier. As the subcarrier SNRs for different cable modems are different, DOCSIS 3.1 allows adaptation of data transmission to cable modems by permitting multiple profiles. These provisions enable intelligent bit loading practices in DOCSIS 3.1. This article addresses the problem of assigning profiles to cable modems based on similarity in subcarrier SNRs. Simple algorithms are proposed to group together cable modems suitable for the same modulation profile. It is shown that as the number of allowed profiles increases, higher average data rate can be achieved.

INTRODUCTION

Hybrid fiber coax (HFC) systems must continue to meet the growing demands for faster Internet connections. While all-fiber networks have been proposed for decades as a long-term solution, copper-based telephone and coaxial cable networks have continued to evolve to provide faster services at much lower cost. This has been possible as fiber has migrated slowly into access connections where its high construction costs can be sustained by sharing over many users. As fiber is brought closer to home, new higher-speed twisted pair and coaxial cable standards are introduced. Recognizing the need to ensure high data rates to customers, the cable industry has introduced DOCSIS 3.1 [1]. Through larger channel bandwidths and more advanced modulation and coding, DOCSIS 3.1 will support maximum speeds of up to 10 Gb/s downstream and 1 Gb/s upstream over HFC networks.

Cable systems prior to DOCSIS 3.1 used single-carrier quadrature amplitude modulation (SC-QAM).[1] DOCSIS 3.1 makes better use of available bandwidth with multicarrier modulation to effect larger bandwidth efficiency. DOCSIS 3.1 uses orthogonal frequency-division multiplexing (OFDM) with low density parity check (LDPC) codes. Unlike traditional OFDM, DOCSIS supports different bit loading per subcarrier, which makes it similar to the discrete multi-tone (DMT) modulation that has been used in digital subscriber line (DSL) systems for 20 years to gain maximum transport efficiency. The support for different bit loading per subcarrier allows for rate optimization that was previously not possible. Considerable work has been done on dynamic spectrum management (DSM) for DSL systems. With DOCSIS 3.1 allowing adaptive modulation, dynamic operation is now possible for cable systems too. The provision of variable bit loading and multiple profiles can be used to maximize benefit to service providers and end users. This article addresses an interesting transmission optimization problem in that context.

A typical cable modem termination system (CMTS) serves a large number of cable modems (CMs). A CMTS has many ports that connect to one or more HFC systems. One or more ports serve a serving group (SG) that includes multiple CMs. Each CM has its own path to the CMTS port despite use of a shared cable for the majority of the path. The channel and noise characteristics vary between each CM and the CMTS. Good performance can be achieved if each transmitted symbol has subcarrier modulations individually tuned for the intended receiver. However, DOCSIS 3.1 restricts the number of bit profiles each CM and CMTS can use for transmission. Considering the downstream case, this means that the CMTS has to communicate with multiple CMs with different signal-to-noise ratio (SNR) profiles using the same bit loading per subcarrier. This in turn means that the CMs need to be grouped together by some measure of similarity so that each group can use the same profile. Figure 1 illustrates the concept.

The DOCSIS 3.1 standard outlines the permissible number of profiles and modulations, the procedures involved in measuring subcarrier SNRs, assignment of profiles to CMs, and other relevant procedures. The decision of grouping and profile selection is otherwise left to the discretion of the vendors. This article describes several techniques for grouping and profile assignment, and demonstrates that doing so results in good system performance. The next section summarizes and explains parts of the

Haleema Mehmood and John M. Cioffi are with Stanford University.

Syed Rahman is with Huawei Technologies.

[1] Support for SC-QAM is maintained in DOCSIS 3.1 for backward compatibility.

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DOCSIS 3.1 standard that are relevant to profiles and their usage. The third section describes and presents simulation results for simple grouping algorithms.

DOCSIS 3.1 Profiles: Specification

DOCSIS 3.1 has moved from single-carrier modulation and narrow channel bandwidths to multi-carrier modulation (OFDM) over much wider channel bandwidths (up to 192 MHz downstream and 96 MHz upstream per channel). OFDM divides the spectrum into a set of orthogonal subcarriers and transmits one modulation symbol on each subcarrier in parallel. OFDM opens the possibility of assigning different modulation orders (bit loading) to different subcarriers. This results in highly efficient usage of the available bandwidth. The system is able to assign a higher modulation order to subcarriers with higher SNRs and a lower modulation order to subcarriers with lower SNRs.

Bit Loading Profile

A bit loading profile is a lookup table that assigns modulation orders to each of the subcarriers in an OFDM symbol independently. DOCSIS 3.1 supports modulation orders upstream from binary phase shift keying (BPSK) to 4096-QAM (CMTS support for 2048-QAM and 4096-QAM is recommended but not required). For downstream, DOCSIS 3.1 allows modulation orders from 16-QAM to 16384-QAM, excluding 32-QAM (with optional support for 8192-QAM and 16,384-QAM). The bit loading profile also includes the option for zero bit loading with no data transmission, which is important to avoid regions of bandwidth that are severely impaired and essential to performance improvement over previous DOCSIS methodologies. The downstream bit loading profile assigns modulation orders to each of the 4096 or 8192 subcarriers of the OFDM symbol independently. The upstream spectrum is divided into groups of subcarriers called minislots. All the subcarriers within a minislot have the same modulation order. The upstream bit loading profile assigns modulation orders to each of the minislots independently according to a loading algorithm.

DOCSIS 3.1 allows multiple bit loading profiles. The bit loading pattern or table is independent from profile to profile. Bit loading profiles are primarily used for data subcarriers. Non-data subcarriers such as pilots, physical link channel, and exclusion bands override the values defined in the bit loading profile and follow a different set of rules. DOCSIS 3.1 requires a minimum of two downstream channels and two upstream channels. Each OFDM channel has its own independent set of bit loading profiles.

The CMTS is responsible for two fundamental tasks pertaining to bit loading profiles:

1. Creation of a set of bit loading profiles for each OFDM channel based on channel conditions
2. Assignment of one or more of the created bit loading profiles for each OFDM channel to each of the CMs

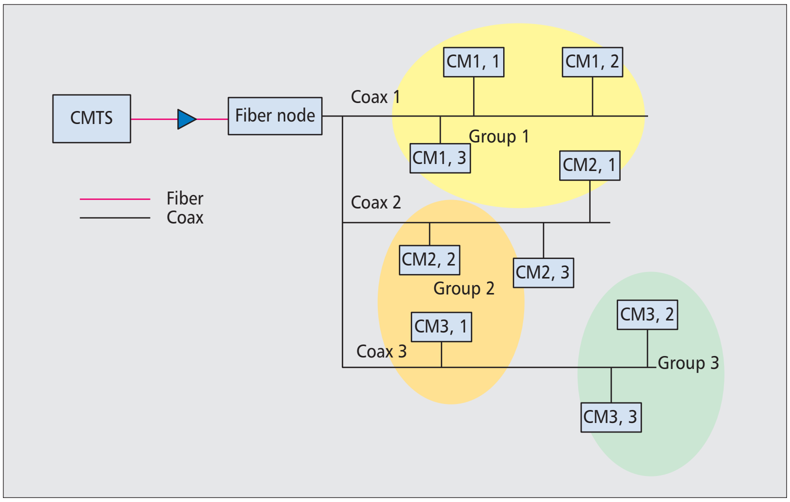

Figure 1. Grouping of CMs in DOCSIS 3.1.

Number of Bit Loading Profiles

A typical CMTS is connected to a large number of cable modems through ports that connect to HFC networks. These provide a logical serving group with a few hundred CMs in each SG. The channel and noise characteristics in the downstream direction vary for each CM. For upstream, noise is the same because the largest component of noise is at the common upstream receiver, but the channel characteristics still differ for different CMs. This channel variation is somewhat compensated by pre-equalization. If the system uses a single bit loading profile for all users, the bit loading for each subcarrier will be based on the minimum SNR across all users for that subcarrier. This limits the data rate of all users to a bare minimum, reducing the overall capacity of the system drastically. On the other hand, if each CM has a separate bit loading profile, for a system with 1000 CMs there will be 1000 bit loading profiles each upstream and downstream, resulting in huge system complexity, latency, memory requirements, and overhead. As a compromise, the number of bit loading profiles was selected to a reasonably low value. In DOCSIS 3.1, there are up to 16 downstream profiles and up to seven upstream profiles in the CMTS, and up to five downstream profiles and two upstream profiles in the CM.

Creation of Bit-Loading Profiles

The CMTS decides both the downstream and upstream bit loading profiles. This decision requires knowledge of channel and noise characteristics of all paths between the CMTS and CMs. For this purpose, the downstream transmission includes known pilots that span the entire OFDM bandwidth and are transmitted once every OFDM frame. An OFDM frame is a group of 128 OFDM symbols. The CM uses these known pilots to estimate the subcarrier SNR values. The CM averages several such estimates obtained over each OFDM frame. Upon request from the CMTS, the CM sends these averaged subcarrier SNR values (with 0.25 dB granularity) to the CMTS. The CMTS uses these subcarrier SNR values from all the CMs to create a set of downstream bit loading profiles. For

The modulation on any subcarrier is, however, constrained by the capability of the weakest CM on that subcarrier within the group. The CMTS needs to take that into account while deciding profile assignment.

upstream, the CMTS commands the CM to send known pilots spanning the entire OFDM bandwidth in a single OFDM *probing* symbol. The CMTS then uses these pilots to estimate the subcarrier SNR values. The CMTS averages several such estimates to obtain average subcarrier SNR values. The CMTS repeats this process for all the CMs and uses the received subcarrier SNR values to create a set of upstream bit loading profiles.

DOCSIS 3.1 uses flat power spectral density across each channel bandwidth. The modulation on any subcarrier is determined by the subcarrier SNR for a particular CM. The CMTS needs to decide which CMs should use the same profile and what those profiles should be. Generally, CMs with similar SNR conditions can be assigned the same bit loading profile. The modulation on any subcarrier is, however, constrained by the capability of the weakest CM on that subcarrier within the group. The CMTS needs to take that into account while deciding profile assignment. Particular algorithms for grouping and profile assignment are vendor-specific and beyond the scope of the DOCSIS 3.1 standard.

Profile Usability Test

Prior to registration, all CMs have only one common downstream profile (boot profile) and one common upstream profile (IUC 13) available to them. The boot profile is very robust (with low data rate) so that the downstream data sent on the boot profile can be received and decoded by all CMs. Similarly, IUC 13 is very robust so that the upstream data sent by any CM using this profile can be received and decoded by the CMTS. Once the CM is registered, the CMTS conducts physical-layer performance tests of the CM on various downstream or upstream profiles so that appropriate profiles (with higher data rates) can successfully be assigned to the CMs.

To test a CM's capability of receiving and decoding a particular downstream profile, the CMTS sends a profile testing request to the CM for that profile. This test profile is not used by the CMTS to send downstream traffic to that CM. The CM uses the traffic of other CMs on this particular test profile to conduct its own performance testing. If the profile to be tested is not already in use in the downstream transmission or carries insufficient traffic, the CMTS transmits test codewords on the test profile. The CM reports its reception conditions of the test profile to the CMTS in the form of "usable/not-usable" along with several other statistics such as received subcarrier SNR values and correctable and uncorrectable codeword errors. Based on the values reported, the CMTS can decide to assign the tested profile to the CM.

To test the suitability of a particular upstream profile, the CMTS sends a grant to the CM and commands the CM to send an "upstream data-profile-testing burst" using a particular upstream bit loading profile. The CM transmits the upstream data-profile-testing burst using the commanded bit loading profile in the allotted grant. The CMTS computes received subcarrier modulation error ratio (MER) and other metrics such as forward error correction (FEC) codeword error count to verify the capability of the CM to use the new bit loading profile.

Mapping of Data to Different Bit Loading Profiles

In the downstream direction, the CMTS maintains a separate buffer for each bit loading profile. A look-up table in the forwarding engine or QoS engine assigns a bit loading profile to each packet and forwards the packet to the appropriate buffer. Packets with the same bit loading profile are sequentially mapped to FEC codewords. The CMTS broadcasts the location and the bit loading profile of each codeword through a separate NCP (next codeword pointer) message. All CMs decode the NCP message. Each CM will decode only those codewords that are associated with its own operating bit loading profile.

In the upstream, the CMTS broadcasts MAP (Media Access Protocol) messages, specifying the assigned minislots and the upstream bit loading profile (IUC) for each upstream burst.

Grouping and Profile Assignment

The previous section outlined the specifications relevant to profile assignment in DOCSIS 3.1. This section presents simple approaches to make efficient use of the provision.

Our analysis lets the total number of allowed bit loading profiles be M, to be used for N users in a given SG. The task at hand is to use N different channel (SNR) profiles corresponding to the N CMs to derive M bit loading profiles (where M is much less than N) in such a way that some measure of data rate is optimized. This task needs to be accomplished independently both for generating upstream and downstream bit loading profiles and for each OFDM channel. The following discussion focuses only on downstream profile assignment for a single channel. A baseline profile for startup is also assumed but not discussed here.

The grouping of CMs under a fixed number of bit loading profiles can be done with various optimization criteria such as:

1. Maximize the sum data rate over all users.
2. Ensure a certain minimum data rate for all users.
3. Maximize the data rate of the worst (minimum data rate) group.
4. Group users according to willingness to pay or data rate demand.

The discussion in this article focuses only on the first criterion, that is, maximizing the sum rate or average rate over all users in the system.

System Model

There are N users in the system, as previously defined. Here, a system is intended to refer to a single SG being serviced by a single port of the CMTS. Each user in the system employs multicarrier modulation with K subcarriers, which are modeled as K independent channels. The number of bits a user can transmit on a subcarrier depends on the SNR on that subcarrier. The SNR in turn depends on the channel-gain-to-noise ratio and power loading on that subcarrier. For DOCSIS 3.1, the power spectral density is flat across the channel bandwidth, so the modulation on any subcarrier is determined only by the channel gain and noise. Each user's loading is thus based on the K-dimensional vector of channel-gain-to-noise ratios. This directly translates into K-dimensional vectors of bit loading for each user, which will be used in the following discussion.

Single Profile for a Group of CMs

Consider the case of multiple CMs that a CMTS wants to communicate with using only one bit profile. The CMTS has to determine the bit profile that all CMs can receive with a certain maximum probability of error. The bit allocation that satisfies that requirement on a certain subcarrier for all users is the minimum of all user bit allocations on that subcarrier. The single profile for a group of users is the minimum modulation on each subcarrier within all of the CMs in the group.

Multiple Profile Selection and Grouping

For the case of choosing M different profiles for N users, two approaches are considered and presented. One approach groups the users together in terms of proximity of vector SNRs. Since the bit capacity of any subcarrier is dictated by its SNR, it makes sense to group users based on overall similarity of SNRs over all subcarriers. Each group can then use the same bit loading profile. A vector grouping or clustering approach is therefore proposed based on the well-known K-means [3] clustering algorithm.

The second approach groups users together with the criterion of maximizing the sum rate of the system. This approach has an explicit goal of improving average data rate of the system, unlike K-means which does it only implicitly. The Bisection Sorting (BiSort) algorithm is proposed (see below), which tries to achieve the objective with low complexity.

K-Means (Vector Quantization)

Consider the K-dimensional problem of grouping N users into M groups, based on the bit capacity of each user on the K subcarriers. The first approach proposed is to use any simple well-known vector quantization (VQ) [4] algorithm. VQ is used to group vectors based on distance from a "centroid." Given a vector source, a distortion measure, and the number of centroids, it finds the groups and centroids that result in the smallest average distortion.

For the set of N K-dimensional input bit vectors, using the well-known K-means clustering algorithm for VQ:

1. Select M bit vectors at random and let them be the initial M centroids.
2. Using the Euclidean distance measure, cluster the vectors around each centroid. This is done by taking each input vector and finding the Euclidean distance between it and each centroid. The input vector belongs to the cluster of the centroid that yields the minimum distance.
3. Compute the new set of centroids. This is done by obtaining the average of each cluster along each dimension. Add the components of each vector and divide by the number of vectors in the cluster.
4. Repeat steps 2 and 3 until the centroids do not change.
5. Once the groups are determined, assign a bit profile per group by finding the minimum bit loading per subcarrier within all CMs in the group.

Using this technique, it is possible to group together N bit profiles into M profiles. This is a greedy algorithm and may fall into local minima. Running the algorithm multiple times can help get better results. The algorithm can also be modified to minimize distance from the minimum in each cluster instead of average Euclidean distance from the centroid. The Euclidean distance measure was chosen because it is a computationally simple and widely used distance metric, but other metrics can be used as well. Also, this kind of clustering does not take into account any particular objective related to bit rates or revenue optimization.

The Euclidean distance measure was chosen because it is a computationally simple and widely used distance metric, but other metrics can be used as well. Also, this kind of clustering does not take into account any particular objective related to bit rates or revenue optimization.

BiSort

The basic problem this algorithm addresses is user grouping and profile assignment with the objective of maximizing the sum rate of the system. Different approaches can be used for such an optimization:

1. Exhaustive search over all possible groupings
2. Formulating and solving the optimization problem
3. Reducing the problem to fewer dimensions and solving a less complex problem

The first approach has huge computational complexity even for moderate problem sizes. An exhaustive search is therefore not feasible even if the computation is done offline.

For the second approach, the optimization problem is non-convex and can be solved using a heuristic approach. A more complete description is beyond the scope of this article.

Consider the third approach on which BiSort is based. It is a simple algorithm that reduces the problem to a single dimension and gives good performance with low complexity.

Assume that the users can be ordered or sorted from best to worst; that is, if a user can achieve a certain rate R, it can also achieve any rate less than R as part of a group with any of the other users, as dictated by the weakest user in the group. This assumes certain channel behavior: if user 1 can achieve rate R_1 and user 2 can achieve rate R_2, each acting alone, $R_1 > R_2$ implies that the channel-gain-to-noise ratio for user 1 is greater than or equal to that for user 2 on all subcarriers. Such sorting of users reduces the problem of grouping to just one dimension. Even for exhaustive search in this case, the size of the problem is much smaller. However, an even simpler algorithm is proposed and shown to give good performance. The BiSort algorithm is explained below.

Let the users be ordered initially according to total individual rates. Using bisection, split the users into two groups. Find the splitting point by

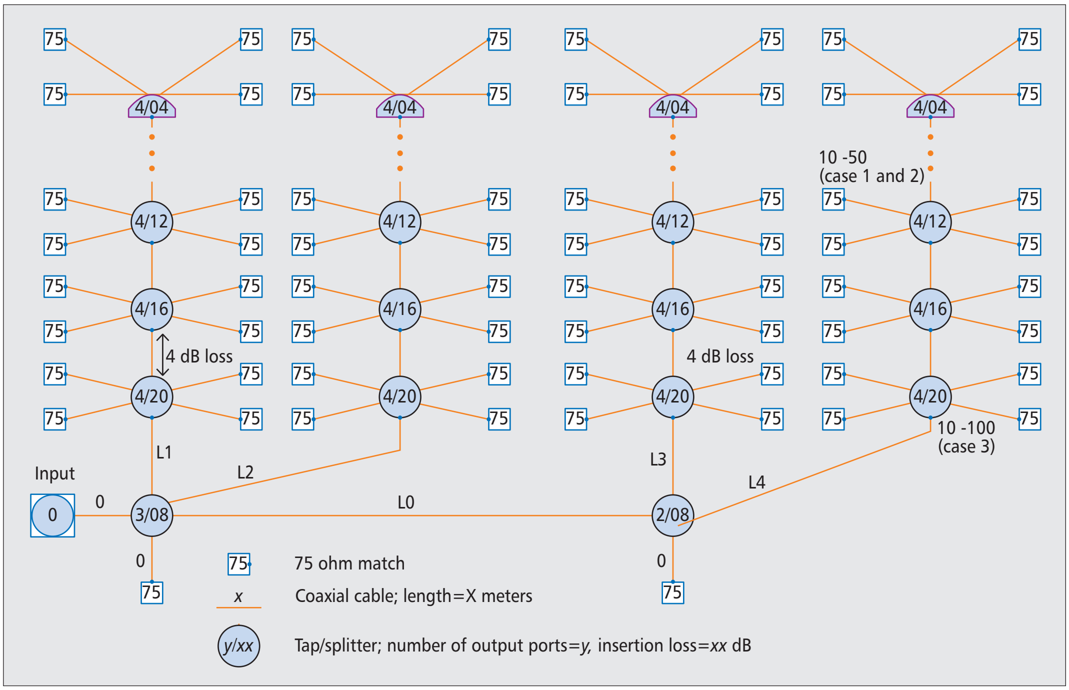

Figure 2. Network layout for simulation.

finding the rate point in the middle that maximizes the sum rate over all users.

Repeat this partitioning step for each group and keep splitting groups into two until the desired number of groups, M, is obtained.

This technique has the underlying assumption that most CM channels differ mainly by an attenuation factor and otherwise have low variation compared to each other. The assumption may not always be true but still gives a good basis for grouping of CMs. Sorting of users based on total data rate hides per subcarrier characteristics and trades off performance for complexity. It uses a single measure for CM performance, simplifying the problem by reducing its size from 4*K* or 8*K* dimensions to a single dimension.

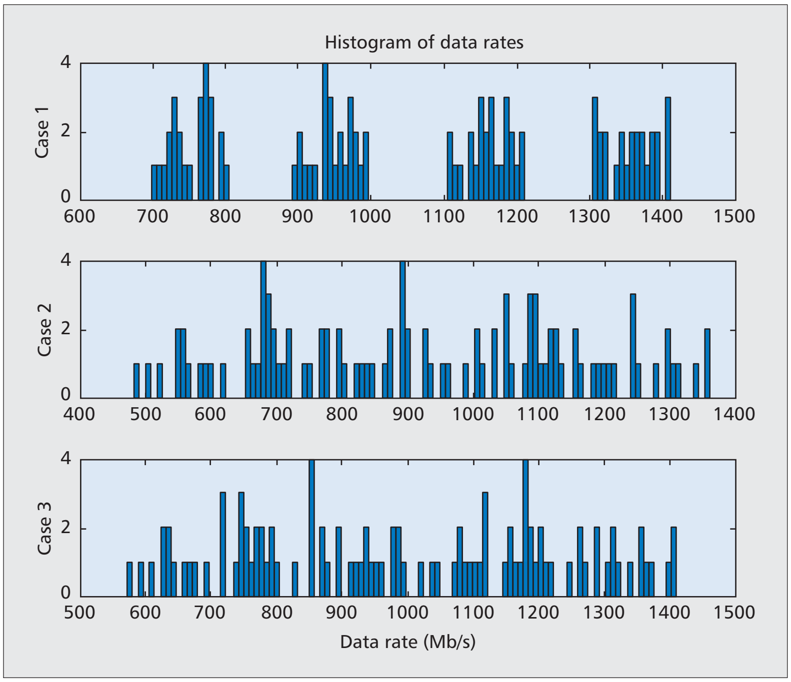

Figure 3. Histogram of data rates with individual profiles.

Performance Results

Simulation results are presented for downstream. The 4*K* mode [1] is used with 192 MHz bandwidth, symbol duration of 22.5 μs (including the cyclic prefix) and 3801 active subcarriers.

The scenario under consideration assumes a single CMTS port serving a network with four major coaxial branches constituting an SG. Each branch has 25 users with a total of 100 users in the SG under consideration. The scenario is depicted in Fig. 2. It is assumed that the attenuation and tilt to the last active are compensated by actives along the path, and node 0 is the input node. Furthermore, the common industry practice of using taps with decreasing insertion loss along the feeder cable is assumed. The tap values in the figure are to demonstrate that idea and are not the actual values used for simulation. This means that there is small difference in signal level at the taps in each branch. However, each branch has a different initial length than the first tap, and this difference will show in the final channel response. The coaxial drop cable to each home also has unaccounted loss and tilt.

As for noise and impairments, a downstream noise floor of –150 dBm/Hz to –155 dBm/Hz is assumed. Intermodulation distortion from 20 analog and 20 digital channels is also simulated. It is further assumed that all CMs in each branch see the same dominant multipath profile. The multipath specification is chosen from the standard specification [1]. A single downstream channel is simulated starting at 422 MHz.

For simulation results, three cases are considered. The length from the drop point to home for all 100 branches is assumed to be between 10 and 50 m for cases 1 and 2, and between 10 and 100 m for case 3. The three cases also differ in the type of coaxial cable used for each case but are otherwise the same. Figure 3 shows a histogram of data rates of the 100 users for three different cases, assuming each user is the only user in the SG and has its own individual profile. This represents the best case performance for each user where it can utilize the entire channel bandwidth without time sharing with any other user. This figure is intended to give an assessment of the data rate capability of individual CMs.

In case 1, the data rates for users in the four branches are clearly separable. This gives the impression that users can be simply grouped together based on coaxial cable branches. The other two cases, however, show that this may not be true for most cases. Case 2 differs from case 1 only in the type of cable used in the network. Here the data rates for users in each branch are not distinguishable. This is expected because the channel characteristics of each individual CM depend on many factors, distance being one important factor of many.

Figure 4 shows the grouping results for different algorithms for case 3 in terms of the data rate per group and the number of users in each group. Here again, the data rate depicted is for each user acting alone using the entire bandwidth. The figure shows that using a higher number of profiles gives more granularity and flexibility, and segregates users that can achieve higher rates from those that can only have limited performance. Different algorithms group users differently, and thereby different SG performance results are achieved. Branch-4 shows the data rates if each branch is considered a single group and $M = 4$. *K*-means-x and BiSort-x show the performance for the *K*-means and BiSort algorithms, respectively, with $M = x$.

Figure 5 shows results of different grouping algorithms and subsequent profile assignment for each of the three cases. A comparison of SG data rate, which is simply normalized individual sum rates (i.e., data rate from equal time sharing of resources between all CMs in the SG) is presented. The highest data rate is achieved if all CMs have individually tuned profiles. Reducing the number of profiles to a single profile severely impairs the performance in all three cases.

The results show that in case 1, *K*-means-4 and Branch-4 give similar results. This is because the users in each branch are clearly similar in data rate performance. However, increasing the number of profiles improves the average data rate performance. That is true for all three cases. Real systems cannot be expected to be clearly group-able into 16 profiles. *K*-means and BiSort algorithms provide simple low-complexity ways to manage grouping and profile assignment for any underlying network characteristics. Table 1 presents the results in numeric form.

These results show how simple grouping techniques help to intelligently decide profile assignment. Such techniques are essential for deriving maximum benefit from provisions in the new standard. They also show that for all cases, the SG data rate for any grouping is higher than a single profile data rate. That is, significant gain can be derived by the use of multiple profiles compared to a single profile.

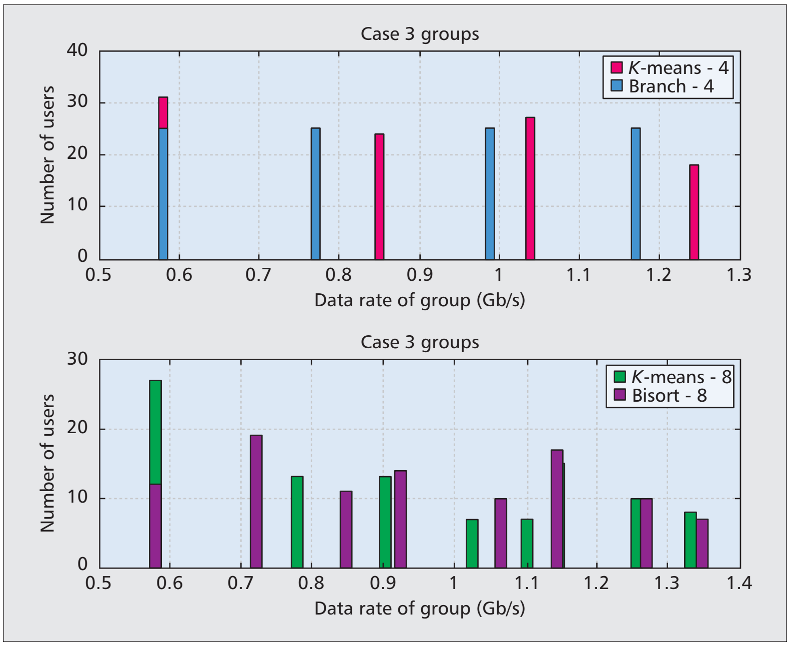

Figure 4. Grouping details for case 3.

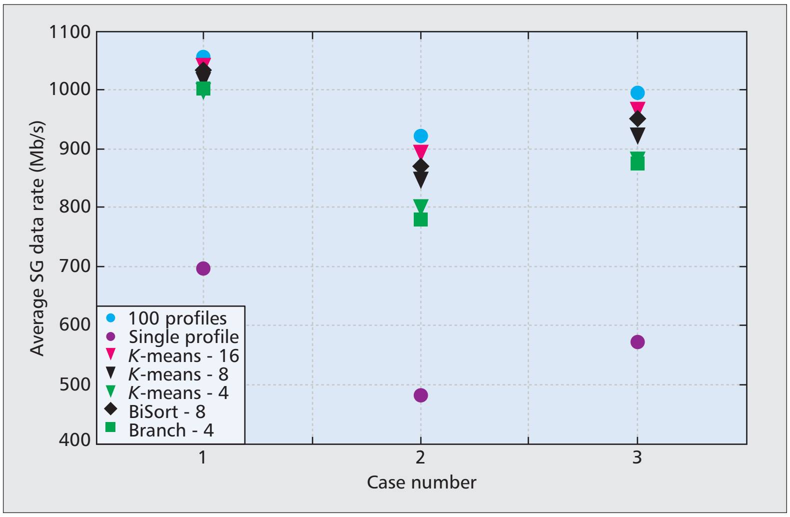

Figure 5. Serving group data rate results.

SG data rate (Mb/s)	Case 1	Case 2	Case 3
100 profiles	1055.5	921.5	992.5
K-means-16	1044.7	896.6	969.5
BiSort-8	1033.2	870.7	951.6
K-means-8	1022.0	851.0	927.8
K-means-4	1002.5	804.9	888.4
Branch-4	1002.5	780.1	874.8
Single profile	698.2	482.6	572.2

Table 1. Serving group data rate results.

These results show how simple grouping techniques help to intelligently decide profile assignment. Such techniques are essential for deriving maximum benefit from provisions in the new standard. They also show that for all cases, the SG data rate for any grouping is higher than a single profile data rate.

Conclusion

DOCSIS 3.1 multiple profile assignment allowance has opened the way for intelligent network management. Users can be grouped together based on similarity in SNR conditions, or the criterion of optimizing data rates or revenue. Vendors and network operators have flexibility in managing profile assignment based on the optimization criterion of their choice. Algorithms can range from computationally complex to quite simple with corresponding trade-off in system performance. With access to real data from live networks, better management decisions can be made, and more efficient grouping algorithms can be developed and tested.

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Biographies

Haleema Mehmood (hmehmood@stanford.edu) is a Ph.D. candidate in the Department of Electrical Engineering at Stanford University. Her research interests are in the area of digital communication and signal processing. She specializes in the application of dynamic spectrum management techniques to wireline and wireless network optimization problems with particular interest in physical layer management for higher-layer utility maximization.

Syed Rahman is currently a member of the cable access research team at Huawei R&D Center, Santa Clara, California. He has over 18 years of industry experience in physical layer algorithm development and digital signal processing involving both wired and wireless systems. He received his M.S and Ph.D. in electrical engineering from the University of Illinois, Chicago. He earned his B.S in electronics and telecommunications from Osmania University, India, with distinction.

John M. Cioffi received his B.S.E.E. in 1978 from the University of Illinois, and his Ph.D. in electrical engineering in 1984 from StanfordUniversity. He has been a professor of electrical engineering at Stanford since 1986-present, now emeritus. He founded Amati Com. Corp. in 1991 (purchased by TI in 1997). He is currently Chairman and CEO of ASSIA Inc. His specific interests are in the area of high-performance digital transmission. Selected awards include the IEEE Alexander Graham Bell Medal (2010); Member of the Internet Hall of Fame (2014); an International Marconi Fellow (2006); and Member, U.S. National and U.K. Royal Academies of Engineering (2001, 2009).

An Experimental RF Noise Cancellation Analysis for Cable Access Systems

Jim Chen, Guangsheng Wu, Xiaoshu Si, David F. Hunter, Pan Wensheng, Ying Shen, and Shihai Shao

ABSTRACT

This article presents the design of an experimental RF noise cancellation system to demonstrate the reduction of interference signals from LTE or other RF devices that overlap with the DOCSIS® or Cable TV spectrum.

The main components of the demonstration system include a four-path RF reconstruction circuit board, SNR monitoring and control module, an adjustable Gradient Control Algorithm in software, and two DOCSIS 3.1 FPGA boards.

The configuration for this demonstration system includes an antenna attached to the RF reconstruction circuit. The antenna is used to receive a reference of the interfering signal. The amplitude and phase of the reference signal is dynamically adjusted to cancel the undesirable component of the RF signal.

The main experimental analysis is focused on a 20MHz wide LTE signal in 700MHz bands. The designed system is able to achieve an optimal noise cancellation result by adjusting the amplitude and phase of the reference signal to minimize the interference signal transmitted in the coax cable.

INTRODUCTION

Mobile network operators have been rolling out Long Term Evolution (LTE) based wireless broadband services using RF bands that overlap frequencies used in the cable TV networks. The 700MHz overlap bands, shown in Fig. 1, are a source of interference to cable TV networks. Verizon, AT&T, and others are rapidly deploying 700MHz 4G LTE networks throughout the U.S. These 4G wireless networks have been growing with tens of thousands of cell base stations, towers and antennas. The 4G LTE wireless networks employ an OFDM based digital modulation scheme. The popularity of 4G LTE networks raises the concern of noise ingress into the cable TV plant infrastructure and customer premise equipment.

CableLabs undertook a study to quantify the risk of interference to cable TV customer premise equipment (CPE) from hand-held LTE devices [1]. The study results show that most of the currently deployed cable CPE devices have sufficient shielding to reject LTE interference. However, an area of LTE interference concern is with retail-grade coaxial cable, RF splitters, and loose connectors. These influences can't be removed by increasing the shielding on set top boxes or cable modems. This led CableLabs to develop a method to remove the interference from the desired signal. The remaining sections of this article describe the method, an experimental implementation, and test results.

DESIGN AND ANALYSIS

Figure 2 shows the block diagram of the RF noise cancellation system we designed. One of the DOCSIS 3.1 FPGAs is used to act as a cable modem termination system (CMTS) to generate downstream data, and one FPGA is implemented with the gradient control module which acts as a receiving cable modem, and we designed a hardware board to perform four-path signal reconstruction.

In the diagram, $r'_I(t)$ is the interference signal from the LTE device and $r_u(t)$ is the useful cable signal, where $r_R(t)$ is a combined interference signal of $r'_I(t)$ and useful cable signal $r_u(t)$. The designed RF interference cancellation circuit receives a copy of reference signal $r_I(t)$ through an attached receiving antenna. The received signal $r_I(t)$ flows through the RF reconstruction circuit board, where the signal amplitude, phase, and delay can be adjusted to reconstruct the interference signal $r_C(t)$. The reconstructed signal $r_C(t)$ is then used as a base to perform noise cancellation against the interference signal $r'_I(t)$ transmitted in the coax cable. The fixed delay lines used in the reconstruction path can produce few fixed delay values.

The adjustable delay component can be added to $r'_I(t)$ to prevent the $r'_I(t)$ delay in the coax cable to be less than the delay of $r_I(t)$ transmitted in the RF reconstruction circuit path.

A four-path RF reconstruction circuit board is designed as part of the cancellation system. The reference signal $r_I(t)$ can flow through four parallel paths, each with adjustable amplitude, phase shift, and delay, to produce a combined

Jim Chen, Guangsheng Wu, Xiaoshu Si are with Futurewei Technologies Inc..

David F. Hunter is with CableLabs.

Pan Wensheng, Ying Shen, Shihai Shao are with University of Electronic Science and Technology of China.

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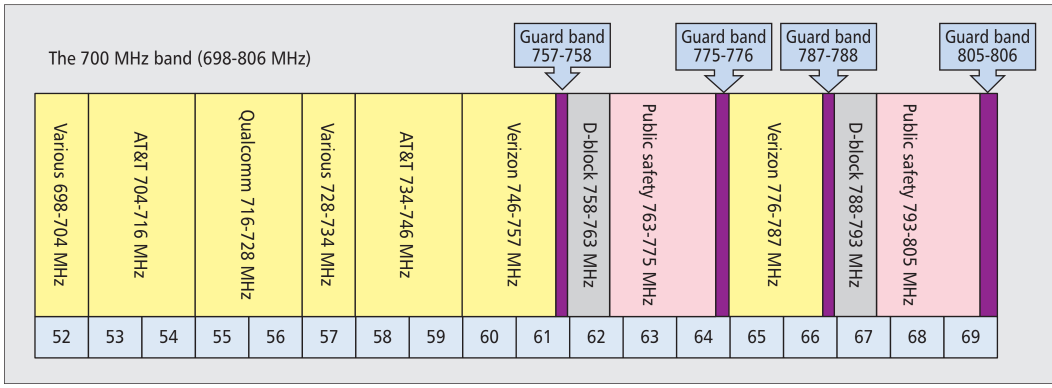

Figure 1. Overlap frequencies in 700 MHz band.

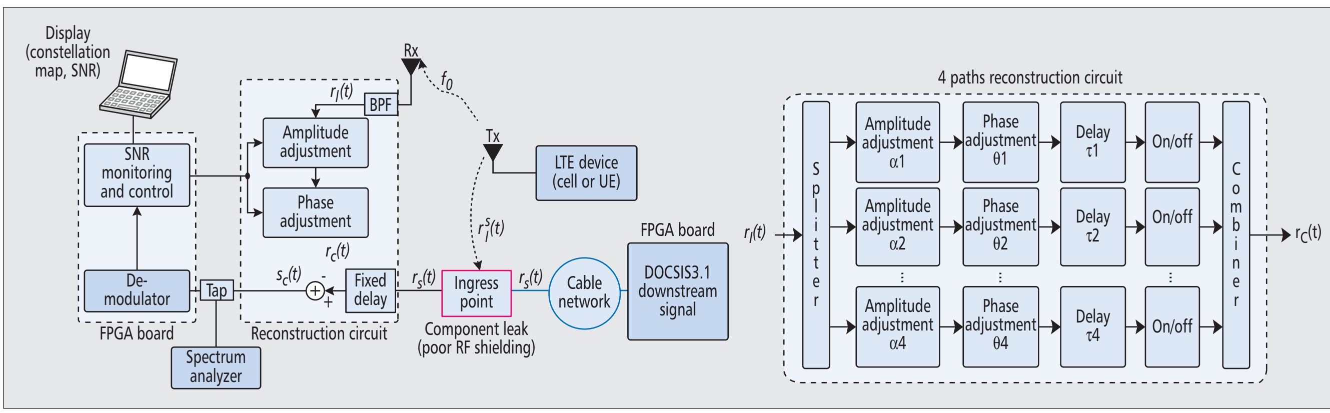

Figure 2. RF noise cancellation system diagram with left figure shows ingress data flow of the LTE signal transmitted in the experimental system and right figure shows the designed 4-path reconstruction circuits.

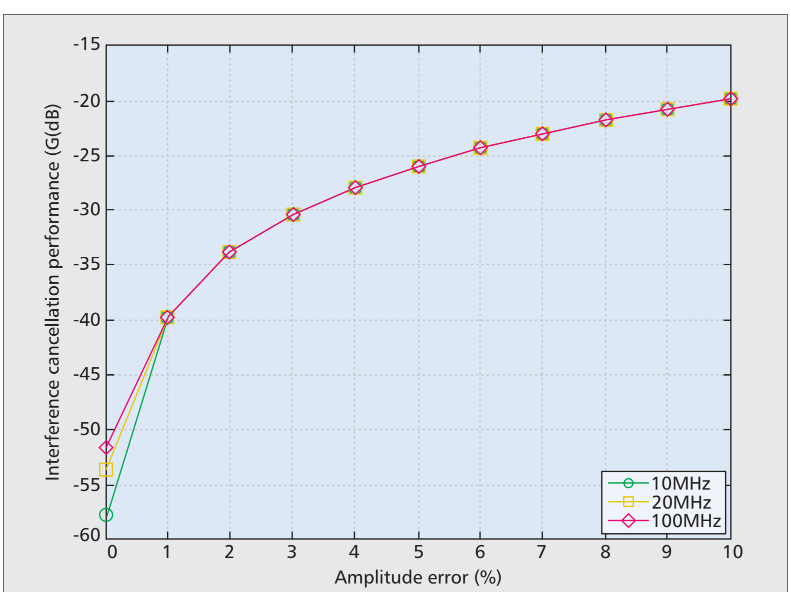

Figure 3. Interference cancellation performance with amplitude error.

reference signal of $r_C(t)$ as an output. The interference cancelled signal $s_C(t)$ can then be obtained by subtracting the interference signal $r_C(t)$ from $r_R(t)$ [2].

The SNR (signal to noise ratio) monitoring control module can track the SINR (signal to interference noise ratio) with a gradient control algorithm based on improved stochastic approximation [3], and the gradient control algorithm is a software module implemented in the DOCSIS 3.1 FPGA board which acts as a receiving cable modem in the cancellation system setup.

The gradient control algorithm uses adjustable amplitude a_i, phase shift q_i, and delay t_i to achieve an optimal reconstructed reference signal and obtain the best SINR. By adjusting the reference signal with amplitude attenuation and phase shift, we can reduce the residual interference energy to almost zero so that a maximum interference cancellation capability can be achieved.

Cancellation Performance Simulation and Analysis

In this section we analyze potential error sources that may impact the output accuracy of the SNR tracking and monitoring module designed in the cancellation system. The error sources can be delay, amplitude, or phase adjustment. To achieve a maximum accuracy of signal reconstruction, we need to examine the corresponding impacts between the interference cancellation capabilities and the error sources.

Figure 3 shows the simulation results related to the cancellation performance with amplitude adjustment error. From the simulation results, we can see the relationship of amplitude adjustment error and the associated cancellation effect. Figure 3 on shows the variation of the cancellation capability with amplitude adjustment errors where the signal bandwidths are in 10MHz, 20MHz, and 100MHz, respectively. With a fixed bandwidth, the data shows interference cancellation capability decreases when amplitude error increases. When the amplitude error is less than 1 percent, the interference cancellation capability in 100MHz is lower than 10MHz or 20MHz signal. If the amplitude error is greater than 1 percent, the interference cancellation capability of all three bandwidths is almost the same.

Figure 4 shows the simulation results related to the cancellation performance with phase adjustment error.

The variation of interference cancellation capability with phase adjustment error for bandwidths in 10MHz, 20MHz, and 100MHz is shown in Fig. 4. The results show that phase error impact for 10MHz or 20MHz is more severe than 100MHz. For example, a 20MHz LTE signal with 0.2° phase error can reduce the interference cancellation capability by about 4.6dB, and a 100MHz LTE-A signal with 0.2° phase error will reduce the interference cancellation capability by only 0.5dB.

Adjustable Parameters

The RF interference reconstruction board is used to reconstruct the signal $r_I(t)$ received by an Omni antenna. After amplitude adjustment and phase shift, the reconstructed reference signal $s_C(t)$ can be obtained and coupled with coaxial cable. The Cable Modem FPGA board has a demodulation module with SINR tracking and control functions. The gradient control algorithm implemented in the FPGA can be tuned with adjustable amplitude and phase to maximize the cancellation effect. To meet the experimental analysis requirements, a range of adjustable parameters can be set with values as shown in Table 1.

Experimental Tests and Results

The RF interference cancellation testing was conducted using the experimental system as shown in Fig. 5. The tests were set up to verify the performance of RF interference cancellation and the influence of interference signal power related to the interference cancellation capability. These tests were conducted in an indoor laboratory environment.

The LTE signal is generated by a signal generator device with added RF amplifier and attached Omni antenna. The LTE signal is tunable in the range of 10dBm to 27dBm [4]. The distance between the transmitting antenna and cable ingress point can be adjusted from 1 to 100 feet in order to generate power strength between 10V/m and 0.01V/m. To achieve an optimal effect of interference cancellation, we can connect the LTE signal generator directly to the ingress point to pick up a stronger signal.

In a real world scenario someone using an LTE-based handset within five feet of a potential ingress point can create field strengths on the order of 2 V/m. Research on RF splitters and coaxial cables, as described in [1], showed that poor quality components can cause

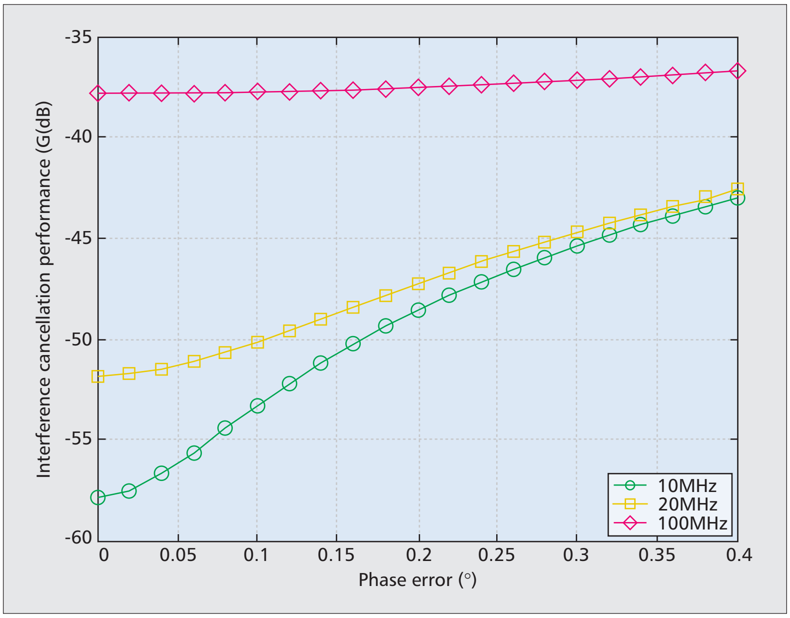

Figure 4. Interference cancellation performance with phase adjustment error.

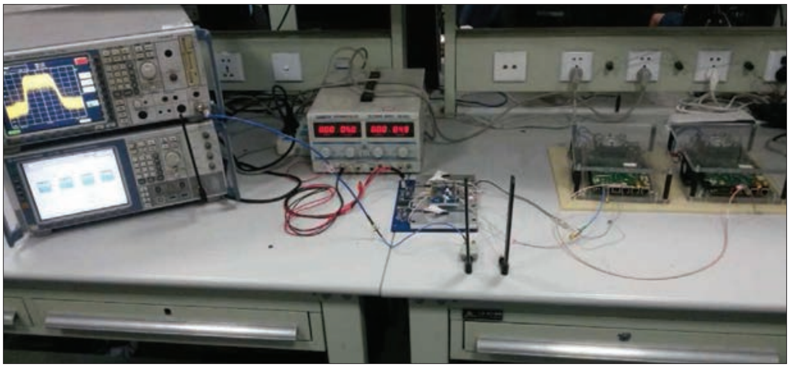

Figure 5. Experimental test setup.

Parameter	Range
Frequency range	500MHz~1200MHz
Interference bandwidth	20MHz
Number of reconstruction paths	1-4
Adjustable delay line of each path	[0, 4, 6, 20]ns
Gain adjustable range	>25dB
Amplitude adjustable steps	0.001dB
Amplitude error adjustment	0.0005dB
Adjustable phase range	360°
Adjustable phase steps	0.1°
Phase error adjustment	0.05°

Table 1. Adjustable parameters.

reduced susceptibility so that an LTE field strength of 0.1 V/m would cause interference and adversely affect the performance of a CPE device without a noise cancellation system. These components act as electrically small and leaky antennas to receive the interfering signal. On the RF splitters in particular the ingress point was through the backing covers that were glued and not soldered to the component case. This allowed the currents flowing on the outer shield to reach the inner conductor. The frequency response of these types of components will be different than the antennas used in the lab testing. The frequency response differences, between the antenna used to receive a copy of the interfering signal and the leaky RF components, will need to be compensated for by the multi-tap reconstruction circuit.

	Item	Parameter
Condition	Interference signal	LTE
	Received interference power (dBm)	–60, –55, –50, –45, –40, –35
	Center frequency (MHz)	725
	Bandwidth (MHz)	20

Table 2. Experimental testing parameters.

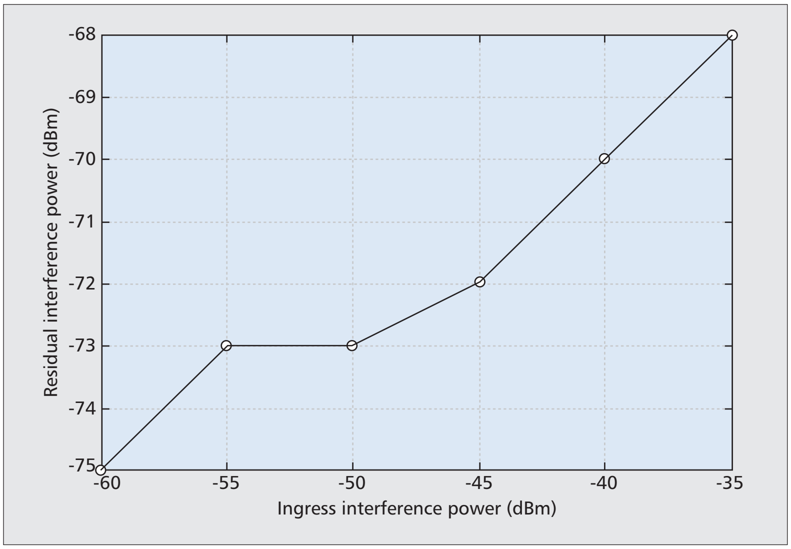

Figure 6. RF cancellation performances vs. ingress interference power.

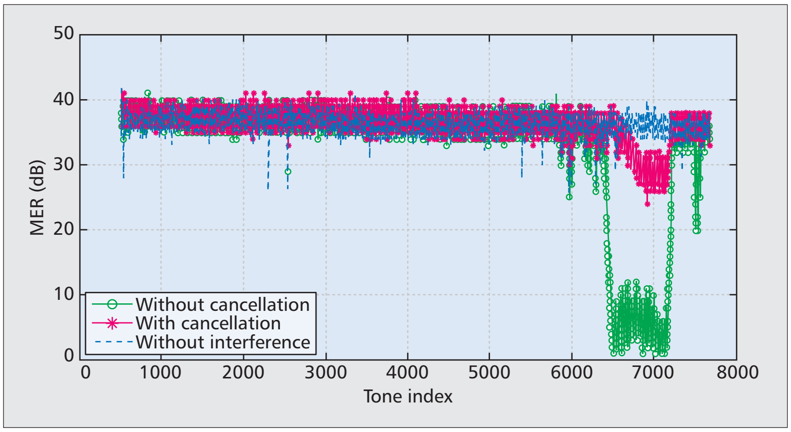

Figure 7. MER performance effect compared with cancellation and without cancellation.

The test procedure we used for the experimental tests can be summarized as follows:

1 Generate 20MHz LTE signal with power of P_t from the signal generator.

2 Turn off interference reconstruction circuits. Record the received interference power P_I.

3 Turn on interference reconstruction circuits to start the RF interference cancellation process. Record the stable residual interference power P_r.

4 Calculate the RF interference cancellation capability.

5 Change the transmission power and repeat the above steps (1) to (4).

Testing parameters used in the above testing procedure are summarized in Table 2.

Figure 6 shows the experimental test results of residual power after cancellation with interference level in the range of –60dBmW ~ –35dBmW.

The test results can be read as follows:

- When the ingress interference power is –35dBm, the residual power after RF cancellation is –68dBm and the cancellation effect of 33dB is achieved.
- With the reduction of ingress power from –35dBm to –60dBm, the cancellation effect decreased to 15dB.
- Due to transmission delay, amplitude, and phase adjustment accuracy to the experimental system, we observed a performance gap between the theoretical analysis and actual test results.

The ingress interference power represents the power of the received reference signal. When the ingress power is weak, the residual interference power will be close to the thermal noise floor. For –75dBm ingress noise, the tested residual power is –85 dBm with PSD of –158 dBmw/Hz, which is close to cable noise floor (cable noise floor at the receiver is around –164 dBm/Hz). With the increase of interference power, the gap between the residual power and cable noise floor is increased and the gap between test results and theoretical simulation results are decreased.

Figure 7 shows the MER performance with and without the cancellation system. From the test result, we observed a significant performance improvement when the experimental cancellation system is applied.

Conclusion

A number of studies have shown that LTE deployment on sub 700MHz or 800MHz frequencies can cause interference to cable systems. This article proposed a novel design of using multi-path RF cancellation circuit to solve the LTE interference issue. It provides detailed analysis and experimental results to show that the RF cancellation process can be effectively operated to mitigate LTE interference in cable systems. The article concludes that:

- The interference cancellation performance increases when the level of the interference power is increased.
- The interference cancellation capability can be limited by the signal transmission delay, amplitude. or phase adjustment error in the testing system.
- A good MER performance can be achieved by applying the designed cancellation system.

Ongoing work will be conducted to evaluate the performance gain associated with replacing the delay lines in the reconstruction circuit with dynamically adjustable components and adding additional reconstruction paths to address multi-path rich environments.

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Biographies

JIM CHEN (chen.jim@huawei.com) is a system architect at Huawei, Santa Clara, California, where he focuses on the research of cable access technologies. Prior to this, he managed the ADC DSLAM Access Product Line and previously worked as a lead system architect in Bell Labs' Switching System Division. He holds a M.S. in computer science from Illinois Institute of Technology

GUANGSHENG WU (wuguangsheng@huawei.com) is a research engineer at Huawei focusing on advanced cable access technologies for MSO HFC networks. His study covers PON and cable broadband access technologies. He has worked on EPOC and DOCSIS3.1 technologies for system architecture and physical layer in Huawei. Prior to joining Huawei, he engaged in EPON technology research and product development and received his Ph.D. from HuaZhong University of Science and Technology, Wuhan, China.

XIAOSHU SI (sixiaoshu@huawei.com) is a research engineer at Huawei focusing on advanced cable access technologies for MSO HFC networks. He is an expert in cable physical layer and worked on EPOC/DOCSIS3.1 physical layer algorithm and prototype development in Huawei. He received his Ph.D. from Wuhan University, Wuhan, China.

DAVID HUNTER (d.hunter@cablelabs.com) is a principle architect focusing on RF transmission systems, both wired and wireless. His background includes developing RF communication systems, including satellite, mobile, and cable TV. He earned an M.S in electrical engineering from the University of Colorado.

WENSHENG PAN (panwens@uestc.edu.cn) is pursuing a Ph.D. degree at the National Key Laboratory of Science and Technology on Communications, University of Electronic Science and Technology of China. He received the B.E. and M.S degrees in communication engineering from the University of Science and Technology of China in 1998 and 2005, respectively. His research interests include RF design, power amplifier design, power amplifier linearization, digital predistortion, and crest factor reduction.

SHEN YING (shenying@uestc.edu.cn) received the BS, MS, and Ph.D. degrees in communications and information systems from the University of Electronic Science and Technology of China, Chengdu, China, in 2002, 2006, and 2009, respectively. He is currently working at the National Key Laboratory of Science and Technology on Communications, University of Electronic Science and Technology of China, Chengdu, China. His research interests include full-duplex and MIMO systems.

SHIHAI SHAO (ssh@uestc.edu.cn) received the B.E. and Ph.D. degrees in communication and information system from the University of Electronic Science and Technology of China, Chengdu, China, in 2003 and 2008, respectively. Since 2008 he has been with the National Key Laboratory of Science and Technology on Communications, University of Electronic Science and Technology of China. His research interests include full-duplex, spread spectrum, and MIMO detection.

Ongoing work will be conducted to evaluate the performance gain associated with replacing the delay lines in the reconstruction circuit with dynamically adjustable components and adding additional reconstruction paths to address multi-path rich environments.

Active Queue Management in DOCSIS® 3.1 Networks

Greg White

ABSTRACT

An important new feature in the DOCSIS 3.1 specification is active queue management (AQM). AQM provides a solution to the problem of providing good application layer quality of experience when multiple applications share a network connection. The need for AQM arises due to the presence of packet buffering in network elements and the mechanics of the TCP congestion avoidance algorithm. Based on simulated performance and implementation considerations, a variant of the Proportional Integral Controller Enhanced (PIE) algorithm, called DOCSIS-PIE, is now included in DOCSIS 3.1 and has recently been added to the DOCSIS 3.0 specification as well. Implementation of DOCSIS-PIE is mandatory for implementation in DOCSIS 3.1 cable modems, and recommended for implementation in DOCSIS 3.0 cable modems. In addition to the mandatory/recommended algorithm, DOCSIS 3.1 and 3.0 cable modem vendors are free to support additional AQM algorithms of their choosing. For managing downstream traffic, the DOCSIS 3.1 specification mandates that the CMTS support an AQM technology, but does not require a specific algorithm. This article discusses the underlying problem that is addressed by AQM, the selection of DOCSIS-PIE as the algorithm of choice for DOCSIS 3.1 cable modems, and the expected performance of the algorithm in simulated network conditions.

INTRODUCTION

WHY IS LATENCY IMPORTANT?

Packet forwarding latency can have a large impact on the user experience for a variety of network applications. The applications most commonly considered as latency-sensitive are real-time interactive applications such as voice over Internet protocol (VoIP), video conferencing, and networked "twitch" games such as first-person shooter titles. However, other applications are sensitive as well; for example, web browsing is surprisingly sensitive to latencies on the order of hundreds of milliseconds.

There are established models for the degradation in user experience for VoIP caused by latency. In the model used for estimating VoIP quality in this article [1], latency has a minimal impact on the VoIP mean opinion score (MOS) (approximately 0.005 MOS points per 20 ms of latency) as long as one-way latency remains below 177 ms. Beyond the threshold of 177 ms, each additional 20 ms of latency reduces VoIP quality MOS score by a more significant 0.13 MOS points.

While online games do not have similar well vetted models for the impact that network parameters have on user experience, a number of researchers have studied the topic, and some data exists to indicate that access network latencies should be kept below 20 ms in order to provide a good user experience.

Loading a web page involves an initial HTTP GET method to request the download of an HTML file, which then triggers the download of dozens or sometimes hundreds of resources that are then used to render the page. While many servers may be involved in providing the page contents, generally speaking, the majority of the resources are served from a small number of servers (four or five). Web browsers will typically fetch the resources from each server by opening up multiple (typically six) TCP connections to the server and requesting a single resource via each connection. Once each individual resource is received, the browser will close the TCP connection and open a new one to request the next resource, thus keeping the same number of connections open at a time. The result of this hybrid parallel-serial download is that the page load time is in some cases driven by the serial aspect, that is, the number of sequential downloads (one completing before the next can start), of which there may be a dozen or more. Round-trip latency can impact the page load time due to the fact that completion of each resource download is delayed by any additional round-trip time (RTT) in the network. Thus, increases in the effective RTT between the client and the server(s) can increase page load time by 10×–20× that amount.

MEGABITS MYTH?

Contrast the above with the sensitivity of page load time to link bandwidth. At the rates cable modem customers are getting today, web page

The author is with Cable Television Laboratories, Inc. (CableLabs).

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load times have reached the point of diminishing returns. In fact, any improvement in link rate beyond about 6 Mb/s returns almost imperceptible improvements in page load time.

Similarly, many other network applications operate at data rates well below what is commonly provisioned for cable modem service.

There is a lot of focus on bandwidth: it is the top-line number that has been used to market high-speed data service. For the foreseeable future that will probably be the case, but when it comes down to the user experience for the actual applications broadband customers are using, improvements in latency may be more important at this point than improvements in bandwidth.

Some sources of latency are hard to address. There is the propagation delay from the user to the server or, for a VoIP session, between two users. There is not much that can be done about the speed of light, but routing paths can be made as short as possible, and content delivery networks can reduce the physical distance and number of hops for some content.

On the other hand, there is a significant issue that has gained a lot of attention in technical circles in the past few years pointing to the fact that a lot of network elements have more buffering memory in them than is really good for application performance.

"Bufferbloat"

Every network element supports buffering of some amount of packets that are destined to be forwarded on the next link. This buffering is important to ensure good utilization of the network link, especially in cases where the incoming traffic rate exceeds the outgoing link rate. In these bottleneck situations, the buffer serves to absorb high-rate traffic bursts so that they can then be played out on the slower outgoing link. Without buffering, most of the packets in the high-rate burst would simply be dropped.

From the perspective of egress link utilization, larger buffers reduce the chance that the egress link will go idle. For bulk TCP traffic (file transfers), user experience is driven by how quickly the file transfer can complete, which is directly related to how effectively the protocol can utilize the network links, again supporting the view that more buffering is better.

But the downside to large buffers is that they result in excessive latency. While this is not an issue for bulk file transfers, it is clearly an issue for other traffic, and the issue is exacerbated by TCP itself.

The majority of TCP implementations use loss-based congestion control, which means the TCP ramps up its congestion window (effectively ramping up its sending rate) until it sees packet loss, cuts its congestion window in half, and then starts ramping back up again until it sees the next packet loss, and that saw-tooth continues. In a lot of networks, especially wired networks, packet loss does not come from noise on the wire. It comes from buffers being full, and when a packet arrives at a full buffer it has to be discarded. This is how TCP automatically adjusts its transmission rate to match the available capacity of the bottleneck link.

The result of this saw-tooth behavior being driven by buffer exhaustion is that the buffer at the head of the bottleneck link is going to saw-tooth between partially full and totally full. Depending on the particular flavor of TCP congestion control (Reno, New Reno, CUBIC, etc.) the portion of time spent in the full (or nearly full) state will vary, and if there are multiple TCP sessions sharing that bottleneck link, the average buffer occupancy will increase. Furthermore, if the buffer is oversized, its average occupancy will be higher as well.

In DOCSIS networks, the cable modem is generally at the head of the bottleneck link for upstream traffic. Historically, and still typically today, cable modems have had a much bigger buffer than is needed to keep TCP working smoothly. Dischinger *et al.* [2] measured buffering latencies on the order of 2–3 s in deployed broadband modems.

Those two factors together — the modem being at the head of the bottleneck link and having an oversized buffer — plus the fact that TCP is going to try to keep that buffer full, results in high upstream latency through the modem whenever there is an upstream TCP session.

The term "Bufferbloat" [3] has been coined to refer to the practice (sometimes inadvertent) of sizing network buffers to be significantly greater than needed to ensure good link utilization, and the resulting significant degradation of interactive applications in the presence of concurrent TCP traffic.

The result of Bufferbloat is that applications other than upstream TCP suffer. Even though the other applications might be low bandwidth, and TCP will back off to accommodate them on the link, their packets arrive to a full or nearly full buffer that may take hundreds of milliseconds or even seconds to play out. This can make web browsing perform poorly, and make VoIP, video chat, or online games unusable. In addition, this could potentially affect downstream TCP performance as well, since the upstream TCP acknowledgments would experience similar latencies. However, this last effect was identified some time ago, and as a result all cable modems have for years supported some kind of TCP acknowledgment prioritization scheme that allows upstream TCP acknowledgments to bypass the large queue.

One reason this situation has persisted in DOCSIS cable modems is that modems, going back to those compliant with the DOCSIS 1.1 specification, have supported multiple service flows. The presumption on the part of modem developers has been that if operators are concerned about latency for certain traffic flows, they can create a separate service flow (which provides a separate buffer) to carry that traffic. Unfortunately, this is not a feasible solution in the vast majority of cases.

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Active Queue Management

As awareness of the topic of Bufferbloat has risen, so too has interest in methods to resolve it. Active queue management (AQM) appears to be the most promising approach because significant network-wide benefits can be derived by

N	F1	F2	Fs	W	VG	C	T
1	0	0	—	1	1	0	0
2	1	1	infinite	1	1	0	0
3	3	3	5MB	1	1	0	0
4	3	3	5MB	1	1	6	0
5	1	0	27MB*	1	1	0	0
6	3	3	5MB	4	4	0	0
7	3	3	5MB	4	4	6	0
8	5	5	0.25MB	4	4	6	0
9	3	3	5MB	4	1	0	100

N: traffic load index
F1: number of simultaneous FTP uploads with 20ms RTT
F2: number of simultaneous FTP uploads with 100ms RTT
Fs: FTP file size
W: number of simultaneous web users
VG: number of simultaneous VoIP/gaming sessions C: CBR data rate (Mb/s)
T: number of torrent (LEDBAT) connections
*Filesize and repetition pattern chosen to exercise DOCSIS "powerboost" feature

Table 1. Traffic scenarios.

implementing it in a relatively small number of bottleneck network elements (e.g., broadband modems). Current AQM approaches seek to detect the "standing queue" created by TCP and, once detected, send TCP a congestion signal (by dropping a packet). The modern algorithms do this without the need to be tuned for network conditions.

DOCSIS 3.1 Active Queue Management

From June 2013 through January 2014, CableLabs worked with the developers of DOCSIS equipment to define an AQM algorithm that would be mandatory for implementation of a cable modem compliant with the DOCSIS 3.1 specification. The DOCSIS 3.1 AQM Working Group evaluated several existing candidate algorithms (extending one of these to improve performance) and two new algorithms developed by CableLabs [6].

Among the candidate algorithms, a version of the Proportional Integral Controller Enhanced (PIE) algorithm [4] optimized for implementation in DOCSIS cable modems was the most attractive candidate due to implementation complexity and alignment with the DOCSIS 3.0/3.1 medium access control layer.

PIE has a distinct advantage over the other algorithms in that the most important parts of the algorithm lend themselves to implementation in software in DOCSIS 3.1 cable modems. This has a couple of advantages. One advantage is that it reduces the development risk for each DOCSIS 3.1 cable modem silicon vendor, since the algorithm does not need to be extensively tested prior to taping out the system on a chip. Another is that it reduces risk for the DOCSIS 3.1 platform in that it allows the algorithm to be modified in the future in devices that are in the field. An additional benefit of PIE is that the algorithm has the potential to be implemented in existing DOCSIS 3.0 cable modems.

CableLabs' DOCSIS 3.1 specification [7] mandates that cable modems implement a specific variant of the PIE AQM algorithm. This specific variant is referred to as DOCSIS-PIE [5]. CableLabs' DOCSIS 3.0 specification has been amended to recommend that cable modems implement the same algorithm. Both specifications allow that cable modems can optionally implement additional algorithms that can then be selected for use by the operator via the modem's configuration file. These requirements on the cable modem apply to upstream transmissions.

Both specifications also include requirements (mandatory in the DOCSIS 3.1 specification and recommended in the DOCSIS 3.0 specification) that the cable modem termination system (CMTS) implement active queue management for downstream traffic; however, no specific algorithm is defined for downstream use.

Performance Simulation

To illustrate the performance benefits expected from AQM in DOCSIS 3.1 networks, simulations have been performed using the Network Simulator (ns2). These simulations focus on scenarios that are anticipated to be particularly relevant for DOCSIS 3.1 deployments in the 2016 timeframe.

Service Model

The configured data rates for the service are extrapolated for 2016 as follows (parameters defined in [7]).

Upstream:
- Maximum sustained traffic rate (MSR): 20 Mb/s
- Maximum traffic burst: 3 MB
- Peak traffic rate (burst rate): 25 Mb/s

Downstream:
- MSR: 100 Mb/s
- Maximum traffic burst: 33 MB
- Peak traffic rate (burst rate): 150 Mb/s

Traffic Models

Broadband Usage Scenarios

Broadband user activity is modeled using nine traffic loads as shown in Table 1. These traffic loads are skewed toward representing peak upstream usage times for a user, rather than being representative of the broadest range of activity levels.

In the web user model the client fetches a single file (representing the html file) and then upon completion of this file transfer proceeds to download 100 resources (of log-normally distributed size) that are spread evenly across four servers. The client maintains six active TCP connections to each server until all 25 resources have been requested from that server. Extrapolating from http://httparchive.org/trends.php for 2016, the total page size (sum of all 101 resources) was configured to be 3.8 MB. The web user downloads the web page, waits 5 s, and then repeats. The metric of interest here is page load time, calculated from the initiation of the TCP connection to download the initial file, to the completion of the TCP session that downloads the final resource. Notably, the web model does not include the Domain Name Service (DNS) lookups that would be present in many real-world page loads. While the number of DNS lookup packets is extremely small relative to the total number of packets involved in loading a page, they are very sensitive to packet loss. The retry timeout for DNS clients is commonly 5 s, so a single DNS packet loss would increase the page load time by approximately that amount.

A single traffic type is used to model both VoIP and online gaming. This traffic type consists of UDP packets of 218 bytes at 50 pkt/s. Packet loss and per-packet latency is monitored, and from these a VoIP MOS score is estimated using a derivation of the International Telecommunication Union (ITU) E-Model for G.711 [1], under the assumption of a 60 ms de-jitter buffer at the receiver and 20 ms of latency outside of the access network.

The upstream torrent traffic is rate-limited as an aggregate to 50 percent of the upstream MSR as an approximation of typical client behavior.

Model Used for TCP Performance Metrics

To assess performance of TCP applications, a model somewhat inspired by Ookla Speedtest.net was used. This scenario is an interesting one because it both represents a commonly utilized methodology for assessing TCP performance in real networks, and is a common scenario in which the user's experience is directly driven by the average TCP upload throughput. In many other upload cases (email, cloud storage and cloud backup, etc.) uploads happen more or less in the background, with the user's interaction (when there is direct interaction) ending with the initiation of the upload task (e.g. clicking "send" on the email message) rather than on its completion.

In the Ookla test, upstream TCP throughput is measured via the use of two simultaneous upstream TCP sessions that transfer data for a total of approximately 10 s. The closest server is chosen by default, but the user can select to run a test to any server in the world. Correspondingly, the simulation uses two TCP sessions, but they are not terminated at 10 s; instead, they are allowed to continue for up to 100 s. Four values of RTT are simulated (20 ms, 50 ms, 100 ms, and 200 ms). The data point from the simulation set that most closely represents typical Ookla throughput results is the average throughput for a 10 s transfer using an RTT of 20 ms, but the other results provide interesting insights into other file transfer conditions.

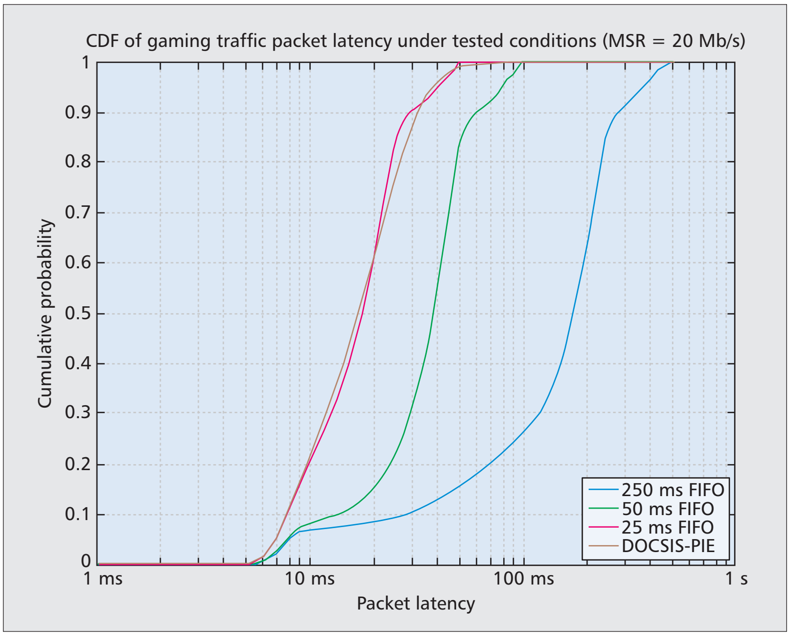

Figure 1. Packet latency statistics for VoIP/gaming traffic.

Shared Channel Congestion Models

Congestion of the shared DOCSIS channel by other users is modeled in order to examine the ability of AQM to respond to changes in available link capacity:

- No congestion: Channel capacity exceeds the peak traffic rate of 25 Mb/s.
- Light congestion: Channel capacity varies among 16.5, 20, 22.5, and 25 Mb/s.
- Moderate congestion: Channel capacity varies among 18.5, 19, 20, and 22.5 Mb/s.
- Heavy congestion: Channel capacity varies among 10, 12, 18, and 20 Mb/s.

For each congestion case, the channel capacity remains at each value for 10 s, and changes in a repeating pattern that exercises all 12 possible rate transitions.

Downstream Queuing

The new CMTS requirements include mandatory support for AQM for DOCSIS 3.1 CMTS and recommended support of AQM for DOCSIS 3.0 CMTS, but no specific algorithm is required. For purposes of this simulation, downstream traffic sees a drop-tail queue at the CMTS with 100 ms of buffering at the MSR (1.25 MB of buffering).

Upstream Queuing

To illustrate the performance of the DOCSIS-PIE AQM algorithm, it is compared against three simple first-in first-out (FIFO) queue configurations. The FIFO queue is representative of earlier generations of DOCSIS modems (e.g., DOCSIS 3.0) or of a situation in which the operator has disabled AQM on a particular service flow. In all, four queuing scenarios are modeled.

250 ms FIFO: In the first of the three FIFO configurations, the cable modem supports a

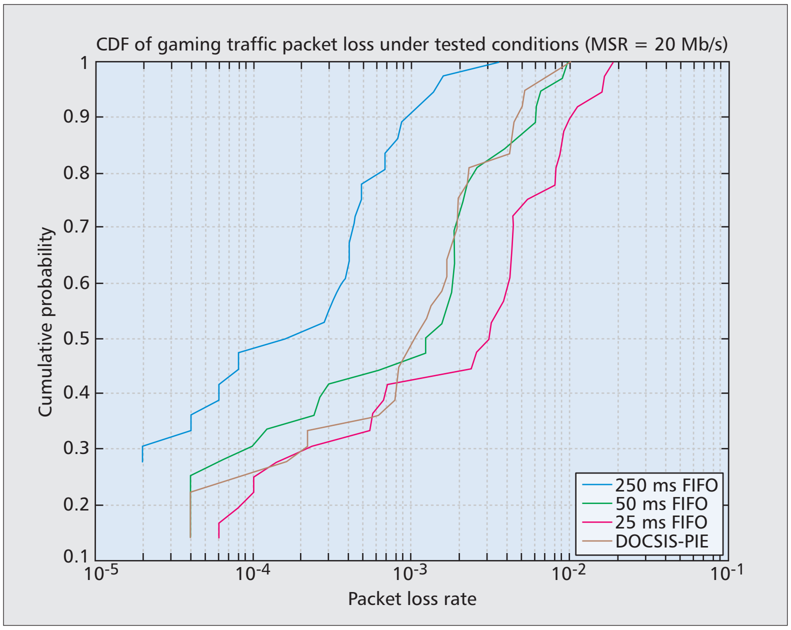

Figure 2. Packet loss statistics for VoIP/gaming traffic.

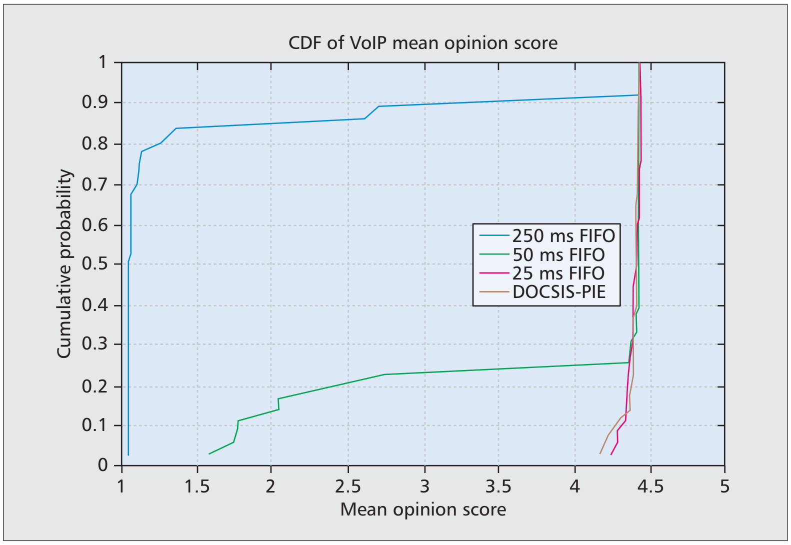

Figure 3. VoIP MOS statistics.

buffer size that is equivalent to 250 ms at the MSR, or 625 kB. This buffer size approximates (or perhaps even understates) the default buffering condition present in a DOCSIS 3.0 modem.

50 ms FIFO: The second FIFO configuration is one in which the cable modem implements a buffer size equivalent to 50ms at the MSR (125 kB). This models a case where the operator has explicitly configured the DOCSIS 3.0 upstream buffer size to a more appropriate size for the service offering.

25 ms FIFO: The third FIFO configuration is one in which the cable modem implements a buffer sized to be 25 ms at the MSR (62.5 kB). This value was chosen to give the closest match to the gaming latency performance of the DOCSIS-PIE algorithm, and could represent a DOCSIS 3.0 service that is optimized for online gaming.

DOCSIS-PIE: The fourth option represents the default condition for a DOCSIS 3.1 cable modem, in which the DOCSIS-PIE AQM algorithm is enabled.

Application Performance

VOIP/Gaming Traffic Performance

Online rapid action games, such as first-person shooter or sports games, require low latency in order to provide a good user experience. In addition, excessive packet loss can impact performance as well.

Figure 1 shows aggregated packet buffering latency results for simulations of the four queuing options in the various network and traffic load scenarios. As can be seen, both the 25 ms FIFO and DOCSIS-PIE options give good (and nearly identical) latency performance for gaming traffic, achieving a median latency of less than 20 ms and a 90th percentile packet latency of approximately 30 ms. The other FIFO options (50 ms buffer and 250 ms buffer) would provide a marginal or poor user experience for games that require low latency.

Figure 2 shows the statistics of packet loss for gaming traffic. While none of these options appear to show excessive packet loss, the DOCSIS-PIE algorithm does excel relative to the 25 ms FIFO option by dropping approximately 1/3 as many packets.

In terms of VoIP user experience, recall that MOS is a 5-point scale with the following values: 5–excellent; 4–good; 3–fair; 2–poor; 1–bad. The MOS estimator used for this analysis assumes a G.711 codec with a maximum MOS of approximately 4.4, and takes into account the latency, jitter, and packet loss experienced by the stream.

The MOS estimator shows that DOCSIS-PIE AQM and 25 ms FIFO both provide excellent performance, whereas the longer FIFO queues result in unacceptable audio quality in certain cases. In fact, the 250 ms FIFO only results in good audio quality in traffic scenario 1 (no TCP upload traffic).

Web Performance

Figure 4 shows the statistics of web page load time. Both 25 ms FIFO and DOCSIS-PIE provide very good web performance, with 90th percentile page load times of approximately 3 s. The 50 ms FIFO also performs fairly well, with a 90th percentile page load time of less than 4 s.

TCP Throughput Performance

In terms of TCP throughput, all of the queuing approaches provide good performance when the session has a short RTT. As the RTT increases, performance degradation can be seen, particularly with the 25 ms FIFO scenario.

Figure 5 shows the averaged TCP performance over moderately short timescales for the four values of RTT. In many locales, and in particular in the United States and other developed regions, RTTs in the 10–50 ms range are likely the most common for upload RTT (due in part to the proliferation of geographically distributed data centers).

As mentioned above, the data point that most closely represents the result one would expect from using speedtest.net is the average throughput at 10 s in the 20 ms RTT case. At that data point, all of the queuing algorithms perform nearly identically. In the longer RTT cases (50ms, 100ms, 200ms) the 25 ms FIFO results show markedly degraded performance relative to the other queuing options.

It is worth noting that the 200ms RTT case has a high bandwidth-delay product (BDP) of 500 kB, equivalent to over 330 packets. When in congestion avoidance, a single Reno-based TCP session (e.g. Windows or Mac OS X) will take a long time to recover from a congestion window decrease. For example, in a case where 50 ms of buffering results in a packet drop, the congestion window will drop from ~400 packets to ~200 packets, and then will take 40 s to recover. A small number of unlucky packet drops can thus have a significant effect on throughput in this case. More advanced TCP implementations such as cubic should recover much more quickly in this situation.

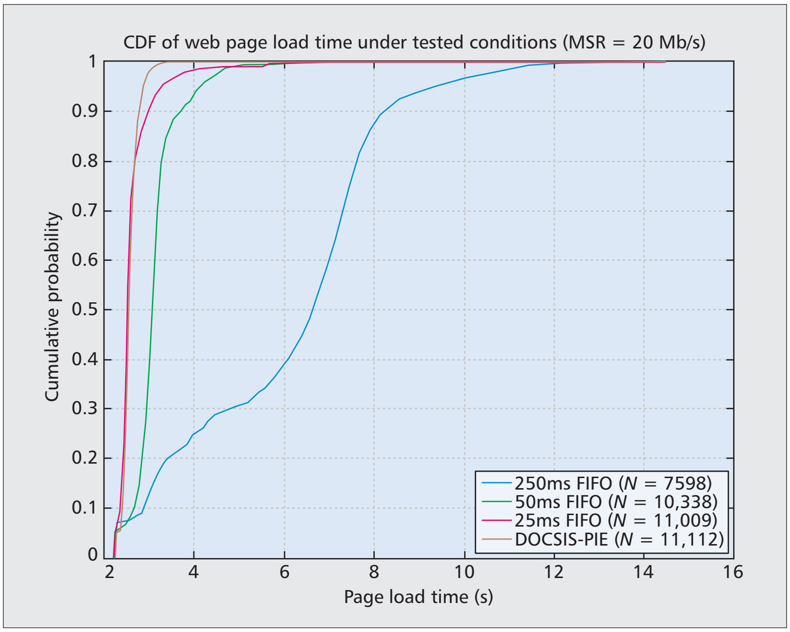

Figure 4. Web page load time statistics.

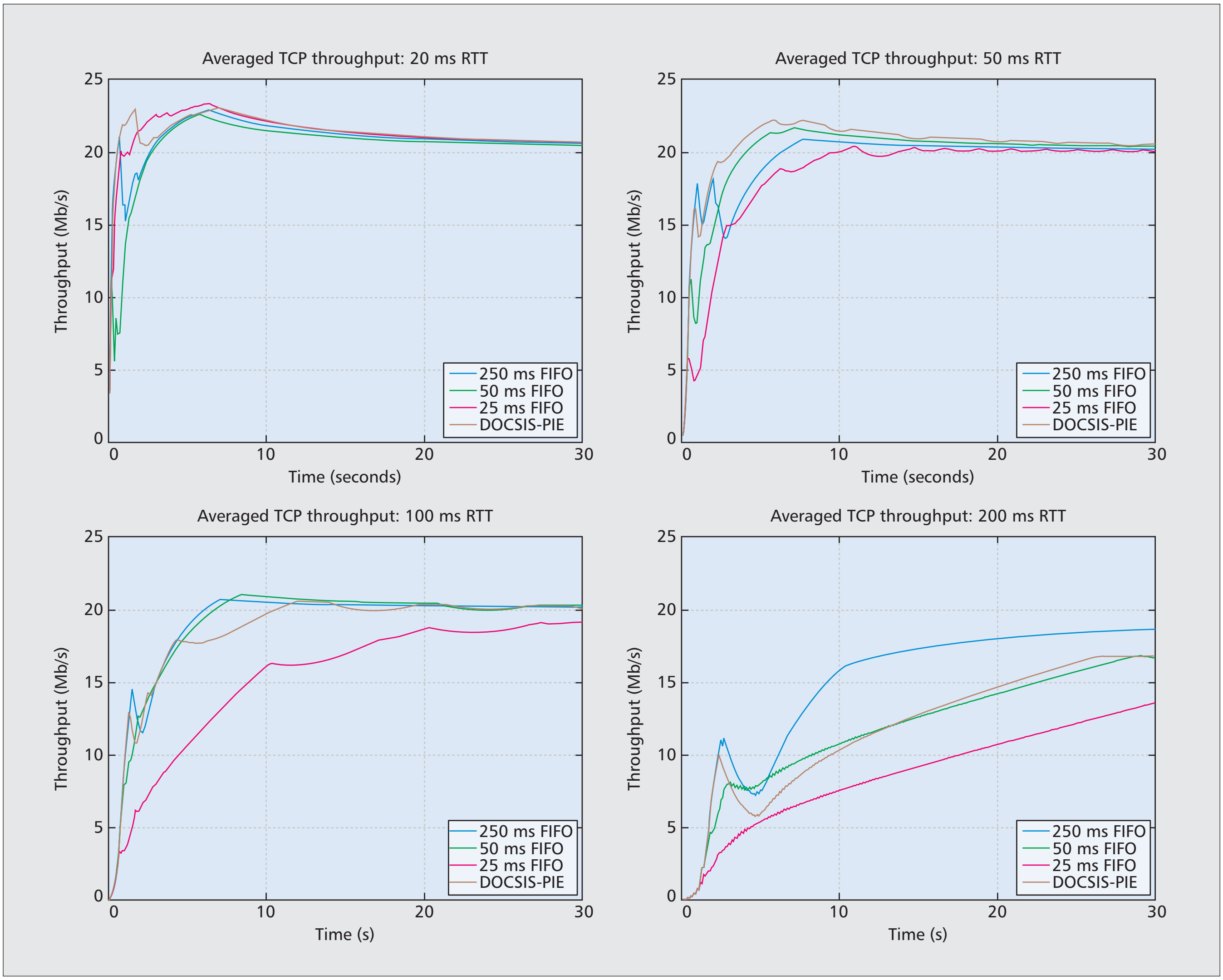

Figure 5. TCP performance

The AQM algorithm implemented on DOCSIS 3.1 cable modems will reduce upstream latency (in loaded conditions) from hundreds or thousands of milliseconds down to tens of milliseconds. In addition, AQM algorithms on the CMTS will do the same for downstream traffic.

Performance Summary

DOCSIS-PIE is shown to provide VoIP quality, gaming latency, and web page load time performance that meet or exceed those provided by 25 ms FIFO, but with approximately 1/3 of the packet loss rate, and without the severe degradation of TCP performance caused by the short FIFO. Moreover, the DOCSIS 3.1 default configuration significantly outperforms the default DOCSIS 3.0 configuration on nearly every measure.

For other protocols and applications beyond those tested, extrapolations from this data can be made. Applications that support loss-based rate control similar to TCP (e.g. TCP-friendly rate control or Stream Control Transmission Protocol) would be expected to have similar average throughput performance with AQM as without.

Conclusion

The DOCSIS 3.1 specification brings an assortment of new technologies and capabilities to cable data networks that will serve to bring major improvements in the performance and reliability of broadband cable Internet service.

By improving the latency performance of broadband connections, the active queue management functionalities in DOCSIS 3.1 equipment will have the potential to substantially improve the user experience for interactive applications.

The AQM algorithm implemented on DOCSIS 3.1 cable modems will reduce upstream latency (in loaded conditions) from hundreds or thousands of milliseconds down to tens of milliseconds. In addition, AQM algorithms on the CMTS will do the same for downstream traffic. The result will be a significant reduction in sluggish web browsing performance, and much enhanced reliability for online games and audio and video telephony applications.

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Biography

Greg White [M'91] (g.white@cablelabs.com) obtained his B.S. degree from Carnegie Mellon University in 1992 and his M.S. degree from the University of Wisconsin — Madison in 1994, both in electrical engineering. From 1995 to 1999, he held various positions at Motorola Land Mobile Products Sector Research and Motorola Labs. In 1999 he joined Cable Television Laboratories, Inc. (CableLabs) and served as the lead architect of the DOCSIS 2.0 and DOCSIS 3.0 specifications. He is presently a Distinguished Technologist at CableLabs, and his current area of interest is in developing advanced technology for improving broadband network performance and user experience.

THE FUTURE OF WI-FI

Edward Au

Minho Cheong

Chiu Ngo

Carlos Cordeiro

Weihua Zhuang

This feature topic is a continuation of our November 2014 feature topic on the Future of Wi-Fi. In this second part of the feature topic, five papers selected from a pool of high-quality submissions are introduced. We hope our readers will find these articles useful for understanding recent developments and for inspiring their own work.

Our feature topic begins with an article entitled "802.11 WLAN: History and New Enabling MIMO Techniques for Next Generation Standards," by Kim and Lee. It examines the IEEE 802.11 standard that defines wireless local area networks (WLAN). A detailed technical overview of the WLAN system and recent developments focusing on multiple-input multiple-output (MIMO) technologies are also presented.

The next article, entitled "Emerging Technologies for WLAN" by Jones and Sampath, reviews a few key features of the published IEEE 802.11ac-2013 amendment, and potential emerging technologies being considered in a few amendments that are under development, including IEEE 802.11ah, IEEE 802.11ai, and IEEE 802.11ax. Further, the authors present channel measurement and prototype performance data to demonstrate the gains of MIMO and MU-MIMO, which are two important features of IEEE 802.11ac, in an indoor environment.

The third article of this feature topic focuses on Wi-Fi based indoor positioning, which has recently attracted increasing attention from both academia and industry. Yang and Shao illustrate in their article "WiFi-Based Indoor Positioning" their proposed approach and compare its performance with a few existing mechanisms in terms of cost, complexity, and system performance.

The last two articles continue to cover another very popular topic from Part I of our feature topic: the interworking of Wi-Fi and cellular systems. The first article is contributed by Zhang *et al.* and is entitled "Coexistence of Wi-Fi and Heterogeneous Small Cell Network Sharing Unlicensed Spectrum." The authors examine the potentials and challenges associated with coexisting Wi-Fi systems and heterogeneous cellular networks sharing the unlicensed spectrum. The authors also present a network architecture that can be applied in small cells to exploit unlicensed spectrum used by Wi-Fi systems. In the second article, "Enhanced Capacity & Coverage by Wi-Fi LTE Integration," Ling *et al.* discuss a few solutions that integrate Wi-Fi with LTE access networks, in order to enable Wi-Fi networks to serve more users with higher throughput demands and as an effective traffic offload solution for cellular operators, all coming with the benefits of using the unlicensed spectrum.

BIOGRAPHIES

EDWARD AU [SM] (edward.ks.au@gmail.com) is a senior staff member at Marvell Technology Group, where he is responsible for product certification and standardization of Wi-Fi. He chairs several technical task groups related to location, power saving, and smart grid technologies in the Wi-Fi Alliance and a study group on next generation 60 GHz in IEEE 802.11. He has a strong research record having published tens of papers and patents. He also serves as editor of various IEEE journals, and served as a Track/Symposium Co-Chair of IEEE conferences. He is the recipient of 2013 Top Editor Award of *IEEE Transactions on Vehicular Technology*.

MINHO CHEONG [SM] is a Managing Director at Newracom, Inc., ETRI's spin-off company that develops solutions for the Korea Wi-Fi ecosystem. Dr. Cheong was a project leader, special fellow, and head of delegates of IEEE 802 at ETRI, and worked on R&D on 4G systems, multi-Gbps nomadic systems, and next generation WLAN. He was the coordinator of Korea's standardization for next generation WLAN, Chair of the VHT Working Group at TTA, and PHY co-chair and functional requirements editor of IEEE 802.11ac and IEEE 802.11ah. His research interests include OFDM, MIMO, and interference cancellation, on which he has filed over 100 patents. He was appointed as "Nation-wide Outstanding Research Group" by the Prime Minister of Korea in 2007. He was the recipient of the Silver Prize in a "Human-Tech. Thesis Contest" in 2004 and the Grand Prize in "DSP Design Contest" in 1997.

CHIU NGO [SM] is head of the Standards and Technology Enabling group, Samsung Electronics Silicon Valley R&D Center. As a senior director, he leads Samsung's U.S. standardization activities in connectivity, content, and service delivery for consumer electronics. He has actively participated in various standardization organizations and holds Samsung's position on several Boards of Directors. He has co-authored more than 40 published papers and holds more than 150 U.S. patents He is a Chartered Engineer of IET.

CARLOS CORDEIRO [SM] is a principal engineer in the Platform Engineering Group within Intel Corporation, USA. He is the overall lead of Intel's standardization programs in Wi-Fi and in the area of short-range multi-Gbps wireless systems using millimeter frequencies. In the Wi-Fi Alliance, Dr. Cordeiro is a member of the Board of Directors and serves as the technical advisor, in addition to chairing the technical task group on 60 GHz. He was the technical editor to the IEEE 802.11ad standard. Due to his contributions to wireless communications, he received several awards including the prestigious Global Telecom Business 40 under 40 in 2012 and 2013, the IEEE Outstanding Engineer Award in 2011, and the IEEE New Face of Engineering Award in 2007. He is the co-author of two textbooks on wireless published in 2006 and 2011, has published about 100 papers in the wireless area alone, and holds over 30 patents. He has served as an editor for various journals.

WEIHUA ZHUANG [F] has been with the University of Waterloo, Canada, since 1993, where she is a professor and a Tier I Canada Research Chair in Wireless Communication Networks. Her current research focuses on resource allocation and QoS provisioning in wireless networks, and on smart grid. She is a co-recipient of several best paper awards from IEEE conferences. Dr. Zhuang was the editor-in-chief of *IEEE Transactions on Vehicular Technology* (2007-2013), and the Technical Program Symposia Chair of the IEEE Globecom 2011. She is a Fellow of the Canadian Academy of Engineering, a Fellow of the Engineering Institute of Canada, and an elected member of the Board of Governors and VP–Mobile Radio of the IEEE Vehicular Technology Society. She was an IEEE Communications Society Distinguished Lecturer (2008–2011).

802.11 WLAN: History and New Enabling MIMO Techniques for Next Generation Standards

Joonsuk Kim and Inkyu Lee

Joonsuk Kim is 802.11ac vice-chair.

Inkyu Lee is with Korea University.

ABSTRACT

IEEE 802.11, which designs wireless local area networks (WLAN), is one of the most successful standards in wireless communication systems. In this article, we review the history of WLAN standards, and provide technical overviews of the recent development of WLAN systems. Especially, as the original inventor and the proposer, we focus on beamforming and compressed feedback schemes, which have been adopted in 802.11 WLAN standards, to improve the throughput for a multiple-input multiple-output (MIMO) system. These techniques are essential to maximize the downlink system throughput for multiple user transmission as well as for single user transmission. Also, we present discussions on new technologies to further enhance user throughput, which are currently considered for future WLAN systems.

INTRODUCTION

Wireless local area networks (WLANs) have been widely deployed in communication networks for decades. As portable devices become prevalent, a demand to access internet without any constraint on location has been growing. This wireless connectivity has become more important features, especially for smartphones and tablets, as internet traffic increases. While we have witnessed the explosive growth of wireless communications recently, the history of WLANs is not that long.

IEEE created a project called 802 in 1980 and many working groups have been made under the project. For WLAN, the first specification was published under 802.11 working group in 1997. Initially, the specification defined only the data rate of 1 or 2 Mb/s operated at 2.4 GHz band using either frequency hopping spread spectrum (FHSS), infrared (IR) or direct sequence spread spectrum (DSSS). Later, they decided to expand the standard under two task groups, at either 2.4 GHz band and 5 GHz band, which became 802.11b and 802.11a, respectively. The complementary code keying (CCK) mode was introduced in 802.11b to support up to 11 Mb/s, while orthogonal frequency division multiplexing (OFDM) was applied for 802.11a which supports up to 54 Mb/s. Both amendments were published in 1999.

While it was expected to use different systems depending on applications with lower or higher data rates, the WLAN market based on the 5 GHz band was not fully utilized due to its high cost of radio frequency (RF) implementation in early 2000. On the other hand, there was a demand for higher data rates than 11b systems at the 2.4 GHz band. In 2002, in order to support the data rate up to 54 Mb/s in the 2.4 GHz band, 802.11g task group (TGg) decided to adopt the same physical layer (PHY) and media access control (MAC) specification of 802.11a [1].

With growth of the usage of internet browsing and multimedia contents, a demand on higher data rates has never stopped growing in WLAN. As a result, IEEE 802.11 committee approved to create another task group, 802.11n, to work on a new amendment for high throughput (HT). The goal of 802.11n was to achieve the throughput of 100 Mb/s at the MAC layer. In order to realize such a throughput enhancement, the 802.11n employed multiple-input multiple-output (MIMO) techniques including spatial-division multiplexing (SDM) [2] with up to 4 streams, space-time block coding (STBC) and transmit beamforming (TxBF).

Utilizing multiple antennas at the transmitter and the receiver, SDM allows multiplexing of multiple data streams across spatial dimensions, and the throughput increases as much as the number of data streams [3]. In addition, TxBF improves the received signal strength by emphasizing the dominant modes of transmission for the channel. However, the consensus on the channel state information (CSI) feedback for TxBF has not been made, which ended up with four different feedback mechanisms, consisting of one implicit feedback method and three explicit feedback schemes, as all optional modes.

After the completion of 802.11n in 2009 [4], 802.11ac added more evolutionary changes on MIMO methods. This new amendment, initiated by very high throughput (VHT) study group,

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adopted more advanced MIMO techniques such as downlink multi-user (MU)-MIMO and allowed up to 160 MHz bandwidth and up to 8 streams. These techniques enable to achieve the network throughput higher than 1 Gb/s. However, the CSI feedback overhead becomes more challenging because the performance of MU-MIMO is sensitive to CSI mismatch over fading channels. Also, implicit feedback turned out to be infeasible due to its frequent calibration operation. These issues were resolved by choosing the compressed explicit feedback [5] as a sole mandatory mode. Finally, debates on the CSI feedback for TxBF over five years came to an end.

In this article, we provide a technical overview of the recent development of WLAN technologies. Also, we illustrate the future direction of the WLAN research, focusing on multiple antenna schemes currently being discussed for the next generation standards. This article starts with a basic description of the preamble of 802.11 frames, which handles control signals from a PHY perspective. Especially, we address the control signals related to MIMO features, including beamforming and MU-MIMO. We include more details on TxBF and its feedback mechanism. Also, we introduce the future WLAN techniques which further improve the performance of MIMO systems. We conclude this article with discussions.

PHY Preamble

Legacy Portion for Basic Information

The packet starts with the PHY preamble that consists of short training field (STF) followed by long training field (LTF) and a signal field (SIG). This general preamble structure is common for all 802.11 specification with variations under very important requirement: backward compatibility. If this requirement is not met, a legacy station (STA) may often get lost to access the media by non-understandable packets without any knowledge when such medium-busy status would last.

For 802.11n, this requirement led to define a mixed-mode format that starts with a legacy preamble, as 802.11a/g, followed by new preamble portion (also known as HT portion). Similarly, 802.11ac also chooses a mixed-mode format with new portion (also known as VHT portion). Both amendments enable an efficient preamble design for MIMO operation. Figure 1 illustrates the time samples of the preamble with 2 transmit antennas in 802.11ac.

The preamble starts with legacy short training field (L-STF), which consists of 10 repeated time samples. It is typically employed for detection of incoming packets, timing synchronization, gain control setting and coarse frequency offset estimation.

After the L-STF, legacy long training field (L-LTF) is located, which triggers all the tones available in operating bandwidth for channel estimation. The L-LTF is also used for fine frequency offset estimation.

After the L-STF and the L-LTF, legacy signal field (L-SIG) follows, which contains the modulation and coding set (MCS) and its length information. For 802.11n and 11ac, these information is overwritten in the V/HT-SIG, because the L-SIG does not support higher MCS and longer length for 11n/ac. The MCS setting in the L-SIG for 11n/ac packets is only to calculate the packet duration with the length field. Thus, even though legacy devices do not understand the V/HT portion, they can at least collect information on the duration of the packet.

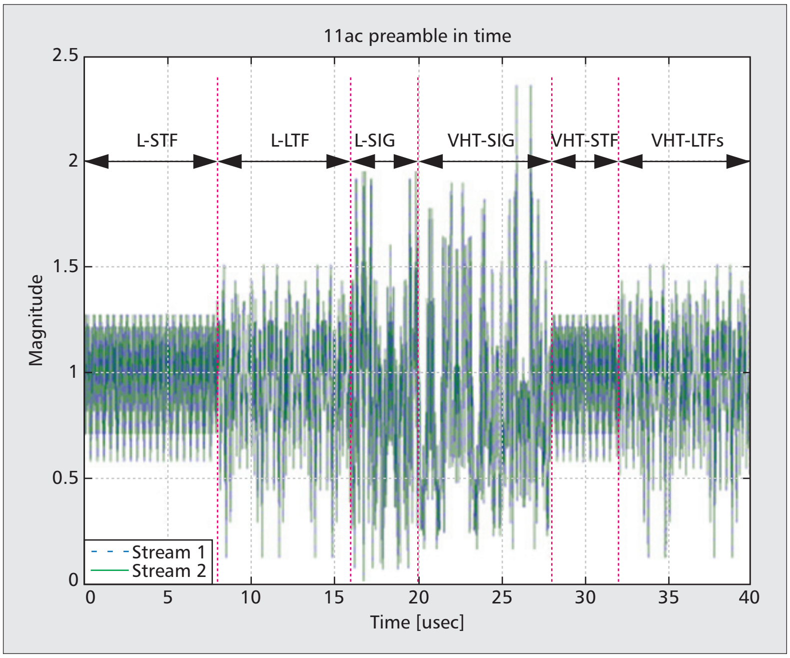

Figure 1. Time samples of the preamble for an 802.11ac system with two transmit antennas.

VHT portion for MIMO information

After 20 μsec long legacy portion of the preamble, more control signals are defined in high throughput signal field (HT-SIG) for 11n and very high throughput signal field (VHT-SIG) for 11ac. Such control signals are MCS (up to 64 QAM with 5/6 coding rate for 11n (MCS7), or up to 256 QAM with 3/4 coding rate for 11ac (MCS9)), coding schemes (convolutional or low-density parity check (LDPC)), STBC mode, short guard interval (SGI) mode, bandwidth (up to 40 MHz for 11n, or up to 160 MHz for 11ac), and so on. The V/HT-SIG is rotated 90 degree with different combination over two OFDM symbols, which enables receivers to identify whether the preamble format is non-legacy mode, 11n or 11ac.

After the V/HT-SIG, a couple of training fields exist to set up the MIMO operation, because it is the point where beamforming may be applied. As this spatial processing changes the beam pattern for certain directions, the receiver may see sudden changes in the dynamic range of the signal strength, which may end up with operating in the non-linear region of an RF amplifier. In order to adjust the gain control to find the sweet spot of the RF scaling for MIMO, there is one OFDM symbol long V/HT-STF.

The V/HT-LTFs are training sequences to estimate MIMO channels. To preserve orthog-

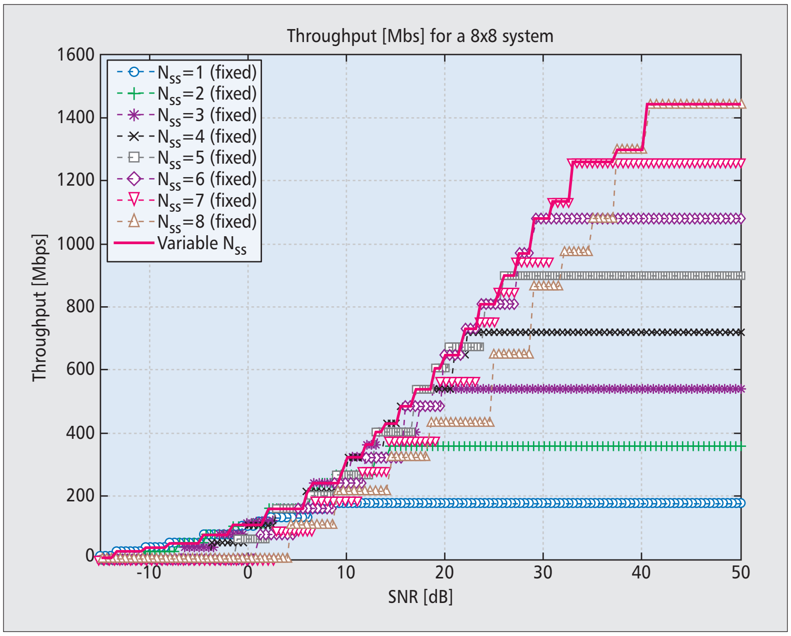

Figure 2. Throughput of 802.11ac systems with up to 8 streams with long guard interval of 0.8 μsec.

onality between antennas, the V/HT-LTF is designed as the original legacy training sequence multiplied by the Hadamard matrix. Unfortunately, the Hadamard matrix is available only for an exponent of two, such as 2, 4, or 8. For six transmit antennas in 11ac, a new orthogonal 6 × 6 matrix based on fast Fourier transform (FFT) is introduced. Since the length of V/HT-LTF in the unit of OFDM symbols should be no less than the number of transmit antennas, for the odd number of transmit antennas, we apply the Hadamard matrix with one dimension higher than the number of transmit antennas. Figure 2 shows an example of throughput performance using channel estimation with up to 8 streams. In this plot, each dashed line represents the throughput with a fixed number of streams.

For 802.11ac, VHT-SIG fields are split into two fields, VHT-SIGA and VHT-SIGB [6]. The VHT-SIGA, shown as in Fig. 3, is located at the same location as HT-SIG in 11n, and the VHT-SIGB is placed after the VHT-LTF field. While the VHT-SIGA has common information for all users, the VHT-SIGB includes user-specific information, such as MCS and the packet length per user, for MU-MIMO. The VHT-SIGB may not be correctly decoded for non-recipients, because this field is beamformed for MU-MIMO. More detailed descriptions regarding MU-MIMO operations follow later.

Contents of Signal Field for MU-MIMO Operations

With a new MU-MIMO feature introduced in 802.11ac, the VHT-SIG should indicate necessary information for recipients to process MU-MIMO packets. Especially, it should identify which stream belongs to which STA, for simultaneous transmission to up to 4 STAs. There have been a couple of proposed methods, however, most of them fail to meet the requirement of giving no impact on the packet detection.

Finally, the Group ID was proposed [7] and accepted unanimously by TGac. The Group ID, which is pre-assigned by the Group ID management frame [8], notifies the order of users over multiple streams. For example, let us assume Group ID = 1 in the VHT-SIGA, where 'Group ID = 1' was announced in advance by the management frame for a group of 4 STAs, [STA-A, STA-B, STA-C and STA-D] in the order. Then, the number of streams for the ith STA, N_{sts_i} in the VHT-SIGA indicates which streams belong to whom, where the STA number 1 to 4 corresponds to A to D. For example, if $N_{sts1} = 2$, $N_{sts2} = 1$, $N_{sts3} = 0$ and $N_{sts4} = 1$, STA-A, B and D would decode the first two, the 3rd and the 4th streams only, respectively. In this example, STA-C has no stream to decode, and thus drops the packet without further process.

One of the benefits of Group ID is to allow recipients to identify the destination of the packet earlier in the PHY preamble, which avoids processing the payload all the way to decode the MAC address in the MAC header. This enables power savings for WLAN STAs, which do not have to decode the payload to find whether the packet is intended for itself or not. However, for single user (SU) transmission, which is signaled by Group ID = 0, such mechanism is missing. In order to have the same benefits for the SU operation, partial associate identification (PAID) was also introduced [9]. By receiving the PAID which represents partial information (9 bits) out of 12 AID bits, recipients can recognize the destination of the packet similarly, and drop the packet earlier if the PAID does not match with the parts of its own AID. The first 3 bits of $N_S$ts field remain valid to signal the number of streams for a SU receiver, as shown in Fig. 3.

Also, the SU coding bit in the VHT-SIGA2 indicates either LDPC or convolutional coding for SU mode. For MU mode, this bit determines the coding scheme for user 1, while other bits in MU-coding-2 to MU-coding-4 identify a coding scheme for user 2 to user 4. In addition, the beamformed bit for SU mode is set to recommend recipients not to use a smoothing algorithm for channel estimation, because the channels may be discontinuous over tones due to beamforming. For MU mode, this bit is reserved since MU packets are always beamformed.

Beamforming

Beamforming is one of advanced MIMO techniques to enhance the throughput significantly. Antenna coordination for directional beams is enabled with an aid of CSI feedback, and thus it is essential to efficiently deliver such information from a beamformee to a beamformer. Out of 4 CSI feedback schemes for 802.11n, the compressed explicit feedback was acknowledged with its benefits [5] and finally accepted as a sole feedback format for 802.11ac, eliminating the other feedback schemes [10]. In this section, the details of the compressed explicit feedback scheme [11] is explained.

Interpretation of a Beamforming Matrix

With TxBF, the transmit signals are steered to enhance the quality of streams in terms of channel characteristics and crosstalk avoidance by applying a transmit beamforming matrix **V**. The beamforming matrix **V** can be designed in many different ways, but one popular way is to apply a singular value decomposition (SVD) technique. For $M_R \times M_T$ MIMO systems with M_T transmit antennas and M_R receive antennas, the $M_S \times 1$ transmitted signal can be beamformed by adopting a beamforming matrix **V** of size $M_T \times M_S$ which transmits M_S streams.

In order for a beamformer to properly implement beamforming, prompt and accurate CSI feedback is important. However, minimizing the feedback overhead size is often an issue while maintaining good quality of feedback information. To reduce the total number of bits in the feedback channel, we may utilize the orthonormal property of the beamforming matrix **V** to represent in angle domain.

For instance, with a 2 × 2 MIMO system with $M_S = 2$, **V** can be expressed as

$$V = \begin{bmatrix} \cos\psi_1 e^{j\phi_1} & \cos\psi_2 e^{j\phi_2} \\ \sin\psi_1 & \sin\psi_2 \end{bmatrix} = \begin{bmatrix} e^{j\phi_1} & 0 \\ 0 & 1 \end{bmatrix} \begin{bmatrix} \cos\psi_1 & -\sin\psi_1 \\ \sin\psi_1 & \cos\psi_1 \end{bmatrix} \qquad (1)$$

where ψ_1, ψ_2, ϕ_1 and ϕ_2 are the phase parameters, and the last row of **V** is chosen to be positive real as a reference phase using the column-wise phase invariant property. In order to have $\mathbf{V}^*\mathbf{V} = \mathbf{V}\mathbf{V}^* = \mathbf{I}$, where $(\cdot)^*$ denotes complex conjugate transpose, these phase values need to satisfy either $\psi_1 - \psi_2 = \pi/2$, $\phi_1 = \phi_2$ or $\psi_1 + \psi_2 = \pi/2$, $\phi_1 = \phi_2 + \pi$, where one of two conditions for angles would be sufficient. Thus we can choose one by limiting the range of angles to be $\psi \in [0, \pi/2]$ and $\phi \in [-\pi, \pi]$ such that $\psi_1 + \psi_2 = \pi/2$, $\phi_1 = \phi_2 + \pi$. These conditions lead us to decompose **V** as a multiplication of two matrices as shown in Eq. 1, with one diagonal matrix for the phase and the other with sinusoid functions for rotation.

Generalized Decomposition for V

The idea of representing **V** in the angle domain can be generalized for higher dimension of the matrix [11]. From Eq. 1, **V** can be expressed as a special form of Givens rotation with a complex diagonal matrix. Let us define $\mathbf{I}_N$ as an identity matrix of size N, and $[\mathbf{A}]_{ii}$ as the (i, i)th element of a matrix **A**. Then, an arbitrary $M_T \times M_S$ unitary matrix **V** can be decomposed as

$$V = \prod_{i=1}^{M_{\min}} \left[D_i \prod_{l=i+1}^{M_T} G^*_{l,i}(\psi_{l,i}) \right] \times \tilde{I}, \qquad (2)$$

where $M_{\min} = \min(M_S, M_T - 1)$, $\mathbf{D}_i = \mathrm{diag}\,(\mathbf{I}_{i-1}, e^{j\phi_{i,i}}, \cdots, e^{j\phi_{M_T-1,i}}, 1)$ is an $M_T \times M_T$ diagonal matrix, $\tilde{\mathbf{I}}$ denotes an $M_T \times M_S$ diagonal matrix with $[\mathbf{I}]_{i,i} = 1$ for $i = 1,\cdot s, M_S$, and $\mathbf{G}_{l,i}(\psi_{l,i})$ indicates an $M_T \times M_T$ Givens rotation matrix with rotation angle $\psi_{l,i}$ as

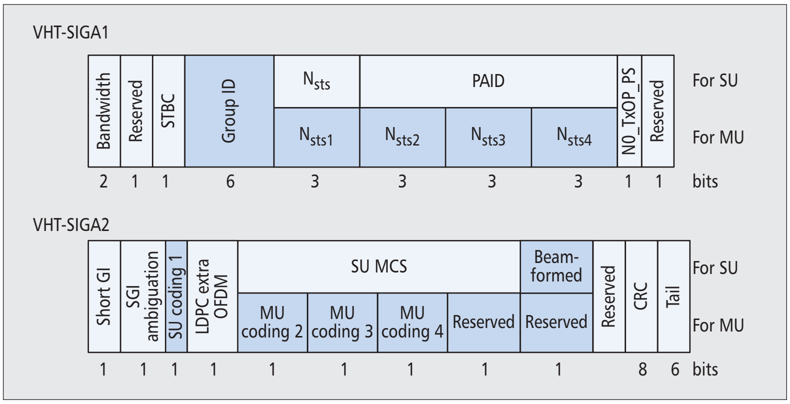

Figure 3. Bit definition in the VHT-SIGA fields for single-user and multi-user operations.

$$G_{l,i}(\psi_{l,i}) = \begin{bmatrix} I_{i-1} & 0 & 0 & \cdots & 0 \\ 0 & \cos\psi_{l,i} & 0 & \sin\psi_{l,i} & 0 \\ 0 & 0 & I_{l-i-1} & 0 & 0 \\ 0 & -\sin\psi_{l,i} & 0 & \cos\psi_{l,i} & 0 \\ 0 & 0 & 0 & \cdots & I_{M_T-l} \end{bmatrix}$$

with non-identity elements located at the lth and ith rows and columns. Note $\phi_{*,i}$ in $\mathbf{D}_i$ are found to make elements of the ith column real, before Givens rotation by $\mathbf{G}_{l,i}$ is performed .

When a beamformee finds an arbitrary beamforming matrix $\tilde{\mathbf{V}}$, which is column-wise phase-invariant, each column of $\tilde{\mathbf{V}}$ can be phase shifted by a diagonal matrix $\tilde{\mathbf{D}} = diag(e^{j\theta_1}, \cdots, e^{j\theta_{M_T}})$, where θ_i can be arbitrarily chosen to make the last row of $\mathbf{V} = \tilde{\mathbf{V}} \times \tilde{\mathbf{D}}$ real. Therefore, the minimum set of parameters to determine the beamforming matrix **V** is ψ_{li} for $i = 1, 2,\cdots, M_{\min}$ and $l = i + 1,\cdots, M_T$, and ϕ_{ji} for $i = 1, 2,\cdots, M_{\min}$ and $j = i,\cdots, M_T - 1$, where θ_i are not required to be fed back for beamforming.

Quantization and Grouping for Adjacent Tones

When **V** is decomposed in sequence based on Eq. 2, it may rely on the previously quantized values of angles in the steps of Givens rotation for each column. Technically, these quantization errors can be accumulated with index i in Eq. 2. However crosstalk between streams is still minimized. On the other hand, if the quantization is performed after all decomposition process is done, the quantized angles found in each step do not rotate the matrix toward nullifying off-diagonal terms in each decomposed matrix.

Therefore, how to quantize the angles needs to be carefully approached. Also, angle resolution for ϕ and ψ should be properly chosen to minimize the quantization error. It was shown in [11] that the choice of $\phi_b = \psi_b + 2$ is optimal, where ϕ_b and ψ_b are the number of bits for ϕ and ψ, respectively. Considering different dynamic range as defined earlier, this implies that the same angle resolution for ϕ and ψ is required to minimize the quantization noise.

In addition, by utilizing the correlation property of the channel between tones, we can further reduce the network overhead of the feedback packet. Considering the coherent bandwidth of the channel and the tone width of an OFDM packet, we may choose one feedback value for a group of tones [12], where a beamformer may interpolate angles to recover those missing tones. For the channel with smaller delay spread, which has larger coherent bandwidth, we may be able to reduce the overhead by grouping more tones.

Performance of Beamforming

Here, we present the performance of the beamforming techniques based on compressed explicit feedback as illustrated in the previous subsections. We use channel D and channel E [13], where they have root mean square (rms) delay spread of 50 nsec and 100 nsec, respectively. Table 1 summarizes the total number of bytes required to deliver the beamforming matrix to reach within a 0.5 dB performance penalty (or 1 dB in *italic font*) with bandwidth of 20 MHz (or 40 MHz) due to a quantization loss for the worst case of channel E for grouping. For 2 × 2 systems operating with BPSK and 1/2 rate-coding,

$M_T \times M_S$	2 × 2	3 × 3	4 × 2	4 × 4
Cartesian	260 (540)	468 (972)	416 (864)	832 (1728)
Givens	52 (108)	156 (324)	195 (406)	312 (648)
Group – 2	26 (54)	78 (162)	98 (203)	156 (324)
Group – 4	13 (27)	39 (81)	49 (102)	78 (162)

Table 1. The number of information bytes required to deliver the beamforming matrix V for 20 MHz (or 40 MHz) bandwidth with channel E.

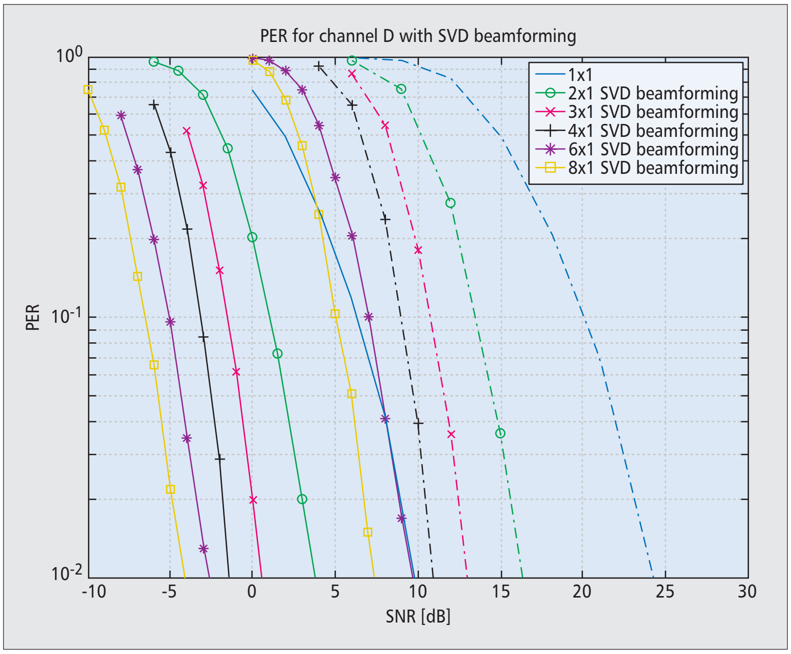

Figure 4. PERs for 802.11ac systems with MT transmit antennas and a single receive antenna using SVD beamforming.

the compressed explicit feedback with 4-tone grouping can reduce the network overhead to only 5 percent, compared to a conventional Cartesian method without a significant performance degradation.

Figure 4 illustrates a packet error rate (PER) performance gain with SVD beamforming for channel D, using compressed explicit feedback with grouping of 4 tones. In this plot, solid lines and dashed lines represent PER performances with MCS0 (BPSK with 1/2-rate) and MCS4 (16QAM with 1/2-rate), respectively. Regardless of MCS choice, beamforming with compressed explicit feedback shows significant performance gain.

Extension to Multiple Users

Beamforming can also be applied to transmit multiple streams simultaneously to multiple users, by nullifying interference between users. In MU-MIMO as shown in Fig. 5, where a beamformer collects CSI from users based on the compressed explicit feedback, a group of spatially separated users can be chosen, which is signaled with Group ID described earlier, and then a steering matrix is recalculated. Typically, the steering matrix found by the beamformer may be different from the beamforming matrix **V** fed back from a user, since it is designed to minimize interference to other users. The bit resolution for **V** in MU-MIMO is also selected higher than SU feedback, as the performance of the steering matrix is more sensitive to the channel mismatch.

This steering matrix is applied starting from the VHT-STF, where the designated user may receive its own streams with relatively much higher strength than other streams for other users. In order to suppress the interference at the receiver, the VHT-LTF contains training sequences for other streams as well. The number of the VHT-LTFs is the total number of streams for all participating users. For MU operations, MCS per user is included in the VHT-SIGB, which is user-specific information, however not decodable by other users because it is nullified by the steering matrix.

Future Techniques

In this section, we describe some promising techniques which allow to further enhance the spectrum efficiency of WLAN systems. These advanced techniques include, however are not limited to, sub-band transmission with group of tones based on orthogonal multiplexing division multiple access (OFDMA), uplink MU-MIMO and full duplex transmission.

Sub-band OFDMA

For downlink transmission, it is difficult to get a real benefit of the increased bandwidth due to commonly used secondary channels in neighboring basic service sets (BSSs) and restrictions on non-continuous channel assignments. We can employ the OFDMA technique, however with a unit of group of tones to reduce the implementation complexity, to maximize resource utilization in wider bandwidth and increase multiplexing flexibility. The sub-band OFDMA can assign subsets of subcarriers to individual users as a multi-user version of the popular OFDM scheme. Compared to the legacy operation which cannot use remaining secondary channels when interference is detected in the secondary channel, the sub-band OFDMA enables to access concurrent channels from multiple users and also utilizes non-continuous channels within a given FFT size.

The sub-band OFDMA technique can be combined with MIMO schemes [2] and then it leads to higher data rate and better reliability by utilizing SDM and spatial diversity methods in the MIMO systems. In addition, employing the resource allocation strategies including proper scheduling of sub-channels to users and transmit power control for each sub-band, the MIMO-OFDMA systems can improve channel utilization efficiency. Such systems exploit advantages of both techniques, providing simultaneously flexibility in resource allocation and increased system performance [16].

Uplink MU-MIMO

Currently, MU-MIMO technique is supported only for downlink to achieve higher aggregated throughput from an access point (AP) to STAs. However, for future WLAN services, it is expected to have more balanced bi-directional traffic between the AP and STAs. In order to achieve uplink efficiency enhancements, MU-MIMO transmission, as shown in Fig. 5, can be applied to uplink as well as downlink. As the counterpart of the downlink MU-MIMO, users simultaneously transmit their data to an AP. This allows us to reduce the number of contentions and the overhead for the back-off time and polling.

To implement the uplink MU-MIMO, the challenges are to determine how to synchronize the timing offset between users and how to compensate frequency offsets which are different between users by nature. Specifically, an AP receives uplink data from users with different arrival timing. However, if the contention period length is sufficiently greater than the sum of channel delay and time difference, then system performance is not deteriorated by uplink MU transmission. In [14], the timing offset problem was solved by expanding the guard interval. Also, the frequency offset requirements in 11ac specification need to be below 30.7 percent and 64 percent of the subcarrier spacing at 2.4 GHz and 5 GHz, respectively. Employing compensation schemes such as guard subcarriers among uplink data streams and frequency offset adjustments, the frequency offset problem can also be alleviated. Then, the implementation complexity of the uplink MU-MIMO may become reasonable for both the AP and users, and thus the uplink MU-MIMO technique may be feasible to improve the uplink spectrum efficiency.

Full Duplex

Existing 802.11 technologies only adopt separate transmission and reception at different time, which is called half duplex transmission. On the other hand, full duplex transmission allows simultaneous transmission and reception over the same time and frequency, which could significantly improve the spectrum efficiency up to two times higher. In this case, self-interference cancellation is critical to realization of the full duplex transmission.

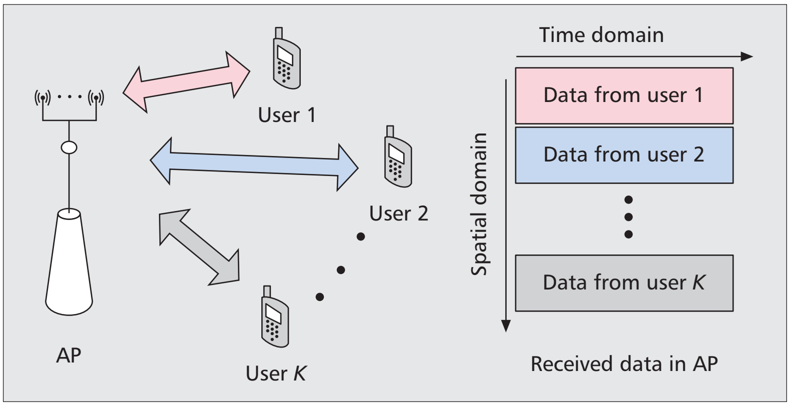

Figure 5. Schematic diagram of multi-user MIMO transmissions.

In order to solve this problem, there are three main cancellation techniques:

1 Antenna level cancellation: the antenna layout minimizes leakage from a transmit antenna into a receive antenna,
2 Analog cancellation: a noise cancellation circuit is applied to adjust the amplitude and phase of the interference reference signal to match the self-interference,
3 Digital cancellation: the scaled and rotated version of the ideal transmit signal is subtracted from the received signal.

By exploiting these techniques, the self-interference can be made to a marginal level \citeDuplex, which helps to improve the spectrum efficiency of future WLANs.

Other Future Considerations

In addition, there are still some other promising techniques that can improve the performance of future WLAN systems. These include cooperative beamforming, 3D beamforming, and relay techniques. Since many BSSs are often deployed by overlapping with each other, conventional carrier sense multiple access (CSMA) may incur performance degradations due to numerous neighboring interference signals. In this case, we can utilize the cooperative beamforming which eliminates the interference to adjacent APs by coordinating cooperative transmissions among APs. Also, the 3D beamforming which considers the three dimensional channel model enables to adjust the radiation beam pattern in both elevation and azimuth planes, and this allows us to increase the link capacity and decrease inter-node interference. Finally, relay methods which forward the signal received from the source to the destination can be adopted to improve link reliability and extend coverage.

Conclusions

In this article, we have examined the IEEE 802.11 standard which defines WLAN. First, we have reviewed the generations of WLAN, starting from 802.11a and 11b to 802.11ac which is

Existing 802.11 technologies only adopt separate transmission and reception at different time, which is called half duplex transmission. On the other hand, full duplex transmission allows simultaneous transmission and reception over the same time and frequency, which could significantly improve the spectrum efficiency up to two times higher.

being published. Then, we have provided technical overviews of the WLAN systems including the PHY preamble structure. Also we have presented the recent developments focusing on MIMO techniques and the compressed explicit feedback scheme which were adopted in 802.11n and 11ac. Finally, we have added future MIMO methods which are possibly considered for next generation WLAN.

Acknowledgements

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Biographies

Joonsuk Kim [S'96, M'01, SM'08] (joonsuk@ieee.org) received the B.S. (summa cum laude) from Seoul National University, and M.S. and Ph.D. from Stanford University, in electric engineering. He joined Broadcom in 2000 and contributed on 802.11a/b/g/n/ac chip development for Broadcom to be the number one leading company in the worldwide WLAN market, as a Sr. Principal Systems Scientist. He served as a chair of Beamforming and Link Adaptation ad-hoc group of IEEE 802.11n, and a vice-chair of IEEE 802.11ac task group. He received IEEE Standards TGn Leadership Awards in 2010, and TGac Leadership Awards in 2014. He has authored more than 100 technical papers/contributions to IEEE and more than 70 U.S. patents granted and issued.

Inkyu Lee [S'92, M'95, SM'01] (inkyu@korea.ac.kr) received the B.S. degree in control and instrumentation engineering from Seoul National University, South Korea, in 1990, and the M.S. and Ph.D. degrees in electrical engineering from Stanford University, California, in 1992 and 1995, respectively. He is currently a department head of the School of Electrical Engineering, Korea University, Seoul, South Korea. He has published over 120 journal papers with the IEEE, and has 30 U.S. patents granted or pending. His research interests include digital communications and signal processing techniques applied for next-generation wireless systems. He has served as an associate editor of IEEE Transactions on Communications and IEEE Transactions on Wireless Communications. He was a chief guest editor of IEEE Journal on Selected Areas in Communications (Special Issue on 4G Wireless Systems) in 2006. He received numerous awards including the Best Young Engineer Award of the National Academy of Engineering of Korea (NAEK) in 2013. He is a member of NAEK.

Emerging Technologies for WLAN

VK Jones and Hemanth Sampath

ABSTRACT

New technologies continue to be introduced for WLAN applications at a robust pace. We review the value proposition for some of the key features of 802.11ac such as larger bandwidth, higher order modulation, and MIMO and MU-MIMO transmission modes, explaining how each feature translates to improved user experience. We present channel measurements and prototype performance data to demonstrate the gains of MIMO and MU-MIMO in an indoor environment. Next, we discuss the value proposition of some key features of 802.11ah. Measurement data of 802.11ah performance is provided, showing how it is a compelling technology for the growing Internet of Things market. We conclude with a preview of emerging technologies that promise to improve user experience — 802.11ai for fast network acquisition and 802.11ax for high-efficiency networking in dense indoor and outdoor networks.

INTRODUCTION

The WLAN market is experiencing unprecedented growth. Almost every home and office in advanced technology markets utilizes a WLAN, and deployments are rapidly proliferating in public areas of congregation like cafés, hotels, and airports. Wireless operators are embracing WLAN for cellular offload as attach rates in smartphones have reached 100 percent. In addition, there is rapid proliferation of WLAN enabled devices across many consumer electronic categories, as consumers demand that their entertainment devices be "connected."

New applications for WLAN continue to emerge. Wireless audio and video streaming are becoming increasingly popular with consumers. Service providers desire WLAN to support multiple high definition (HD) streams in a home, moving content to and from devices like tablets, set-top boxes, digital media adapters, media centers, or televisions. Phone and tablet manufacturers want devices to interact wirelessly to local services or peripherals such as video cameras. In addition, there are emerging use cases like Miracast [1] that provide means for "mirroring" a small screen to a larger video screen.

In order to pave the way for new device categories and new application use cases, there are exciting new technologies emerging for WLAN that will address the need for increased network capacity, longer range, lower power consumption, and ease of use.

The authors are with Qualcomm.

KEY TECHNICAL FEATURES OF 802.11AC

MANDATED USE OF THE 5GHZ BAND

The WLAN market is now transitioning from IEEE 802.11n to 802.11ac [2], due to the promise of higher throughput and more reliable performance available in the 5 GHz unlicensed band. 802.11ac mandates the use of the 5 GHz band, a band with more significant spectrum availability compared to the commonly used 2.4 GHz band. While 802.11a mandated use of the 5 GHz band, the attractiveness of 802.11a was limited due to the fact that 802.11g offered similar data rates. 5 GHz was optional for 802.11n; thus, lower-cost 2.4-GHz-only solutions have been more popular than dual-band devices. In contrast, the higher bandwidth modes of 802.11ac cannot be used in the 2.4 GHz band, so the transition from single-band 802.11n to dual-band 802.11ac is being driven by a desire for higher data rates.

In the United States today, there is approximately 500 MHz of available unlicensed spectrum between 5.15 and 5.725 GHz. In Europe and Japan, there is approximately 400 MHz of spectrum available. In India and China, there is 260 and 100 MHz of spectrum available. In contrast, the 2.4 GHz band can accommodate only three non-overlapping 20-MHz-wide channels, which has led to many competing devices per channel and heavy levels of interference. The larger spectrum availability in the 5 GHz band provides for more network capacity, and leads to fewer competing devices per channel and thus reduced interference compared to traditional 802.11g and single-band 802.11n networks.

MAXIMUM DATA RATE

One key feature of 802.11ac is the increase in data rate compared to 802.11n. This is achieved through the use of expanded channel bandwidth and higher-order modulation. Figure 1a shows the peak data rate and the per-spatial stream (SS) data rate for various WLAN standards that have evolved over the years. 802.11b and 802.11g support peak data rates of 11 and 54 Mb/s, respectively. 802.11n increased the peak data rate to 600 Mb/s. 802.11ac has further increased the peak data rate to 6.9 Gb/s, over ten times that of 802.11n.

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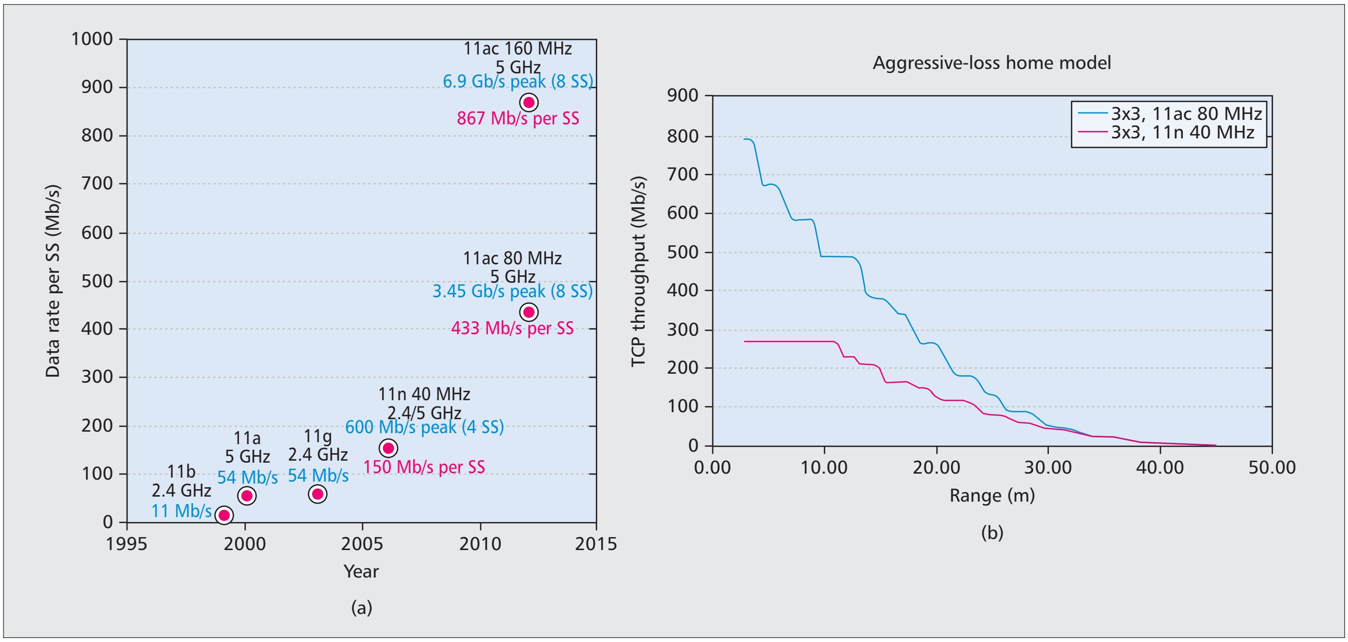

Figure 1. a) Peak data rate and the per-spatial stream (SS) data rate of WLAN technologies; b) rate-over-range of three spatial stream 802.11ac and 802.11n devices.

Wider Bandwidth and Higher Order Modulation

802.11ac introduces three new expanded channel bandwidth modes: 80, 160, and 80+80 MHz, while also defining support for 20 and 40 MHz bandwidth modes to match channel bandwidth modes of 802.11n and 802.11g. The 80 and 160 MHz transmissions use contiguous spectrum, while the 80+80 MHz mode allows the construction of the transmitted signal to occupy separate 80 MHz segments. The 80 MHz transmissions use 234 data tones per orthogonal frequency-division multiplexing (OFDM) symbol, which more than doubles the data rate compared to an 802.11n 40 MHz transmission that only uses 108 data tones. The 160 and 80+80 MHz transmissions use exactly twice the number of data tones as the 80 MHz transmission, thereby doubling the data rate further.

802.11ac introduces the use of 256-quadrature amplitude modulation (QAM) [2]. Two forward error correction (FEC) coding rates are defined: 3/4-rate and 5/6-rate. As a comparison, 802.11n supports up to 64-QAM with these same coding rates. Thus, 802.11ac achieves a 33 percent increase in peak data rate over 802.11n.

The combination of higher-order modulation and increased channel bandwidth enables an 802.11ac device to support approximately three to six times higher data rate compared to an 802.11n device for the same number of antennas or spatial streams (SSs). 802.11n achieves a maximum of 150 Mb/s per SS (108 data tones in 40 MHz of bandwidth with a maximum of 5.0 bits per tone). This results in a maximum data rate of 600 Mb/s, assuming the maximum supported 4 SS multiple-input multiple-output (MIMO) transmission of 802.11n. 802.11ac reaches 433 Mb/s per SS using 80 MHz channel bandwidth (234 data tones with 6.67 bits per tone), and 867 Mb/s per spatial-stream using 160 or 80+80 MHz bandwidth. This results in a maximum data rate of 6.9 Gb/s, assuming the maximum supported 8 SS MIMO transmission of 802.11ac.

While 6.9 Gb/s is an eye-catching maximum data rate, a more commonly cited advantage of 802.11ac is the ability to cross the 1 Gb/s barrier with small form-factor devices [6]. A two-antenna 802.11ac device (using a maximum of two SSs) can support a maximum data rate of 1.73 Gb/s. Furthermore, an 802.11ac device can surpass the data rate of an 802.11n device, but with much lower complexity and cost. For example, an 802.11n device requires three antennas (three spatial streams) to achieve a similar data rate (450 Mb/s) as a single-antenna 802.11ac device.

Rate-Range

In addition to increasing the maximum data rate, these enhancements also lead to improved rate-over-range performance of 802.11ac compared to 802.11n. Figure 1b shows simulated performance of both technologies, using a path loss model validated with measurements from a large home. Both technologies support three spatial streams, and have three transmit and receive antennas (3×3). The figure shows the TCP/IP throughput vs. the distance between the wireless devices.

From Fig. 1b, it can be seen that the 802.11ac devices can connect at twice the range of the 802.11n devices, at the maximum TCP/IP throughput of the 802.11n device (approximately 280 Mb/s). For an end user, this translates to 802.11ac devices experiencing higher throughputs across most locations in a home/office environment. Another observation from Fig. 1b is that the peak rate of the 802.11ac device is three times that of the 802.11n device. For the end user, this translates to 802.11ac-enabled devices experiencing much higher throughput for in-room and peer-to-peer scenarios.

Improved 802.11ac data rates translate to improved application layer throughputs for the same latency. For example, a single-antenna 802.11ac 80 MHz device (433 Mb/s data rate) can support approximately 200 Mb/s application throughput with a 20 ms latency limit. In comparison, a single antenna 802.11n 40 MHz device (150 Mb/s data rate) can support approximately 70 Mb/s throughput with the same latency constraint. Improved 802.11ac data rates also translate to a reduction in application layer latency relative to 802.11n. A single-antenna 802.11ac device (433 Mb/s data rate) can support 100 Mb/s throughput with less than 5 ms latency. In comparison, a single-antenna 802.11n 40 MHz device (150 Mb/s data rate) requires more than 100 ms latency.

Another important benefit of improved data rate is improved battery life due to lower joules-per-bit consumption in a variety of applications like video streaming, Miracast [1] and file transfers. In a typical commercial handset with an integrated CPU running a file transfer application, the measured time for a file transfer was 0.3 s using 802.11ac with 80 MHz bandwidth. That same file size consumed 0.84 s when using 802.11n with 40 MHz bandwidth. This reduction in device on time to complete a task resulted in 2.3 times lower energy consumption when using 802.11ac relative to 802.11n, despite the fact that the instantaneous power consumption for 802.11ac was slightly higher than when using 802.11n.

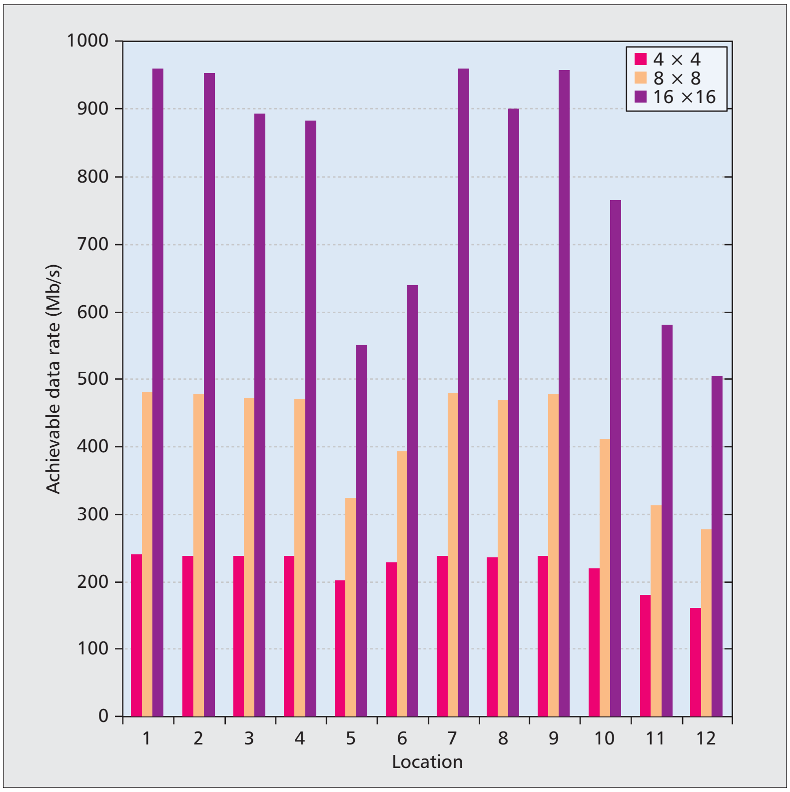

Figure 2. MIMO channel measurements in indoor environments.

Eight Spatial Streams

802.11ac introduces support for up to eight SSs, compared to 802.11n, which defines up to four SSs. In 802.11ac, equal modulation is applied to all SSs for a particular user. Specifically, the transmitter bits are encoded, interleaved and modulated according to one of 10 prescribed modulation and coding scheme (MCSs) and then spatially mapped to physical antennas. The spatial mapping between the SSs and antennas is implementation-specific, and may be frequency-dependent and include transmit steering or precoding matrices [2].

At Qualcomm Incorporated, we conducted extensive indoor channel measurements [3, 4] in an enterprise setting to assist in the construction of the IEEE 802.11ac channel model. Figure 2 illustrates some results from a measurement campaign in one of our office buildings. More than 3500 MIMO channel measurements were made in 12 locations, spanning both line-of-sight (LOS) and non-line-of-sight (NLOS) scenarios, with up to 98 dB of path loss. Measurements were made with up to 16 antennas at both ends of the link, so a comparison can be made of achievable throughput for various numbers of antenna configurations. Figure 2 shows results for 4 × 4, 8 × 8, and 16 × 16 achievable data rates. The achievable data rate was computed by recording the MIMO channel realization across 64 tones in a 20 MHz bandwidth, computing the receiver post-processing signal-to-interference-plus-noise ratio (SINR) assuming a minimum mean square error (MMSE) receiver, and mapping it to the 802.11n physical layer (PHY) rate table using 52 tones for data.

It can be seen from Fig. 2 that data rate scales with MIMO channel dimension beyond 4 × 4 to a large number of antennas. This is attributed to the rich indoor scattering environment, leading to a large number of supported spatial modes even for a 16 × 16 antenna configuration. In most locations, antennas spaced 1/2 wavelength apart observe channel realizations with low correlations that provide almost linear growth of achievable data rate vs antenna dimension. While eight-SS support may seem particularly challenging to accommodate in a small form-factor device, it should be noted that using co-located cross-polarized antennas can save significant space, and that many retail WLAN access points (APs) have more than 10 antennas today. In fact, with additional channel measurements, we observed that using co-located cross-polarized antennas lead to minimal loss in channel capacity in an indoor environment. It was our desire to exploit this spatial capacity, while recognizing the feasibility issues of using more than two antennas at a station device, which led us to promote multi-user MIMO (MU-MIMO) for 802.11ac.

Multi-User MIMO

In 802.11n, a transmission to a device happens using single-user (SU) MIMO modes, where the data rate to a device scales with the minimum number of antennas of each devices. An 802.11n access point (AP) must transmit data using time-division multiplexing (TDM) to different devices, attempting to divide up the network throughput between stations. Unfortunately, 802.11n network capacity is then limited by lower-cost devices that have a smaller number of antennas.

802.11ac MU-MIMO transmission modes

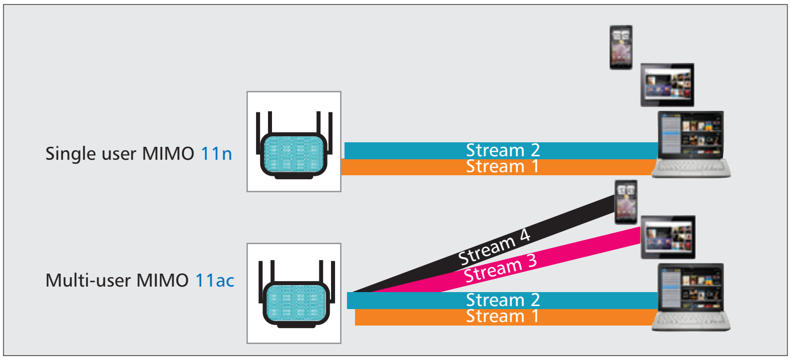

Figure 3. Illustration of MU-MIMO 802.11ac and SU-MIMO 802.11n.

allow simultaneous transmissions to multiple devices using space-division multiplexing (SDM), as depicted in Fig. 3. This significantly improves the spectral efficiency of a WLAN when there are stations with limited numbers of antennas. Essentially, MU-MIMO captures the maximum spatial capacity without requiring all individual stations to have a large number of antennas. The 802.11ac standard allows a MU-MIMO transmission to be sent to up to four simultaneous stations. Each station may receive up to four SSs, but the total number of SSs in a MU-MIMO transmission does not exceed eight total SSs summed across all stations.

As an example, consider a WLAN network where the AP, utilizing four antennas, serves downlink traffic to a laptop with two antennas and a handset with one antenna. Even though an AP may support four SS MIMO (i.e., 1.73 Gb/s data rate in 80 MHz), only two SS transmissions (i.e. 867 Mb/s) can be supported to the laptop, and only a single SS can be transmitted to the handset (i.e., 433 Mb/s) due to the antenna limitations at the stations. Without MU-MIMO, the AP must TDM data to both devices and incur some additional medium access contention overhead. Hence, the network capacity with both devices drops to less than the average of the two rates (e.g., less than 650 Mb/s). The single-antenna handset device creates a capacity bottleneck for the two-antenna device.

Using MU-MIMO, the AP can transmit data to both devices simultaneously, avoiding additional contention overhead and increasing the network data rate to 1300 Mb/s. MU-MIMO delivers MIMO network capacity, but without the requirement of many antennas at the stations.

An important benefit of an 802.11ac network is the reduction of station-side complexity. Thus, 802.11ac can be viewed as a technology that lowers network deployment costs. This is because in an 802.11ac MU-MIMO network, performance is not sacrificed with the addition of stations with fewer antennas. Fewer antennas means lower-cost station devices with fewer analog chains and with digital circuitry built for fewer SSs. As a result, an 802.11ac device can have a smaller number of antennas than an 802.11n device, but support much higher network throughput compared to an 802.11n device. A Qualcomm over-the-air MU-MIMO prototype has demonstrated that a lower-complexity 802.11ac network with a 4 × 4 AP and three single-antenna stations has similar network throughput as a more complex 802.11n network with a 4 × 4 AP and three 3 × 3 station devices, even when both types of networks use the same channel bandwidth.

Transmit beamforming and MU-MIMO require knowledge of the channel state to compute a precoding or steering matrix that is applied to the transmitted signal to optimize reception at one or more stations. The 802.11ac standard employs an explicit channel sounding and feedback protocol that works as follows. Upon gaining access to the medium, the AP transmits a Null Data Packet (NDP) Announcement Frame, identifying the stations for which it wants to collect channel state information for a subsequent MU-MIMO transmission, followed by an NDP frame. The first identified station in the NDP Announcement Frame then estimates the downlink channel from the NDP and feeds back the channel state information (CSI) to the AP. The remaining identified stations send CSI feedback upon subsequently being polled. The AP can then calculate the MU-MIMO precoding weights and use them in subsequent MU-MIMO data transmissions to the relevant stations. One positive consequence of the 802.11ac MU-MIMO specification is that it led to a unification of an underlying transmit beamforming feedback mechanism. This fact accelerated the industry adoption of transmit beamforming, and cleared up the confusion surrounding the multiple defined methods of 802.11n.

MU-MIMO Performance

During the development of the 802.11ac specification, we validated the performance of MU-MIMO over the air (OTA), using custom software and utilizing a four-antenna 802.11n mini-PCIe card containing the Qualcomm WCN 1320 chip (Fig. 4a).

Similar to how 802.11ac is defined, the AP transmitted one NDP to multiple stations. Each station in the network then estimated the downlink channel from the NDP and sequentially transmitted back the CSI data to the AP using an 802.11n packet. The AP used the CSI to calculate the MU-MIMO MMSE precoder and then transmitted a single precoded SS to each station simultaneously. The OTA prototype was equipped with a tool to plot a raw packet-by-packet data rate over time. Physical layer data rates per packet were calculated by determining the receiver post-processing SINR and mapping SINR to an appropriate 802.11n or 802.11ac MCS. Much of the testing used the four-antenna AP device in Fig. 4a, but to test the full capabilities of the 802.11ac MU-MIMO specification, we also built a custom eight-antenna AP that could enable transmissions of up to eight SSs distributed across multiple stations.

Figure 4b shows a representative snapshot of 802.11n SU-MIMO, 802.11ac SU-MIMO, and 802.11ac MU-MIMO PHY data rates. The network configuration uses a four-antenna AP, three one-antenna stations, 40 MHz bandwidth (BW), and long guard interval. Figure 4b shows data collected at in-room distances between the AP and stations, so there was a high signal-to -

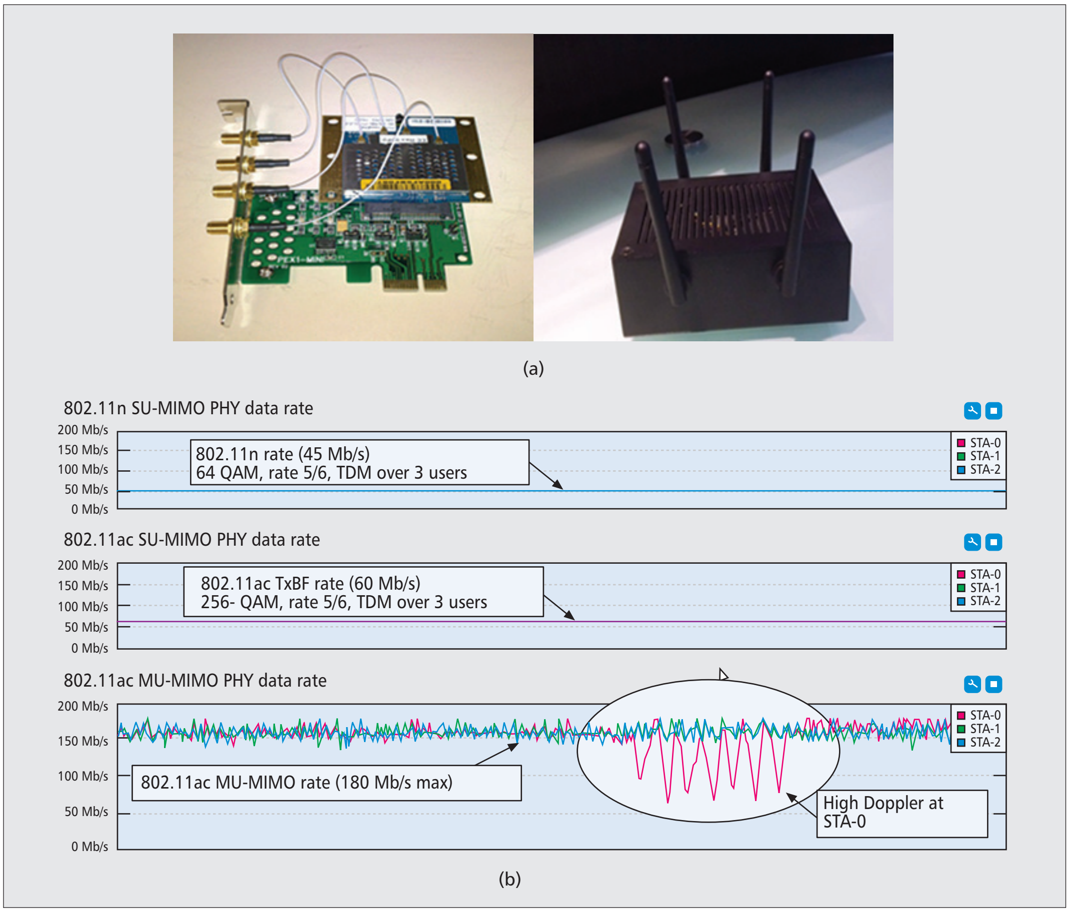

Figure 4. a) 4 × 4 802.11n mini–PCIe card and MU–MIMO access point; b) time snapshot from OTA MU–MIMO prototype in LOS, high SNR scenario.

MU-MIMO provides an improvement of approximately three times the data rate over baseline 802.11n transmissions in this scenario. These gains were realized even when the AP and stations were in LOS channel conditions, and even with stations placed as close together as possible.

noise ratio (SNR) for each link. It should be noted that no MAC layer adaptive rate control or rate averaging across packets was employed. It can be seen from Fig. 4b that the 802.11n SU-MIMO transmissions are saturated at the maximum 802.11n MCS of 64-QAM rate 5/6 (135 Mb/s), leading to an effective data rate per station (STA) of 45 Mb/s, including the TDM factor of three. The 802.11ac SU-MIMO transmissions are saturated at the maximum MCS of 256-QAM rate 5/6 (180 Mb/s), leading to an effective data rate per STA of 60 Mb/s, including the TDM factor of three.

MU-MIMO provides an improvement of approximately three times the data rate over baseline 802.11n transmissions in this scenario. These gains were realized even when the AP and stations were in LOS channel conditions, and even with stations placed as close together as possible. This is in alignment with channel measurement data presented earlier, where it was explained that even a half-wavelength antenna spacing leads to a very high number of available spatial dimensions. In practice, despite close proximity or LOS channel conditions, there remains enough local scattering from walls, floors, ceilings, and the device itself to create a rich multipath environment.

Another important finding, illustrated in Fig. 4b, is that MU-MIMO transmissions are robust to a noisy or rogue station that may send outdated or otherwise inaccurate CSI feedback (due to either Doppler or poor implementation). In the data-rate-versus-time plot, STA-0 was subjected to a high Doppler event by rapidly waving the handheld device. As a result, the CSI feedback from STA-0 was typically outdated once the MU-MIMO transmission occurred. This reduction in CSI fidelity for STA-0 directly impacts the STA-0 data rate and not that of other STAs in the MU-MIMO group. Intuitively, this can be explained as follows. In high SNR regimes, MMSE-based transmit precoding schemes approach the zero-forcing precoder, where transmissions to a given station occur in the null-space of other stations. If a station's CSI (and hence its null-space information) is incorrect at the AP, only that station suffers from interference as transmissions to other stations will no longer be in the null-space of the station.

Figure 5a shows a representative plot of the median system throughput over a two-hour period of time vs. the total number of MU-MIMO SSs (Nss) used across all stations. We see that maximum system throughput is obtained when the total number of MU-MIMO SSs is approximately 75 percent of the total number of AP antennas. For an AP with four antennas, throughput is maximized with three total SSs. An eight antenna AP can maximize throughput using six total SSs. For less than 75 percent loading, system throughput is proportional to the

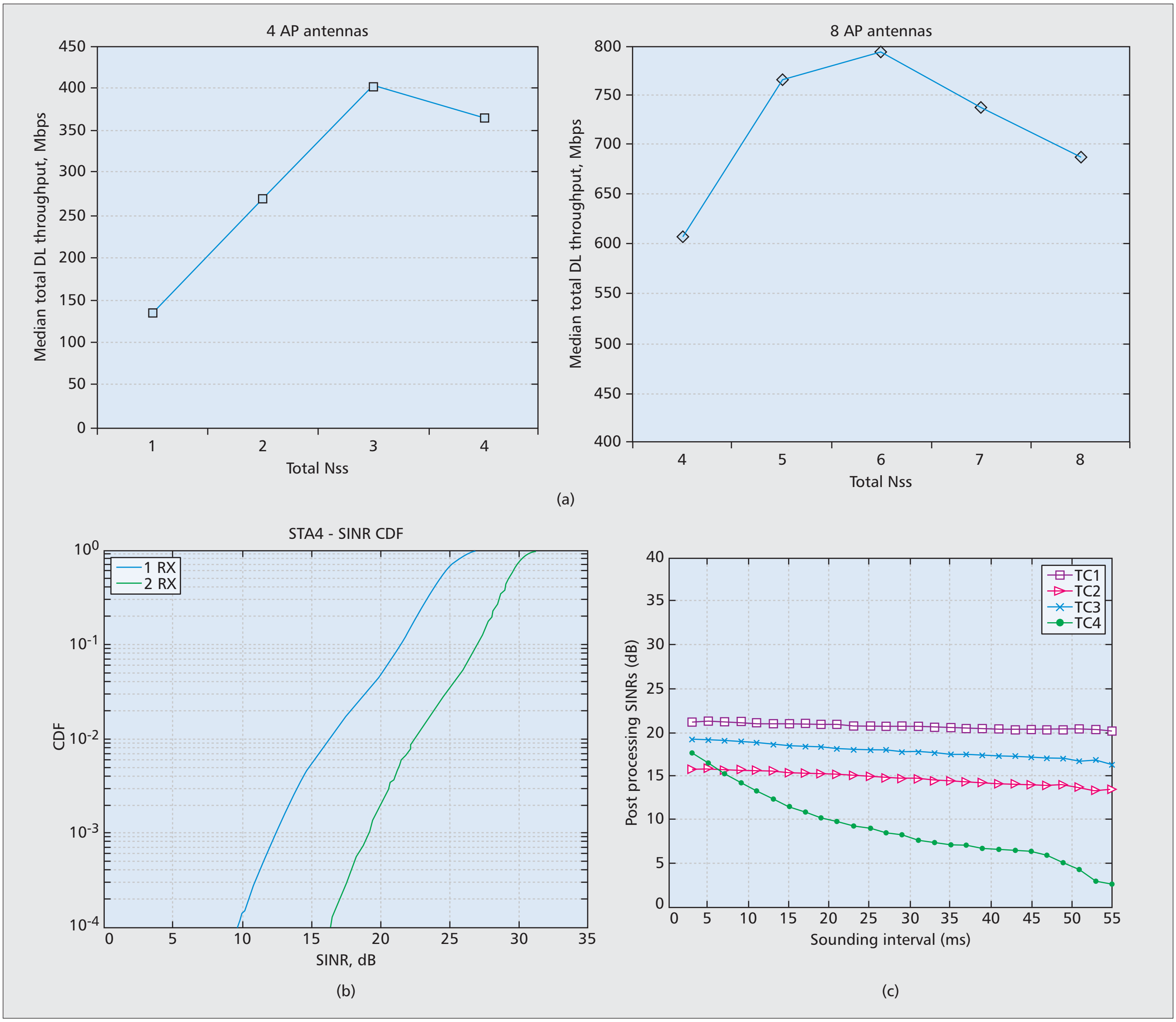

Figure 5. a) Median throughput vs. total number of MU-MIMO spatial streams (Nss) across all STAs; b) CDF of MU-MIMO post-processing SINRs with 1 Rx antenna and 2 Rx antennas (interference nulling enabled); c) MU-MIMO post-processing receiver SINRs, over time, for different user test cases (TC): TC1 — watching a movie; TC2 — typing on the keypad; TC3 — holding the device to the ear; TC4 — walking.

number of SSs, and inter-stream interference tends to be negligible. However, beyond 75 percent loading, the throughput loss due to inter-stream interference exceeds any additional throughput gains due to increased number of spatial streams. That level of inter-stream interference is dependent on the noise floor of the station devices, along with the transmit error floors due to common analog impairments in an AP. This demonstration of MU-MIMO employed devices that have 3 dB margin to the error vector magnitude requirements in the 802.11ac specification.

If a station has additional antennas beyond the number of spatial streams assigned to it in a MU-MIMO transmission, the surplus antenna(s) can be used to cancel interference created by MU-MIMO streams intended for other stations [5]. This is enabled by the design of the long training field(s) (LTFs) in 802.11ac, allowing the receiver to estimate the channel corresponding to all MU-MIMO spatial streams. Figure 5b demonstrates the improvement of post-processing receiver SINR due to interference nulling. The figure shows the cumulative distribution function (CDF) of post-processing SINR collected from a network configuration where three stations each receive one spatial stream from a four-antenna AP. One of the stations has two receive antennas and uses the extra antenna to cancel any interference from other streams. The SINR is logged from the stations during a two-hour period in a busy lab environment. In this scenario, interference nulling provides a 5 dB median SINR gain for the station with two receiver antennas.

Figure 5c demonstrates another observation about the performance of MU-MIMO with handheld devices, which will exhibit repeated small movements associated with stationary use cases such as watching a movie, typing the keypad, and holding the device to the ear. For reference, performance with a station experiencing walking speeds was also recorded. For each test case, multiple tests were run across a variety of LOS and NLOS conditions, with varying path loss. The stations are actual handset mockups in order to encourage users to interact as if they were using a real phone. In addition, different test subjects were used to gather diverse data. For each test, post processing SINRs were recorded for a minute of time. The main observation is that for typical stationary use cases, degradation in the post-processing SINR is at most 3 dB for a CSI feedback delay of 50 ms. Therefore, MU-MIMO works well with handheld devices and typical use cases.

For the higher Doppler rate walking test, where the test subject is walking with the mockup antenna placed to the user's ear, the degradation in SINR is ~10 dB for a CSI feedback delay of 50 ms. With immediate CSI feedback, the performance degradation was less than 3 dB. Thus, for mobility use cases, immediate feedback is required for good performance.

Future technology enhancements like uplink MU-MIMO can be used to reduce CSI feedback overhead in order to improve downlink MU-MIMO performance even in mobility use cases.

802.11ah for Internet of Things and Extended-Range WLAN

Introduction

The wireless industry is poised for high growth in the wireless sensor market, with applications across home and industrial automation, healthcare, energy management, and wearable devices. The home automation category includes devices such as temperature, moisture, and security sensors. They are often small form-factor devices that are battery-powered and headless, but require whole-home coverage (including attics, backyards, basements, and garages). Unfortunately, the market is fragmented with multiple non-interoperable technologies, some with coverage issues and some with non-user-friendly network configuration and deployment issues.

The IEEE 802.11ah Task Group is developing a specification for the license-exempt bands below 1 GHz, targeting such lower-data-rate and longer-range applications for devices commonly referred to as the Internet of Things. 802.11ah enables a "lower band of Wi-Fi," augmenting the traditional 2.4 and 5 GHz bands used today.

802.11ah Features and Supporting Measurements

The 802.11ah Draft Specification [7] defines mandatory 1 and 2 MHz BW modes that are globally interoperable. These modes are ideally suited for devices that require low power consumption and long-range connection to an AP. In addition, there are optional 4, 8, and 16 MHz modes for the more traditional WLAN use cases, which can be used in regulatory domains that allow larger BW. The 802.11ah modes enable a wide range of data rates between 150 kb/s to 78 Mb/s per spatial stream. The lowest data rate of 150 kb/s is enabled by the use of 1 MHz bandwidth, 2× repetition coding, 1/2-rate FEC coding, and binary phase shift keying (BPSK). For four SSs, the maximum data rate is 312 Mb/s. 802.11ah inherits the advanced FEC and spatial diversity schemes from 802.11ac. This rich data set and robustness enable a wide variety of use cases pertaining to the Internet of Things such as sensors, wireless audio, security cameras, wireless video, and Internet connectivity.

Qualcomm Incorporated has conducted measurement studies across several single-family (2400–5000 ft^2) U.S. homes [8] to demonstrate the improved range characteristics of 802.11ah modes in the 900 MHz unlicensed band. Our measurements revealed more than a 10 dB range advantage of 802.11ah over 802.11n and 802.11b. In the measurement campaign, the APs were placed in typical locations, and measurements were made at multiple locations around the home, including backyards, garages, and basements. At each location, the median path loss and SNR were measured in both the 900 MHz and 2.4 GHz bands. These measurements were combined with 802.11n (20 MHz) and 802.11ah (1, 2, 4, and 8 MHz) receiver sensitivity performance targets to estimate the supported physical layer data rates at each location. The data rate computation assumed an AP with two antennas and stations with a single antenna. In addition, an 8 dB obstruction loss was applied to account for possible realistic obstructions like human bodies and furniture. The data rates reflect uplink performance assuming only 4 dBm of transmit power from stations. Figure 6 shows the 802.11ah rate vs. range improvement over 802.11n and 802.11b from three representative homes with varying square footage.

It can be seen that 802.11ah has improved coverage and higher data rates in locations with larger path loss due to the lower-BW physical layer modes [8] and better propagation characteristics at sub-1 GHz frequencies. 802.11n and 802.11b devices experience significant coverage holes for locations with larger range, and would require higher transmit power for whole home coverage. That higher output power comes at a significant disadvantage of higher power consumption for the 802.11n and 802.11b stations. By exploiting the better range capabilities of 802.11ah to reduce output power, 802.11ah stations can achieve multiple-year battery operation for applications that require low duty cycle.

In addition to the link budget benefits described above, 802.11ah accommodates larger delay spread and Doppler spread, making it a favorable technology for outdoor use. Due to longer symbol times, the delay spread tolerance is 10 times that of 802.11ac. Furthermore, 802.11ah has an improved pilot design that enables robust channel tracking throughout a packet.

802.11ah introduces several innovations to enable low-power applications. Included in this list are smaller frame formats that save power [9], new traffic priority rules, and scheduled access for battery-operated devices that improves

The 802.11ah Draft Specification defines mandatory 1MHz and 2MHz BW modes that are globally interoperable. These modes are ideally suited for devices that require low-power consumption and a long-range connection to an AP. In addition, there are optional 4, 8 and 16 MHz modes for the more traditional WLAN use-cases.

latency and reduces collisions. Furthermore, there are more efficient sleep modes that allow a device to remain asleep longer and wake-up more efficiently [10]. In addition, an 802.11ah network can support thousands of devices due to new efficient paging and scheduling mechanisms. 802.11ah also has mechanisms to enable a relay operation that can be used to further extend coverage.

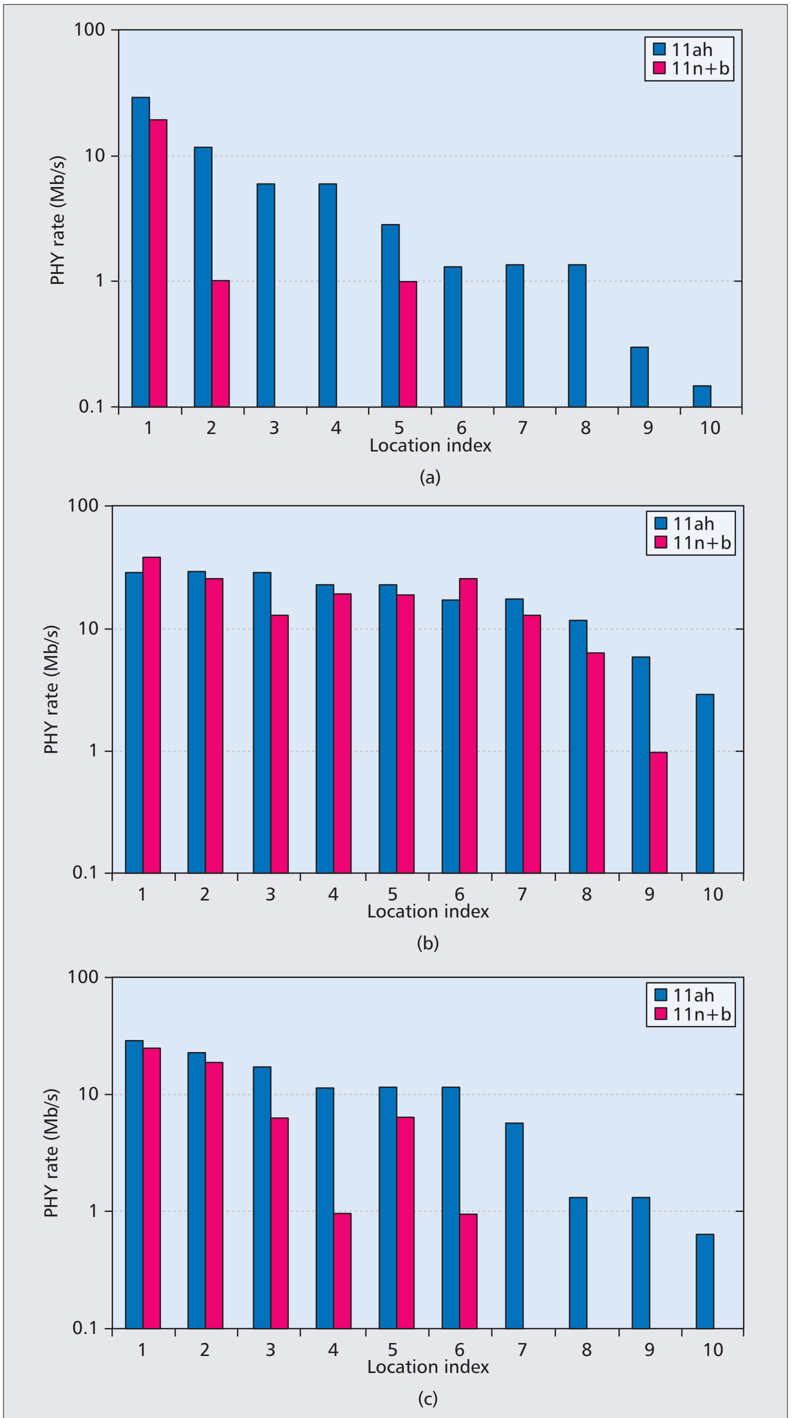

Figure 6. Rate and coverage advantage of 802.11ah over 802.11n and 802.11b in single-family homes: a) 2230 ft^2 home with basement and yard space (2.4 GHz path losses: 76–104 dB); b) 3500 ft^2 single-family home with two floors (2.4 GHz path losses: 72–94 dB); c) 4249 ft^2 home with basement, backyard, and front yard (2.4 GHz path losses: 76–98 dB).

The above mentioned technology enhancements, coupled with an ecosystem of interoperable WLAN devices (including smartphones), make 11ah a compelling technology for the Internet-of-Things.

Emerging Technologies

Due to the overwhelming success of WLAN, improved connection times and throughput performance in crowded networks is an area of concern in the future. Thus, industry efforts have begun to address performance in these scenarios.

802.11ai for Optimized Connectivity Experience

The IEEE 802.11ai Task Group is developing a specification [11] for fast initial link setup (FILS), which promises to address operator concerns about poor offload experience to WLAN. These concerns include significant connection signaling overhead, slow handoff mechanisms between hotspot APs, and "probe storms" that lead to service outage.

802.11ai enhances network connection experience through faster initial link setup time, enabled by defining steps to complete association and authentication with a four-way handshake and IP address assignment in just two round-trip messages [12]. A FILS discovery frame is defined that leads to more efficient and more frequent beacon transmissions, leading to more rapid network detection. AP-to-AP handoff is improved in dense networks by means of a Neighbor AP advertisement and FILS parameter advertisement in beacons and probe responses. Neighbor AP advertisement contains elements such as basic service set identification (BSSID), service set identification (SSID), channel information, and target beacon transmit time (TBTT) offset of neighboring APs. Knowledge of these parameters significantly reduces device network scanning time. FILS parameter advertisement consists of elements such as subnet identifier (ID), authentication domain, and IP address domain of the AP. Knowledge of these parameters helps the device identify the correct APs and enable faster link setup time.

Qualcomm Incorporated has prototyped elements of 802.11ai on commercially available handsets and APs. Assuming a 200 ms authentication server delay, it was observed that 802.11ai significantly reduced handset connection time to a new AP with a different SSID and subnet ID, from approximately 8 s to less than 0.5 s.

802.11ai also reduces probe storms by introducing alternative methods of connection to inefficient Probe Request and Response messages. New efficient messaging is introduced including Broadcast Probe Response, Selective Probe Response based on Probe-Request content, and shorter Probe Response containing only the changed parameters since the last association. The FILS discovery frame, which is sent much more frequently than the legacy beacon, also reduces Probe Request and Response traffic.

802.11ax for High-Efficiency WLAN

To address inefficiencies of WLAN in dense indoor and outdoor networks, and improve robustness to interference in the traditional 2.4 and 5 GHz bands, a new IEEE 802.11 Task Group called 802.11ax has been formed. The Task Group is in the early stage of specification development with a projected completion date around the 2019 timeframe. Evaluation metrics include average and 5 percent per station throughput, area throughput, and packet delay and error rate requirements of applications. The Task Group will take into consideration new application trends with bidirectional, uplink-intensive, and peer-to-peer traffic such as video-conferencing, user-generated uploads from wearable devices, smartphones and other consumer electronic devices, display, docking, and scalable peer-to-peer networks.

A key technical direction for the Task Group is to increase parallelization of traffic in the spatial and frequency domains, to achieve at least four times average medium access control (MAC) throughput increase per station over 802.11ac networks. Technologies under consideration include uplink MU-MIMO, orthogonal frequency-division multiple access (OFDMA), an improved OFDM numerology with longer symbol duration and cyclic prefix for outdoor channel support, and enhancements to the legacy clear channel assessment (CCA) schemes.

The Task Group will take into consideration new application trends with bi-directional, up-link intensive and peer-to-peer traffic such as video-conferencing, user-generated uploads from wearable devices, smartphone and other consumer electronic devices, display, docking and scalable peer-to-peer networks.

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Biographies

VK Jones (vkjones@qca.qualcomm.com) is a vice president, technology, at Qualcomm Incorporated where he leads the Wireless LAN System Engineering organization. He came to Qualcomm in 2006 through the acquisition of Airgo Networks Incorporated, a company he co-founded in 2001. Prior to Airgo Networks, he was chief scientist and co-founder of Clarity Wireless, which in 1998 was acquired by Cisco Systems, where he helped establish the first wireless business unit. Prior to founding Clarity Wireless, he served as acting assistant professor, mechanical engineering, Stanford University. He received his Ph.D. and M.S.E.E. from Stanford University in 1995 and 1991, respectively. He received his B.S.E.E. from Purdue University in 1990, where he won numerous basketball championships for Delta Tau Delta.

Hemanth Sampath (hsampath@qti.qualcomm.com) is a principal engineering manager and WLAN systems engineering lead at Qualcomm Research Center. His team focuses on design, standardization, prototype, and product support for next generation WLAN technologies. Prior to his current role, he was the systems lead for a pre-LTE (UMB) chip development project within Qualcomm. From 2001 to 2004, he held technical positions at Iospan Wireless Inc. and Marvell Semiconductor. He received his Ph.D. and M.S.E.E. from Stanford University in 2001 and 1998, respectively. He received his B.S.E.E. from the University of Maryland at College Park in 1996.

WiFi-Based Indoor Positioning

Chouchang Yang and Huai-Rong Shao

ABSTRACT

Recently, several indoor localization solutions based on WiFi, Bluetooth, and UWB have been proposed. Due to the limitation and complexity of the indoor environment, the solution to achieve a low-cost and accurate positioning system remains open. This article presents a WiFi-based positioning technique that can improve the localization performance from the bottleneck in ToA/AoA. Unlike the traditional approaches, our proposed mechanism relaxes the need for wide signal bandwidth and large numbers of antennas by utilizing the transmission of multiple predefined messages while maintaining high-accuracy performance. The overall system structure is demonstrated by showing localization performance with respect to different numbers of messages used in 20/40 MHz bandwidth WiFi APs. Simulation results show that our WiFi-based positioning approach can achieve 1 m accuracy without any hardware change in commercial WiFi products, which is much better than the conventional solutions from both academia and industry concerning the trade-off of cost and system complexity.

INTRODUCTION

Recently, indoor positioning has attracted a lot of attention from the research as well as industry communities. This is partly driven by the increasing market demand for indoor location-based services. Due to the limitation of GPS signal strength indoors, nearby anchors with known positions are generally needed for an indoor positioning system. In general, there are two components in indoor positioning systems. One is nearby anchors with the knowledge of their own location information, while the other is a positioning device with the objective of identifying its location by processing wireless signal through nearby anchors. Although the potential needs of indoor navigation services have been already addressed from industry aspects, the ultimate solution in terms of accuracy, reliability, real time, and low cost remains open.

There are several factors that make the design of indoor positioning systems challenging:
- *Cost*: The need to deploy nearby anchors and the development of localization devices to identify objects are costly.
- *Numbers of nearby anchors*: Unlike the traditional navigation GPS system, which uses 31 active satellites in orbit, the indoor environment and space sometimes limit the number of anchors.
- *Complicated indoor environment*: The wireless signal used to measure the distance and angle indoors usually suffers significantly from obstacles, such as walls, objects and/or human beings, which lead to multipath effects.

Due to its wide deployment, we expect WiFi to become a prominent tool for indoor positioning. In this article, we use WiFi access points (APs) with multiple antennas as nearby anchors, while the positioning device could be any mobile platform with WiFi capability such as smartphones or tablets. Since the number of WiFi APs is limited and the system bandwidth is narrow (up to 40 MHz for 802.11n and 160 MHz for 802.11ac), we propose a novel method of sending multiple messages to improve time of arrival (ToA) measurement and angle of arrival (AoA) estimation.

The article is organized as follows. First, we review the conventional indoor localization methods and applications. We then discuss our proposed improved estimates of ToA and AoA, respectively. Next, we describe the application of the proposed mechanisms to WiFi indoor positioning and summarize its performance via simulations. Finally, we conclude our work.

INDOOR LOCALIZATION APPROACHES AND APPLICATIONS

In general, the approaches used in indoor localization can be classified into four categories:
1 Time of arrival
2 Angle of arrival
3 Hybrid ToA/AoA
4 Received signal strength (RSS) and fingerprint

Each of these approaches has its own advantages and limitations.

TIME OF ARRIVAL

Time of arrival is the travel time between a transmitter and a receiver. The distance can be calculated using travel time multiplied by the speed of light. To measure the travel time in the air, this approach usually requires synchronization between transmitters and receivers. In addition, it requires at least three anchors to have the plane-domain (2D) localization as depicted in Fig. 1a and four anchors for 3D localization. The positioning performance is decided by a sig-

Chouchang Yang is with the University of Washington.

Huai-Rong Shao is with Samsung Research America.

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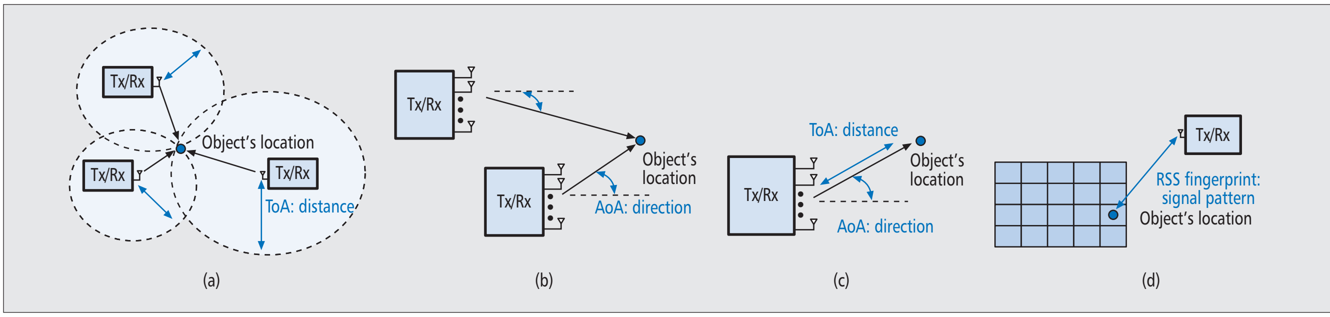

Figure 1. a) Time of arrival approach; b) angle of arrival approach; c) hybrid ToA/AoA approach; d) received signal strength and fingerprint approach.

nal's bandwidth as well as sampling rate [1]. When the arrival wireless signal is sampled in receiver as depicted in Fig. 2, the first sample that captures the arrival signal is not exactly the ToA. In other words, when a signal's bandwidth is not wide enough, the ToA measurement may result in a wide error range of distance. For example, a wireless system with 10 MHz bandwidth as well as sampling rate can only measure the time duration up to 1×10^{-7} s resolution. Therefore, the maximal error in distance is up to $3 \times 10^{8} \times 10^{-7} = 30$ m. When the wireless system has 1 GHz bandwidth, the receiver can measure up to 1×10^{-9} s resolution such that the maximal error in distance is lower than 30 CM. Current popular solutions applied to ToA are ultra wideband (UWB) systems [2]. The accuracy a UWB system can achieve is up to 1 cm [3]. However, it requires very wide bandwidth as well as special hardware design to support localization, which results in very high hardware cost.

Angle of Arrival

Angle of arrival measurement is the method that can determine the incoming signal direction from a transmitter on the antenna array. By exploiting and detecting phase difference among antennas [4], the direction of an incoming signal can be calculated. In order to locate the position, this approach requires two anchors with antenna arrays at different places to obtain the object position as depicted in Fig. 1b. However, due to multi-path affection, the AoA in terms of line of sight (LOS) is hard to obtain. An example of a commercial solution applied to AoA is Quuppa's HAIP system where the positioning accuracy can be achieved from 0.5 to 1 m [5]. However, it requires a specific hardware device including 16 array antennas with a transmitter as nearby anchors and a special tag as a positioning device through Bluetooth enhancement of a wireless signal.

Hybrid ToA/AoA Approaches

Due to the complicated indoor environment and limited number of nearby anchors, a hybrid ToA/AoA approach has been introduced [6]. By utilizing the information measured from AoA and ToA, the number of nearby anchors can be reduced. As illustrated in Fig. 1c, it is possible to localize an object using a single nearby anchor. The hybrid approach suffers the same challenge of signal bandwidth as well as the number of antennas. Nevertheless, hybrid systems leverage the benefit from both mechanisms.

Received Signal Strength and Fingerprint

The received signal strength (RSS) and fingerprint is a site-survey approach for positioning [7]. For RSS-alone approaches, the received signal strength ratio reflects the distance information. Through calibration of transmitter power with a corresponding free-space channel model established by measuring distance and power at each point in advance, the coarse distance can be estimated for each anchor node. To further improve location precision, RSS with fingerprint combination is proposed. In general, due to the multi-path effect, each location receives a unique signal through the combination of various rays from different paths. Thus, the signal property, such as frequency response, and signal strength regarding the I/Q channel has its own fingerprint. By associating the signal fingerprint with the target, the anchor can deduce possible location from a pre-measured fingerprint database. This mechanism only requires one anchor node instead of multiple anchors for positioning. In Fig. 1d, the signal fingerprint in each grid is measured in advance. When the object sends the signal toward a nearby anchor (Tx/Rx), the anchor can choose the most similar fingerprint from the database regarding the received signal to do localization. The localization accuracy will depend on not only the size of grid area but also the signal difference among grids. When the difference between signal fingerprints is small, the uncertainty increases.

A commercial solution applying RSS indication (RSSI) alone is Apple's iBeacon technology, and one applying RSS fingerprint-based is Apple's WiFiSLAM. For iBeacon, the location is calculated via a Bluetooth Low Energy system. The coarse distance can be d as immediate, near, and far, three different statuses that refer to within centimeters, a couple of meters, and more than 10 meters, respectively. The WiFiSLAM solution applies a WiFi RSS fingerprint-based mechanism with gyroscope sensor assistance [8] to record each location's fingerprint. WiFiSLAM can achieve up from 1.75 to 2.5 m accuracy. However, it requires a site survey fingerprint in advance. When the environment is dynamic, such as a shopping mall with moving crowd, the performance can degrade dramatically. Moreover, the calculation loading increases exponentially with the numbers of fingerprint points in the database.

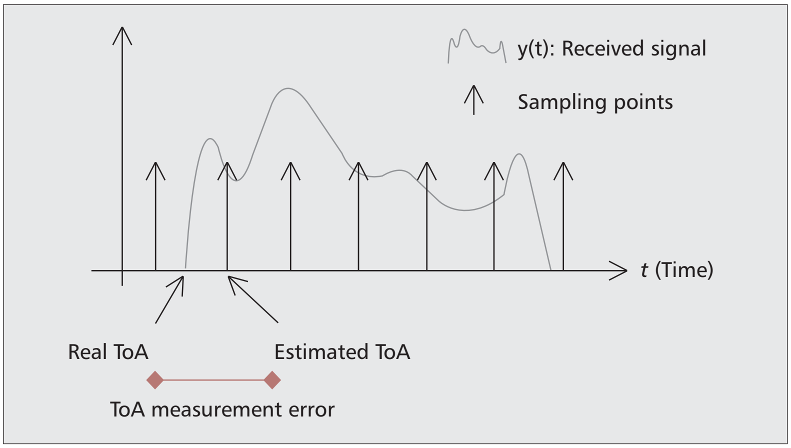

Figure 2. ToA performance vs. transmitter bandwidth.

MINIMAL COST FOR INDOOR LOCALIZATION INFRASTRUCTURE

Several existing solutions have been investigated through various applications such as UWB, WiFi, and Bluetooth-based applications recently. By considering the cost in terms of deploying anchor nodes as well as the coverage area, the most possible solution is expected in WiFi [9] systems. Unlike a UWB system, WiFi infrastructures have been widely deployed within indoor environments. Comparing with other existing infrastructures such as a Bluetooth-based system, each WiFi anchor node can cover a much wider area than Bluetooth, which costs less anchor nodes. In addition, WiFi has been considered as an important medium for high throughput transmission in indoor environments. In other words, the WiFi hardware capability will follow user desires in terms of high transmission speed by increasing sampling rate with multiple-input multiple-output (MIMO) structure, which can benefit indoor localization performance. The existing work in [10] has shown that a WiFi signal can provide promising performance to overcome indoor multipath, where LOS is possible to detect. By having gyroscope sensors' assistance from mobile phones, the multipath effect can be further mitigated [11]. Therefore, we choose WiFi as a possible solution for low-cost and accurate localization.

SUPER-RESOLUTION TOA AND PERFORMANCE IMPROVEMENT

Time of arrival performance is decided by signal bandwidth as well as sampling rate. When the sampling rate is low (narrow bandwidth), the ToA may not be captured precisely. In Fig. 2, the received signal is sampled by the black arrows, while the arrival signal (real ToA) is between two sampling positions, causing the measurement error. The state-of-the-art methods to improve ToA use super-resolution estimation, which is based on subspace decomposition of the autocorrelation matrix and requires the calculation of an inverse matrix with its eigenvectors [12, 13]. Those estimation approaches result in heavy calculation loading, while performance improvement still relies on bandwidth.

In the following, we propose a method that can increase the accuracy of the ToA estimate by sending multiples of the same predefined message to assist estimation. Using the fact that each incoming message is very unlikely sampled at the same place, the collection of multiple received messages in linear time invariant (LTI) channels is able to reconstruct the received signal at higher resolution. We denote the received signal as $y(t)$, while $x(t)$ is the message sent by the sender. Then the received signal can be described as $y(t) = \mathrm{x(t)} \otimes h(t)$, where $h(t)$ is the channel impulse response representing the environment effect, and $\otimes$ is the convolution operation. Given the sampling time period as T_s, the received signal after sampling through an analog-to-digital converter (ADC) can be described as $y_d[n] = y(n \times T_s + \tau) + w[n]$. The $y_d[n]$ and $w[n]$ are the nth sample point in terms of received signal and additive white Gaussian noise (AWGN), while τ is the relative starting point of the sampling position and $\tau \in [0, T_s]$.

If the channel is time invariant and the transmitter sends multiple but equal predefined messages, the received signal after sampling can be described as in Fig. 3. The black, red, and blue arrows represent the sampling of three different arrival messages from the received signal $y(t)$. Since the transmitter has sent the same message multiple times, the receiver will always receive the same signal $y(t)$. One can notice that the sampling positions regarding each incoming signal are not at the same positions. Therefore, the collection of multiple messages with right order can possibly reconstruct the received signal, while we assume that the channel is invariant and the noise is negligible compared to the received signal power level. Taking Fig. 3 as an example, the combination of black, red, and blue samples is able to reconstruct received signal $y(t)$ at higher resolution than using one message alone. Although using multiple messages is able to assist reconstruction of the received signal, how to arrange and make messages in the right order is still not clear. We applied the frequency transform property as in the following to achieve the reconstruction goal.

Time Shift Property

$$\text{If } y(t) \underset{\text{Inverse Frequency Transform}}{\overset{\text{Frequency Transform}}{\rightleftharpoons}} Y(f)$$

$$\text{Then } y(t-\tau) \underset{\text{Inverse Frequency Transform}}{\overset{\text{Frequency Transform}}{\rightleftharpoons}} Y(f)\exp(-j2\pi f\tau)$$

The $\exp(-j2\pi f\tau)$ results in the phase rotation $-2\pi f\tau$ in the I/Q domain. The concept behind the above equation means that the time difference between the same signal received at different times, $y(t)$ and $y(t - \tau)$, can be estimated through the phase shift property in the frequency domain. By repeatedly sending the message from the transmitter, the ith message after sampling in the receiver is described as $y_d^{(i)}[n] = y[n \times T_s + \tau_i]$. If we have the ith and jth messages after sampling as $y_d^{(i)}[n] = y[n \times T_s + \tau_i]$ and $y_d^{(j)}[n] = y[n \times T_s + \tau_j]$, the time difference among these two messages is $\delta\tau = \tau_j - \tau_i$. For fast Fourier transform (FFT) size N, $\delta\tau$ can be calculated by applying the time shift property at subcarrier K as

$$\delta_t = \frac{\angle Y_d^j[K] - \angle Y_d^i[K]}{2\pi \frac{K}{NT_s}} \text{ for } K \in 0,1,2,\ldots,N-1 \quad (1)$$

where $\angle\cdot$ is the phase from the I/Q domain in the receiver and $Y_d^j[K] = \text{FFT}\{y_d^j\}$. Due to the cyclic property of discrete time frequency transform, the time difference measuring range is determined by NT_s/K.

Since the receiver can always map the sampling signals to the bit sequence, the resolution of timing difference between two received messages from bit mapping is determined by sampling period T_s and sampling rate. However, further time difference under T_s resolution is hard to obtain. Comparing this with Eq. 1 implies that the time difference can be measured at better resolution beyond the sampling rate limitation. In fact, the measured performance in terms of time difference will depend not only on noise level but also on the choice of subfrequency K. The reason is that different subfrequencies K decide the different measurement ranges such that the same error ratio of phase shift results in different measuring error ranges as well. Therefore, for a predefined message with multiple subfrequencies, we can combine their individual phase shifts to jointly estimate a reliable time difference measurement.

The phase information from the rotation in the frequency domain reflects the time shift between messages. Thus, we can arrange messages with their relative time difference to reconstruct high resolution in terms of received signal. We illustrate this idea using Fig. 3. If the transmitter has sent out three identical messages to the receiver, the received signal at different times may be sampled at slightly different places. These three arrival messages of samples are denoted as $y_d^{(1)}$, $y_d^{(2)}$ and $y_d^{(3)}$. Each message of samples can apply Eq. 1 to allocate the relative place from their time difference. Below, we demonstrate how to estimate ToA in terms of the first message. By placing the other two messages with respect to the first message, we can have high resolution with respect to signal $y(t)$. If the first sample that captures the arrival signal $y(t)$ is from the third message, the ToA can be estimated using the time difference ζ_3 plus the first message timestamp t_q as following $t_q + \zeta_3$. Thus, when the transmitter has sent multiple predefined messages, we can always obtain the ToA through this approach.

One may consider that object's movement could break the linear time invariant (LTI) property. In fact, an indoor object's movement is usually slow, and the wireless channel between a sender and a receiver still remain static over a very short time duration. For example, given a scenario where the object's speed is 20 km/h, the movement in 0.1 ms is 5.6×10^{-4} m. Within this tiny movement, the channel environment between the sender and the receiver can be treated as the same. Therefore, any multiple messages sent within this short time duration are

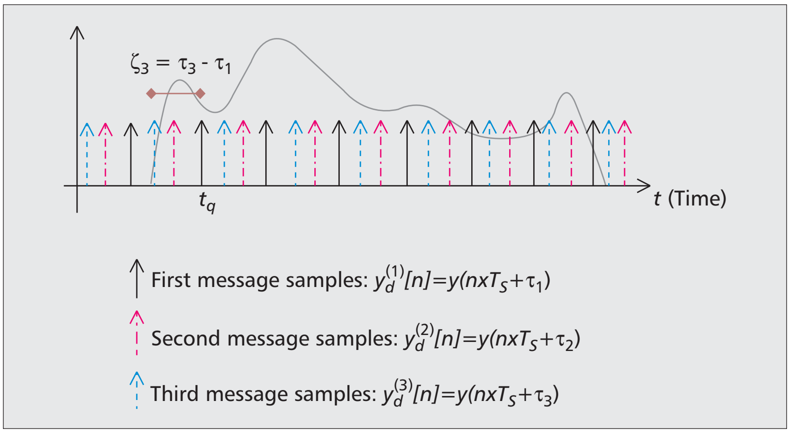

Figure 3. Multiple messages for reconstruction of the received signal.

under an LTI channel. Another possible factor that may break the LTI property is a crowded WiFi channel due to the delay caused by channel contention. However, in a WiFi system, there is a sequence of preambles (Physical Layer Convergence Protocol, PLCP, preambles) and multiple OFDM symbols in each packet. Those preambles and orthogonal frequency-division multiplexing (OFDM) symbols can provide multiple subcarriers' messages in each single packet. Therefore, for each received packet, the receiver can estimate the possible arrival time through the sequence of messages. Unless there is no received packet due to heavy collision, the estimated time can be obtained. The more packets the recipient receives under the LTI property, better the performance it can achieve. The receiver can use cyclic redundancy check (CRC) code and channel state information to ensure that the LTI property exists among multiple packets.

The signal reconstruction relies on finding the sample of a message that is closest to the arrival signal. We denote this message as *the nearest ToA message* (the blue arrows in Fig. 3), a sample of which is relatively near the arrival signal. In the next section, we show how to utilize *the nearest ToA message* to estimate the AoA information.

AoA Approaches and Constraint Improvement

Generally, AoA is measured using the phase difference of arrival signal among multiple antennas. When incoming messages suffer from the multipath effect, the received signal will suffer from the combination of rays with different AoAs. Traditionally, the approaches to identify the angle with respect to each ray from different paths use multiple signal classification (MUSIC) algorithms [10, 14]. Those techniques are subspace decomposition of the autocorrelation matrix. By finding eigenvectors and an inverse matrix, the algorithms can return angle information with each ray. However, which ray represents the LOS is still not clear, so the LOS AoA information cannot be obtained. Another approach is the joint angle and delay estimation

Generally, AoA is measured by using the phase difference of arrival signal among multiple antennas. When incoming messages suffer from the multipath effect, the received signal will suffer from the combination of rays with different AoAs.

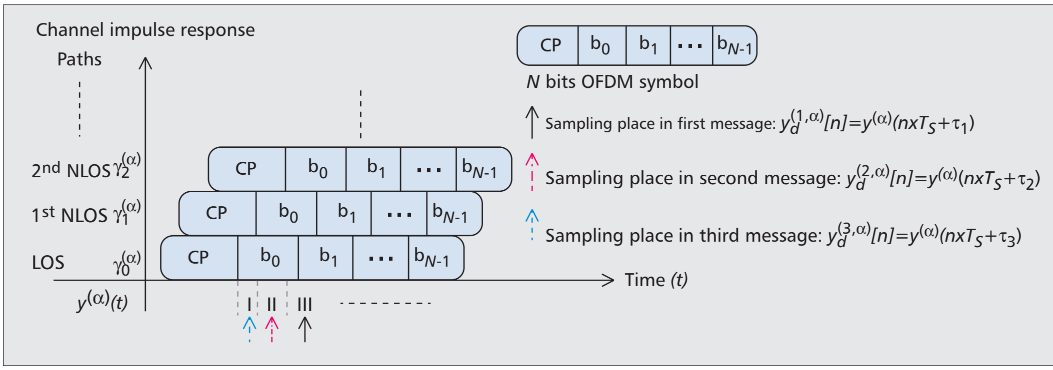

Figure 4. AoA estimation through channel impulse response.

(JADE) method [15]. Although this approach is able to find the AoA with respect to LOS, it requires a many times larger matrix than a MUSIC algorithm for calculation. Moreover, the number of antennas needs to be larger than the number of multipaths. Therefore, the complexity is extremely high.

Since the multipath environment disturbs the AoA measurement, we propose a novel method that can recover from the multipath effect such that AoA can be obtained from the LOS signal. Considering a receiver equipped with N_A antennas, the transmitter sends the predefined message $x(t)$ to the ath antenna of a receiver through wireless multipath channel $h^{(a)}(t) = \gamma_0^{(a)}\delta(t) + \Sigma_{i=1}^{L} \gamma_i^{(a)}\delta(t - \eta_i)$ for $a = 1, 2, \ldots, N_A$. The received signal $y^{(a)}(t)$ can be described as $y^{(a)}(t) = x(t) \otimes h^{(a)}(t)$. We denoted $y_d^{(i,a)}[n]$ as the ith received message after sampling in the ath antenna.

In Fig. 4, the received signal $y^{(a)}(t)$ is the combination of multiple OFDM symbols with each of their delaying times η_i from the multipath effect. The $\gamma_0^{(a)}$ is the channel coefficient of LOS, while $\gamma_1^{(a)}$ is NLOS for $l = 1, 2, \ldots, L$. The complex channel coefficient from LOS among antennas $[\gamma_0^{(1)}, \gamma_0^{(2)}, \ldots, \gamma_0(N_A)]$ indicate the AoA. Hence, if we can obtain channel state information in terms of each ray, we are able to extract the LOS and hence AoA. For OFDM systems, the channel state information can be obtained through channel estimation techniques from pilot assistances of the ith message thus:

$$\hat{h}^{(i,a)}[n] = \mathrm{IFFT}\left\{ H_E\left(\frac{Y_d^{(i,a)}(K)}{X(K)}\right)\right\} \qquad (2)$$

where N is the FFT size, $X(K)$ is the pilot in the K sub-frequency domain, and H_E is the channel estimation approach to estimate the whole frequency response through pilot interpolation. However, the first element in channel impulse response from channel estimation may not be the LOS. The reason is that the sampling positions of a message affect channel estimation. We use Fig. 4 to explain this phenomenon.

When the sampling position is located in zones II and III as $y_d^{(1,a)}$ and $y_d^{(2,a)}$ in Fig. 4, the first ray of channel impulse response as well as LOS will merge and reverse to the last element in $\hat{h}^{(i,a)}[n]$. Thus, it results in $\hat{h}^{(i,a)}[0] \neq \gamma_0^{(a)}$, for $i = 1, 2$. The reason is that the sampling position in zones II and III combines a replica of itself from non-LOS (NLOS), which results in the estimation of channel impulse response not in the right order. In zone I, the sampling position only includes the cyclic prefix (CP) rather than replicas from NLOS, which can be eliminated from channel estimation. Thus, to have LOS in the first element, we can utilize *the nearest ToA message*, the relative sampling position of which is near the arrival time. Hence, by using *the nearest ToA message* in channel estimation, we can obtain channel impulse response with correct order such that $\hat{h}^{(Nearest,a)}[0] = \gamma_0^{(a)}$. Then the LOS angle can be estimated through $\gamma_0^{(j)}$ for $j = 1, 2, \ldots, N_A$ antennas.

The traditional approaches in [10, 15] require a certain number of antennas to overcome the multipath effect. When the amount of multipath increases, more antennas are required to estimate AoA. For example, Quoppa's solution requires 16 antennas with frequency hopping (Bluetooth) to overcome the multipath effect. Our proposed approach does not need to increase the number of antennas. Our minimal requirement for antennas can be only two.

The WiFi-Based Indoor Positioning System

Due to the fact that WiFi technology is widely deployed in indoor environments, we apply our proposed mechanisms to a WiFi system with an access point (AP) with multiple antennas as nearby anchors. The positioning devices could be mobile platforms with WiFi capability such as smartphones and tablets. When there is only one WiFi AP, we apply a hybrid AoA/ToA system to locate the target's positioning. When the number of nearby WiFi APs are two or more, we applied AoA to obtain higher accuracy location. This design can provide accurate positioning service even when the number of nearby anchors is limited.

Since WiFi bandwidth is not as wide as that of UWB, we apply the multiple message approach to assist ToA measurement as well as AoA estimation. In our proposed system, each WiFi AP is equipped with N_A antennas, while the user's mobile phone is equipped with a single antenna only.

DISTANCE MEASUREMENT

In our system, we apply the round-trip time (RTT) approach to obtain distance without requiring time synchronization between transmitters and receivers. In the traditional RTT measuring approach, the transmitter sends the message toward the receiver and records the transmit timestamp. Then, when the receiver responds the message back to the transmitter, the RTT can be calculated from the interval between sending and arrival times. However, this RTT time also includes the receiver processing delay time, and a measurement error occurs. In our scheme, the receiver does not require responding to the message immediately. The transmitter first sends out a burst of messages and records the transmit timestamp of the first message as t_S. Then the receiver uses our proposed ToA approach to measure the arrival time stamp t'_S regarding the first message and allocates the response time started in $t'_S + i \times T_U$, where the T_U is the agreement between transmitter and receiver, and i is the arbitrary number decided by the receiver for allocation convenience. Once the transmitter estimates the first message from receipt with arrival times stamp t_R, the RTT is

$$\text{RTT} = (t_R - t_S) \text{ modulo } T_U$$

One can notice that the ToAs in t'_S and t_R are calculated at each side individually. Hence, the LTI property does not need to hold for the whole RTT estimation process. It only needs to hold for a short time duration while calculating t'_S or t_R. Here, we choose $T_U = N \times T_S$, since each OFDM length is NT_S. Therefore, the maximal measuring distance in a ToA is $NT_S \times C/2$ where C is the speed of light. Take IEEE 802.11n as an example, where bandwidth is 40 MHz and FFT size is 128; the maximal measuring distance under this scenario is up to 480 m. Since WiFi indoor communication distance is usually much less than 480 m, the positioning range from the ToA is quite reliable. By preallocating packet sending in $i \times N \times T_S$, we can reduce the process delay to nearly zero.

ANGLE MEASUREMENT

In our system, the mobile phone sends multiple messages toward the WiFi AP, where the AoA is measured by using the channel estimation technique. The AoA can be obtained by any pair of antennas. When the number of antennas is more than two, the AoA can be jointly estimated for better performance.

LOCALIZATION PROCEDURE

In previous sections we have illustrated how to find out the distance and angle in our system. In the following, we describe the localization procedure:

- **User device**: Requests positioning service of the WiFi AP
- **WiFi AP**: Starts sending requests for RTT measuring
  1. After granting the positioning request, the WiFi AP starts to send burst M messages and records the sending timestamp of first message for later RTT time calculation.
  2. Each message's content is the same and contains multiple subfrequency pilots of an OFDM symbol.
- **User device**: Measures the arrival time and responds with a burst of messages for RTT measurement
  1. The user device reconstructs the received signal by re-ordering the M messages with relative time difference.
  2. The ToA of the first received message can be obtained from the reconstructed signal.
  3. The user device chooses arbitrary i for the sending time $i \times N \times T_S$ to send the burst M messages where each message contains multiple subfrequency pilots.
- **WiFi AP**: Measures the RTT and AoA in terms of user device
  1. The ToA of the first received message can be obtained from the reconstructed signal.
  2. The distance is the RTT multiplied by the light speed divided by 2 where the RTT is the time difference between sending time and received time modulo by $i \times N \times T_S$.
  3. The AoA is measured by using the message nearest the ToA to estimate channel state information.
  4. The WiFi AP returns its own reference location as well as the distance and direction back to the user device.
- **User device**: Uses AoA/ToA to locate its own position and sends requests to other WiFi APs
  1. If the user device only obtains a position from a single WiFi AP, the device calculates its own position by using the WiFi AP reference location with distance and direction angle, as shown in Fig. 1a.
  2. If the user device has positions from more than one WiFi APs, the device only uses AoAs to deduce its own position by combining directions with each WiFi AP, as shown in Fig. 1b.

PERFORMANCE ANALYSIS AND COMPARISONS

We investigate the performance of our proposed system by simulations. We assume sampling rates as well as bandwidth are 20 M/40 MHz, and FFT sizes are 64/128. In addition, we assume each AP has four antennas, while the user device is a smartphone with a single antenna. Among burst messages, the time between them is assumed to be Ts/M where M is the number of burst messages. We consider M = 10/20 messages and 16/32 pilots assistance for FFT size 64/128. We demonstrate our system performance through the IEEE 802.11n TGn channel model with different numbers of messages in terms of ToA/AoA.

In Fig. 5, we compare our approaches with a state-of-the-art MUSIC mechanism where the y-axis represents the distance between AP and device in meters, while the x-axis describes signal strength in dB. Since the MUSIC algorithm only

When SNR is low, the noise factor plays an important role to affect phase measurement. Hence, the time shift property, which refers to the time difference between messages, is not precise, and the performance is degraded.

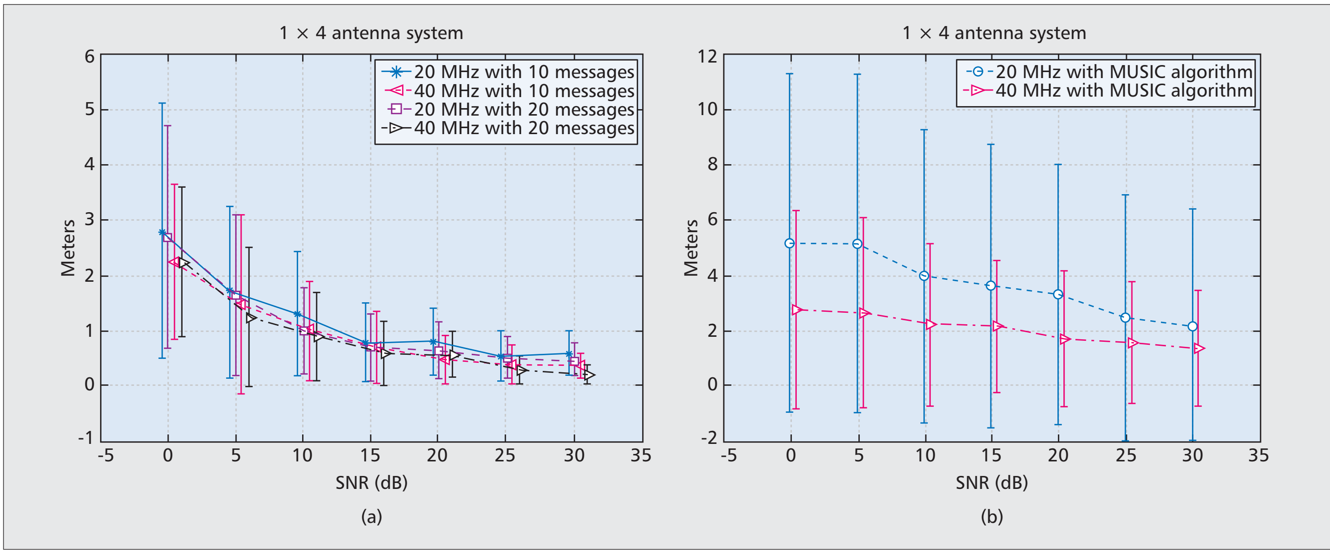

Figure 5. a) Proposed ToA approach performance; b) current state-of-the-art ToA performance (MUSIC algorithm).

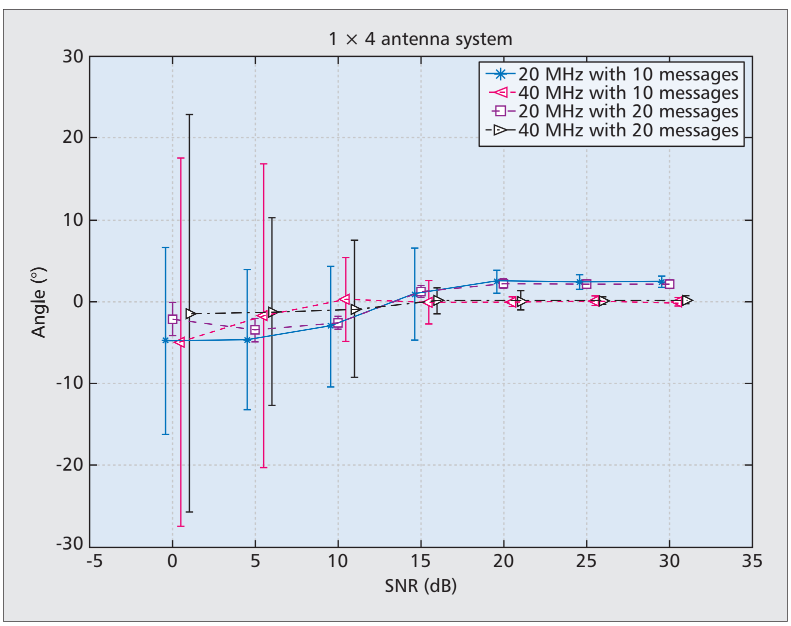

Figure 6. Proposed AoA approach performance.

utilizes one message, the performance is still limited by signal bandwidth. Our proposed solution can provide much better performance through multiple messages. One can notice that 20 messages in 20 MHz bandwidth has similar performance as 10 messages in 40 MHz bandwidth.

For AoA performance in Fig. 6, the angle in the y-axis is calculated by estimating the incoming signal angle with the array of antennas. Figure 6 shows that the AoA performance can be improved by a larger number of messages as well as a wider bandwidth when SNR is high. When SNR is low, the 40 MHz performance is worse than 20 MHz bandwidth. The reason is that FFT size 128 has longer length, and its channel estimation is more sensitive in low SNR, which results in larger errors in AoA. When SNR is higher, the AoA for the message with 32 pilots is much better than the message with 16 pilots.

The positioning error from AoA depends on the distance between the WiFi AP and the mobile device, since the measurement error of angle multiplied by the distance becomes a positioning error. We consider a scenario where the SNR is 20 dB and a WiFi AP's maximal indoor communication distance is 50 m. By utilizing 10 predefined messages, a single WiFi AP can achieve 2.2 and 1 m positioning range for 20 and 40 MHz bandwidth, respectively. When multiple WiFi APs are used, the position performance can reach 2.2 and 0.5 m, respectively. When SNR is low, the noise factor plays an important role to affect phase measurement. Hence, the **time shift property**, which refers to the time difference between messages, is not precise, and the performance is degraded. Nevertheless, our proposed mechanism achieves superior performance under current WiFi AP hardware conditions.

Conclusion

We illustrate different mechanisms and their applications used in indoor localization in terms of cost and complexity. The conventional ToA/AoA approaches to overcome bandwidth limits and numbers of multipaths are introduced and discussed. We show that our proposed approach can improve ToA resolution compared to the super-resolution method under limited signal bandwidth without heavy calculation loading. Moreover, our proposed AoA approach through multiple message assistance can reduce the need for multiple antennas. We demonstrate the performance by applying the mechanism to a WiFi-based indoor positioning system where it can support localization with just a single WiFi AP. When the number of nearby WiFi APs is two or more, the position accuracy can improve further by AoA joint positioning. Finally, simulation showed that our proposed approach provides superior performance without changing hardware settings in a practical WiFi AP setting.

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Biographies

Chouchang Yang (ccjack@uw.edu) received his B.S. and M.S. degrees from National Cheng Kung University, Tainan, Taiwan, R.O.C., in 2004 and 2006, respectively. He is currently working toward his Ph.D. degree at the University of Washington. His current research interests include LTE, next-generation WiFi communications, network communication, RFID, physical layer system design, and anonymous network development.

Huai-Rong Shao [M'99, SM'04] (hr.shao@samsung.com, hrshao@ieee.org) has been leading various research projects since joining Samsung Research America at San Jose, California in 2005. He holds more than 100 patents in the United States and other countries, and has published about 50 peer reviewed top journal/conference papers. He has led the technology development of various international standards such as IEEE 802, Wireless Gigabit Alliance (WiGig), WiFi Alliance (WFA), and ECMA. He serves as the MAC Co-Chair of the IEEE 802.11ah Task Group, and is on the Board of Directors and a Technical Editor for WiGig. Before joining Samsung, he worked as a researcher at Microsoft Research Asia from 1999 to 2001 and a research scientist at Mitsubishi Electric Research Laboratories (MERL) in Boston, Massachusetts, from 2001 to 2005. He received his B.Sc degree in 1994 and Ph.D. degree in 1999 in computer science and technology from Tsinghua University, Beijing, China. His research interests are system and protocol design for big data, cloud computing, software defined network, and Internet of Things; new technology development in wireless and wired communication areas, particularly MAC design for WiFi and cellular; and multimedia communication, particularly compressed/uncompressed video transmission.

Coexistence of Wi-Fi and Heterogeneous Small Cell Networks Sharing Unlicensed Spectrum

Haijun Zhang, Xiaoli Chu, Weisi Guo, and Siyi Wang

Haijun Zhang is with Beijing University of Chemical Technology and the University of British Columbia.

Xiaoli Chu is with the University of Sheffield.

Weisi Guo is with the University of Warwick.

Siyi Wang is with Xi'an Jiaotong-Liverpool University and University of South Australia.

ABSTRACT

As two major players in terrestrial wireless communications, Wi-Fi systems and cellular networks have different origins and have largely evolved separately. Motivated by the exponentially increasing wireless data demand, cellular networks are evolving towards a heterogeneous and small cell network architecture, wherein small cells are expected to provide very high capacity. However, due to the limited licensed spectrum for cellular networks, any effort to achieve capacity growth through network densification will face the challenge of severe inter-cell interference. In view of this, recent standardization developments have started to consider the opportunities for cellular networks to use the unlicensed spectrum bands, including the 2.4 GHz and 5 GHz bands that are currently used by Wi-Fi, Zigbee and some other communication systems. In this article, we look into the coexistence of Wi-Fi and 4G cellular networks sharing the unlicensed spectrum. We introduce a network architecture where small cells use the same unlicensed spectrum that Wi-Fi systems operate in without affecting the performance of Wi-Fi systems. We present an almost blank subframe (ABS) scheme without priority to mitigate the co-channel interference from small cells to Wi-Fi systems, and propose an interference avoidance scheme based on small cells estimating the density of nearby Wi-Fi access points to facilitate their coexistence while sharing the same unlicensed spectrum. Simulation results show that the proposed network architecture and interference avoidance schemes can significantly increase the capacity of 4G heterogeneous cellular networks while maintaining the service quality of Wi-Fi systems.

INTRODUCTION

In recent years, the mobile data usage has grown by 70–200 percent per annum. More worryingly, the bursty nature of wireless data traffic makes traditional network planning for capacity obsolete. Amongst both operators and vendors alike, small cells (e.g., picocells, femtocells and relay nodes) have been considered as a promising solution to improve local capacity in traffic hotspots, thus relieving the burden on overloaded macrocells. A lot of research and development efforts have been made to efficiently offload excess traffic from macrocells to small cells, especially in indoor environments [1].

Due to the scarcity of licensed spectrum for cellular networks, small cells are expected to share the same spectrum with macrocells even when they are deployed within the coverage area of a macrocell [2]. A frequency reuse factor of 1 in 3G HSPA+ and 4G LTE/LTE-A systems has proven to yield high gains in network capacity. If without notable amounts of extra spectrum made available for mobile communications, future cellular networks will unsurprisingly continue to explore aggressive frequency reuse methods. Accordingly, the envisaged large-scale deployment of small cells is likely to be hampered by the potentially severe co-channel interference between small cells and the umbrella macrocell and between neighboring small cells in dense deployment.

In view of this, the wireless industry is examining the efficient utilization of all possible spectrum resources including unlicensed spectrum bands to offer ubiquitous and seamless access to mobile users [3]. The unlicensed 2.4 GHz and 5 GHz bands that Wi-Fi systems operate in have been considered as important candidates to provide extra spectrum resources for cellular networks. The initially targeted 5 GHz unlicensed band has potentially up to 500 MHz of spectrum available. In USA, Korea and China, deploying LTE-A in unlicensed spectrum does not require changes to the existing LTE-A standards (e.g., 3GPP Rel-10). In most other countries, the regulatory requirements of `Listen Before Talk' in unlicensed spectrum mandate standard modifications (e.g., candidates for 3GPP Rel-13).

Nowadays, most mobile devices such as smartphones and tablets support Wi-Fi connectivity, while the proliferation of Wi-Fi access points continues. The Wi-Fi access point density in developed urban areas has reached over 1000 per square km. Widely deployed Wi-Fi systems are playing an increasingly more important role in offloading data traffic from the heavily loaded cellular network, especially in indoor traffic

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hotspots and in poor cellular coverage areas. Very recently, the FCC voted to make 100 MHz of spectrum in the 5 GHz band available for unlicensed Wi-Fi use, giving carriers and operators more opportunities to push data traffic to Wi-Fi. Wi-Fi access points have even been regarded as a distinct tier of small cells in heterogenous cellular networks. However, since Wi-Fi systems are wireless local area networks (WLANs) based on the IEEE 802.11 standards, they have usually been designed and deployed independently from the cellular networks. Now that the wireless industry is seeking to explore the unlicensed spectrum currently used by Wi-Fi systems for LTE/LTE-A and future cellular networks' usage as well, the coexistence and interworking of Wi-Fi and heterogeneous cellular networks become an area requiring extensive research and investments. The joint deployment of Wi-Fi and cellular networks in the unlicensed spectrum can increase the overall capacity of a heterogeneous network, provided that the mutual interference between Wi-Fi and cellular systems is properly managed so that both can harmoniously coexist.

Benefits promised by the coexistence of Wi-Fi and cellular networks in unlicensed spectrum have started to attract interest from the research community. In [4], the authors proposed a quality of service (QoS) based strategy to split the unlicensed spectrum between Wi-Fi and femtocell networks. Although the unlicensed spectrum splitting scheme considers fairness between Wi-Fi access points and femtocells, the split use of the spectrum between two systems prohibits a high cross-network throughput. In [5], the authors investigated the deployment of a heterogeneous vehicular wireless network consisting of IEEE 802.11b/g and IEEE 802.11e inside a tunnel for surveillance applications, and specifically evaluated the handover performance of the hybrid Wi-Fi/WiMAX vehicular network in an emergency situation. In [9], time-domain resource partitioning based on the use of almost blank subframes (ABSs) was proposed for LTE networks to share the unlicensed spectrum with Wi-Fi systems. Qualcomm has recently proposed to deploy LTE-A in the unlicensed 5 GHz band currently used mostly by Wi-Fi. The main idea is to deploy LTE-A as supplemental downlink (SDL) in the 5725–5850 MHz band in USA, with the primary cell always operating in the licensed band. Verizon and Ericsson are also exploring similar ideas. Huawei and CMCC have investigated the availability, commonality and feasibility of integrating the unlicensed spectrum to International Mobile Telecommunications-Advanced (IMT-A) cellular networks [3]. LTE-Unlicensed (LTE-U) was first proposed by Qualcomm and Ericsson as a technology to run LTE in unlicensed spectrum in congested areas. Since February 2014, NTT DoCoMo and Huawei have been researching LTE-U, which they refer to as Licensed-Assisted Access using LTE (LAA-LTE). They have demonstrated on pre-commercial multi-cell networks that LAA-LTE achieves better coverage and capacity in the 5 GHz unlicensed spectrum than Wi-Fi alone. However, there are still concerns that LTE-U may completely take over the Wi-Fi bands in dense deployments.

It is worth noting that technical issues related to the coexistence of Wi-Fi and heterogeneous cellular networks in unlicensed spectrum, such as efficient spectrum sharing and interference mitigation, have not been sufficiently addressed. In [10], the authors proposed an integrated architecture exploiting the opportunistic networking paradigm to migrate data traffic from cellular networks to metropolitan Wi-Fi access points. In [10], Bennis *et al.* introduced the basic building blocks of cross-system learning and provided preliminary performance evaluation in an LTE simulator overlaid with Wi-Fi hotspots. For the unlicensed spectrum sharing deployment of Wi-Fi and LTE-A systems, the co-channel interference between Wi-Fi tier and LTE-A tier can be mitigated by using ABSs, in which the interfering tier is not allowed to transmit data, and the victim tier can thus get a chance to schedule transmissions in the ABSs with reduced cross-tier interference [12]. Moreover, it has been shown that by estimating the number of co-channel transmitters and knowing the deployment density of network nodes in a region, the average channel quality at any point in a coverage area can be inferred [13].

In this article, we present a network architecture to support the coexistence of Wi-Fi and heterogeneous cellular networks sharing the unlicensed spectrum. Based on the network architecture, we first provide an in-depth review of the ABS mechanism used for mitigating the co-channel interference from small cells to Wi-Fi systems and present a spectrum-sensing based fair ABS scheme without priority. We then propose an interference avoidance scheme based on small cells estimating the density of nearby Wi-Fi transmissions to facilitate the unlicensed spectrum sharing between small cells and Wi-Fi access points. Simulation results are provided to evaluate the performance of the proposed network architecture and interference avoidance scheme in facilitating coexistence of Wi-Fi and 4G heterogeneous cellular networks in unlicensed spectrum.

In recent years, regulatory bodies are considering the possible coexistence of multiple disparate radio access technologies (RATs) on the same frequency band. This includes both the licensed bands (e.g., TV spectrum) and unlicensed bands (e.g., amateur spectrum).

Heterogeneous Network Architecture

Figure 1 shows the network architecture where several Wi-Fi access points and small cells coexist in the coverage area of a macrocell. The macrocell, small cells, and Wi-Fi access points share the same unlicensed spectrum for providing radio access to users, millimeter-wave radio is used for small-cell backhaul links, and device-to-device (D2D) communications are supported based on Wi-Fi Direct or LTE Direct. As shown in Fig. 1, the control plane (C-plane) and user plane (U-plane) are split on the radio links associated with small cells. Specifically, the C-plane of user equipments (UEs) associated with a small cell is provided by the macro evolved Node B (eNB) in a low frequency band, while the U-plane of UEs associated with a small cell is provided by their serving small cell in a high frequency band. For UEs associated with the macrocell, both the C-plane and U-plane of their radio links are provided by the serving

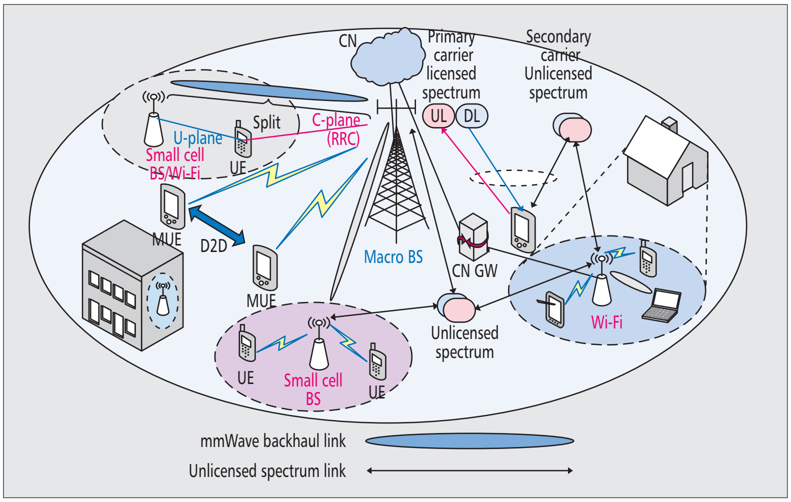

Figure 1. Network architecture of LTE-A and wi-fi coexistence.

macrocell. Since the C-plane of small-cell UEs is managed by the macrocell, the radio resource control (RRC) signallings of small-cell UEs are transmitted from the macrocell, and the handover signalling overhead between small cells and the macrocell can be much reduced [6]. Regarding Wi-Fi access points as a type of small cells, the C-plane and U-plane split can be applied in a Wi-Fi/Macrocell scenario, where the C-plane of UEs associated with a Wi-Fi access point is provided by the macro eNB in a low licensed frequency band, while the U-plane of UEs associated with a Wi-Fi access point is provided by their serving Wi-Fi access point in a high unlicensed frequency band. The interworking of Wi-Fi and cellular networks benefits from the split of C-plane and U-plane in terms of mobility robustness, service continuity, reduction in cell-planning efforts, energy efficiency, etc.

When considering the backhaul issues of small cells, expensive wired backhaul links may not always be feasible, especially for the dense deployment of small cells. In the meanwhile, in-band wireless backhaul solutions using the licensed spectrum may not be feasible either, because of the scarcity of the licensed spectrum [7]. Recently, the millimeter-wave bands, such as the unlicensed 60 GHz band and the low interference licensed 70 GHz and 80 GHz bands, have been considered as promising candidates for the small cell backhaul solution. This is motivated by the huge frequency bandwidth (globally harmonised over more than 6 GHz millimeter-wave spectrum) that can be exploited, and the spatial isolation supported by highly directional beams. At the same time, the new IEEE 802.11ad standard, a.k.a. WiGig, uses the unlicensed 60 GHz millimeter-wave band to deliver data rates of up to 7 Gb/s. This adds a large amount of new frequency bandwidth to existing Wi-Fi products, such as IEEE 802.11n operating in the 2.4 GHz and 5 GHz bands, and IEEE 802.11ac operating in the 5 GHz band.

For providing radio access, the macrocell, small cells and Wi-Fi access points share the unlicensed spectrum. The network architecture in Fig. 1 integrates the coexistence of Wi-Fi and cellular networks and facilitates smart management of data traffic in mobile operators' networks. For instance, the data traffic could be dynamically routed to the optimal radio interface for a particular application and user, with network congestion, reliability, security, and connectivity cost taken into account. As can be seen from Fig. 1, the primary carrier always uses licensed spectrum to transmit control signaling, user data and mobility signalling, while the secondary carrier(s) use unlicensed spectrum to transmit best-effort user data in the downlink and potentially the uplink.

Handover Procedure between Wi-Fi and LTE/LTE-A

Seamless mobility is one of the key aspects of interworking between Wi-Fi and LTE/LTE-A systems. During the handover process, there should be no package loss or radio link failure in order to ensure the user's QoS. In current Wi-Fi systems, interworking between Wi-Fi and LTE/LTE-A systems is not supported, although it is badly needed due to users' frequent mobility between the coverage areas of Wi-Fi access points and cellular networks. In 3GPP Rel-11, trusted WLAN access to the Enhanced Packet Core (EPC) is based on S2a-based Mobility over GPRS Tunnelling Protocol (SaMOG), which is enhanced in 3GPP Rel-12 to provide traffic steering and mobility between LTE-A and Wi-Fi networks and to optimize the use of network resources. The 3GPP standard TS 23.401 describes seamless and non-seamless handover solutions between 3GPP and non-3GPP access networks. These standards enable users to continue using data services when they pass across macrocells, small cells and Wi-Fi hotspots. In the network architecture shown in Fig. 1, a UE can handover between Wi-Fi and cellular networks through the core network (CN) gateway (GW). In Fig. 2, the CN GW based handover procedure between a source Wi-Fi (macrocell) and a target macrocell (Wi-Fi) is given. Since the C-plane of Wi-Fi users is managed by the macrocell, the RRC signallings of Wi-Fi users are transmitted from the macrocell, and the handover signalling overhead between the Wi-Fi access point and the macrocell can be much reduced.

Almost Blank Subframes Allocation

Coexistence without Priority

In recent years, regulatory bodies are considering the possible coexistence of multiple disparate radio access technologies (RATs) on the same frequency band. This includes both the licensed bands (e.g., TV spectrum) and unlicensed bands (e.g., amateur spectrum). In countries such as the USA, Canada, and the UK, regulatory efforts are being made to permit the operation of white spaces devices (WSDs). For example, the IEEE 802.22 fixed point-to-point cognitive radio transmissions in TV white spaces, and more recently the IEEE 802.16h wireless broadband protocols.

In the licensed bands, there is a clear notion of the primary and secondary users, whereby spectrum sensing techniques are employed by secondary users to avoid causing interference to primary users. This can be achieved by identifying primary transmissions using spectrum sensing or geo-location database operations. However, there is a lack of research activities examining how secondary users associated with different RATs can avoid or mitigate co-channel interference to each other. In the unlicensed bands without the concepts of primary and secondary users, a similar challenge exists between different RATs sharing the same spectrum.

In this article, we refer to the coexistence of two RATs without priority ranking as *coexistence without priority*. More specifically, we focus on allowing cellular communications to co-exist with Wi-Fi communications on an equal basis, i.e., no discrimination between primary and secondary users. What is new here is the coexistence of two disparate RATs that were not designed to be in coexistence, together with the impact of this on the interference map. Whilst the coexistence of contention based systems have been explored (e.g., IEEE 802.11 and IEEE 802.15 systems) [8], the coexistence of a non-contention system (LTE) with a contention system (Wi-Fi) is not well explored, especially when no priority ranking between them is given. In fact, a reasonable suspicion is that the allocation based transmission protocols of LTE may completely block the collision based protocols of Wi-Fi. Coupled with the growing density of small cells, this lack of interpretability on the same spectrum band can cause severe capacity issues.

In multiple RAT coexistence, communication protocols can operate in either their default normal mode or a coexistence mode. The latter is triggered when another RAT is sensed nearby and action is needed. Coexistence mechanisms can be divided into two groups:
- Those that require message exchange between nodes or RATs
- Those that do not

In general, cross-RAT coordination is difficult due to the disparate protocol development processes and vendor differences. Therefore, in the following sub-section we will review a non-collaborative coexistence mechanism that allows LTE to co-exist with Wi-Fi in the unlicensed spectrum.

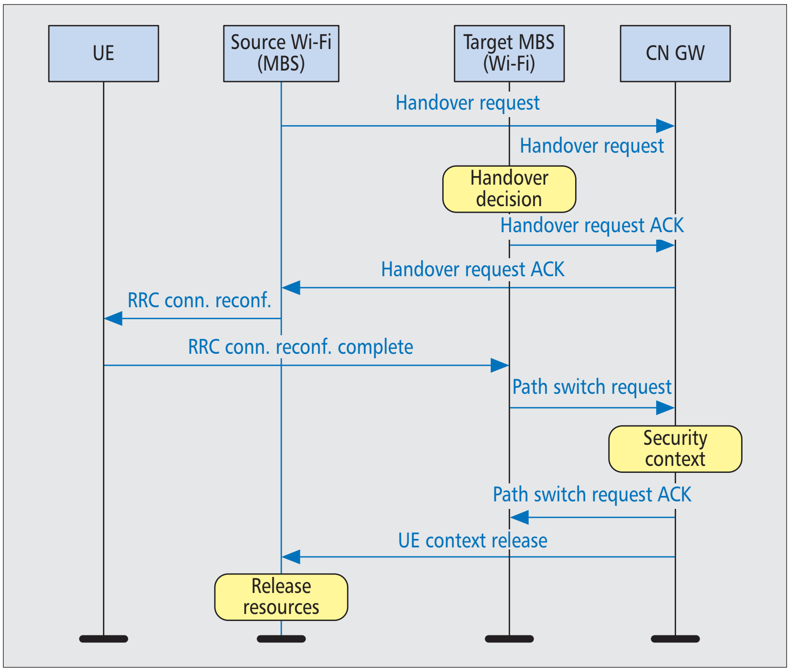

Figure 2. Handover procedure between Wi-Fi and LTE/LTE-A.

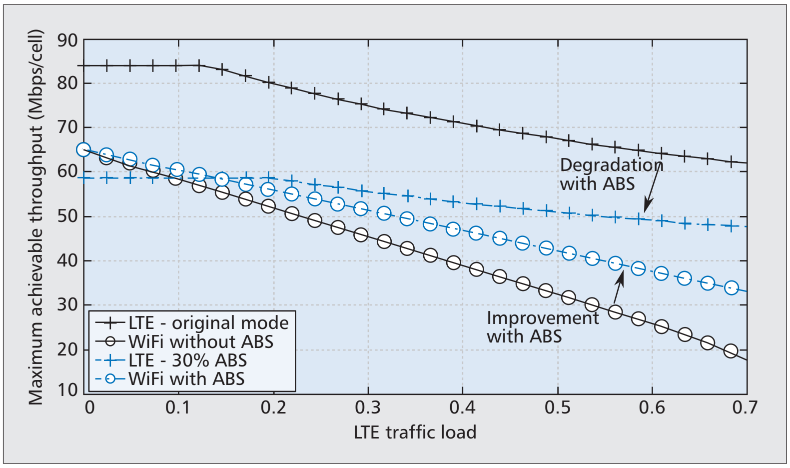

Figure 3. Maximum achievable throughput (Mb/s per cell) for an LTE or Wi-Fi cell under different normalized traffic loads (normalized to cell capacity), with LTE operating with: a) original mode, and b) ABS.

Random Almost Blank Subframe Allocation

In [9], autonomous (without coordination) coexistence between LTE and Wi-Fi was achieved by LTE transmitting ABSs under a 3GPP Rel-10 time-division-duplex (TDD) scheme. The ABSs are subframes with reduced power or content. They are backwards compatible with 3GPP Rel-8 and Rel-9 in that several synchronization channels remain (e.g., common reference signals). For interference avoidance between cells of the same RAT, ABSs are triggered by coordination messages between eNBs via the X2 interface. The frequency of ABS transmissions can be adapted to the time-varying interference environment. For interference avoidance between cells of different RATs, coordination messaging between cells of different RATs is challenging for the previously mentioned reasons. Therefore, ABSs are transmitted randomly at some rate without coordination and without the need for backwards compatibility with previous releases on the unlicensed bands [9]. The central conceit to this idea is that during the random ABSs, the Wi-Fi access points can detect the channel vacancy and transmit following its contention based protocol. Accordingly, the allocation nature of LTE transmissions can be suppressed in a way that avoids coordination or spectrum sensing, but at the cost of decreased LTE spectral efficiency and network capacity.

In Fig. 3, the maximum achievable throughput for an LTE or Wi-Fi (IEEE 802.11n) cell under different normalized traffic loads are plotted, with LTE operating either under the original mode or with 30 percent of the subframes randomly selected as ABSs. The parameters

Parameter	Value
LTE Carrier Frequency	2100 MHz
Wi-Fi Carrier Frequency	2400 MHz
LTE Cell Density	3 per km2
Wi-Fi Density	300 per km2
LTE Cell Transmit Power	40 W
Wi-Fi Transmit Power	1 W
Pathloss	3GPP Urban Micro
Peak LTE Throughput	84 Mb/s (64 QAM SISO)
Peak IEEE 802.11n Throughput	65 Mb/s (64 QAM SISO)

Table 1. Simulation parameters.

used in the simulation can be found in Table 1. The results show that under the original mode, LTE cell capacity saturates at around 84 Mb/s due to discrete modulation and coding schemes employed (64QAM with Turbo coding), while Wi-Fi cell capacity saturates at 64 Mb/s. As the LTE traffic load increases, the capacity of all cells falls due to increased radio resource usage and the resulting increased interference. LTE cell capacity decays slower than Wi-Fi capacity. The *super-linear* degradation of Wi-Fi capacity may finally lead to 0 Mb/s. By employing 30 percent ABSs in LTE, the Wi-Fi capacity is improved significantly at high LTE traffic loads, while the LTE cell capacity falls by 10–24 Mb/s. Note that the capacity degradation rates for both LTE and Wi-Fi become slower with LTE ABS transmissions, because random ABSs mitigate both cross-tier and co-tier inter-cell interference. It is worth noting that the overall aggregate capacity of LTE and Wi-Fi is actually reduced with ABS, indicating that the random ABS mechanism benefits fairness instead of overall capacity.

In summary, ABSs can be transmitted randomly by LTE transmitters to allow the spectrum-sharing coexistence of allocation-based LTE transmissions and contention-based Wi-Fi transmissions with an improved fairness between them, but at the cost of decreased overall aggregate capacity of LTE and Wi-Fi networks.

Interference Avoidance with Neighborhood Wi-Fi Density Estimation

Inference Framework

It has been shown that avoiding co-channel interference in a network with a high interference intensity can improve the long-term system throughput [12]. However, coordinating interference avoidance on the radio resource management (RRM) level typically requires a large volume of coordination information exchanged between multiple base stations (BSs) via the X2 interface. Specifically, each BS in an OFDMA system needs to know whether its neighboring BSs are transmitting on each available radio resource block. This level of coordination taxes the *backhaul capacity*, while any *delay* in information sharing may cause the interference avoidance performance to falter.

In [13], an interference estimation technique that does not require information sharing between BSs or UEs was devised based on each BS sensing the spectrum and estimating the number of co-channel transmissions in a defined observation zone. By estimating the number of co-channel transmitters and knowing the cell density in the region, the average channel quality at *any random point* in a coverage area can be inferred. As the expressions are tractable, the computational complexity is extremely low. The methodology can be applied to a K-tier heterogeneous network by leveraging a stochastic geometry framework and an opportunistic interference reduction scheme, which was shown to approach the interference estimation accuracy achievable by information exchange on the X2 interface [13].

The inference framework assumes that each cell is equipped with a spectrum sensing device.[1] On each frequency band f, the sensor at each cell (located at distance h from the BS) is able to detect the power density P_f from all co-channel transmitters in an unbounded region. Given knowledge of the spatial distribution of co-channel cell deployments [15], the density of co-channel transmissions λ_f can be inferred from the P_f measurements [13]:

$$\lambda_f \propto \frac{\sqrt{P_f / P}}{Q(h,\alpha)} \tag{1}$$

where α is the pathloss distance exponent, P is the average transmit power of the BSs, and the function $Q(h, \alpha)$ is given in [13]. Without loss of generality, this inference framework can be applied to a K-tier heterogeneous network comprised of macrocells, femtocells and Wi-Fi access points. Figure 4 illustrates how a femtocell infers the number of co-channel transmitters in a 3-tier heterogeneous network by sensing the received power spectrum. In this illustrative example, the estimated transmitter activities on the considered frequency band are: 50 percent of LTE macrocells, 100 percent of LTE femtocells, and 25 percent of Wi-Fi access points. This spectrum sensing mechanism is not able to know which cells are transmitting, but it provides a statistical notion for a BS to infer the channel quality of a served user.

Based on the inferred density of co-channel transmissions λ_f in the vicinity, the signal-to-interference ratio (SIR) on frequency band f at distance d away from the sensing BS is estimated as:

$$SIR_{f,d} \propto P_f^{-1}\left[\frac{Q(h,\alpha)}{Q(d,\alpha)}\right]^2, \tag{2}$$

where the constant of proportionality is the received signal strength.

[1] Low cost spectrum sensing equipment for 2–5GHz is now readily available.

Simulation Results

Figure 5 shows the simulation results of peak cell capacity versus normalized cell traffic load for a variety of static and dynamic interference mitigation schemes [13, 14]. The baseline is a hard frequency reuse 1 (HFR1) scheme, which shows a peak capacity of 58 Mb/s/cell when the cells are unloaded (i.e., minimum inter-cell interference). This value falls steadily to 40 Mb/s/cell for fully loaded cells without interference mitigation (i.e., maximum inter-cell interference). A similar trend exists for HFR3. The soft frequency reuse with a power backoff factor 0.5 (SFR P0.5) performs better than the previous two schemes. We can see from Fig. 5 that the TDD-based sequential game coordinated (SGC) interference avoidance scheme achieves a much higher peak cell capacity than the HFR and SFR schemes at low and medium cell traffic loads. The uncoordinated interference avoidance scheme proposed in [13] provides the highest peak cell capacity at high traffic loads and achieves over 90 percent the peak cell capacity of the SGC interference avoidance scheme that requires channel state information.

Discussion and Challenges

The disadvantage with a TDD-based coordinated interference avoidance scheme is the need for two prerequisites [14]:

- Cell pairing or clustering
- Static or dynamic assignment of cell priority

Effective cell pairing often involves the association of cells that are dominant interferers to each other. However, this may not always be the case. For example, the antenna bore-sight of BS *A* is pointing at BS *B*, but the antenna bore-sight of BS *B* is pointing at a direction away from BS *A*. Alternatively, two cells that are closest to each other will be paired together. Cell priority assignment refers to the process of assigning different transmission priorities to cells. Random access, traffic weighted, and QoS weighted cell priority assignments have been considered in the literature.

Conclusion

In this article, we have looked into the potentials and challenges associated with coexisting Wi-Fi systems and heterogeneous cellular networks sharing the unlicensed spectrum. We have introduced the network architecture for LTE/LTE-A small cells to exploit the unlicensed spectrum already used by Wi-Fi systems. The ABS mechanism and an interference avoidance scheme have been presented to mitigate the interference between Wi-Fi and LTE/LTE-A systems when both transmitting in the same unlicensed spectrum. Simulation results have shown that with the proper use of ABS mechanism and interference avoidance schemes, heterogeneous and small cell networks can improve their capacity by using the unlicensed spectrum used by Wi-Fi systems without affecting the performance of Wi-Fi.

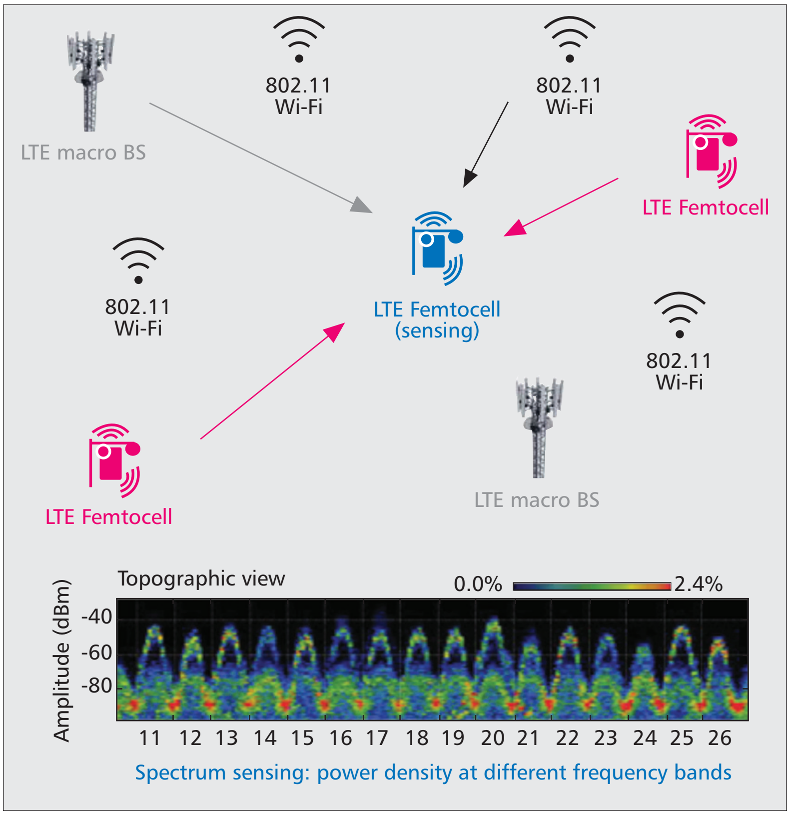

Figure 4. Illustration of a femtocell inferring the number of co-channel transmitters from a 3-tier cellular and Wi-Fi heterogeneous network by sensing the received power spectrum [13].

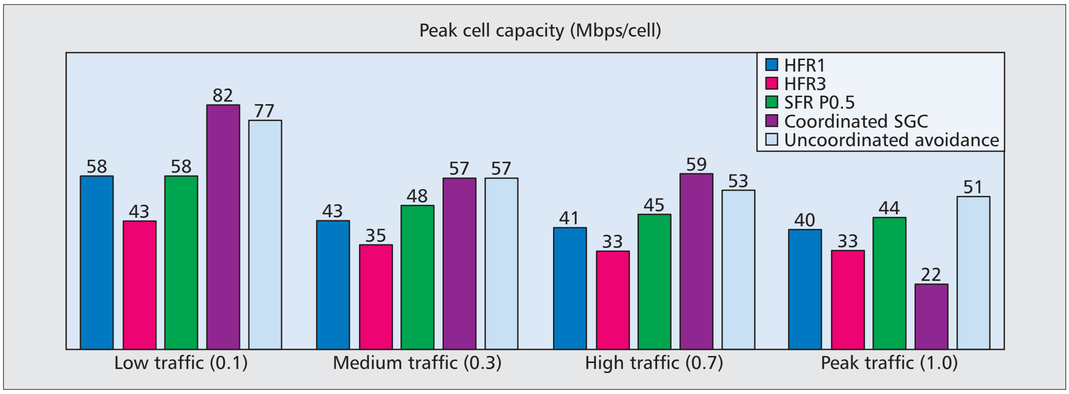

Figure 5. Plot of peak cell capacity versus different normalized cell traffic loads for a variety of static and dynamic interference mitigation schemes [13, 14].

Simulation results have shown that with the proper use of ABS mechanism and interference avoidance schemes, heterogeneous and small cell networks can improve their capacity by using the unlicensed spectrum used by Wi-Fi systems without affecting the performance of Wi-Fi.

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Biographies

Haijun Zhang [M'13] (dr.haijun.zhang@ieee.org) received his Ph.D. degree in School of Information and Communication Engineering (SICE), Beijing Key Laboratory of Network System Architecture and Convergence, Beijing University of Posts Telecommunications (BUPT). Currently, he is an Associate Professor in College of Information Science and Technology, Beijing University of Chemical Technology. He is also a Postdoctoral Research Fellow in Department of Electrical and Computer Engineering, the University of British Columbia (UBC). From September 2011 to September 2012, he visited Centre for Telecommunications Research, King's College London, London, UK, as a joint Ph.D. student and Visiting Research Associate. He has published more than 50 papers and authored 2 books. He served as Symposium Chair of the 2014 International Conference on Game Theory for Networks (GAMENETS'14) and Track Chair of 15th IEEE International Conference on Scalable Computing and Communications (ScalCom2015). His current research interests include Wireless Resource Allocation, 5G, Heterogeneous Small Cell Networks and Ultra-Dense Networks.

Xiaoli Chu [M'05] (x.chu@sheffield.ac.uk) is a Lecturer in the Department of Electronic and Electrical Engineering at the University of Sheffield, UK. She received the B.Eng. degree in Electronic and Information Engineering from Xi'an Jiao Tong University in 2001 and the Ph.D. degree in Electrical and Electronic Engineering from the Hong Kong University of Science and Technology in 2005. From Sep. 2005 to Apr. 2012, she was with the Centre for Telecommunications Research at King's College London. She is the leading editor/author of the book Heterogeneous Cellular Networks — Theory, Simulation and Deployment, Cambridge University Press, May 2013. She is Guest Editor of the Special Section on Green Mobile Multimedia Communications for IEEE Transactions on Vehicular Technology (Jun. 2014) and the Special Issue on Cooperative Femtocell Networks for ACM/Springer Journal of Mobile Networks & Applications (Oct. 2012). She is Co-Chair of Wireless Communications Symposium for the IEEE International Conference on Communications 2015 (ICC'15), was Workshop Co-Chair for the IEEE International Conference on Green Computing and Communications 2013 (GreenCom'13), and has been Technical Program Committee Co-Chair of several workshops on heterogeneous networks for IEEE GLOBECOM, WCNC, PIMRC, etc.

Weisi Guo [M'11] (weisi.guo@warwick.ac.uk) received his M.Eng., M.A. and Ph.D. degrees from the University of Cambridge. He is currently an Assistant Professor and Co-Director of Cities research theme at the School of Engineering, University of Warwick. In recent years, he has published over 50 IEEE papers and won a number of awards and grants. He is the author of the VCEsim LTE System Simulator, and his research interests are in the areas of heterogeneous networks, self-organization, energy-efficiency, and molecular communications.

Siyi Wang [M'13] (siyi.wang@xjtlu.edu.cn) received his Ph.D. degree in wireless communications from the University of Sheffield, UK in 2014. He was a research member of the project "Core 5 Green Radio" funded by Virtual Centre of Excellence (VCE) and UK EPSRC. He is currently a lecturer at Xi'an Jiaotong-Liverpool University (XJTLU). He has led a team and won the second prize of the IEEE Communications Society Student Competition "Communications Technology Changing the World". He has published over 25 IEEE papers in the past 3 years in the area of 4G cellular networks, molecular communications and mobile sensing, and won an IEEE best conference paper award. His research interests include: molecular communications, indoor-outdoor network interaction, small cell deployment, device-to-device (D2D) communications, machine learning, stochastic geometry, theoretical frameworks for complex networks and urban informatics.

Enhanced Capacity and Coverage by Wi-Fi LTE Integration

Jonathan Ling, Satish Kanugovi, Subramanian Vasudevan, A Krishna Pramod

The authors are with Alcatel Lucent.

ABSTRACT

Wi-Fi provides cost-effective data capacity at hotspots in conjunction with broadband cellular networks. The hotspots are required to capture a large number of users and provide high data rates. Data rates, over the Wi-Fi interface, are influenced by the media access protocol, which loses throughput due to contention based protocol delays and unintended collisions when a large number of users are active. The hotspot range that determines the number of users that can associate is limited by the lower power of the client rather than the access point. By diverting the traffic destined to the access point via another access network, both range and efficiency can be improved. This uplink redirection or diversion is achieved by simultaneous use of the Wi-Fi and LTE radio interfaces. Three options, loose, tight, and hybrid integration, are presented to providing enhanced capacity and coverage.

INTRODUCTION

Large scale adoption of smart phones and tablets means that wireless data networks must provide high data rates anytime and anywhere. Wi-Fi networks are characterized by high peak rates, but lower efficiency for small packets, and by limited coverage. On the other hand, 3GPP cellular LTE networks are characterized by ubiquitous coverage and high spectral efficiency. LTE small cells are beginning to be deployed as a complement to LTE macro cells to address the need for higher capacity. To utilize unlicensed spectrum allocated at 2.4 GHz and 5 GHz, the LTE network may also be augmented with Wi-Fi access points (APs) co-located with the LTE small cells or deployed separately in their neighborhood. Optimization by co-location allows sharing backhaul and reduces operational costs. Optimizing the data traffic over the cellular and Wi-Fi interfaces also provides benefits. For example, serving best effort traffic over the Wi-Fi interface and QoS sensitive traffic over the LTE interface enhances user experience in both traffic classes by leveraging the large bandwidth available on the Wi-Fi link and by freeing up resources on the LTE link.

In Wi-Fi/LTE heterogeneous networks, current schemes for integration [1] allow users access to one or the other, or concurrent use of both radio interfaces, but for different applications [1, 2]. For example, a web browsing user may be switched from LTE to Wi-Fi access when Wi-Fi becomes available.

The switching between access networks is currently managed by the device independently or assisted by policies given by the cellular service provider. Policies are stored in an access network discovery and selection function (ANDSF), in the core network and communicated to the client's device. The client device is responsible for execution of the policy, i.e. selection of the appropriate air interface. Standardization efforts are ongoing in 3GPP to further supplement ANDSF based policies with radio access network (RAN) information [3]. Figure 1 describes the evolution of these mechanisms, from policy selection based on UE, to selection augmented by RAN signaling, and then to the simultaneous use of both interfaces by an application in order to further reduce service delays and consistently maintain user experience. In this article we describe three solutions that utilize Wi-Fi and LTE access interfaces simultaneously, providing gains beyond the federation bandwidth. This has been called super-aggregation, meaning that the benefits of the combined system are greater than the sum of their components [4].

Experience on the Wi-Fi network can be uneven due to excessive loading [5–7]. Due to contention based access and the imperfect nature of radio channel sensing, packet collisions can occur between any of the co-channel transmitters: between APs transmitting to multiple clients on the downlink and the clients transmitting to the one of the APs in the uplink; between the multiple clients themselves; and between neighboring APs. The overall coverage of Wi-Fi is limited by the relatively low transmit power of the client, so a mobile user will easily move in and out of hotspot coverage. Therefore, capacity of a Wi-Fi system can be improved by redirecting uplink traffic from the client devices that are capable to the LTE radio interface, thus removing the source of contention, collisions, and range limitation.

In the sections that follow as both background and motivation, Wi-Fi capacity and coverage are discussed. We present three levels of network integration:

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The LTE small cell provides higher data rates to users than the LTE macrocell because of better channel quality at the receiver due to the proximity of the user to the small cell. The Wi-Fi cell provides additional capacity but has limited range over which users receive benefits.

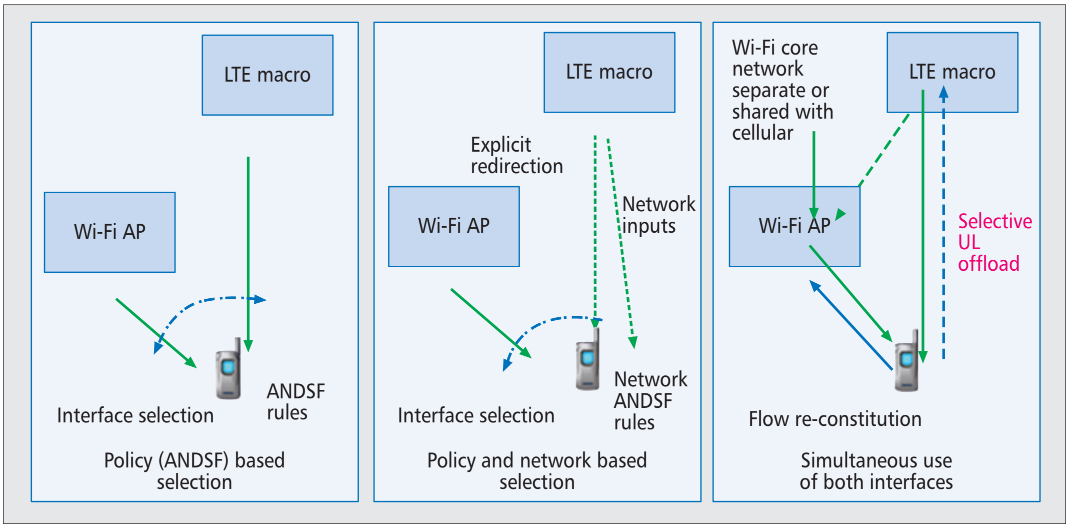

Figure 1. Evolution of Wi-Fi LTE integration schemes.

(1) Loose integration using a high layer control protocol that interacts with the application and both radio interfaces.
(2) Tight integration with Wi-Fi and where the control and aggregation function is part of the LTE radio link controller.
(3) Hybrid integration where the LTE network provides an alternative route back to the Wi-Fi AP.

Proposal (1) is a modification of the IETF MPTCP standard. Proposals (2) and (3) are expected to be new proposals to 3GPP. To complete the discussion, we offer thoughts on system optimization and concluding remarks.

Wi-Fi Capacity

Uplink and downlink transmissions in Wi-Fi time-share the wireless channel using a polite media access control (MAC) protocol called Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) [9]. According to the protocol, each potential transmitter, either an access point or client, must defer transmission until the shared channel is deemed to be clear. CSMA/CA prevents collisions and allows Wi-Fi to be robust and scalable for large planned and unplanned/ad-hoc deployments.

However, CSMA/CA has been reported to be the cause of performance degradation for large numbers of users and dense deployments [5, 7, 10]. We have observed similar trends in simulations for increasing the number of clients. Using the event driven simulator NS-3, experiments were run where clients attempt to send as much traffic as possible on both uplink and downlink. That is, there is always data available to send. As the number of clients is increased the total throughput degrades due to packet collisions, as shown in Fig. 2. The simulator modeled 802.11a clients at a fixed radio transmission rate of 54 Mbps, and was modified from capturing packets on a first-come first serve basis, to a realistic model that captures packets based on signal-to-noise and interference ratio (SINR). In the experiment we also varied the contention window size and the maximum number of packet retries. Increasing the contention window increases the overhead but reduces the probability of collision. Increasing the number of retries reduces the probability that a packet will be lost, and thus need to be transmitted by a higher layer protocol. Increasing the contention window resulted in lower throughput for a small number of clients and an improvement at a high number of clients with high collisions. Adjustments to both parameters, however, do not truly mitigate the drop in throughput, especially when one considers single client throughput as the baseline.

There are also efforts within the Wi-Fi standards groups to enhance throughput and performance. The IEEE standard 802.11ac provides higher bandwidths to individual clients by bonding across available channels [11]. Since channels are taken away from other users, 802.11ac provides full benefits in sparse networks. Meanwhile, the IEEE 802.11 Task Group is addressing dense networks. The Dual Wi-Fi [6] proposal carves out blocks of spectrum to separate uplink and downlink from the common Wi-Fi channel based on relative uplink/downlink load. While this solves the issue of contention between uplink and downlink and enhances spectral efficiency, it reduces the available bandwidth for either of the links since a portion of it has to be avoided for exclusive use by the other link.

Wi-Fi Coverage

Consider a heterogeneous network deployment where macro cell coverage is provided by LTE, and an embedded small cell provides service using both LTE and Wi-Fi. The LTE coverage represents a wide area across which capacity can be distributed among a large number of users. The LTE small cell provides higher data rates to users than the LTE macrocell because of better channel quality at the receiver due to the proximity of the user to the small cell. The Wi-Fi cell provides additional capacity but has limited range over which users receive benefits.

Wi-Fi range is usually determined by the client with lower uplink transmit power, rather than the AP. Clients typically transmit at 40 mW due to cost and safety reasons, whereas APs often have the capability to transmit up to the maximum regulatory power, e.g. 4W EIRP at 2.4 GHz in the United States. Given the higher downlink power, and if not constrained by the uplink, range can be increased by about 3× and area coverage by 10× according to typical distance dependence of inverse 4th power. Note that for cellular networks such large asymmetry in UL and DL range is typically not present.

Range has direct impact on a large network's deployment costs. Indoor APs have lower cost because the building or home owner provides the backhaul and electricity, and the APs themselves have no need for environmental hardening. Radio signals are strongly attenuated by exterior walls, and thus signals from indoor transmitters are received weakly outdoors [12]. Thus for full Wi-Fi coverage, separate indoor and outdoor APs must be installed [13].

By using a higher power air interface for the uplink, and if Wi-Fi were used in one-way or downlink only mode, then Wi-Fi range could be increased. A lower cost Wi-Fi based network with coverage extending to outdoors using indoor APs, or vice-versa, would be possible. We describe the architecture to achieve this in the upcoming section on integration.

Integration of Wi-Fi and LTE

We present solutions to enhance efficiency and range of Wi-Fi, by redirecting uplink traffic from the Wi-Fi network interface to the LTE network interface. Depending on the architectural variant, some traffic may still be transported on the Wi-Fi uplink, e.g. management packets. Downlink traffic is selectively split between the Wi-Fi and LTE networks, providing benefits of offload to the cellular network. The uplink traffic, being routed via the LTE radio link, is completely scheduled and hence offers better performance to the user. When the AP only delivers downlink traffic, there is no contention to resolve using the CSMA protocol, and in a planned network with no co-channel neighbors, Wi-Fi can operate on a completely scheduled basis. Downlink rates can be guaranteed since the channel use is fully under the control of the scheduler. Guaranteed rates can be extremely helpful in providing high quality video. TCP downlink applications also experience improved throughputs by avoiding uplink TCP acknowledgements (ACKs) over the unpredictable and congested Wi-Fi link, which could have otherwise lead to unnecessary shrinkage of the TCP transmit window due to lost/delayed TCP ACKs. Re-direction should occur when the contention due to uplink traffic on the Wi-Fi interface begins to impact channel availability. Where redirection of both management and data packets is achieved, extended coverage is possible since Wi-Fi will not be limited by the lower power of the client.

Three distinct Wi-Fi/LTE integrated solutions are presented. In loose integration, network connections can be managed separately without knowledge of each other's presence. Loose integration requires only modification to operating software in the client device and server. It operates with the least knowledge of the network state, provides performance improvement, and is easily implementable. In tight integration both networks are closely coupled to potentially provide the highest performance, but at high implementation complexity. We also present a hybrid solution where networks are minimally aware of each other, and demonstrate how simple routing of packets over the alternate network can improve coverage. In terms of complexity, hybrid integration falls between loose and tight integration.

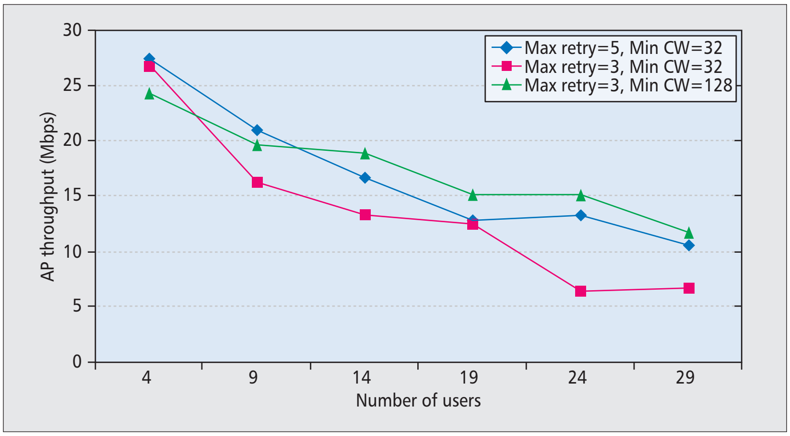

Figure 2. Degradation of Wi-Fi total throughput with increasing number of users for different operating parameters.

Loose Integration

Multipath TCP (MPTCP) is an IETF draft standard that defines a method of joining independent TCP sub-streams to provide a reliable connection between client and server applications [8]. MPTCP provides bandwidth federation and redundancy. Redundancy can be used to support user mobility over wired and wireless networks. The current specification provides control, scheduling, and compatibility with existing Internet routers. Note that in the TCP/IP networking reference model, TCP and MPTCP operate on the transport layer and communicate upward to the application and downward to the network layer that handles routing. Refer to [9] for additional details.

We consider augmenting MPTCP with a rule that specifies how an interface is used when it is available. The rule controls how traffic is optimized, and may also utilize statically configured information such as air-interface overhead for a particular packet size. Figure 3 illustrates the architecture for loose integration using MPTCP. Two different modes of operation are shown: one on the left for achieving downlink traffic aggregation that can be used to offload downlink traffic from LTE to Wi-Fi; and the other on the right is used to offload uplink traffic from Wi-Fi to LTE for improving efficiency on the Wi-Fi air interface.

By changing the uplink "cost" on the Wi-Fi TCP subflow, along with using existing metrics such as measured round-trip time and bandwidth, MPTCP would schedule the majority of the uplink traffic on the LTE TCP subflow. This would increase Wi-Fi efficiency, but not as much as if all uplink signaling were shifted. That is, on

In theory, to improve responsiveness MPTCP could utilize cross layer information, e.g. Wi-Fi air-time usage, packet retry counters, and signal strength, but this comes at the cost of impacting the robustness and simplicity provided by layering network services and protocols.

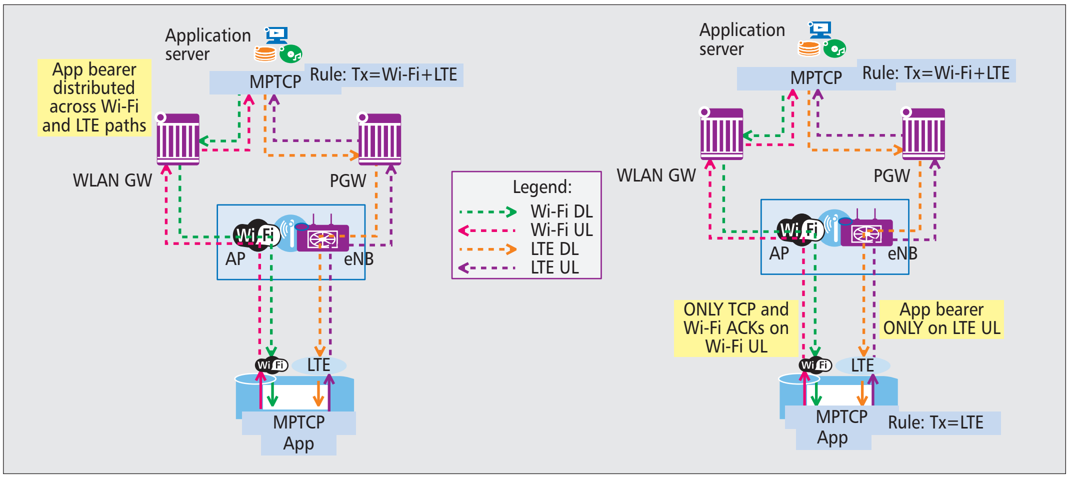

Figure 3. Architecture for loose Wi-Fi LTE integration.

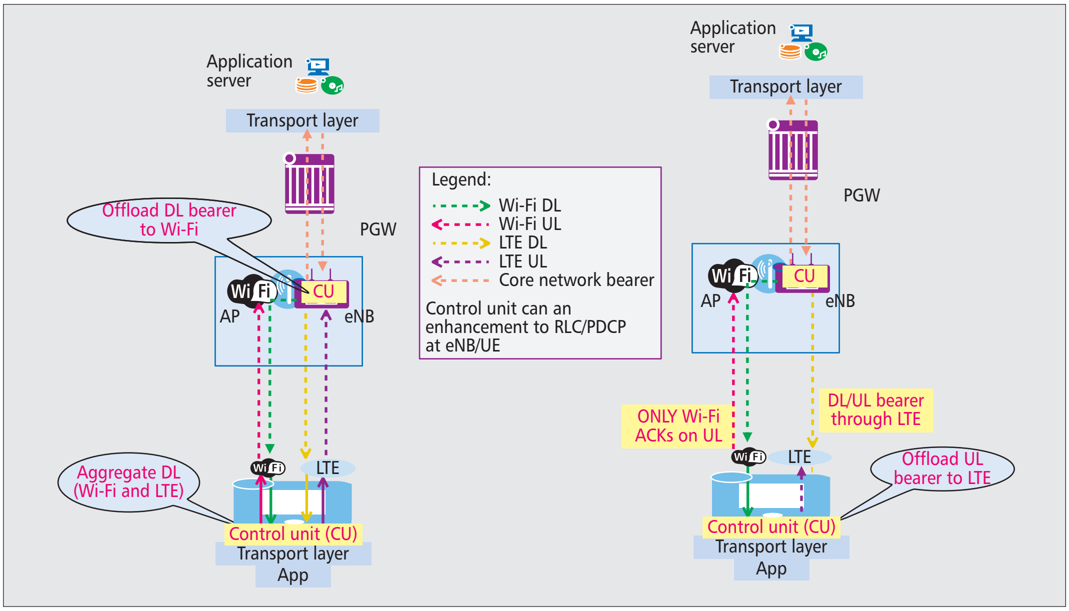

Figure 4. Architecture for tight Wi-Fi LTE integration.

the Wi-Fi uplink TCP ACKs are still being generated in response to downlink TCP packets. While shifting the ACKs to a different interface appears feasible, it does break the independence of TCP subflows. Finally, in theory, to improve responsiveness MPTCP could utilize cross layer information, e.g. Wi-Fi air-time usage, packet retry counters, and signal strength, but this comes at the cost of impacting the robustness and simplicity provided by layering network services and protocols.

Tight Integration

Figure 4 illustrates the tight architecture where traffic, also known as a bearer in 3GPP, is distributed across the radio technologies via a control unit (CU) at the base station (termed eNB in 3GPP). Traffic enters and exits the LTE RAN via the PGW. As example use cases, for the tight integration solution, two modes of operation are shown in Fig. 4. The mode on the left demonstrates downlink bearer aggregation across Wi-Fi and LTE interfaces for providing cellular offload; the mode on the right illustrates offload of uplink TCP traffic to LTE from Wi-Fi for enhanced Wi-Fi efficiency. The two key functional units of this system are a central control entity at the eNB that meets each user's quality of experience (QoE) requirements by determining the split of the bearer between the Wi-Fi and LTE interfaces in the downlink, and a control unit at the client device that handles aggregation and feedback for the downlink traffic and routes uplink traffic.

The existing radio link controller at the base station can be upgraded to handle both links. A similar control function operates at the client to distribute data across the radio interfaces. Both of these obtain feedback on the channel capacity and buffer drain rates from the existing network stack. Using this information, the control unit can quickly respond to dynamic RF conditions and user loads to tailor the bearer distribution between the Wi-Fi and LTE interfaces. Due to the direct availability of the feedback at the con-

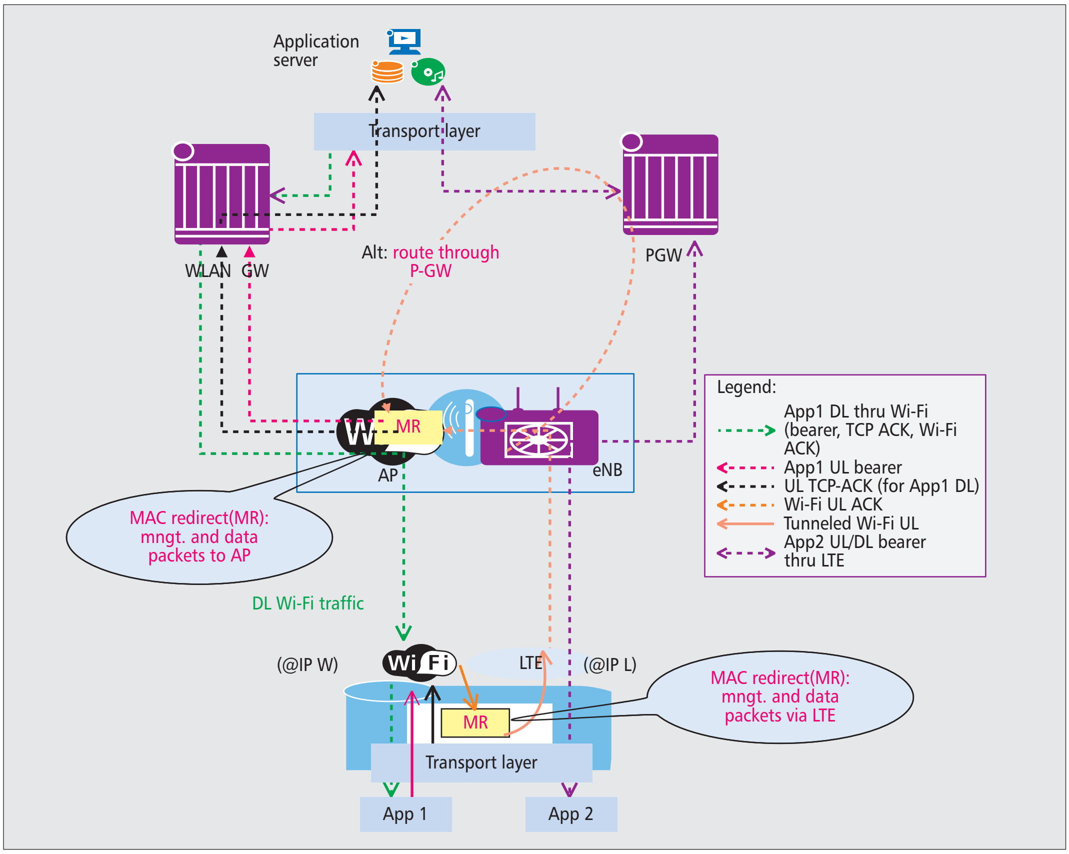

Figure 5. Architecture for hybrid integration. App1 is in hybrid mode. App2 uses LTE only.

The integrated Wi-Fi/LTE system, with all uplink transmissions (data and management) transferred from Wi-Fi to LTE, enhances the capacity of Wi-Fi as well as the range due to the higher uplink transmit power on the LTE interface as compared with Wi-Fi.

trol unit, it can respond faster to the access link conditions as compared to the loose integration solution, where the transport layer has to infer the congestion in the links using secondary effects such as round trip delays and TCP ACK loss. In this architecture, both the transport signaling, e.g. TCP ACKs, and application data in the uplink can be transferred to the LTE air link.

Hybrid Integration

While the loose and tight integration schemes provided enhanced Wi-Fi performance by offloading different components of the application and management traffic in the uplink, the Wi-Fi protocol based uplink control signaling such as MAC layer ACKs, probe and association requests for Wi-Fi link setup, need to be carried over the Wi-Fi uplink. Therefore, the range at which the device can remain associated with the Wi-Fi AP is still limited by the Wi-Fi uplink. Further, elimination of the Wi-Fi protocol's control signaling, such as ACKs, from the Wi-Fi uplink can reduce interference caused to the neighboring APs and hence improve performance in dense deployments. A method to deal with this is explained in the hybrid integration solution

Consider the use of Wi-Fi with an independent IP path for offload of cellular traffic, also called a non-seamless wireless offload (NSWO) in 3GPP [14]. Both interfaces are active and available, and each will be independently assigned an IP address. Applications can be connected to one or the other interface, or even both.

In the hybrid integration solution, as illustrated in Fig. 5, the Wi-Fi interface is assisted by the LTE network to transport the uplink Wi-Fi packets (uplink bearer such as application data and TCP control signaling, uplink control and management frames such as Wi-Fi MAC layer ACKs) via a tunnel to the Wi-Fi AP. As an illustration, take an application that uses only the IP interface via Wi-Fi, such as App1 in Figure 5. The downlink data flows only over the Wi-Fi radio and uplink traffic is tunneled via the LTE interface to the AP. From the perspective of the AP, all uplink traffic appears to flow to and originate from the Wi-Fi interface. The tunnel traverses the LTE RAN through the eNB and terminates at the Wi-Fi AP either through the LTE packet core, which is the regular path for a LTE bearer, or via a direct path between the eNB and the AP. While the direct path requires enhancements to the eNB, it offers the advantage of lower latency over the tunneled link. APs terminate the IP tunnel by injecting the packets into the networking stack at a point where they would normally be obtained from the Wi-Fi radio itself. Otherwise, both networks remain independent and unaware of each other. As an illustration of the independence of the two interfaces, we show in Fig. 5 another application on the device, i.e. App2, which uses the IP path via the LTE interface, operating independent of the hybrid mode of operation for App1 that uses LTE assisted Wi-Fi link.

The integrated Wi-Fi/LTE system, with all uplink transmissions (data and management) transferred from Wi-Fi to LTE, enhances the

Aspect \ Integration type	Loose	Tight	Hybrid
Network functionality impacts	Transport layer support for MPTCP/Proxy at UE and application server. No Impact to LTE RAN or Wi-Fi core network.	LTE protocol stack changes at UE and eNB. Otherwise no impact to Wi-Fi or LTE network.	Wi-Fi MAC redirection module at UE and AP. No impact to LTE RAN or Wi-Fi core network.
Wi-Fi uplink offload	TCP data	IP packets including TCP data and ACKs	Wi-Fi MAC layer packets: data and control
Wi-Fi range extension	No	No	Yes

Table 1. Comparison of Wi-Fi LTE integration schemes.

capacity of Wi-Fi as well as the range due to the higher uplink transmit power on the LTE interface as compared with Wi-Fi. By offloading the complete uplink traffic to an alternate interface, interference to neighboring co-channel APs is reduced, making it well suited for operation in dense deployments. When APs are sparsely placed, the capacity of a heterogeneous network is enhanced by being able to serve users in a larger range and higher downlink capacity from the APs. The hybrid integration mode can be used in conjunction with loose and tight integration mode to achieve further path optimization by enabling TCP ACKs and the application bearer to directly reach their protocol end-points using the LTE interface without the need to traverse the tunnel via the AP, wherein only the Wi-Fi protocol control and management information uses the uplink tunnel to reach the AP.

System Level Considerations

Existing multi-radio, Wi-Fi/LTE clients can be enhanced by software upgrade to support the Wi-Fi/LTE super aggregation schemes. For the readers' convenience they are summarized in Table 1. These (enhanced) integrated Wi-Fi mode clients will need to co-exist with (legacy) existing standard multi-radio and single radio Wi-Fi devices. The integrated Wi-Fi mode may be attached to a separate service set identifier (SSID) to distinguish itself from standard access. That is, the Wi-Fi AP will broadcast at least two SSIDs: one for standard Wi-Fi access and the other for integrated access. For hybrid integration, its SSID would be transmitted at full power; for standard access, the SSID would be transmitted at a power close to that of the client. When in range of both SSIDs the connection manager would determine whether to enable integrated mode or stay in the standard connection mode.

Overall Wi-Fi efficiency can be improved by designating short time intervals as downlink only periods. During this period downlink transmissions for integrated Wi-Fi and standard clients are scheduled. Since the newer devices in the integrated Wi-Fi mode have their uplink redirected to LTE, they can fully leverage the improved Wi-Fi performance without any negative impact from the uplink transmissions of the legacy mode devices. This is an improvement over the standard Wi-Fi mode devices whose uplink will have to wait until the downlink-only period is completed. In order to reserve this period, the network allocation vector (NAV) of nearby clients must be updated. The AP should broadcast a clear to send to self (CTS-To-Self) 802.11 management message for airtime reservation of up to 32 ms for the data in its buffer. In the integrated mode, contention is reduced, therefore the contention window can be tuned to operate at the minimum levels to control the protocol overhead.

LTE systems can either be Frequency Division Duplex (FDD) or Time Division Duplex (TDD). In FDD LTE systems, separate frequency bands are reserved for uplink and downlink. In TDD LTE systems, the resources (time slots) on a common frequency band are shared between uplink and downlink, with the uplink to downlink resource ratio being tune-able based on relative traffic load. Therefore, TDD LTE systems offer further flexibility by allowing appropriate tuning of the uplink-downlink resource ratio to accommodate the additional uplink load offload from Wi-Fi to LTE.

Conclusions

Wi-Fi cost-effectively addresses the tremendous demand for data from wireless devices by leveraging unlicensed spectrum. Wi-Fi uplink spectral efficiency and coverage limitations were discussed, motivating solutions that integrate Wi-Fi with LTE. The expanding availability of LTE with the ubiquitous presence of multi-radio devices offers different options for integration, with different levels of implementation complexity and the corresponding trade-off in benefits. Integration enables Wi-Fi networks to serve more users with higher throughput demands and as an effective traffic offload solution for cellular operators.

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Biographies

JONATHAN LING (Jonathan.Ling@alcatel-lucent.com) is a researcher at Bell Laboratories currently focusing on Wi-Fi/LTE integration. He has 20 years of experience in proposing enhancements and evaluating performance of wireless systems, and over 40 publications. He measured MIMO radio channels, investigated heterogeneous network capacity scaling, and developed a compressed sensing algorithm for channel estimation. He earned a B.Sc degree in electrical engineering from Rutgers University, Piscataway, NJ, and an M.Sc. in computer science and Ph.D. in electrical engineering, both from Stevens Institute, Hoboken, NJ.

SUBRAMANIAN VASUDEVAN (subramanian.vasudevan@alcatel-lucent.com) is Director of the Advanced Performance group and CTO Prime for Small Cells in the Wireless CTO at Alcatel-Lucent. He leads a team building state-of-the-art modeling for wireless technology that develops and refines analyses and performance characterizations of new technology concepts to guide technology strategy, customer engagement, innovation, and standardization studies. Previously, he led the CDMA air interface and access networks standards group in Alcatel-Lucent's Wireless Standards and Intellectual Property Organization. He earned a Bachelor's degree in electrical engineering from the Indian Institute of Technology, Bombay, India, and a M.S and Ph.D from the University of Colorado, Boulder, in the Department of Electrical and Computer Engineering.

SATISH KANUGOVI (satish.k@alcatel-lucent.com) is a distinguished member of technical staff (DMTS) in the Wireless CTO division at Alcatel-Lucent. He has approximately 15 years of experience in the domain of wireless research, standardization and product development spanning across a multitude of 2G/3G/4G technologies. He is currently part of the advanced performance group involved in innovation and analysis of schemes to guide the technology strategy for future wireless networks. In his earlier roles, he was the ALU delegate for 3GPP2 standards and wireless RAN system engineering. He has approximately 20 patents and publications in the area of wireless systems. He did his BE in electronics and communication form Delhi College of Engineering in 1999. He is Senior Member of IEEE and formerly chair of the India chapter of Alcatel Lucent Technical Academy.

KRISHNA PRAMOD ADHARAPURAPU (Krishna_Pramod.A@alcatel-lucent.com) is a distinguished member of technical staff (DMTS) in the Wireless–Small Cells division at Alcatel-Lucent. He has approximately 14 years of experience in product development spanning across wireless/wireline technologies, in the platform and OA&M areas. He is currently part of the platform team of small cells products and is involved in feature development and testing of small cells. He has filed three patents. He did his B.E. in electrical and electronics engineering from BITS (Birla Institute of Technology and Sciences), Pilani, India.

NETWORK TESTING

Ying-Dar Lin *Erica Johnson*

In the last call, we received eight submissions and accepted two articles while leaving one article to go through another round of revision and review. For future issues, we have shifted the due date by two months to February 1 and August 1 to shorten the time to publication in the September issue and March issue, respectively.

Each technology has its life cycle from being emergent and then getting deployed, to growing fast and becoming mature. For communications technologies, in the emerging stage it is the basic testing, such as conformance and interoperability, applied to new solutions. In the growing and mature stages the effort would be shifted to performance testing to further improve the deployed solutions. In this issue we have two articles, one on field performance testing of the deployed LTE systems, and another on building a basic measurement tool on the emerging and thus emulated content-centric networks. This series welcomes contributions on new test methodologies and results, addressing either emerging or maturing communications technologies.

The article on handover interruption time in LTE networks ("Measurement and Stochastic Modeling of Handover Delay and Interruption Time of Smartphone Real-Time Applications on LTE Networks") is one of the few studies to profile handover delay and interruption time in a real LTE "operational" network. Unlike most studies with analytical and simulation results, or test results drawn from a lab test bed, the authors conducted, with help from an operator, measurements on 16 locations, including low-populated streets, high-populated streets, subway stations, shopping malls, and moving vehicles. This field test bed consists of E-UTRAN (Evolved Universal Terrestrial Radio Access Network, including UEs (user equipments) and eNBs (evolved Node B) and EPC (Evolved Packet Core) including MME (mobility management entity), S-GW (serving gateway), P-GW (packet gateway), PCRF (policy and charging rules function), etc. The handover process consists of three phases: handover preparation (between UE, source eNB, and target eNB); handover execution (between UE and target eNB); and handover completion (between target eNB, EPC components). The handover execution phase is the most critical part because the service would be interrupted. Thus, its length decides the handover interruption time (HIT). Within that phase, UE runs hard handover to first disconnect from the source eNB, and then connect to the target eNB by a random access (RA) process, and finally receive an ACK (acknowledgement) of the radio resource control (RRC) connection reconfiguration complete message.

The authors reported some interesting results. Delay in EPC is often larger than the delay in E-UTRAN because the number of EPC components is small and could be far away from UEs. The delay from MME to P-GW takes the largest part of the total handover delay at 32.56 percent. This is because P-GWs are multi-purpose and heavily loaded. This could be improved by virtualizing P-GW to more servers, i.e. NFV (network function virtualization) with better scalability. The most critical HIT in the handover execution phase accounts for 22.12 percent of the total handover delay. This is dominated by the random access channel (RACH) period where RACH configurations matter. Key RACH configurations include the number of users in a cell, RACH scheduling period, preamble split, and RACH load. The observations provided in this study could be very useful to operators, enabling them to better configure the key operational parameters, and vendors, to enable them to better engineer the handover procedures in E-UTRAN and EPC.

The emerging content-centric networks (CCNs) or named data networks (NDNs) aim to enable contents addressable by names instead of by IP addresses. Publishers provide contents through the network, while consumers retrieve content by issuing requests with names or name prefixes. Routers need to forward content requests by looking up forwarding information base (FIB) populated by name-based routing protocols. Contents could also be cached in routers, i.e. in-network cache. In summary, the ideas of CCN are based on "naming," "routing," and "caching." The second article on contrace for CCN ("Contrace: A Tool for Measuring and Tracing Content-Centric Networks") presents a tool for measuring and tracing CCN. It is similar to ping and traceroute for IP networks. Contrace can trace the routing path and measure round-trip time. It could also measure the state of cache in routers to get access count and cache lifetime, in order to estimate content popularity and optimize in-network caches

Just like ping and traceroute, which rely on IP and ICMP embedded in all IP routers and hosts, contrace also needs to embed something in CCN routers. Thus, the authors implemented the tool in two programs: a user command program (contrace), and a forwarding daemon (contracted) and embed contracted to all CCN routers. The authors did their experiments on their CCN emulated by seven virtual machines. It has been demonstrated that the tool can trace the routing path of content retrieved by a name prefix in a CCN, just like by an IP prefix in an IP network. It could also trace the end-to-end delay along a path and hop-by-hop delay between CCN routers, and count the cache access and lifetime to compute content popularity. It appears to be a powerful tool for research and development on this emerging technology.

BIOGRAPHIES

YING-DAR LIN (ydlin@cs.nctu.edu.tw) is a distinguished professor at National Chiao Tung University (NCTU) in Taiwan. He received his Ph.D. in computer science from UCLA in 1993. He is the director of Network Benchmarking Lab, which reviews network products with real traffic and is an approved test lab of the Open Networking Foundation (ONF). He is an IEEE Fellow, IEEE Distinguished Lecturer, and ONF research associate. He co-authored the book *Computer Networks: An Open Source Approach* (McGraw-Hill, 2011).

ERICA JOHNSON (erica.johnson@iol.unh.edu) combines business acumen and an in-depth understanding of complex networking technology to direct the University of New Hampshire InterOperability Laboratory (UNH-IOL). In recognition of her ability to drive technical innovation, Fierce Telecom named Erica to the publication's 2011 Women in Wireline. She serves as an IPv6 Ready Logo Regional Officer, IPv6 Forum Fellow, and USGv6 Test Program lead. Erica received her Bachelor of Computer Science and an MBA from UNH in 2001 and 2011, respectively.

Measurement and Stochastic Modeling of Handover Delay and Interruption Time of Smartphone Real-Time Applications on LTE Networks

Donghyuk Han, Sungjin Shin, Hyoungjun Cho, Jong-Moon Chung, Dongseok Ok, and Iksoon Hwang

Donghyuk Han, Sungjin Shin, Hyoungjun Cho, and Jong-Moon Chung are with Yonsei University.

Dongseok Ok and Iksoon Hwang are with Samsung Electronics.

ABSTRACT

For continuous services of mobile user equipment (UE), Long Term Evolution (LTE) systems conduct evolved node B (eNB) switching based on hard handover technology, which breaks a connection before the connection to the target eNB (T-eNB) is made. As handover service interruption seriously degrades network performance, precise knowledge of the handover (HO) performance is necessary in finding out defects of the current system and discovering clues for improvements. Although the performance of LTE handover and its anticipated effect on network services are important evaluation indexes, in existing literature only the theoretical performance is analyzed and very few actual measurements on practical LTE networks have been presented. In this article, the HO delay and handover interruption time (HIT) performance of LTE networks are measured for several cases in accordance with the average number of users in a cell. Based on the internal HO procedures that influence HO delay and HIT, the key parameters are analyzed. In addition, based on the estimated number of users in a cell, a reference probability density function (PDF) that can be used for HIT prediction is presented.

INTRODUCTION

In order to provide high quality of service (QoS) and seamless connectivity for mobile services, cellular networking technologies have significantly improved in recent years. Handover (HO) is one of the most QoS performance degrading activities, so Third Generation Partnership Project (3GPP) has been developing various standardized HO mechanisms to meet higher levels of QoS requirements [1]. The Long Term Evolution (LTE) HO mechanism provides a significantly enhanced performance compared to the conventional mechanism used in the Evolved High Speed Packet Access (HSPA+).

Despite these improvements, the HO mechanism of LTE networks intrinsically induces interruption of service which severely degrades QoS as well as user experience (UX). LTE systems only support hard HO, which results in an unavoidable interruption in service when HO is performed. This is referred to as handover interruption time (HIT), which is the period the user equipment (UE) cannot exchange user plane data during HO. Although HIT has a huge impact on the HO performance in terms of QoS and UX, a practical performance assessment on commercially deployed networks is very hard to conduct, as most of evolved packet core (EPC) elements performing the HO process are complex and are not measurable from the user's perspective. However, if the UE could predict the expected HIT, and the key internal influencing factors were also known, there could be many beneficial techniques that could be developed to improve the HO performance.

Although many previous studies have evaluated LTE HO performance, performance analysis has been only based on theoretical models. Practical performance analysis of LTE including throughput and HO is provided in [2]. However, only measurement results are presented so that it is hard to discuss internal issues in LTE systems. In this article, the field test measurement methods on the HO performance of commercial LTE networks and the analysis results including the key factors impacting large HO delay are presented. In addition, 3GPP LTE standard parameters which affect HIT are analyzed (including network load) and a stochastic model of HIT is presented. Using a proposed HIT model, an adaptive handover offset scheme is discussed, which can achieve reduced HIT and load balancing in LTE networks.

The rest of the article is organized as follows. We describe the network test setup and its results. We analyze HIT and its stochastic characteristics, and we conclude the article.

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The processing delay in EPC elements for bearer establishment, generating encryption keys, and session modification are also considered. However, the link setup time for assigning radio resources is ignored as it has a much smaller duration compared to other delay components.

LTE Network Measurement and Results

When performing HO, the source eNB (S-eNB) decides the type of HO between X2 HO and S1 HO. X2 HO is performed using the X2 interface, which provides direct connection between eNBs. S1 HO is performed when there is no X2 connectivity or when it is not allowed to use X2 interfaces. This article analyzes X2 HO, since HO is always executed via the X2 interface when the eNBs belong to the same pooling area. The HO procedure within 3GPP LTE is composed of three phases: HO Preparation, HO Execution, and HO Completion. Thus, the total HO delay T_{HO} is obtained by summing the time delay for each phase. In the measurement analysis, the transmission delay is obtained based on transmission time of signaling messages. The processing delay in EPC elements for bearer establishment, generating encryption keys, and session modification are also considered. However, the link setup time for assigning radio resources is ignored as it has a much smaller duration compared to other delay components.

Figure 1a shows the network architecture of the 3GPP EPC and the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) [3]. E-UTRAN is composed of eNBs, which are interconnected with each other through X2 interfaces. The LTE architecture and HO procedures are presented in Fig. 1b, followed by the description of each phase including measurement methods.

HO Preparation

The HO Preparation phase is performed without EPC elements so that the preparation messages are directly exchanged among the eNBs and the UE. An UE measures the radio signal strength of the serving and the neighbor cells and reports this information to its S-eNB using a `Measurement Report`. When signal strength of the neighbor cell has been stronger than that of the serving cell for a certain period, as shown in Fig. 1c, the S-eNB will decide to execute handover and initiates the HO Preparation phase by sending a `Handover Request` to the target eNB (T-eNB). A `Handover Request` includes UE context information, QoS parameters, and General Packet Radio Service Tunneling Protocol (GTP) Tunnel information. T-eNB assigns radio resources for the UE and replies to the S-eNB with a `Handover Request ACK`.

The delay for the HO Preparation phase is represented as $T_{HO\ Prep} = 2T_{S\text{–}eNB\text{–}T\text{–}eNB} + t_{eNB}$, where $T_{S\text{–}eNB\text{–}T\text{–}eNB}$ indicates the transmission delay between the S-eNB and T-eNB [4], and t_{eNB} represents the processing delay of the eNB, where t_{eNB} includes the processing of generating security keys and establishing a S1 bearer.

HO Execution

The HO Execution phase is most critical on the handover performance due to the HO interruption/disconnection that occurs. The HO Execution phase is triggered by the S-eNB sending a `Radio Resource Control (RRC) Connection Reconfiguration` to the UE, which is the command to modify the RRC connection. Upon reception of the `RRC Connection Reconfiguration`, the UE disconnects from the S-eNB and detects the synchronization signal from the T-eNB to perform synchronization, and then attempts to initiate the Random Access (RA) process. As eNBs allocate resources for HO attempts, the RA process is rapid and contention-free. When HO requests exceed the pre-allocated resources, the RA process becomes contention-based such that UEs have to compete for channel access. During HIT, downlink packets are delivered to the T-eNB through the forwarding bearer, so packet loss is prevented. When the UE completes the RA process, the UE informs the T-eNB by sending a `RRC Connection Reconfiguration Complete` and the T-eNB replies to the UE with an ACK.

The delay for the HO Execution procedure $T_{HO\ Exe}$ is represented as $T_{HO\ Exe} = T_{HIT} + T_{UE\text{-}eNB}$, where the T_{HIT} is the HIT and $T_{UE\text{-}eNB}$ is the transmission delay between an UE and an eNB. The measurement of HIT is obtained by computing the time difference between the reception of the `RRC Connection Reconfiguration` and the reception of the completion acknowledging `RRC Connection Reconfiguration Complete`. The measurement of the transmission delay between the UE and the S-eNB $T_{UE\text{–}eNB}$ is described in the next subsection.

HO Completion

The HO Completion phase consists of Path Switching, Modify Bearer, and IP Connectivity Access Network (IP –CAN) Session Modification. The T-eNB requests for the MME to switch the GTP tunnel towards the T-eNB using the `Path Switch Request`. MME then sends a `Modify Bearer Request` to the serving gateway (S-GW) to request the S-GW to switch the path to the T-eNB. After Policy and Charging Rules Function (PCRF) makes the authorization and policy decision on the IP-CAN Session Modification procedure and T-eNB completes the path switching, the S-eNB will release the UE Context when it receives an `UE Context Release` from the T-eNB.

The HO Completion delay is expressed as: $T_{HO\ Comp} = 2T_{eNB\text{–}MME} + 2T_{MME\text{–}P\text{–}GW} + T_{IP.CAN} + T_{S\text{–}eNB\text{–}T\text{–}eNB} + t_{S\text{–}GW} + t_{MME} + t_{eNB}$. The HO Completion delay comprises of the transmission delay for exchanging `Path Switch Request` and `Path Switch Request ACK` ($2T_{eNB\text{–}MME}$), the transmission delay for delivering `Modify Bearer Request` and `Modify Bearer Response` ($2T_{MME\text{–}P\text{–}GW}$), the processing delay of IP–CAN Session Modification ($T_{IP.CAN}$), the transmission delay of `UE Context Release` ($T_{S\text{–}eNB\text{–}T\text{–}eNB}$), and the processing delay in the S-GW ($t_{S\text{–}GW}$), MME (t_{MME}), and eNB (t_{eNB}). Most of the signaling messages in the HO Completion phase don't pass through the UE and thus cannot be measured by the UE. Thus other Evolved packet system Mobility Management (EMM) scenarios are tested to measure delay components on $T_{HO\ Comp}$.

The transmission delay of $T_{eNB\text{-}MME}$ is measured by the initial attach procedure. When an UE initially attaches to the LTE network, the UE sends a `RRC Connection Request` and

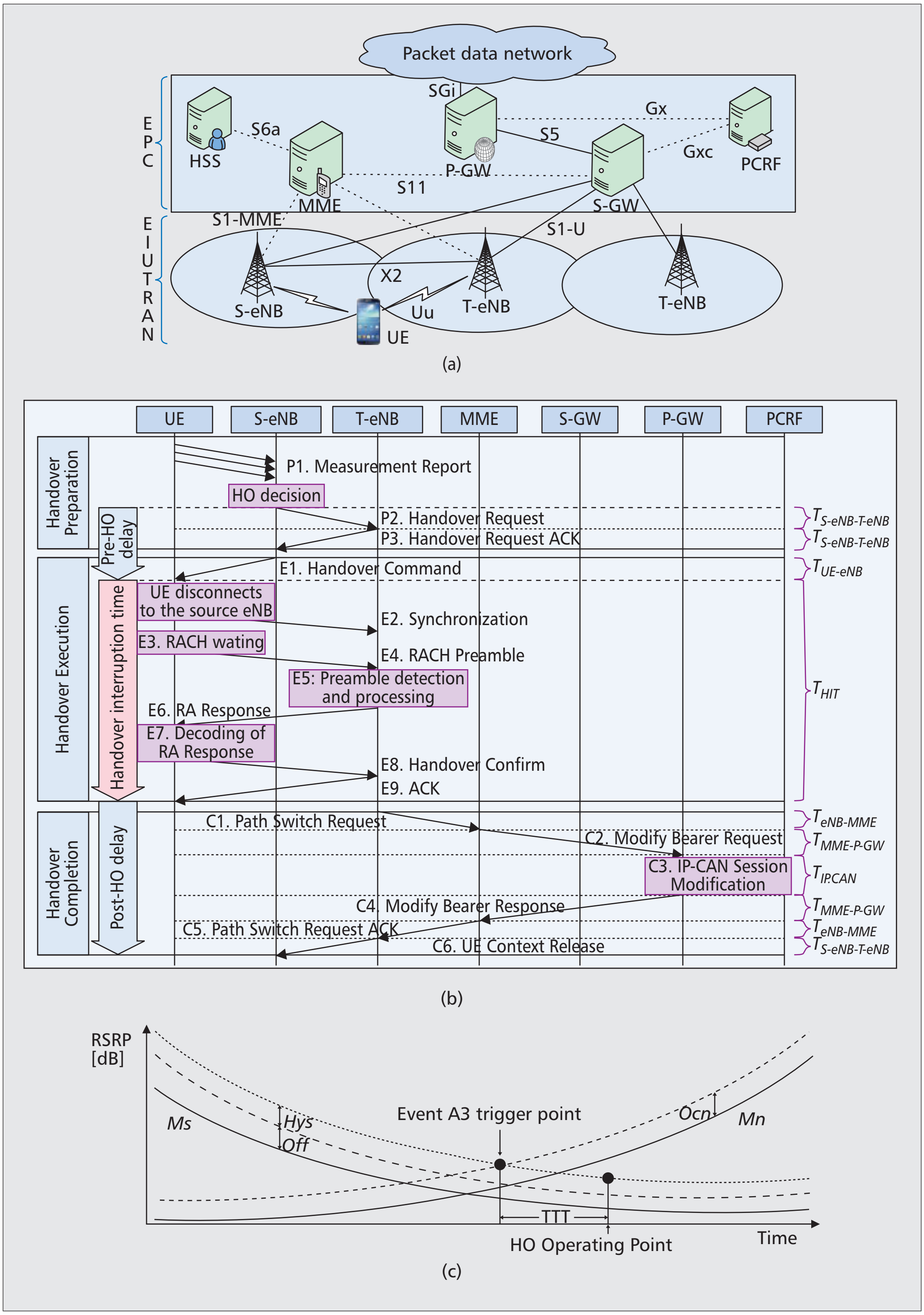

Figure 1. LTE architecture and HO procedure: a) LTE network architecture; b) signaling flow of LTE HO; and c) handover event and related parameters.

In the measurements, network tools as well as Android platform tools are needed. Android Debug Bridge (ADB) is used for Traceroute and Tcpdump commands, where Traceroute confirms the delay between the UE and the P-GW, and Tcpdump executes packet capturing at the Android kernel.

the eNB replies with a `RRC Connection Setup`. The one-way delay $T_{UE\text{-}eNB}$ is obtained from half of the time difference between the two messages. During the initial attach procedure, the UE also sends an `Attach Request` to the MME and the MME responds with a `Security Mode Command`. Thus $T_{UE\text{-}MME}$ is obtained from these message exchanges, then the transmission delay of $T_{eNB\text{-}MME}$ can be obtained as $T_{eNB\text{-}MME} = T_{UE\text{-}MME} - T_{UE\text{-}eNB}$. The Tracking Area Update (TAU) procedure is another EMM scenario that is useful to obtain the transmission delay $T_{MME\text{-}P\text{-}GW}$. TAU is initiated when an UE moves to an unregistered Tracking Area (TA) or when the TAU timer expires. The TAU procedure delay TTAU is composed of the signaling messages exchanged between the UE and the MME $T_{UE\text{-}MME}$,

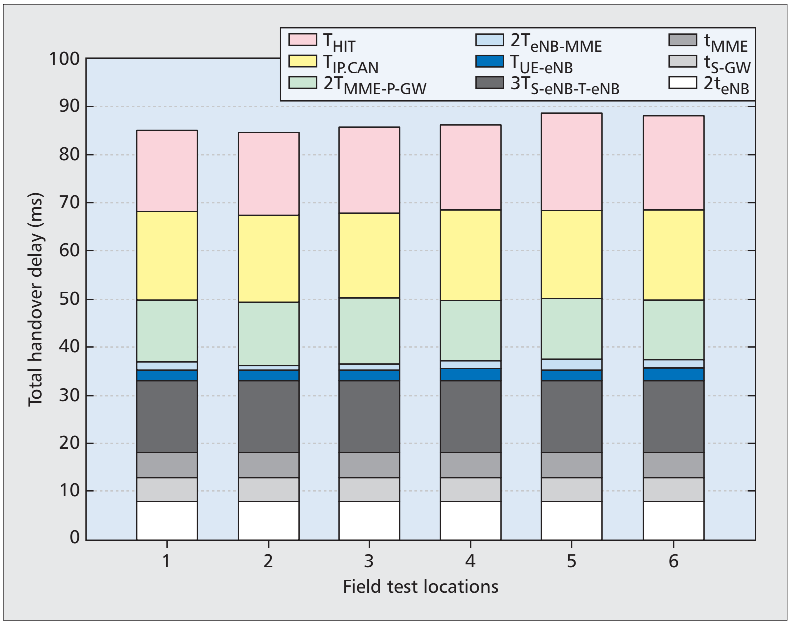

Figure 2. Analysis of each HO delay components.

between the MME and the packet data network gateway (P-GW) $T_{MME-P-GW}$, IP-CAN session modification $T_{IP.CAN}$, and processing delays in the EPC entities (t_{S-GW}, t_{MME}, t_{eNB}), resulting in $T_{TAU} = 2T_{UE-MME} + 2T_{MME-P-GW} + T_{IP.CAN} + t_{S-GW} + t_{MME} + t_{eNB}$. The TAU procedure starts from the UE sending a `TAU Request` and ends when the UE receives a `TAU Accept`, where the procedure delay T_{TAU} is obtained by computing the time difference. By obtaining T_{UE-MME} and the processing delay of EPC entities, the transmission delay $T_{MME-P-GW}$ can be obtained, which allows computing the whole HO Completion procedure delay $T_{HO\ Comp}$. The processing delay of the S-GW includes the downlink path switching of the T-eNB. The total HO delay is presented below.

$$T_{HO} = 3T_{S\text{-}eNB\text{-}T\text{-}eNB} + T_{UE\text{-}eNB} + 2T_{eNB\text{-}MME} + 2T_{MME\text{-}P\text{-}GW} + T_{IP.CAN} + T_{HIT} + t_{S\text{-}GW} + t_{MME} + 2t_{eNB} \quad (1)$$

Testing Environment and Measurement Results

The field test environment of the LTE network is based on 3GPP Release 8: LTE Band 3 with a 20 MHz channel bandwidth. The UE is a Samsung Galaxy Note 1 with Qualcomm Scorpion 1.5 GHz dual-core, Qualcomm MSM8660 modem, and is connected to a laptop providing downlink and uplink message logging using a Qualcomm Extensible Diagnostic Monitor (QxDM). QxDM provides a diagnostic client that displays data transmitted to and from the UE. Thus, it enables a user to obtain most of the HO delay parameters by computing the time difference between the transmission and the reception of signaling messages shown in Fig. 1b. The timestamps of signaling messages are provided by QxDM (which supports a ms resolution) that measures Control Plane messages. In addition, Wireshark (which supports a μs resolution) is used for supplementary analysis on User Plane data.

In the measurements, network tools as well as Android platform tools are needed. Android Debug Bridge (ADB) is used for Traceroute and Tcpdump commands, where Traceroute confirms the delay between the UE and the P-GW, and Tcpdump executes packet capturing at the Android kernel. Wireshark is executed on a Windows kernel that the UE is communicating with, so that the delay can be analyzed. The delay component $T_{IP.CAN}$ was obtained from measurement data of the Policy and Charging Control (PCC) function modeled into an emulator testbed, where the P-GW and the PCRF are implemented with GNU C++ 4.1, and each entity uses OpenDiameter 1.0.7h [5]. In the Seoul (S. Korea) metropolitan area, 19 locations are chosen as field test locations (FT1 to FT19). The first 6 locations are chosen to measure the overall HO procedures. FT1 is a drive test with a moving distance of 15.9 km, FT2 is an university campus, FT3 is a main street of a low populated area, FT4 is a subway station, FT5 is a complex shopping mall, and FT6 is a main street of a highly populated area. For measuring HIT and establishing a HIT stochastic model in terms of the number of active mobile users (in units of people/cell/min), the other 13 locations (FT7 to FT19) selected were all subway stations, since the information of number of active mobile users was available from the communications service provider and these areas experience frequent HO events. The 13 locations selected for the experiments range from the highest to a medium number of active mobile users among the subway stations in Seoul. For each field test location, 150 HO occurrences were measured and analyzed.

Figure 2 shows the amount of time each HO procedures consumes. Looking into the delay components, delay of procedures involving the S-GW and P-GW is much larger than the delay involving only eNB and/or MME. Including the IP-CAN Session Modification process, the P-GW is responsible to 36.08 percent of the total HO delay in average. Due to the very small number of P-GWs deployed in commercial LTE service networks, the P-GW is likely to be far away from the UE. The processing delay is also subject to be large since P-GWs are multirole devices which anchor S-GWs, set QoS policy, and manages accounting data and the IP address of every UE, etc. Thus by increasing the number of P-GWs or by applying a distributed mobility management (DMM) architecture for offloading, the HO delay can be alleviated. For example, 3GPP has been working on Local IP Access (LIPA) and Selected IP Traffic Offload (SIPTO) [6]. In the SIPTO architecture, certain types of traffic are offloaded to the P-GW close to the UE. In the LIPA architecture, a local gateway substitutes the P-GW, which enables a separate network connection. Although the DMM architectures are expected to cope with the load distribution of centralized gateways, mobility support is limited.

On average, HIT takes 21.08 percent of the total HO delay, and has a dominant role in QoS degradation since the UE cannot be serviced by either the S-eNB or the T-eNB during HIT.

[1] Analysis of the measured data reveal that the number of active mobile users is proportional to the transient population of a subway station (which is defined as the number of passengers boarding and alighting per hour), where the floating population information of the Seoul Metro and Korea Statistical Information Service (KOSIS) was used as a reference (http://kosis.kr).

HIT Stochastic Modeling and Internal Analysis

HIT Measurements and Stochastic Modeling

This section presents the analysis on the time distribution and the performance model of HIT. A summary of the measurement test results of FT7 to FT19 is provided in Table 1.

For the statistical analysis on the HIT distribution, the Kolmogorov-Smirnov test (K-S test) is used to select and verify the appropriate probability distribution function (PDF) from the measured data. The K-S test conducts a time-domain signal processing of the comparing signals and provides the level of similarity of the PDF of the signals. K-S test defines the significance level

$$\alpha = 2\sum_{i=1}^{\infty}(-1)^{i-1}e^{-2i^2 n\varepsilon^2}$$

and the distance

$$D = \varepsilon\left(\sqrt{n}+0.12+0.11/\sqrt{n}\right)$$

between the empirical distribution of the measured data and the cumulative distribution function (CDF) of the candidate distribution, where n is the number of measurements and ε is the maximum difference between the CDF of the candidate distribution and the empirical data. After the D and α values are calculated, the distribution of which the K-S distance D is smallest and the largest significance level α is selected as the proper distribution for the measured data. The statistical distribution of HO delay T_{HO} was chosen as the normal distribution with the minimum K-S distance 0.1708, which was better than the K-S distance 9.5126 of the lognormal distribution, and the K-S distance 4.8539 of the exponential distribution. In the same way, the time distribution of HIT T_{HIT} was also chosen as a normal distribution with the minimum K-S distance 0.6798, which was better than the K-S distance 6.9917 of the lognormal distribution and 3.3539 of the exponential distribution.

Using the statistical data obtained from the measurement tests summarized in Table 1, the average and the standard deviation of HIT can be modeled by the expected number of people/cell/min.[1] Since most of the internal process during HIT is RA, as shown in Fig. 1, the number of mobile users per single cell can be a determinant for HIT. Currently, the performance modeling strategies can be classified into three methods: curve fitting, parameterization, and coupling. Among all three methods, curve fitting is the most preferable as it generates a model based on a least square fit of the empirical data without the need of application information or system parameters. Thus, the curve fitting method is used to model the HIT as a function of the number of x people/cell/min. The curve fitting results are shown in Fig. 3. The normal distribution can be expressed as $N\{\mu, \sigma^2\}$, where μ is a mean and σ is a standard deviation of the distribution. Conducting the curve fitting method as in Fig. 3a and Fig. 3b, the HIT PDF $f_{THIT}(x)$ is obtained as in Eq. 2.

Measurement Location	Number of eNBs	People/cell/min	HIT Avg (ms)	HIT Standard deviation (ms)
FT7	25	2.103	17.10	0.516
FT8	15	2.153	17.25	0.547
FT9	18	2.420	17.43	0.616
FT10	15	2.988	17.38	0.611
FT11	32	3.013	17.57	0.677
FT12	7	3.067	17.53	0.624
FT13	41	3.160	17.6	0.611
FT14	30	3.587	17.75	0.721
FT15	27	4.077	17.94	0.827
FT16	39	4.668	18.25	0.829
FT17	12	5.486	18.55	0.912
FT18	21	6.565	18.87	1.064
FT19	15	7.539	19.34	1.202

Table 1. Summary on HIT measurements for various locations.

$$f_{THIT}(x) = N\{0.39x + 16.34, (0.12x + 0.27)^2\} \quad (2)$$

The comparison of the HIT model and the measured data is provided in Fig. 3c.

Analysis on Internal Process of HIT

As shown in Fig. 1, the internal process during HIT is composed of the following components:

1 Synchronization to T-eNB
2 Waiting time due to Random Access CHannel (RACH) scheduling period
3 RACH Preamble transmission
4 Preamble detection and process time at T-eNB
5 Transmission of RA Response
6 Decoding of scheduling grant and timing alignment at the UE
7 Confirming HO completion

After the synchronization, the UE starts RA procedures. Configuration of RACH procedures to reduce its RA delay can result in a significant improvement of mobile services, since the RA delay is directly correlated with HO delays, specifically, HIT. The average of the RACH procedure delay is estimated as follows [7].

Based on Table 2, the average delay due to the RACH scheduling period indicates the time the UE holds on transmission for the RACH transmission schedule, which becomes significantly longer when transmission errors or retransmissions occur. Such transmission errors are likely to occur when the number of users in a cell increases. For HIT internal analysis, measurements of the parameters of RACH are presented in Fig. 4.

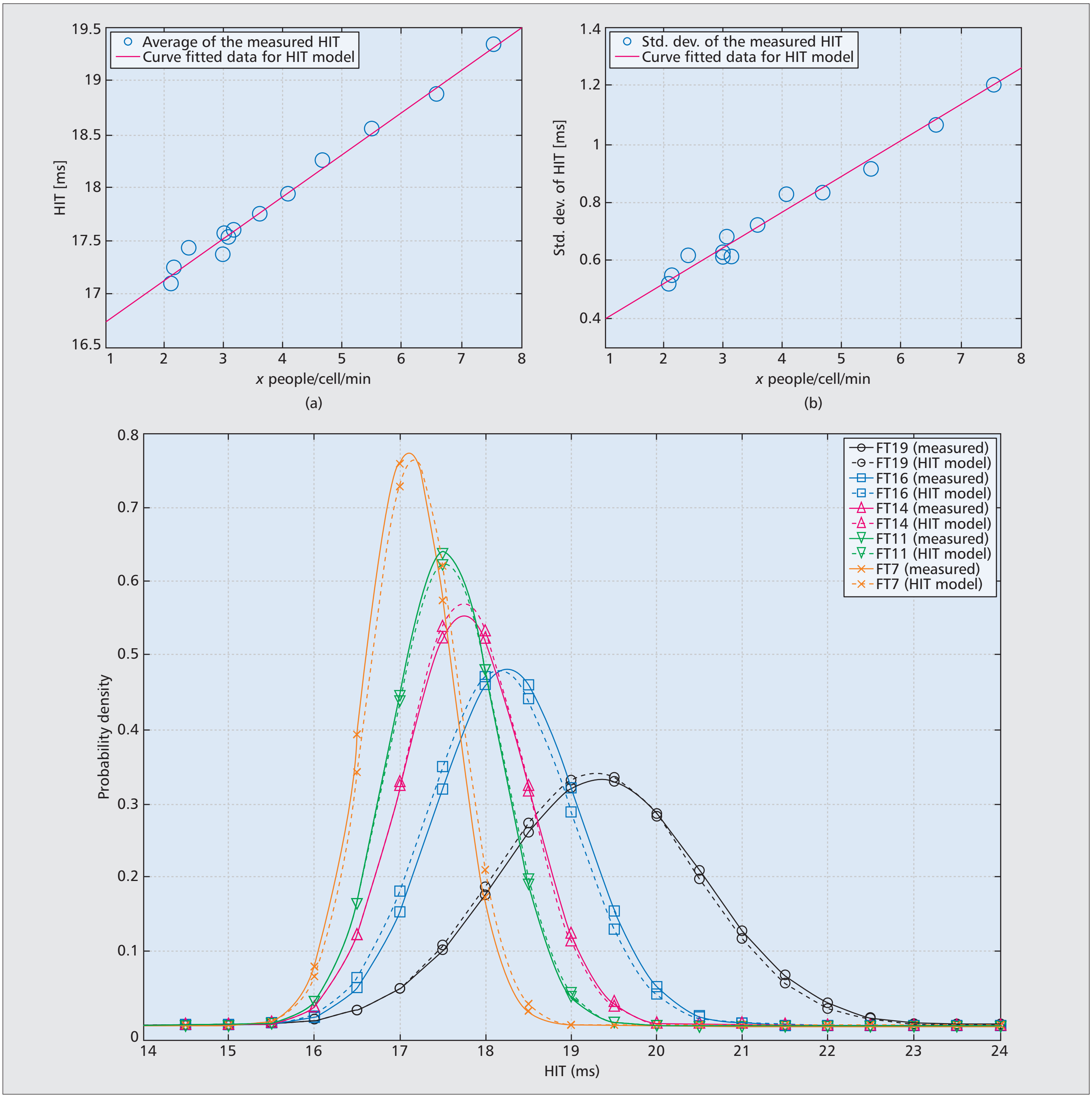

Figure 3. Analysis on HIT modeling: a) curve fitting method for average of HIT; b) curve fitting method for standard deviation of HIT; c) comparison of PDF between the HIT model and measured data.

As shown in Fig. 4a and Fig. 4b, RACH access delay is highly related to the load on RACH, which is represented by the number of preambles/s. If the number of preambles increase to a level that the dedicated RACH resources could not be accommodated, contention-based access is applied so that a RA attempt may collide with another attempt. This leads to an increased RA delay as shown in Fig. 4b. As RA delay depends on the RA probability, it starts to increase when the number of preambles/s is 80, which is the same point the RA probability decreases. This turning point varies with the detection miss probability (DMP) and the preamble split. As shown in Fig. 4c, RA delay linearly increases with the RACH scheduling period as an UE must wait for the RACH schedule. When the RACH period is short, the UE proceeds RACH steps for a brief time, hence RA delay is reduced. Preamble split determines the allocation of dedicated RACH resources. The UE will not experience collision of a sending preamble, since dedicated preambles are used for HO (i.e., contention-free RA). As the allocation ratio increases the RA delay decreases due to having plenty of resources for handover signaling. However this would result in a side effect that the UE on other cases of RACH (e.g., initial access of an idle UE and reestablishment after a radio link failure (RLF)) suffers increased RA delay. The standard value of preamble split is 1/8, and the RA delay drastically increases when the preamble split drops less than 1/16, as indicated by ⓐ and ⓑ in Fig. 4d, respectively.

Based on this analysis, LTE network configuration parameters can be optimized to reduce HIT and provide more reliable QoS. As increasing RACH load exerts a bad influence on HIT, it can be considered in network planning and eNB deployments. As QoS degradation of the mobile application is highly relevant to HIT, it is favorable to reduce both the length (i.e., duration) and the number of HITs. The length of HIT can be reduced by optimal configuration of RACH resources (e.g., RACH configuration and RACH preamble split) and HO parameters. The information required for RACH optimization includes the RACH information reported from UEs, and Physical RACH (PRACH) configuration exchanged between eNBs via X2 interface [8]. When the proposed HIT model is utilized for RACH optimization, it is expected to achieve higher performance in LTE HO. For example, the HIT information helps to decide how to increase RACH resources for HO (contention-free RA) when the HIT is longer than the target specified by the operator, or longer than ones neighbors' average HIT, which can be exchanged by X2 interfaces. The opposite decision is made when the delay of contention-based RA is in poor condition.

In addition, the proposed HIT model helps the decision of HO Operating Point (HOP) in Fig. 1c to achieve lower HIT and load balancing. HOP is determined by HO parameters defined in [9] and affects HO failure as well as QoS of an UE. HO failure is caused by a RLF and is classified into Late, Early, and Wrong Cell HO failure. When an UE is moving at a high velocity and HOP lag occurs due to strict HO conditions, RLF occurs in the S-eNB before HO (Late HO failure). Early HO failure occurs when the hysteresis parameter (*Hys*) and Time-to-Trigger (TTT) is too small. Thus, optimization of HO parameters is important to reduce HO delay as well as to avoid HO failure.

Some of the HO parameters that need to be optimized are shown in Fig. 1c, which describes a representative HO event. In Fig. 1c, *Mn* is the reference signal received power (RSRP) of the neighbor cell, *Ocn* is the cell specific offset of the neighbor cell, *Ms* is the RSRP of the serving cell, and *Off* is the offset parameter. An UE enters HO Event A3 when the condition $Mn + Ocn - Hys > Ms + Off$ is satisfied, and HO occurs when the condition holds for TTT. Using the HIT model $f_{T_{HIT}}(x)$, eNB increases its *Ocn* values and attracts UEs when the expected average HIT is small, and decreases it otherwise. This adaptive control of *Ocn* can be expressed by $Ocn = c(HIT_{ref} - HIT_{avg})$, where c is a non-negative coefficient, HIT_{avg} is the average HIT obtained from the HIT model, and HIT_{ref} is the reference value. HIT_{ref} can be set to meet a specific QoS requirement (e.g., the estimation value of HIT in Table 2 [7], an average HIT of neighbor cells, or a threshold for a specific real-time application).

The HO performance of LTE has shown great improvement over former mobile communication technologies. Especially, the packet loss problem has been eliminated due to packet forwarding features. The focus of this article is based on reducing the HIT which would help improve multimedia services. For example, ITU-T G.1080 [10] defines traffic error duration requirements for the video codec of H.262 to be less than 16 ms, where adaptive control of the investigated parameters could lead to an improvement in QoS during HO for such applications.

Component in Fig. 1	Description	Time [ms]
E2	Radio Synchronization to the T-eNB	1
E3	Average delay due to RACH scheduling period	2.5
E4	RACH Preamble	1
E5-E6	Preamble detection and transmission of RA response (Time between the end RACH transmission and UE's reception of scheduling grant and timing adjustment)	5
E7	UE Processing Delay (Decoding of scheduling grant and timing alignment)	5
E8-E9	Transmission of RRC Connection Reconfiguration and ACK	3.0
	Total HIT [ms]	17.5

Table 2. Delay analysis of RACH procedure components.

Conclusion

In this article, measurements and stochastic modeling of the key time parameters in LTE HO are presented. The contributions of this article include the following:
- The proposal of a field test methodology that enables estimation of LTE HO delay parameters.
- Evaluation of HO delay parameters as well as the overall HO performance on commercially deployed LTE networks, which is proven to increase proportionally with the number of users in a cell.
- The proposal of a system concept to reduce HIT using a developed stochastic HIT model.
- Analysis on HIT based on various RACH load and RACH configurations.

From the network testing results, HO delay and HIT characteristics are analyzed and some configurations to be optimized are presented. From the test results, it is found that HIT is normally distributed for a variable RACH load (people/cell/min), where the average HIT increases by 10 percent and the standard deviation of HIT increases by 89 percent when the RACH load increases from 2.103 to 7.539 (people/cell/min). Based on the field test results, it is shown that HO in LTE networks may result in a QoS degradation of real-time mobile applications (such as mobile video). Based on the analy-

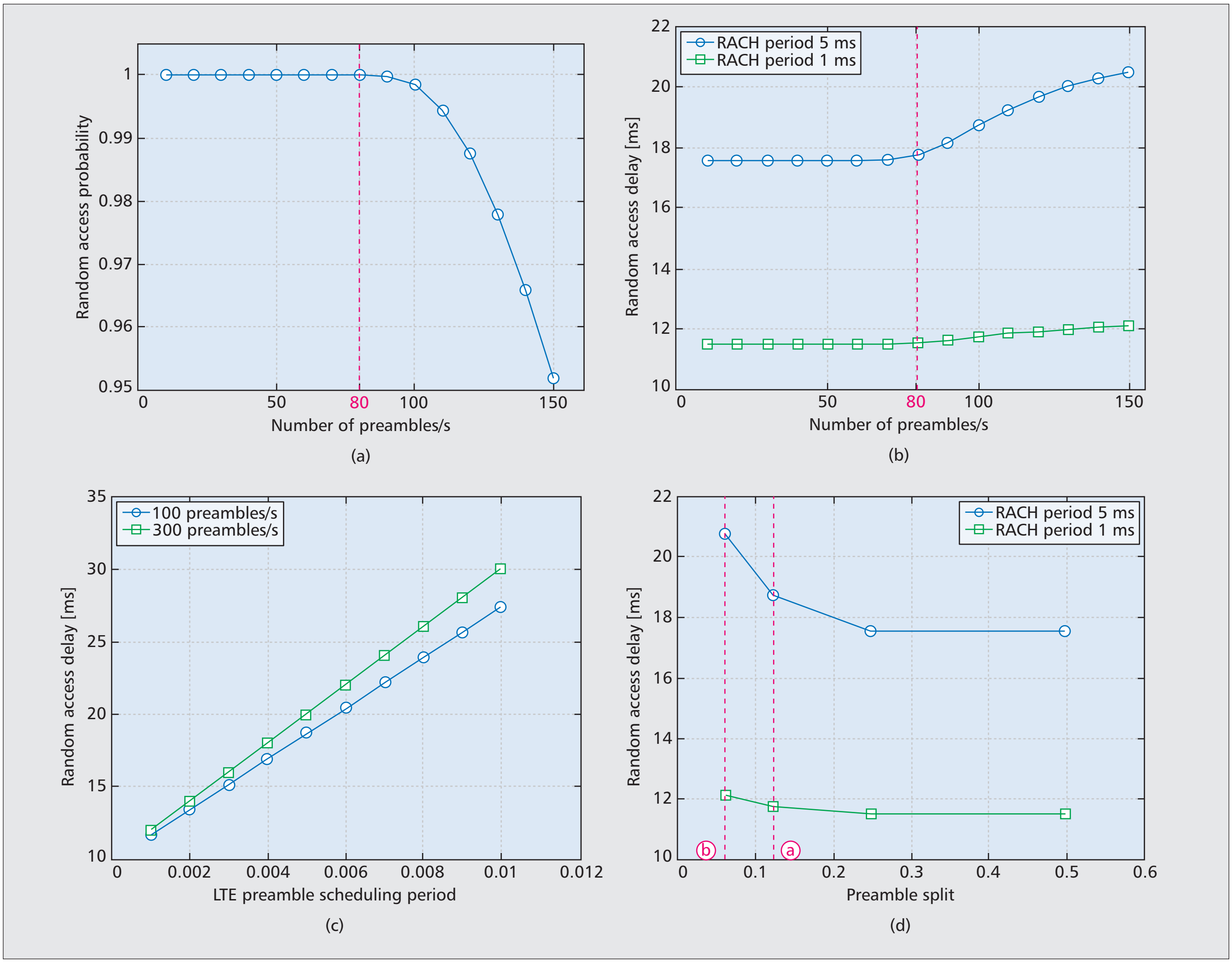

Figure 4. RA performance analysis: a) RA probability according to the number of preambles; b) RA delay according to the number of preambles; c) RA delay according to preamble scheduling period; d) RA delay according to preamble split.

sis conducted in this article, some remedies for improvement include:

1 Increase the number of gateways or to share the duties of them during the HO process

2 Optimize the RACH configuration and HO parameters

As the RACH configuration is a critical factor for reducing HIT, configuration parameters (e.g., preamble split and RACH load) and their impacts on HIT were analyzed. A large preamble split may provide performance improvements only for HO, but may result in detrimental effects on other RACH procedures, thus prudent design is needed. Analysis shows that RACH load is a key parameter that service providers need to consider, since the HIT will increase when the RACH load increases. To enable adaptive control, the practical effects on HIT are measured and stochastically modeled in terms of RACH load, which is represented by the number of UEs per cell.

Acknowledgments

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BIOGRAPHIES

DONGHYUK HAN (asrun@yonsei.ac.kr) received his B.S. degree from Yonsei University, where he is currently working towards a Ph.D. degree in the School of Electrical and Electronic Engineering. His research focuses on mobile networks and broadband QoS networking, including routing and handover protocols. Throughout his Ph.D. studies he worked on projects with Qualcomm, Samsung Electronics, LG Electronics, and ETRI, etc., in developing routing, scheduling, and handover protocols for improved QoS control.

SUNGJIN SHIN (ssjin@yonsei.ac.kr) received the B.S. degree from the School of Electrical and Electronic Engineering, Yonsei University, Seoul, Korea, in 2011. He is currently a Ph.D. graduate student in the School of Electrical and Electronic Engineering and a Research Member in the Communications and Networking Laboratory (CNL) of Yonsei University. His research interests include smartphones, mobile wireless networks, and horizontal/vertical handover.

HYOUNGJUN CHO (soarer@yonsei.ac.kr) received his B.S. degree from the School of Electrical and Electronic Engineering, Yonsei University, Seoul, Republic of Korea, in 2012. He is currently a graduate student in the School of Electrical and Electronic Engineering and a research member of the Communications and Networking Laboratory (CNL) at Yonsei University. His research focuses on SDN, NFV, mobile and IoT networks, and MPTCP.

JONG-MOON CHUNG (jmc@yonsei.ac.kr) received his B.S. and M.S. degrees from Yonsei University and Ph.D. degree from the Pennsylvania State University. Since 2005, he has been a professor in the School of Electrical & Electronic Engineering, Yonsei University. From 1997~1999, he served as an assistant professor/instructor in the Department of Electrical Engineering, Pennsylvania State University. From 2000~2006, he was with the School of Electrical & Computer Engineering, Oklahoma State University (OSU) as a tenured associate professor. He is an Editor of the IEEE Transactions on Vehicular Technology and Co-Editor-in-Chief of the KSII Transactions on Internet and Information Systems (TIIS).

DONGSEOK OK (dongseok.ok@samsung.com) received his Bachelor's degree and Master's degree in computer engineering from the Pusan National University in 2008 and 2010. He was a software developer at LG Electronics from 2010 to 2012. He is currently a software engineer at the Software R&D Center, Samsung Electronics, Korea. His research interests include test automation for wireless network and architecture and protocol for Internet of Things.

IKSOON HWANG (is73.hwang@samsung.com) received B.S., M.S., and Ph.D. degrees in electronics engineering from Yonsei University, Seoul, Korea in 1995, 1997, and 2001, respectively. From 2002 to 2007, he was a chief research engineer at LG Electronics Inc., Korea. From 2007 to 2011, he was a research engineer at TELECOM SudParis, France where he worked on model-based testing of communication protocols. He is presently a principal engineer at the Software R&D Center, Samsung Electronics, Korea. His research interest includes protocol validation, verification and testing, and the protocols and architectures of Internet of Things.

Contrace: A Tool for Measuring and Tracing Content-Centric Networks

Hitoshi Asaeda, Kazuhisa Matsuzono, and Thierry Turletti

ABSTRACT

Content-Centric Networks (CCNs) are fundamental evolutionary technologies that promise to form the cornerstone of the future Internet. The information flow in these networks is based on named data requesting, in-network caching, and forwarding, which are unique and can be independent of IP routing. As a result, common IP-based network tools such as ping and traceroute can neither trace a forwarding path in CCNs nor feasibly evaluate CCN performance. We propose "contrace," a network tool for CCNs (particularly, CCNx implementation running on top of IP) that can be used to investigate the round-trip time (RTT) between content forwarder and consumer, the states of in-network cache per name prefix, and the forwarding path information per name prefix. We report a series of experiments conducted using contrace on a CCN topology created on a local testbed and the GEANT network topology emulated by the Mini-CCNx emulator. The results confirm that contrace is not only a useful tool for monitoring and operating a network, but also a helpful analysis tool for enhancing the design of CCNs. Further, contrace can report the number of received interests per cache or per chunk on the forwarding routers. This enables us to estimate the content popularity and design more effective cache control mechanisms in experimental networks.

INTRODUCTION

Content-Centric Networks (CCNs) [1] or Named-Data Networks (NDNs) [2] are promising approaches toward the future Internet. In a CCN, publishers provide content through the network, and receivers retrieve content by name. In this network architecture, routers forward content requests by means of their forwarding information bases (FIBs), which are populated by "name-based routing protocols." A CCN enables receivers to retrieve content from an "in-network cache": if a router receiving a content request has the requested named data in its cache, it forwards the data toward the receiver without forwarding the content request to the publisher. As such, implementing this new communication technology requires innovative ideas and research in terms of "naming," "routing," and "caching." To evaluate the architectures and protocols, researchers validate their proposals by running actual implementations, such as the CCN prototype CCNx [3], on testbeds such as PlanetLab [4]. However, because requesting and transmitting named data is unique and can be independent of IP routing, common IP-based network tools such as *ping* [5] and *traceroute* [6] are unable to evaluate CCN protocols and architectures. Furthermore, these tools do not allow the state of the in-network cache to be discovered. Hence, we need a network tool that can assist troubleshooting activities in CCNs while helping the design of in-network caching and name-based routing mechanisms.

In this article we discuss the design and implementation of *contrace*, an active network measurement tool for investigating the path and caching condition in CCNs. We assume that such a CCN is formed by a CCNx implementation running on top of IP. Contrace consists of a user program and a forwarding daemon implementation. Contrace "Query" messages are invoked by the contrace user and forwarded by CCN routers toward a specified source or caching routers. Contrace "Response" messages are initiated by a router holding the requested content (i.e. cache) or the publisher, and forwarded toward the contrace user. Thus, contrace facilitates the tracing of a routing path and provides additional information such as the round-trip time (RTT) between routers along the path. In addition, contrace identifies the state of the cache, such as the access count and lifetime of cached content, on a router. Obtaining such information allows us to estimate content popularity as a means of optimizing in-network caching. Furthermore, contrace implements policy-based information provisioning that enables administrators to "hide" secure or private information, but does not disrupt the forwarding of messages. This policy-based information provisioning reduces the deployment barrier faced by operators in installing and running contrace on their routers. We developed and implemented contrace and conducted experiments in a CCN topology created on a local testbed and the GEANT network topology emulated by the Mini-CCNx [7]. The results show that contrace provides the forwarding path information per name prefix, and facilitates investigation of the content popularity and the potential cache capacity in the network.

Hitoshi Asaeda and Kazuhisa Matsuzono are with the National Institute of Information and Communications Technology (NICT).

Thierry Turletti is with INRIA.

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Measuring Performance in CCNs

Overview

CCNs enable consumers (i.e. receivers) to obtain information or content without continuous end-to-end communication channels between hosts. CCNs focus on information dissemination in the network: how consumers retrieve content by name using "interest" packets, and how publishers (i.e. content owners) provide content through the network using "content object" packets. CCNs advocate receiver-driven communication using name prefixes for routing. In this paradigm, consumers initiate communication by sending a request (interest) for specific content, and CCN routers forward the interest toward the content publisher responsible for the requested content. A FIB is a lookup table that is used to determine the incoming interfaces (called incoming Faces) for all content. The FIB entries comprise a name prefix and incoming Face pairs. Each CCN router also maintains a pending interest table (PIT), a lookup table containing name prefixes, and outgoing Faces used to forward received content.

When a CCN router receives an interest, the router examines the content to determine whether it is cached in its local content store (CS). If the named content is in the router's CS, the router forwards the cached content out via the Face on which it arrived. If the named content is not in the router's CS, it searches its PIT to determine whether an interest for the same named content has already been processed. If the same named content is in the router's PIT but the Face receiving the interest has not been set, the router updates the PIT entry by adding the Face as the outgoing Face and discards the interest. If the named content is not in the router's PIT, the router creates a new PIT entry, and forwards the interest via the incoming Faces specified in its FIB.

If any router along the path toward the publisher does not have the requested content in its cache, the interest packet will finally reach the publisher. On receiving an interest, the publisher forwards the content object to its downstream router. Each CCN router on the path that receives the interest forwards the data via the outgoing Faces, stores the forwarded content in its CS if needed, and deletes the corresponding PIT entry.

Motivation

CCN technology changes the data retrieval mechanism from a host-centric model to a content-centric model. Name-based routing paths for forwarding interests and data are organized based on the name prefixes of the content and do not depend on IP routing. The content forwarder is not always the publisher (i.e. content owner); any router or node along the path may become the content forwarder if it has the requested content in its cache. Conceptually, CCN also has high affinity with multipath routing, as it can perform packet-by-packet forwarding, although there are optimization difficulties [8]. The forwarding paths can be determined by the unique CCN strategy and/or routing layer [9]. Although there are standard methods for measuring the performance of IP networks [10] and common IP-based network tools such as ping [5] and traceroute [6], they cannot be directly used to measure or trace the forwarding path in CCNs. Moreover, in-network caching is an innovative core function of CCNs, whereas there are currently no mechanisms to investigate the cache state in routers and the content distribution in the networks.

A network tool for CCNs could be used as an analytical tool to investigate fine-grained cache information such as the number of interests received per cache, and to design forwarding and caching strategies as well as new routing protocols in CCNs.

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Performance Metrics and Forwarding Path Information

First, it is necessary to identify which performance metrics should be measured in CCNs. According to Pentikousis *et al.* [11], CCN performance metrics can be subdivided into "traffic" and "system" metrics. The most important traffic metrics for consumers are the throughput and the RTT between content forwarder (i.e. forwarding router or publisher) and consumer. Although throughput depends on the transport protocol, RTT is the natural metric for charactizing the end-to-end network conditions, because it has no application or transport protocol dependency.

In general, although traffic metrics are important for consumers, publishers and researchers are more interested in system metrics such as memory usage and signaling overhead, as these allow the efficiency of each networking protocol or architecture to be evaluated. In CCNs, however, system metrics that are firmly linked to in-network caching are important to all parties. This is because in-network caching affects the overall content retrieval performance, and the use of a large amount of cache may affect the memory usage of the caching router, which would reduce the forwarding performance.

Furthermore, unlike IP-based routing, both the publisher and the routers are capable of forwarding content in CCN. While the consumer does not generally need to know which content forwarder is transmitting the content to the consumer, operators need to identify the content forwarder and observe the forwarding path information per name prefix for troubleshooting and monitoring networks.

Related Work

Current CCNx components [3] incorporate a command called *ccndstatus*. Ccndstatus shows information about Faces and content store (CS) information, such as neighbor routers addresses, cache list, lifetime, and expiration time per cache in a CCN router. However, ccndstatus does not provide any information about transmission delay or other traffic metrics. Ccndstatus also requires the user to access all routers along the path to examine the in-network cache. Examining all potential forwarding routers for some specific content by accessing each router is not a practical solution for troubleshooting or monitoring networks.

In addition, network tools should allow researchers to investigate the performance of content retrieval or design algorithms/mechanisms for in-network cache or name-based routing in CCNs. Unfortunately, in many cases researchers are forced to use simulators for their

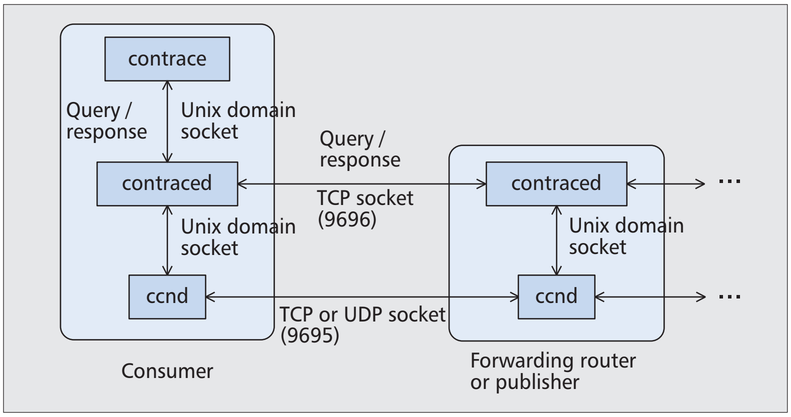

Figure 1. Socket communications among *contrace* command, *contraced*, and *ccnd*.

experiments because there are no tools to satisfy their needs. For instance, Rossi and Rossini [12] were forced to use simulations to analyze the caching performances of:
- Homogeneous cache sizing, in which the CS spaces of all routers are the same.
- Heterogeneous cache sizing, in which a given amount of memory resources are allocated across routers by considering graph-related centrality metrics.

It is preferable to investigate such strategies using the actual running codes on a real (or emulated) network.

Design and Implementation

Concept

Traceroute is a useful tool for analyzing the routing conditions in IP networks as it provides intermediate router addresses along the path between source and destination and the RTT for the path. However, as outlined later, this IP-based network tool cannot trace the name prefix paths used in CCNs. Furthermore, given a source-rooted forwarding path per name prefix, tracing from a source (i.e. forwarding router or publisher) to a consumer is difficult, because we do not know which branch of the source tree the consumer is on. Additionally, it is not feasible to flood the entire source-rooted tree to find the path from a source to a consumer. Conversely, walking up the tree from a consumer toward a source is feasible because CCN routing protocols know the upstream router for a given source. Tracing from a receiver to a source can involve only the routers on the direct path. This idea is inspired from the design of the IP multicast traceroute facility [13].

To determine the state of in-network caching under CCN, the efficiency and performance of in-network caching can be measured by the following metrics for CS in the router:
- Size of the cached content.
- Number of chunks for the content.
- Number of accesses (i.e. received interests) per cache or chunk.
- Lifetime and expiration time per cache or chunk.

A fifth metric of the forwarding router or publisher's IP address is included among the states of in-network caching.

We propose "contrace," a network tool for active measurement in CCNx-based CCNs. Contrace provides:
- The RTT between content forwarder (i.e. forwarding router or publisher) and consumer.
- The states of in-network cache per name prefix.
- The forwarding path information per name prefix.

Contrace is also a useful analysis tool to aid in the future design of CCNs. It can report the number of received interests per cache or chunk on the forwarding routers, indicating the popularity of the content. By comparing this number across all forwarding routers with contrace, we can illustrate the content popularity and evaluate different caching strategies in experimental networks.

The contrace implementation consists of the contrace command and the contraced daemon implementation running on a CCNx router (*ccnd*). Figure 1 shows the socket communications between the contrace command, contraced, and ccnd. Ccnd, which is a component of CCNx, is a daemon that forwards and caches content objects. The contrace command is invoked by specifying the name prefix of the content and/or the publisher or forwarding router. It initiates the request (i.e. creates the Query message) to obtain forwarding path and cache information. Contraced running on the contrace user receives the request and communicates with local ccnd. Contraced retrieves cache and other information from local ccnd via a UNIX domain socket, and forwards the request to the neighbor contraced running on the upstream neighbor router (if it is not the content forwarder or the specified end point). When the request reaches the content forwarder, contraced on the forwarder transmits the reply (by sending the Response message) to contraced running on the downstream neighbor router, and the reply is forwarded back to the contrace user.

Contrace Command

The contrace command enables the contrace user to investigate the forwarding path based on the name prefix of the content (e.g. ccnx:/ news/today/). The usage of contrace is as follows (the descriptions of all the options are shown in Table 1):

Usage: contrace [–R] *name_prefix* [—nocache] [–h *content_forwarder*] [–g *gateway* [–g *gateway* …]] [–s *skip_node* [–s *skip_node* …]] [–p port] [–w *wait_time*]

The name prefix is a mandatory option for the contrace command. If the contrace user does not know the name prefix of the content to be specified, s/he can specify "ccnx:/," the default name prefix. If this default name prefix is given, the user will receive all cache information from the upstream neighbor (adjacent) router(s).

It is also possible to specify both the name prefix of the content and use the "–h" option with the content forwarder, which could be the publisher or forwarding router for the content (e.g., ccnx:/news/today/ –h pub.example.com). If both name prefix and content forwarder are specified, the query message is forwarded in accordance with the IP routes, which could be independent of

Option	Description
[-R] *name_prefix*	Mandatory option. Name prefix of the content (e.g. ccnx:/news/today/) the *contrace* user wants to trace. "ccnx:/" is the default name prefix. The -R option requires an exact match for the name prefix; otherwise, a partial match is allowed.
––nocache	No cache information is requested. Only the router names and RTT along the path are returned.
-h *content_forwarder*	Node name or IP address of the publisher or forwarding router (e.g. -h pub.example.com). If this is specified, Query is forwarded in accordance with IP routing path (without looking up CCN FIB).
-g *gateway* [-g *gateway*] …	Node name or IP address of gateway router(s) to be passed through. Similar to source routing. If this is specified, Query is forwarded in accordance with IP routing path (without looking up CCN FIB). This option must be used with the -h option (an error is returned if this is not specified).
-s *skip_node* [-s *skip_node*] …	Node name or IP address of the forwarding router to be skipped. If the specified router receives a Query, it forwards the Query to the upstream router regardless of its cache contents. This option must not be used with the -h option (an error is returned if both are specified).
-p *port*	TCP port number for Query/Response. Default: 9697.
-w *wait_time*	Timeout value (in seconds) that the *contrace* user waits for a Response. After the timeout, the *contrace* user silently discards the Response. Default: 4 s.

Table 1. Description of contrace command options.

the CCN FIBs. The contrace user would then obtain the forwarding path information and cache information for the specified name prefix from the specified content forwarder. If the default name prefix and content forwarder are specified, the contrace user obtains all cache information from the specified content forwarder.

In addition to the RTT between content forwarder and consumer, when the contrace command specifies the "–h" option and the IP address of the content forwarder (or "–g" option and the IP address of gateway), the RTTs of other alternative paths for the specified name prefix are measured. This is useful for troubleshooting and in the design and evaluation of routing protocols, because the output potentially indicates a better path for the named content.

When the contrace user only wants to check the forwarding path information (i.e. without cache information) and the RTT between content forwarder and consumer, s/he can utilize the "—nocache" option.

After the contrace command is invoked, it waits for the reply from the upstream router, extracts the returned response messages, and shows the information on the standard output. To cancel the request when the message packets are lost during transmission, or if there is no route for the specified name prefix or publisher/router, contrace forces the session to close after the timeout. Following the timeout, even if the contrace user receives a response, it is discarded. The "–w" option specifies the timeout value (in seconds) that the contrace user will wait for the response. The default timeout value is 4 s.

CONTRACED FORWARDING DAEMON

Contraced is a daemon that transmits query and response messages. As seen in Fig. 1, when contraced receives a query message from a contrace program, it communicates with local ccnd via a UNIX domain socket to retrieve information about the upstream router(s) from its FIB. For ccnd to communicate with contraced, a small patch must be applied to ccnd. If the local ccnd is not the content forwarder for the request, contraced forwards the query message to the neighbor contraced running on its upstream neighbor ccnd over TCP (currently using the 9696 port). When the request reaches the content forwarder, contraced on the router or publisher retrieves the cache information from its CS. The response message is then transmitted back to the neighbor contraced that had forwarded the query message. The response message is forwarded toward the contrace user in a hop-by-hop manner.

Contraced can transmit messages to any node running contraced without referring to the CCN FIB. When contraced receives a query message including the "-h" option record (mentioned in the previous section), it forwards the query message to the specified address in accordance with the IP routes. This implementation enables the contrace user to investigate the alternative route or cache information for the specified name prefix.

As introduced earlier, if it is difficult to optimize the transmission timing or select the best route, the CCN strategy or routing layer can use different Faces defined in multipath routes to transmit a request for specific content. To compare the performance of content fetch from the different paths, contraced immediately and simultaneously transmits the query message to these Faces, whenever multipath routes are configured for the name prefix.

MESSAGES AND RECORDS

Contrace uses two message types: Query and Response. Both messages are encoded in Type, Length, and Value (TLV) format. The Type field is eight bits long, the Length field is 16 bits long, and the length of the Value field is variable. The Query and Response Type values are 0x01 and 0x02, respectively. The Query message is forward-

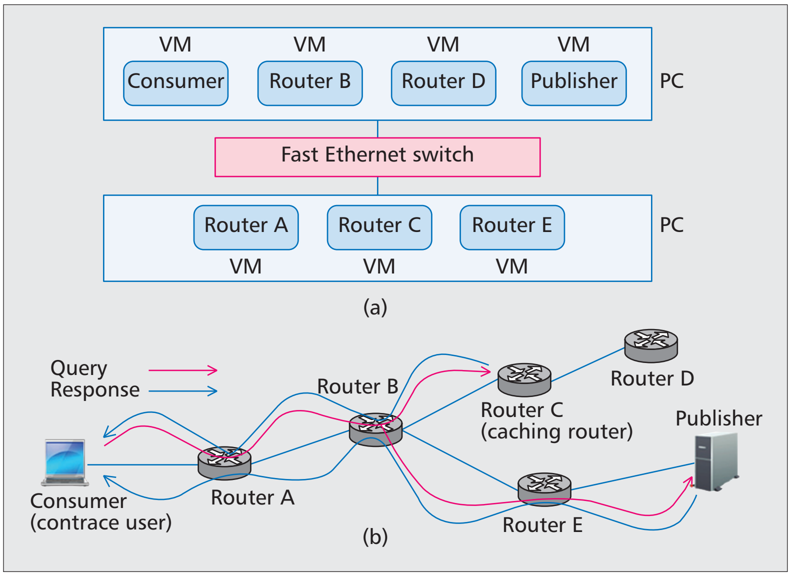

Figure 2. Experimental configuration of the local testbed: a) configuration of seven VMs on two PCs; and b) CCN topology formed by the VMs.

ed in a hop-by-hop manner. When it reaches the content forwarder, the content forwarder turns the Query message into a Response message by changing the Type field value from 0x01 to 0x02.

When Query or Response messages are transmitted, contraced appends one or more "records" to the Response message. The following eight records are defined and transmitted with the same TLV format:

1. Contrace user address
2. Name prefix
3. -h option
4. -g option
5. -s option
6. Timestamp
7. Responder address
8. Cache status

Records 2–5 are appended by the contrace command only if specified. The timestamp record is appended by every contraced along the path upon receiving the Query, and the responder address (and cache status records if requested) are appended by the content forwarder. The contrace user address and responder address records are appended to the Query and Response, respectively. A timestamp record that specifies the forwarding router or consumer's IP address and UNIX time on the local machine is mandatory for the content forwarder.

Response Policy and Configuration

Although contrace gives excellent troubleshooting cues, some network administrators or operators may not want to disclose everything about their network to the public, or may wish to securely transmit private information to specific members of their networks. Contrace provides policy-based information provisioning allowing network administrators to specify their response policy in a configuration file, *contrace.conf*, for each router.

We define the following three options that can be specified in the contrace.conf file for when the router receives a Query message:
- Whether "-h" option is allowed or rejected.
- Whether "-g" option is allowed or rejected.
- Whether cache information is disclosed, partially disclosed (i.e. except the request specifying the default name prefix (i.e. ccnx:/)), or not disclosed at all.

On the other hand, we entail that each router runs contraced and does not disrupt forwarding contrace Query and Response messages. When a Query message is received, the router appends the timestamp record and forwards the request to the upstream router toward the content forwarder, but can hide other information according to the policy configuration.

Cases will arise in which a router along the path does not support contraced. In such cases, a downstream router that supports contrace will reply to the contrace client with a message indicating that the upstream router does not support contrace. Further, when an upstream router rejects a Query message, its downstream router

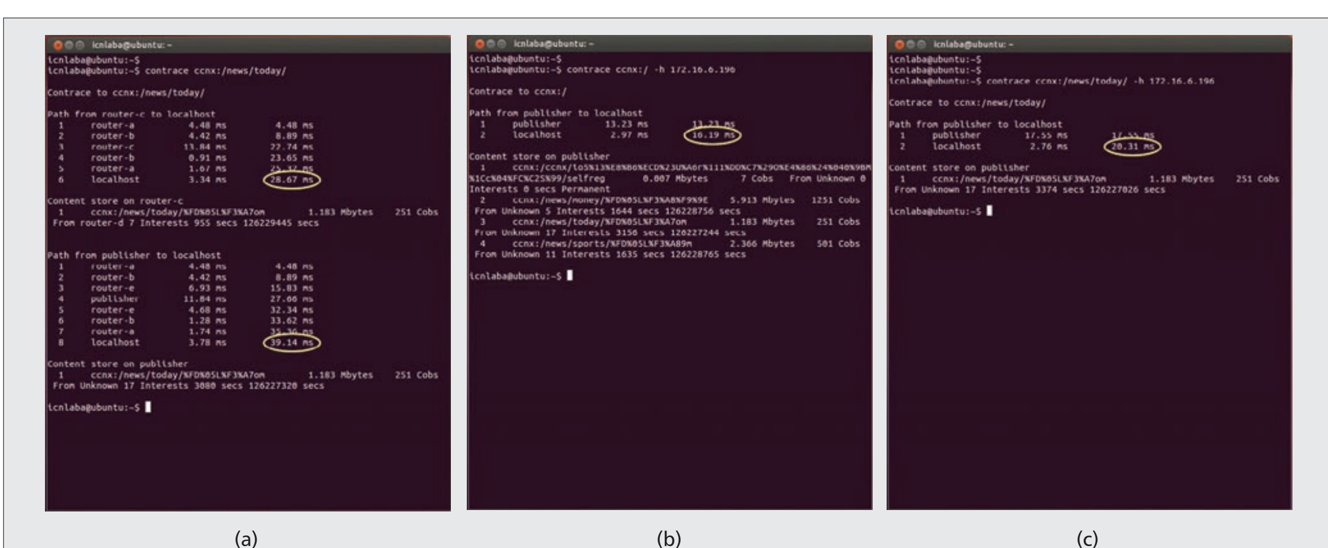

Figure 3. Output of contrace command with different options: a) name prefix; b) default name prefix and content forwarder; and c) name prefix and content forwarder.

will reply to the contrace client with a message indicating that contrace Query is prohibited.

Contrace in Action

First, we deployed seven VMs on two physical PCs. Both the VMs and the PCs ran the Linux operating system (Ubuntu Desktop 12.04 LTS 64-bit version). The VMs communicated with each other via a fast ethernet switch, as shown in Fig. 2a. CCNx 0.8.2 and contrace components (contrace command and contraced) were installed on each VM. The ccnds on the VMs formed the logical CCN topology shown in Fig. 2b.

Figure 3 shows the output of contrace commands input by a consumer (descriptions of all outputs are listed in Table 2). We verified that contrace messages were forwarded in a hop-by-hop manner, and that each forwarding router appended its own timestamp record. Further, when the contrace Response message returned to the contrace user, the contrace program calculated the one-way delays (OWDs) between each router and the RTT from the contrace user to the content forwarder initiating the Response. To ensure the OWDs shown in the output were correct, the Network Time Protocol (NTP) [14] was installed on each router to synchronize their internal clocks. Note that even if the routers' internal clocks are not synchronized, the RTT measured between the contrace user and the content forwarder (yellow oval in Fig. 3) is always correct because both start and end times are given by the contrace user.

In this experiment, router B was configured as a multipath for receiving the content of ccnx:/news/today/. As shown in Fig. 3a, the Query message was transmitted from router B in two directions (to routers C and E) simultaneously. Two Response messages (transmitted from router C and the publisher) were then received by the contrace user. Another observation from Fig. 3a is that router C obtained the cache of ccnx:/news/today/ from router D, not the publisher. From Fig. 3b we can observe that the publisher (with IP address 172.16.6.196) cached four contents, while the output of Fig 3c indicates only ccnx:/news/today/ in its cache information, because the contrace command explicitly specified the prefix.

Next, we investigated the caching performance of CCNx using contrace with a realistic topology on an emulator. Rossi and Rossini [12] assessed the cache hit probability and path stretch $d / |P| \in [0, 1]$, where d is the hop count of the data and $|P|$ is the hop count from the consumer to the publisher (i.e. without caching). We built the GEANT network topology, which consists of 27 nodes, using Rocketfuel [15] on top of the Mini-CCNx emulator [7]. The publisher was assumed to have 100 contents. These were characterized by a Zipf probability distribution with the value of the Zipf exponent $\alpha = 1.25$. Each consumer sends 2500 interests in total, based on the Zipf popularity setting. Under the homogeneous cache sizing approach, each router has equal CS space to satisfy received interests with a probability of about 50 percent. In the heterogeneous approach, the degree centrality (DC) metric was adopted ($DC(n)$ is defined as the number of links incident upon a router n) for its simplicity and good allocation criterion (i.e. the greater the DC of a router becomes, the more CS space the router should have) [12]. We allocated a CS space that can satisfy the received interests with a probability of about 80 percent to each router with $DC \geq 4$.

By running contrace at the consumer, we assessed path stretch, because contrace traces the path from the consumer to the router/publisher having the requested data. Figure 4a shows the measured path stretch, and compares the homogeneous and heterogeneous approaches at four consumers randomly selected. These results illustrate the improvement in path stretch in the heterogeneous case. Furthermore, because contrace can show the number of received interests per cache or chunk on a router, the content popularity (i.e. the number of accesses for each content) can be measured (as shown in Fig. 4b for the publisher). It is also clear that under the heterogeneous approach, the access number of popular content (e.g. content-index 1, 2, and 3) decreases with the improvement in path stretch. We conclude that contrace facilitates investigation of the performance of the CCNx protocol, routing/caching strategy in testbeds and/or emulated networks, and the design of future CCN architectures.

Category	Description	
Requested prefix (starting from "Contrace to prefix name")	Name prefix of content asked for by *contrace* user	
Routing information (starting from "Path from the content forwarder (publisher or forwarding router) to contrace user")	Col. 1	Sequential number of one-way hops on the path
	Col. 2	Node name or IP address on the path
	Col. 3	One-way delay from previous node to this node
	Col. 4	Sum of one-way delays from *contrace* user to this node. If this node is *contrace* user, this value indicates the RTT between *contrace* user and the content forwarder.
Content store (CS) information (starting from "Content store on the content forwarder")	Col. 1	Sequential number
	Col. 2	Name prefix of the cached content
	Col. 3	Size of the cached content
	Col. 4	Number of content objects for the cached content
	Col. 5	Node name or IP address from which the cached content is provided
	Col. 6	Number of received interests
	Col. 7	Lifetime of the cached content
	Col. 8	Expiration time of the cached content

Table 2. Description of contrace command outputs.

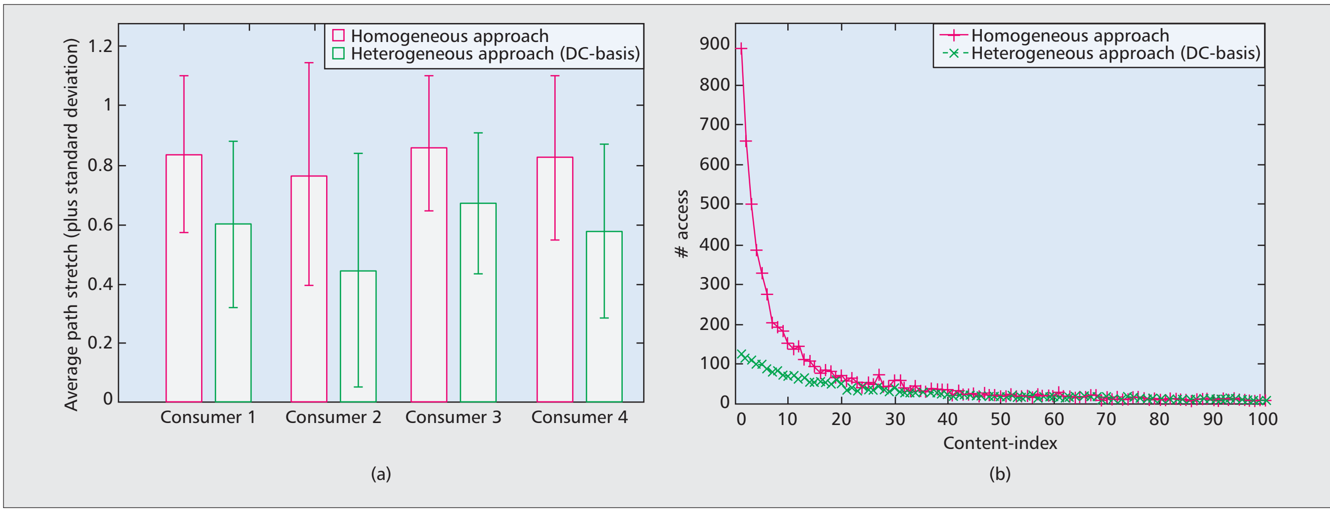

Figure 4. a) Average path stretch measured at the four consumers; b) content popularity measured at the publisher, in the homogeneous approach and the heterogeneous approach measured with contrace running on the GEANT topology emulated by Mini-CCNx.

Conclusion and Outlook

The novel features introduced by the content-centric communication model necessitate specialized network tools that do not currently exist. In this article we proposed "contrace," a network tool for CCNs that can be used to investigate:

- The RTT between content forwarder (i.e. forwarding router or publisher) and consumer.
- The state of an in-network cache per name prefix.
- Forwarding path information per name prefix.

Contrace facilitates troubleshooting of CCNs and can help in the design and the optimization of routing protocols and routing/caching strategies.

We implemented contrace on a CCNx router, and showcased this tool on a testbed and on a large emulated network. Our experiments demonstrated that contrace provides forwarding path information per name prefix, and enables the measurement of CCN traffic and system performance metrics. We also investigated cache information for each name prefix, the content popularity, and the potential cache capacity in experimental networks.

In future work we plan to add new functionalities on contrace, to obtain the cache hit probability at routers along a path, and to visualize characteristics of the dispersal of target content (i.e. where/when requests for target content are concentrated). This will promote the extensive analysis of caching/routing strategies, and may engender a fundamentally new design approach.

Acknowledgement

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Biographies

Hitoshi Asaeda (asaeda@nict.go.jp) is a planning manager at Network Research Headquarters, NICT, and a principal investigator (PI) of the ICN project in NICT. He was with IBM Japan, Ltd. from 1991 to 2001. From 2001 to 2004 he was a research engineer specialist at INRIA. He was a project associate professor at the Graduate School of Media and Governance, Keio University, where he worked from 2005 to 2012. He chairs the ICN WG of Asia Future Internet Forum (AsiaFI). He holds a Ph.D. from Keio University.

Kazuhisa Matsuzono is a researcher at the Network Architecture Laboratory, NICT. He received a Ph.D. degree from Keio University in 2012. He was a post-doctoral fellow at INRIA in 2013. His research interests include transport protocols for multimedia flows, network coding, and information centric networks.

Thierry Turletti is a senior research scientist with the DIANA team at INRIA. He received the MSc and the Ph.D. degrees in computer science from the University of Nice - Sophia Antipolis, France. His current research interests include software defined networking, reproducible network evaluation, and wireless networking. He serves as an associate editor of the Hindawi journal *Advances in Multimedia* and the Springer journal *Wireless Networks*.

SERIES EDITORIAL

RADIO COMMUNICATIONS: COMPONENTS, SYSTEMS AND NETWORKS

Amitabh Mishra *Tom Alexander*

Primarily due to the integration of concepts from networks as diverse as cognitive radio, 5G cellular, sensors, and WLANs, coupled with the desire to provide seamless connectivity across multiple technologies and the proliferation of the number of mobile applications and attached devices, modern wireless networks are becoming (if not already) quite complex and heterogeneous. An emerging communication paradigm encompasses device-to-device multi-hop interactions in a dense urban environment, performing traffic offloading and relaying from one network to the other. Examples include cellular to WLAN handoffs and vice versa, and multi-tier cellular networks. These networks may even belong to different operators.

In emerging wireless networks a call or connection may have to traverse more than one link from its origin to reach a base station (BS). The same holds true at the destination. For example, assume that a call originates on a cognitive link, but the primary user (PU) arrives sooner than the call can complete. As a result, the call will need to be switched to another available link provided a backup link is available. Otherwise, the call may have to be dropped, or some other option must be pursued.

A multiplicity of options exist. If the PU does not return, then the call may move to a mobile receiver via a relay, assuming that relaying is the only means that has been provisioned for mobile to mobile communication, and there is such a receiver for the specific session in the neighboring small cell. It is possible that there is no such receiver; in this case, the system has no choice but to relay the call to a neighboring WLAN, if one is available in the neighborhood.

In addition to WLAN offloading, there may be other options for traffic offloading: for example, utilizing the presence of femto-cells or small cells, or other tiers of a cellular network. If none of these options is available and there is no return of the PU, then the user may alternatively transmit toward a base station directly, or by relaying through a neighboring small cell. Similar options need to be pursued at the terminating end-point as well. Thus, in the near future a user will have more than one choice for mobile network connectivity. The offloading decisions, however, will be made by network operators considering factors such as cost, traffic volume, terminal interfaces, and interference (SINR) among cells in the neighborhood.

As you can guess, traffic offloading, multi-hop relaying, and mobile handoffs from one network to the other will take place in the presence of all sorts of interference and impairments. To facilitate reliable communication in such an environment, we need to research and develop novel interference and impairment mitigation techniques.

In this issue of Radio Communications, we include an article, "The Sector Offset Configuration Concept and Its applicability to Heterogeneous Cellular Networks," which focuses on interference mitigation techniques used when a call moves from a small cell to a serving macro-cell through multiple handoffs. As discussed above, femto-cells are deployed indoors in large numbers today, and small cells (pico- or micro-cells) are deployed outdoors to interwork with macro-cells to provide enhanced coverage.

In general, small cells are configured to transmit on a dedicated frequency carrier, which limits the number of channels that small cells and macro-cells can access. A better solution is to utilize co-channels that are available to both the small and macro-cells. However, this co-channel solution introduces additional cross-tier interference that affects coverage, capacity, and handoff management. Recently, enhanced inter-cell interference coordination (ICIC) techniques have been proposed to overcome cross-tier interference, such as carrier aggregation (CA) in frequency domain, and the almost blank subframe (ABS) approach in the time-domain. The article describes both approaches in detail along with their limitations and effectivenesss.

In this article, authors present a novel technique for cross-tier interference mitigation based on a sector offset configuration that significantly improves cell-edge user throughput without affecting the frequency reuse factor. They show via simulation that the proposed technique provides an improvement to ICIC between small and macro-cells, particularly for users of small cells who are located at the edge of the coverage area. The improvement of ICIC additionally leads to an improvement in the handover performance for both single and multi-carrier cellular networks.

In future issues of this Series, we look forward to bringing you similar timely articles from our community of authors covering emerging trends in wireless communications R&D. We also thank our readership for their time and attention.

BIOGRAPHIES

THOMAS ALEXANDER [M] (talexander@ixiacom.com) is a senior architect at Ixia. Previously, he has worked at VeriWave Inc. (acquired by Ixia), PMC-Sierra Inc, and Bit Incorporated (acquired by PMC-Sierra), and prior to that was a research assistant professor at the University of Washington. He has been involved in various aspects of wired and wireless networking R&D since 1992, in the areas of ATM, SONET/SDH, Ethernet, and (since 2002) wireless LANs. He is also active in standards development, and has served as editor of IEEE 802.3ae, chief editor of IEEE 802.17, and technical editor of IEEE 802.11. He received his Ph.D. degree from the University of Washington in 1990.

AMITABH MISHRA [SM] (amitabh@cs.jhu.edu) is a faculty member at the Information Security Institute of Johns Hopkins University, Baltimore, Maryland. His current research is in the area of cloud computing, data analytics, dynamic spectrum management, and data network security. In the past he has worked on the cross-layer design optimization of sensor networking protocols, media access control algorithms for cellular-ad hoc interworking, systems for critical infrastructure protection, and intrusion detection in mobile ad hoc networks. His research has been sponsored by NSA, DARPA, NSF, NASA, Raytheon, BAE, APL, and the U.S. Army. In the past, he was an associate professor of computer engineering at Virginia Tech and a member of technical staff with Bell Laboratories working on the architecture and performance of communication applications running on the 5ESS switch. He received his Ph.D. in electrical engineering from McGill University. He is a member of ACM and SIAM. He has written 80 papers that have appeared in various journals and conference proceedings, and holds five patents. He is the author of a book, *Security and Quality of Service in Wireless Ad hoc Networks* (Cambridge University Press,2007) and a technical editor of *IEEE Communications Magazine*.

The Sector Offset Configuration Concept and Its Applicability to Heterogeneous Cellular Networks

David López-Pérez, Holger Claussen, and Lester Ho

ABSTRACT

Wireless data traffic is increasing in an exponential manner, and thus vendors and operators are looking for cost-effective solutions to meet user equipment demands. In this article a joint optimization of the deployment and operation of small cells and macrocells is presented through the novel concept of sector offset configuration, which helps to significantly improve user equipment performance. The proposed sector offset configuration complements Release 10 enhanced intercell interference coordination, and further increases its interference mitigation capabilities. Simulation results for a Long Term Evolution network show that compared to a network that uses standardized Release 10 almost blank subframes, the proposed configuration can improve cell-edge user equipment performance by 50 percent. The proposed configuration can mitigate handover failures by 46 percent, since handovers start at a higher signal quality due to alleviated interference conditions. These benefits come at the cost of an increased number of antennas at the macrocell site to achieve the sector offset configuration.

INTRODUCTION

Recent traffic projections for North America show that data traffic is set to grow exponentially over the coming years, and that wireless data traffic is increasing the most rapidly. As a result, vendors and operators face the challenge of meeting user equipment (UE) wireless data traffic demands in a cost-effective manner. This requires reusing the available spectrum as much as possible in an efficient manner.

The transition to a heterogeneous network (HetNet) structure where small cells reuse the spectrum locally and provide most of the capacity, while macrocells provide a wide coverage for mobile UEs, can address this capacity problem [1].

Today, indoor small cells are deployed in large numbers, mainly in the form of femtocells, and many vendors have started to develop outdoor small cells to complement macrocellular coverage [2]. Most of today's small cell deployments are configured to transmit on a dedicated frequency carrier. While this avoids cross-tier interference, it also limits the available spectrum that small cells and macrocells can access, and thus is less efficient than co-channel deployments where small cells and macrocells reuse the available spectrum. However, while co-channel deployments provide better frequency reuse, the additional cross-tier interference can result in coverage, capacity, and mobility management issues for some UEs. Frequency and time domain enhanced intercell interference coordination (ICIC) techniques such as carrier aggregation (CA) and almost blank subframe (ABS) have recently been introduced to mitigate interference in co-channel deployments. However, these techniques still rely on the blanking of some frequency/time resources for the sake of interference mitigation. Due to the interactions between the small cell and macrocell tiers in co-channel deployments, we envision a joint optimization of the deployment and operation of small cells and macrocells as necessary.

In this article a novel sector offset configuration [3, 4] for the macrocell tier is proposed can significantly enhance both average and cell-edge macrocell UE (MUE) throughput without reducing the frequency reuse factor. Moreover, the proposed sector offset configuration provides the basis for improved ICIC between small cells and macrocells, which in turn results in a significantly enhanced performance for cell edge small cell UE (SUE). The proposed sector offset configuration also improves mobility performance, since handovers start at a higher signal quality due to ICIC. Details of two different implementations for the sector offset configuration tailored to multi-carrier and single-carrier HetNets are discussed in detail.

The remainder of the article is organized as follows. First, the proposed sector offset configuration for the macrocell tier is presented, and its implementation for multi-carrier and single-carrier HetNets is discussed, together with its bene-

The authors are with Bell Laboratories, Alcatel-Lucent.

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fits and drawbacks. Then, it is explained how small cells can take advantage of the proposed sector offset configuration. The scenario and system model used for the evaluation of the proposed sector offset configuration are introduced thereafter, followed by a performance comparison based on system-level simulations. Finally, the conclusions are drawn.

Macrocell Tier with Sector Offset Configuration

In order to improve the average and cell-edge UE performance in the macrocell tier, and provide the basis of an improved ICIC between the small cells and macrocells, a novel sector offset configuration [3] is proposed, where:

- The available spectrum is divided into two spectrum fragments. In a high-speed packet access (HSPA) network, these spectrum fragments may be two HSPA carriers, while in a Long Term Evolution (LTE) network, they may also be two different LTE carriers or two subsets of resource blocks (RBs) that belong to the same LTE carrier. If CA is used, they can also be two component carriers. We look at technology-dependent implementation details in following sections.
- Each base station (BS) is equipped with two sector configurations so that the antenna patterns in the first configuration are offset with respect to the antenna patterns in the second configuration, such that the direction of highest gain in the first sector configuration points in the direction of the sector boundary in the second sector configuration.
- Each sector configuration then manages and transmits in a different spectrum fragment.

Figure 1 depicts the traditional and sector offset configurations for a three-sector BS. In the traditional configuration, the coverage of the first and second spectrum fragments fully overlap with each other and share the same cell edges. In the sector offset configuration, the two sector configurations are offset by 60° with respect to each other, and thus the direction of the main antenna beam in one spectrum fragment overlaps with the cell edge of the complementary spectrum fragment. In Fig. 1 it is also important to note that because the same antenna patterns are used in the traditional and sector offset configurations, there is a large overlap among the coverage areas of neighboring antennas in the sector offset configuration. This is a main difference from traditional six-sector BSs, which leads to some benefits in terms of handover performance, as shown later.

As a result of the sector offset configuration, the macrocell performance is enhanced because:

- There are more antennas pointing in different directions.
- ICIC is improved since neighboring antennas transmit in complementary spectrum fragments.

As a consequence, compared to the traditional macrocell sector configuration, the areas suffering from high interference between sectors of a BS no longer overlap in both spectrum fragments. In other words, in the traditional sector configuration shown in Fig. 1a, a UE connected between two sectors of the traditional BS would suffer from a poor signal-to-interference-plus-noise ratio (SINR). However, in the proposed sector offset configuration shown in Fig. 1b, this UE would benefit from an improved SINR since there is an antenna pointing at it, and because of the improved ICIC (i.e. neighboring antennas transmit in complementary spectrum fragments). It is important to note that these benefits apply to both the downlink and the uplink.

Because of the large overlap between neighboring antennas, and due to the improved ICIC, handovers are also started at a higher SINR, so UEs have more time to perform the handover procedure. As a result, handover failures are mitigated.

One way of looking at this sector offset configuration is as a special form of static beamforming, where each UE follows the antenna

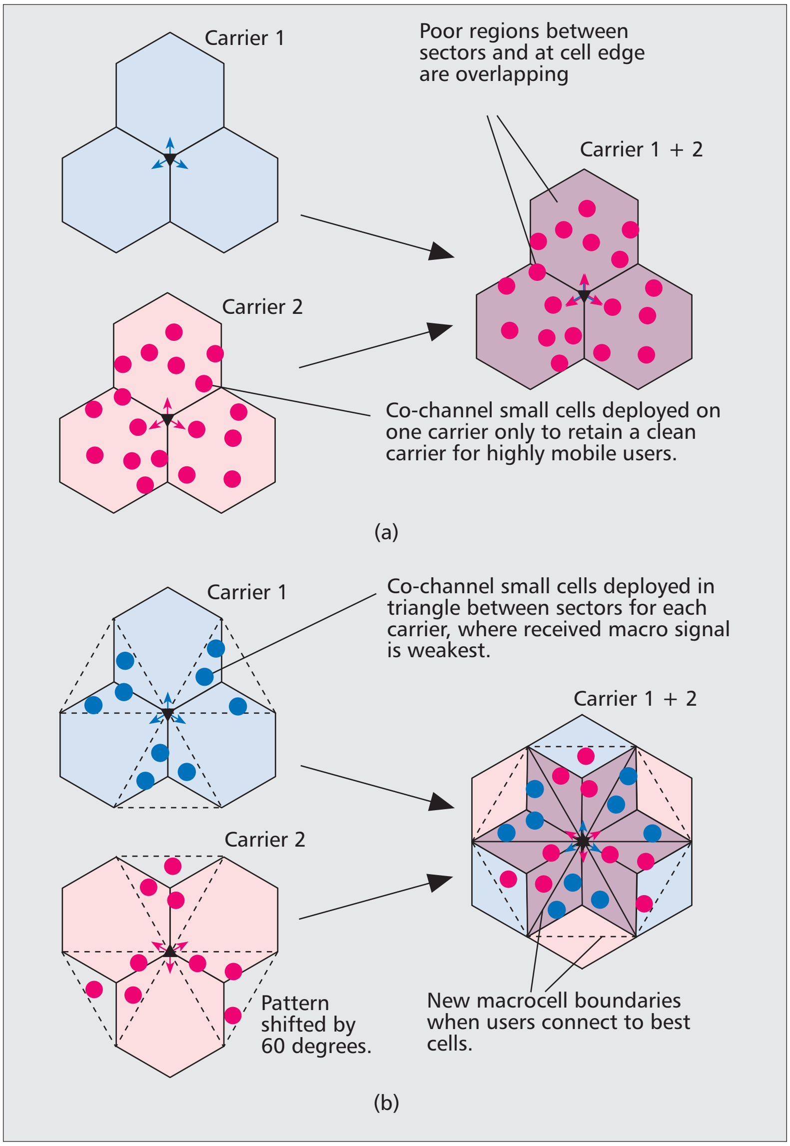

Figure 1. a) Multi-carrier HSPA/LTE HetNet configuration for traditional cell structure with partial frequency reuse; b) proposed configuration with sector offset.

Taking advantage of the flexibility of OFDMA, the sector offset configuration can also be realized in LTE networks while using a single frequency carrier. In this way, the mentioned increased number of handovers in the multi-carrier network implementation can be avoided.

beam via handovers or scheduling decisions, instead of an antenna beam formed in the direction of each UE. This proposed configuration is much simpler, and does not require channel amplitude and phase measurements and feedback for multiple transmit antennas.

Coordinated multipoint (CoMP) transmissions could also be used to enhance the performance of cell edge UEs [5]. However, this requires intensive inter-BS coordination to ensure that the same signal is transmitted to the targeted cell edge UE from different BSs. The sector offset configuration is a much lighter approach to improve cell edge performance, in which the burden of complexity, feedback, and overhead is mitigated.

Network-Level Sector Offset Configuration Realization

At the network level, the sector offset configuration can be realized using two different approaches depending on the number of available carriers.

Implementation in a Multi-Carrier Network — In a multi-carrier network, a straightforward approach to realizing the sector offset configuration is to associate a different frequency carrier to each sector configuration [3]. However, one disadvantage of this implementation is that connecting each UE to the best sector requires additional handovers compared to the traditional approach. Moreover, since the best neighboring sector is always on a different carrier, these additional handovers are inter-carrier handovers, which requires the UEs to perform and report measurements on different carriers. However, in contrast to a traditional cell structure, these handovers occur at higher SINRs due to ICIC when using the sector offset configuration, and are thus less sensitive to delays or other problems that can cause dropped calls.

Implementation in a Single-Carrier Network — Taking advantage of the flexibility of orthogonal frequency-division multiple access (OFDMA), the sector offset configuration can also be realized in LTE networks while using a single frequency carrier. In this way, the mentioned increased number of handovers in the multi-carrier network implementation can be avoided [4].

In more detail, the proposed configuration for single-carrier LTE networks is as follows (Fig. 2):

- The LTE subframe is divided into two fragments in the frequency domain. For example, if there are R available RBs, the first spectrum fragment may contain the first $R/2$ RBs, and the second spectrum fragment may contain the remaining $R/2$ RBs. Other partitioning approaches are possible.
- Neighboring offset antennas are grouped in pairs such that each antenna only belongs to one pair, and each antenna of the pair transmits on a different spectrum fragment. It is important to complementary note that the R RBs transmitted by each antenna pair use the same cell-specific reference signal (CRS) as in the traditional approach, regardless of the orientation of the antennas used to transmit it.

In this way, the two offset antennas of a pair appear as a unique antenna port to UEs; thus, handovers between these offset antennas are avoided. From a UE perspective, this CRS transmission configuration, in which part of the CRS symbols are transmitted over one antenna (a.k.a. antenna 1) and the remaining CRS symbols over the offset antenna (a.k.a. antenna 2), results in received SINR improvements over part of the RBs and received SINR degradations over the remaining RBs. This very same fact allows appropriate antenna-to-UE assignments using off-the-shelf schedulers. UEs that are under the coverage of antenna 1 will see higher SINRs in the CRS symbols of the first $K/2$ RBs than in the remaining RBs, and report in the corresponding channel quality indicators (CQIs) better signal quality.[1] Then, due to the better SINR reported, off-the-shelf schedulers will prefer to assign UEs that are under the coverage of antenna 1 to the first $K/2$ RBs, and UEs who are under the coverage of antenna 2 (offset antenna) to the remaining RBs. This proposed sector offset is compliant with current eNodeBs and UEs.

It is important to note that reference signal received power (RSRP) and wideband SINR measurements are taken by the UEs over the CRS symbols all across the bandwidth. Therefore, the proposed sector offset configuration has an impact on such measurements. Indeed, the split of the transmission of CRS symbols between offset antennas reduces wideband SINRs in the high regime, but improves them at the low regime, thus enhancing performance at the cell edge, as shown later. Physical downlink control channel (PDCCH) messages, which are also scrambled all across the bandwidth, follow this trend as well. On the subject of PDCCH messages, it is worth mentioning that PDCCH messages in LTE Release 11 and beyond do not need to be scrambled all across the bandwidth, and can be transmitted in specific RBs using enhanced physical downlink control channels (EPDCCHs) [6]. UEs under the coverage of antenna 1 will be configured to receive EPDCCHs via RBs in spectrum fragment 1, while UEs under the coverage of antenna 2 will be configured to receive EPDCCHs via RBs in spectrum fragment 2. As a result, the mentioned issue vanishes.

Hardware-Level Sector Offset Configuration Realization

Since today's Multiple Input Multiple Output (MIMO) transceivers are equipped with more than one radio frequency (RF) chain and baseband unit, the only changes required to realize the sector offset configuration are at the BS antennas and the network configuration. This makes it very easy to upgrade existing cellular network deployments. An important point is that this scheme is completely transparent and fully compatible with existing UEs.

Compared to the traditional approach in Fig. 3a, the two antenna element columns with a mechanical offset shown in Fig. 3b can be used to implement the sector offset configuration. Since the signal on spectrum fragment 1 is transmitted via a different antenna than the signal on

[1] CQIs are fed back from UEs to eNodeBs to assist scheduling, and based on measurements taken over CRS symbols. Channel state information-reference signals (CSI-RSs) may also be used to increase the reliability of such measurements in the proposed sector offset configuration and aid channel-dependent scheduling [5].

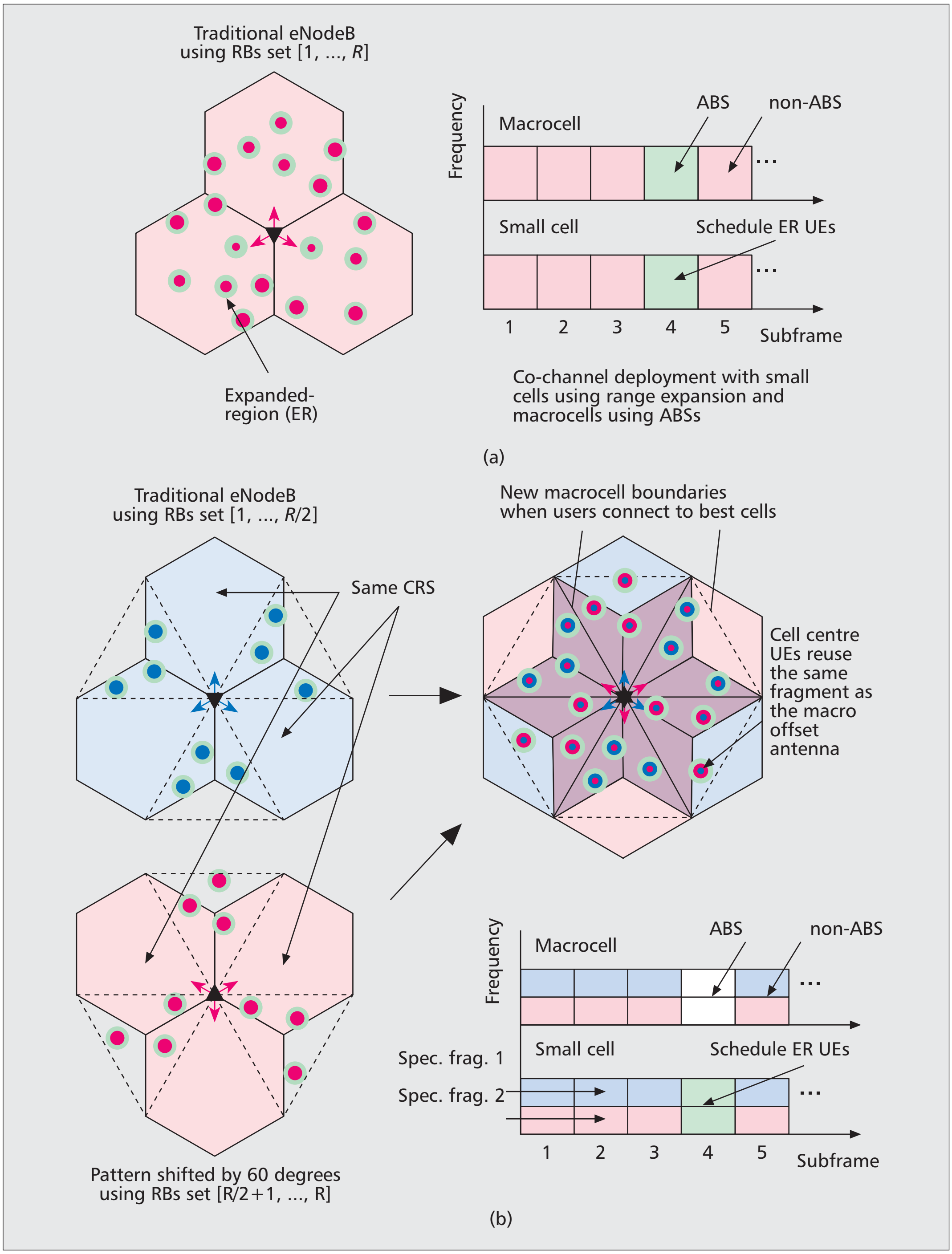

Figure 2. a) Single-carrier LTE HetNet configuration for traditional cell structure with frequency reuse of 1; b) proposed configuration with sector offset.

While the sector offset configuration described here is for eNodeBs with three sectors, it can be extended to higher sectorizations in the horizontal and vertical dimensions with smaller angular offsets. For higher sectorizations, other, more compact and flexible antenna structures can be used.

spectrum fragment 2, two different amplifiers with lower power each are required instead of one wideband amplifier. The energy transmitted on each carrier remains the same. The two antenna element columns can fit in one radome, so that the visual impact at the antenna tower remains compared to a traditional configuration.

While the sector offset configuration described here is for eNodeBs with three sectors, it can be extended to higher sectorizations in the horizontal and vertical dimensions with smaller angular offsets, as described in [3, 4]. For higher sectorizations, other more compact and flexible antenna structures, such as the circular antenna array shown in Fig. 3c, can be used.

Difference between the Sector Offset Configuration and Higher Order Sectorizations

It is important not to confuse the concept of sector offset configuration with that of higher order sectorization, for the following reasons:

• Antennas with a larger half power beamwidth are used in the sector offset configuration than in its equivalent higher order sector-

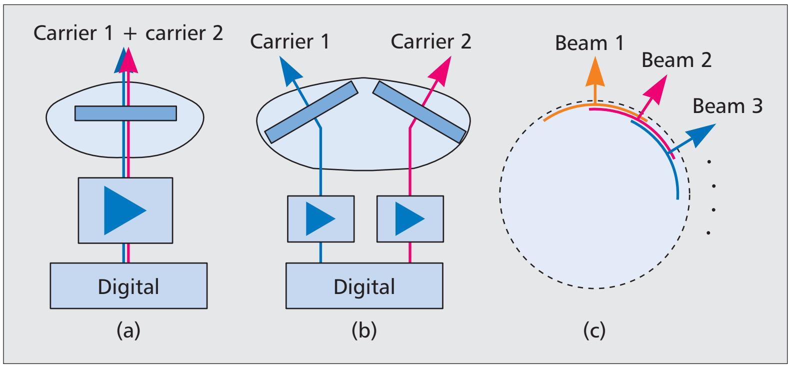

Figure 3. Implementation examples: a) traditional radome; b) sector offset radome; c) circular array.

ization using the traditional configuration. The sector offset configuration in Fig. 1 uses 65 degrees half power beamwidth antennas, while its equivalent six-sector BS using the traditional configuration would use 33 degrees half power beamwidth antennas. This results in a much larger overlapping between neighboring antennas, which facilitates mobility management for mobile UEs (handovers are started at higher SINR). Due to this large overlapping among neighboring antennas in the sector offset configuration, UEs in between the two antennas can also be served by both spectrum fragments, and thus can exploit maximum capacity.

• In the traditional configuration, neighboring antennas transmit over the entire spectrum, while in the sector offset configuration, neighboring antennas transmit over complementary spectrum fragments. This provides a significant interference coordination at the expense of reducing its spectrum reuse in particular areas.

• In the traditional configuration, as many base band units as beams are used, while in the sector offset configuration, the number of base band units can be halved. Neighboring offset antennas transmitting in complementary spectrum fragments may re-use the same base band unit. This could significantly reduce the cost of the sector offset configuration.

Small Cell Tier with Sector Offset Configuration

Small cells can take advantage of the proposed sector offset configuration for macrocells, and also benefit from an improved ICIC. According to the sector offset configuration realizations presented in the previous section, small cells may be configured as presented in the following.

Small Cell Operation in a Multi-Carrier Network

In multi-carrier implementations, a small cell, depending on its location, should be configured such that it only transmits on the spectrum fragment where the umbrella macrocell provides the lowest SINR [7]. For example, within the cell-edge triangular areas between macrocell sectors shown in Fig. 1, small cells should use the spectrum fragment complementary to the umbrella macrocell. These small cells may also reuse the spectrum fragment of the umbrella macrocell to transmit with a reduced power to their cell-center UEs, since they already benefit from good channel conditions due to the short distance between transmitter and receiver. In addition, small cells may be prioritized when their received power and/or SINR exceeds a pre-defined value to maximize offloading to small cells.

The proposed frequency configuration for small cells can be achieved by the following auto-configuration process:

- Measure the received powers from macrocells at the small cell BS on both carriers and identify the maximum value.
- Select the carrier with the lowest measured maximum value for small cell use.

Using this approach, the macrocell performance improves since macrocell UEs are served by the carrier on which the macrocell antenna provides the highest gain, thus avoiding regions with high interference between sectors. A further benefit of this cell structure is that a macrocell carrier always appears clean or with reduced small cell interference at each location, and thus some macrocell UEs can benefit from improved SINRs. Moreover, fast moving macrocell UEs do not necessarily need to handover to a small cell, but can move through it on the clean macrocell carrier. This prevents dropped calls due to handover delays for fast moving UEs.

The small cell performance also improves since small cell UEs are served by the carrier on which the macrocell signal is received at lower power. This results in improved coverage, reduced interference, and more flexibility at what minimum distance to the macrocell a small cell can be deployed effectively.

Small Cell Operation in a Single-Carrier Network

As indicated earlier, the proposed sector offset configuration can also be applied to groups of RBs in LTE networks with only one carrier. Since depending on the small cell location, a subset of RBs may appear clean or with reduced interference from the eNodeB, we propose that small cells use:

- The subset of RBs in which the small cell BS measurements report the lowest SINR for cell-edge UEs.
- The subset of RBs in which the small cell BS measurements report the highest SINR for cell-center UEs [8].

In this case, the scheduler would also automatically assign UEs to the best RBs, preventing the need for handovers and measurements on other carriers. Using the previously presented example, if a small cell BS is under the coverage of antenna 1 of the umbrella eNodeB sector, which transmits over the first $K/2$ RBs, its UEs will see higher SINR in the CRS symbols of the last $K/2$ RBs than in the first $K/2$ RBs. This is because the last $K/2$ RBs suffer from reduced cross-tier interference as they are handled by the coverage of antenna 2 of the umbrella eNodeB sector, whose main antenna lobe is pointing in a different direction. In this way, new scheduling opportunities are created for small cell cell-edge UEs in the last $K/2$ RBs.

Parameters	Value	Parameters	Value
Scenario		Antenna	
Macro BS placement	7 eNodeBs, 800 m inter–site distance	Maximum gain	G_{max} = 15.5 dBi (macro), 7.06 dBi (pico)
Pico BS placement	one at the centre of each hotsopt	H. halfpow. beamwidth	$\alpha = 65°$
Scenario size	1500 m × 1500 m, around central macrocell	V. halfpow. beamwidth	$\beta = 11.5°$ (macro)
Scenario resolution	2 m	Front-to-back ratio	κ = 30 dB (macro)
Transmit power	$P_{tx,n}$ = 21.6 W (macro), 1 W (picos)	Downtilt	$\delta_1 = 8.47°$ (macro)
Noise density	–174 dBm/Hz	Elements & spacing	4 element dipole, $d_{elem.} = 0.6\lambda$ (pico)
UEs		Phase difference	$\delta_{phase} = 95°$ (pico)
UE density	600 per km^2, 50% hotspots, 30% outdoors	Element amplitude	$\alpha_{elem.}$ = [0.9691,1.0768,1.0768,0.8614] (pico)
Hotspots (static)	20 UEs per hotspot within 40 m radius	Handover	
Outdoor UE speed	30 km/h	Time-to-trigger	160 ms
Channel		Hysteresis	2 dB
Carrier frequency	2000 MHz	RSRP measurement interval	40 ms
Bandwidth	5 MHz (1 LTE carrier of 25RBs)	L3 filter reporting interval	200 ms
NLOS path–loss	$G_{Pn} = -21.5 - 39\log_{10}(d)$ (macro) [9]	L3 forgetting factor	1/4
	$G_{Pn} = -30.5 - 36.7\log_{10}(d)$ (pico) [9]	Preparation + execution time	90 ms
LOS path–loss	$G_{Pl} = -34.02 - 22\log_{10}(d)$ [9]	Out of sync./in sync	$\gamma_{n,k}^{ESSM} < Q_{out}$ = –8 dB/g$\gamma_{n,k}^{ESSM} > Q_{in}$ = –6 dB
Shadow fading (SF)	6 dB std dev. [10]	Radio link failure timer	1 s
SF correlation	$R = e^{-1/20d}$, 50% inter–site	Scheduling	
Environment loss	$G_{E,n}$ = –20 dB if indoor, 0 dB if outdoor	Time domain scheduled UEs	U = 10 UEs
Antenna		Avg. throughput forgetting factor	$1/t_c = 0.5$
Height	25 m (macro), 10 m (pico)	UE targeted throughput	$C_{n,k}^{target}$ = 128 kb/s

Table 1. Simulation parameters.

Combination of Sector Offset Configuration with Cell Range Expansion

The proposed small cell operation is compatible with standardized cell range expansion (CRE) and ABSs. As illustrated in Fig. 2a, when using the traditional LTE Rel. 10 eICIC, small cell UEs in the expanded region are scheduled in subframes overlapping with the macrocell ABSs, thus avoiding cross-tier interference.

In the proposed sector offset configuration with CRE and ABS, when eNodeBs transmit ABSs, small cells also allocate their range-expansion UEs in those subframes that overlap with macrocell ABSs, following standard operation. In contrast, when eNodeBs transmit normal subframes with macrocell UE data, we propose that small cells just follow the scheduling strategy defined in the previous section, i.e. cell-edge UEs are allocated to the spectrum fragment with the least interference, while cell-center UEs are allocated to the complementary spectrum fragment.

In order to make the most of the scheduling opportunities offered by macrocell ABSs and the sector offset configuration, we also propose to equip each small cell with a two-level load balancing scheme [8]. The first load balancing step divides the set of connected UEs to a small cell into two subsets:

- The subset of UEs to be scheduled in subframes that overlap with macrocell ABSs.
- The subset of UEs to be scheduled in subframes that do not overlap with macrocell ABSs.

The second load balancing scheme works further in the second subset of UEs and divides it into two further subsets:

- Cell-edge UEs, which will be allocated to the RBs benefiting from cross-tier interference mitigation due the sector offset configuation.
- Cell-center UEs which will be allocated to the remaining RBs.

In both load balancing procedures, the target is to equalize the throughput of the worst UEs in each subset (e.g. five percentile throughput) based on the knowledge of their wideband SINR and available resources. For more details on this two-step load balancing, please refer to [8].

Scenario, System Model and Benchmark

System-level simulations were performed to evaluate the performance of the proposed sector offset configuration in terms of downlink throughput and handover performance. A single carrier LTE HetNet scenario was considered, and Table 1 presents the most relevant simulation parameters.

The simulated scenario was downtown Dublin, Ireland, and consisted of one LTE carrier centered at 2 GHz with 5 MHz bandwidth and seven eNodeBs with three sectors each. The UE distribution consisted of a uniform background distribution with additional hotspots. Hotspots were circular with a 40 m radius and their positions were randomly selected. An outdoor pico eNodeB was deployed at the center of each hotspot. UEs were uniformly distributed within the scenario and the hotspots. There were 600 UEs per square kilometre, with 50 percent of them located in the hotspots. Each hot spot had 20 UEs. Overall, 70 percent of all UEs were placed indoors, and the remaining 30 percent outdoors. This ratio was applied to UEs in both the background distribution and the additional hotspots. Realistic antenna patterns were used at both the macro and pico eNodeBs, whose most representative features are presented in Table 1, and their horizontal and vertical diagrams can be respectively seen in [3] and [7]. The small cell antenna consisted of a vertical four element dipole array, providing an omnidirectional coverage in the horizontal plain and a downtilt of 10.83 degrees in the vertical plane. Path losses, shadow, and multi-path fading were also modelled according to the details in Table 1. UEs were equipped with two receiver antennas. Maxi-

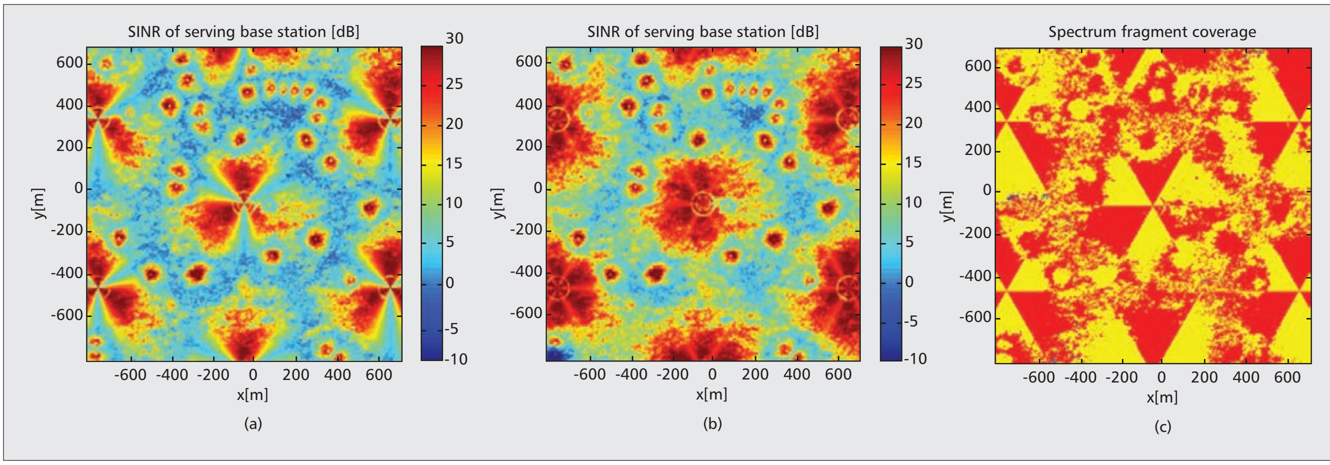

Figure 4. SINR distribution and spectrum coverage for the 3 sector eNodeB case: a) 3 sector traditional conf. SINR distribution; b) 3 sector offset conf. SINR distribution; and c) 3 sector offset conf. spectrum coverage.

mal ratio combining (MRC) was used to combine signals, while exponential effective SINR mapping (EESM) was used to estimate wideband SINRs. Handover results were calculated based on 500 UE routes through the simulated scenarios, with UEs traveling at 30 km/h. For more details on network topology, UE distribution, simulations assumptions and methodology, please refer to Table 1 as well as [7] and [8].

In this paper the proposed sector offset configuration is benchmarked against a traditional deployment with frequency reuse 1, where small cells and macrocells reuse all available RBs. The impact of CRE and ABSs is also analyzed. The simulated single-carrier configurations were:

- Case 1 — Reuse 1: small cells do not use CRE and macrocells do not use ABSs.
- Case 2–4 — Reuse 1 with CRE and ABS: small cells use CRE and macrocells use an ABS duty cycle of 0.0, 0.1, or 0.2.
- Case 5–7 — Sector offset sectorization: small cells and macrocells use the proposed sector offset configuration for single-carrier networks. Small cells also use CRE and macrocells use an ABS duty cycle of 0.0, 0.1, or 0.2. The mentioned load balancing algorithms from earlier are also enabled.

The range expansion bias of each expanded-region small cell was individually tailored to allow each small cell to cover its hot spot of 40 m radius. A maximum range expansion bias (REB) cap of 9 dB was used.

Performace Comparison

SINR Distribution

Figures 4a and 4b show the SINR spatial distribution of the best data channels in the presented scenario when macrocells transmit normal subframes and use the traditional configuration and the proposed sector offset configuration, respectively. It can be easily seen that the proposed sector offset configuration increases the average SINR of the best data channels due to its better macrocell coverage. The 50 percentile and five percentile improvements are 3.96 dB and 2.18 dB, respectively. Figure 4c also illustrates the coverage of spectrum fragment 1 (yellow color) and spectrum fragment 2 (red color) when using the proposed sector offset configuration. A good balance between spectrum fragments is achieved. From this figure, it can be seen how when a small cell is deployed within the yellow macrocell coverage, it can use the red spectrum fragment, which suffers from low macrocell interference, to serve its cell-edge UEs and the yellow spectrum fragment to serve its cell-center UEs. In this way, a large inter-cell interference mitigation is achieved even when the macrocells transmit normal subframes.

Capacity Performance

Let us first focus on the three cases without ABSs ($D_{\mathrm{ABS}} = 0.00$), i.e. cases (1), (2), and (5). When we compare the cases with no sector offset configuration with and without range expansion at the small cells, case (1) and case (2), the average MUE throughput increases by 16 percent due to UE offloading, but the five percentile SUE throughput reduces by 233 percent due to the increased number of SUEs and the high cross-tier interference suffered by expanded-region SUEs. When we compare the cases with cell range expansion with and without sector offset configuration, case (2) and case (5) — no LTE Rel. 10 eICIC is used — the average MUE throughput increases by 17 percent due to the better cross-tier interference mitigation of the proposed configuration. This cross-interference mitigation is more obvious at the cell boundaries, where the five percentile SUE throughput increases by 115 percent since cell-edge UEs are scheduled in the spectrum fragment not used by the umbrella sector offset antenna.

Lets us now focus on the three cases with cell range expansion and LTE Rel. 10 eICIC, i.e. cases (2), (3), and (4). We can see that when the ABS duty cycle increases, cross-tier interference is significantly alleviated at the expense of a reduced macrocell performance due to subframe blanking. When D_{ABS} increases from 0 to 0.20, case (2) to case (4), five percentile SUE throughput increases by 172 percent, while the average MUE throughput decreases by 19 percent. This is a general trend inherent to ABS operation.

	1 no bias	2–4 Rel. 10 eICIC			5–7 Rel. 10 eICIC + sector offset		
	D_{ABS} = 0.00	D_{ABS} = 0.00	D_{ABS} = 0.10	D_{ABS} = 0.20	D_{ABS} = 0.00	D_{ABS} = 0.10	D_{ABS} = 0.20
Throughput/gain–over–case (2)/Gain–over–case (3) or (4). Units: [kb/s]/[%]/[%]							
Macro sector tp (av.)	8968/–3	9274/0	8391/–10/0	7489/–19/0	10635/+14	9630/+4/+15	8604/–8/+15
Picocell tp (av.)	5137/+5	4906/0	5411/+10/0	5703/+16/0	4529/–8	5327/+9/–2	5691/+16/–1
MUE tp (av.)	175.3/–16	208.0/0	188.2/–10/0	168.0/–19/0	244.0/+17	220.9/+6/+17	197.4/–5/+18
MUE tp (5%–tile)	39.7/–31	57.2/0	51.1/–11/0	45.1/–21/0	82.4/+44	73.0/+27/+43	63.8/+12/+41
SUE tp (av.)	463.6/+15	402.1/0	443.6/+10/0	467.8/+16/0	372.7/–8	438.6/–1/–1	468.6/+6/0
SUE tp (5%–tile)	164.0/+233	49.2/0	105.0/+113/0	133.8/+172/0	106.0/+115	138.8/+182/+32	157.6/+220/+18
UE tp (5%–tile)	55.8/+2	54.5/0	62.6/+15/0	59.5/+9/0	91.2/+67	93.9/+72/+50	85.5/+57/+43
Hard handover/Gain. units:[events/UE/km2]/[%]							
Number	242.35/+6	227.55/0	254.98/+12	254.98/+12	240.08/+6	263.95/	263.95/+16
Failure rate [%]	3.71/–71	12.60/0	2.29/–82/0	2.29/–82/0	9.06/–28	1.24/–90/–46	1.24/–90/–46

Table 2. Performance comparison.

In order to assess the gains provided by sector offset while complementing LTE Rel. 10 eICIC, let us compare the cases with the same ABS duty cycle with and without sector offset. Comparing the cases with D_{ABS}=0.10 , case (3) and case (6) (gains marked in red), the proposed configuration increases average MUE throughput, five percentile MUE throughput, and five percentile SUE throughput by 17 percent, 43 percent, and 32 percent, respectively, while average SUE throughput remains constant (–1 percent). Overall five percentile UE throughput increases by 50 percent. MUE performance is increased due to the better MUE alignment with the antenna beams provided by sector offset and the interference mitigation resulting from the sophisticated spectrum fragment reuse. Cell-edge SUE performance is also increased because when the macrocells transmit normal subframes, the SUEs with the worst channel gains are scheduled in the spectrum fragment not used by the umbrella sector offset antenna. Similar gains are observed with D_{ABS} = 0.20 (gains marked in blue).

Note that the proposed configuration with D_{ABS} = 0.10 achieves roughly the same five percentile SUE performance as LTE Rel. 10 eICIC with D_{ABS} = 0.20 and no sector offset. In this case the proposed configuration is thus able to decrease the ABS duty cycle up to a factor of 2, and in turn enhance macrocell UE performance by 31 percent, while maintaining a targeted five percentile SUE performance.

Mobility Performance

Because the number of handovers performed in a HetNet is dominated by the number of small cells, we can see from Table 2 that all compared approaches incur approximately the same number of handovers per UE per hour. In terms of handover failures, we can see that the case with small cell REB and no macrocell ABSs, case (2), provides the worst performance. This is mostly because handovers from small cells to macrocells tend to fail due to high interference conditions. When macrocell ABSs are used, case (3), the handover failure rate decreases by 82 percent compared to the previous case. This is because small cell to macrocell handover failures are significantly mitigated due to the interference mitigation provided through subframe blanking at macrocells. Note that the ABS duty cycle has no impact on the handover performance, since all UEs performing a handover from a small cell were allocated to subframes overlapping with the macrocell ABSs with a higher priority. When the proposed sector offset configuration is adopted in combination with macrocell ABSs, case (6), the handover failure rate is further decreased by 46 percent (compared with the case where macrocell ABSs but no sector offset is used, case (3)). Due to the more uniform SINR distribution provided by the sector offset configuration (more macrocell antennas pointing in different directions), MUEs start their handovers at a higher SINR and thus have more time to perform the handover process. Sector offset configuration thus significantly alleviates handover failures in HetNets.

Conclusion

In this paper the concept of sector offset configuration has been presented together with two possible implementations for multi-carrier and single-carrier networks. The implementation aspects of the proposed sector offset configuration at macrocell BSs have also been discussed, considering the network and hardware configuration. Moreover, it has been explained in detail how small cell BSs can take advantage of the proposed sector offset configuration through self-organization. Extensive simulation results have been presented to quantify the benefits of the proposed configuration. The proposed HetNet configuration with offset sectorization significantly improves the network performance compared to existing state-of-the-art configurations, both in terms of UE throughput and handover failure rate. In more detail, for a single carrier LTE HetNet, cell-edge UE throughput has been shown to improve by 50 percent, while average and cell-edge MUE throughput have been improved by 17 percent and 32 percent, respectively. Handover failure rates decreased by 46 percent when adopting offset sectorization. This comes at the cost of an increased number of antennas at the macrocell site to achieve the sector offset configuration. The concept is fully compatible with current mobile devices and cellular standards, and is in general easy to implement.

Handover failure rates decreased by 46 percent when adopting offset sectorization. This comes at the cost of an increased number of antennas at the macrocell site to achieve the sector offset configuration. The concept is fully compatible with current mobile devices and cellular standards, and is in general easy to implement.

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BIOGRAPHIES

DAVID LÓPEZ-PÉREZ (david.lopez-perez@alcatel-lucent.com) is a Member of Technical Staff at Bell Laboratories, Alcatel-Lucent, and his main research interests are in HetNets, small cells, interference and mobility management as well as network optimization and simulation. Prior to this, David earned his Ph.D. in Wireless Networking from the University of Bedfordshire, UK in Apr. '11. David was Research Associate at King's College London, UK from Aug. '10 to Dec. '11, carrying post-doctoral studies, and was with VODAFONE, Spain from Feb. '05 to Feb. '06, working in the area of network planning and optimization. David was also invited researcher at DOCOMO USA labs, CA in 2011, and CITI INSA, France in 2009. For his publications and patent contributions, David is a recipient of both the Bell Labs Alcatel-Lucent Award of Excellence and Certificate of Outstanding Achievement. He was also finalist for the Scientist of the Year prize in The Irish Laboratory Awards. David has also been awarded as Ph.D. Marie-Curie Fellow in 2007 and Exemplary Reviewer for *IEEE Communications Letters* in 2011. He is founding member of IEEE TSCGCC and author of the book "Heterogeneous Cellular Networks: Theory, Simulation and Deployment" Cambridge University Press, 2012. Moreover, he has published more than 75 book chapters, journal and conference papers, all in recognized venues, and filed more than 30 patents applications. He is or has been guest editor of a number of journals, e.g., *IEEE JSAC*, *IEEE Comm. Mag.*, TPC member of top tier conferences, e.g., IEEE Globecom and IEEE PIMRC, and co-chair of a number of workshops.

HOLGER CLAUSSEN [SM] (holger.claussen@alcatel-lucent.com) is leader of Small Cells Research at Bell Labs, Alcatel-Lucent. In this role, he and his team are innovating in all areas related to future evolution, deployment, and operation of small cell networks to address the exponential growth in mobile data traffic. His research in this domain has been commercialized in Alcatel-Lucent's Small Cell product portfolio and continues to have significant impact. He received the 2014 World Technology Award in the individual category Communications Technologies for innovative work of "the greatest likely long-term significance." Prior to this, Holger was head of the Autonomous Networks and Systems Research Department at Bell Labs Ireland, where he directed research in the area of self-managing networks to enable the first large scale femtocell deployments from 2009 onwards. Holger joined Bell Labs in 2004, where he began his research in the areas of network optimization, cellular architectures, and improving energy efficiency of networks. Holger received his Ph.D. degree in signal processing for digital communications from the University of Edinburgh, United Kingdom in 2004. He is author of more than 70 publications and 100 filed patent applications. He is Fellow of the World Technology Network, and member of the IET.

LESTER HO [SM] (lester.ho@alcatel-lucent.com) is a Distinguished Member of Technical Staff at Bell Laboratories, Alcatel-Lucent, in Dublin, Ireland. His main areas of research are in small cells, self-organizing network techniques, and network optimization. Lester joined Bell Labs in the UK in 2003, where many of his research into SON techniques for small cells can be found in commercial deployments today. He received his Ph.D. in self-organization in wireless networks from Queen Mary, University of London in 2003. He has 29 patents granted, 30 patent filings pending, and over 35 peer reviewed publications.

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Forecasting of Access Network Bandwidth Demands for Aggregated Subscribers Using Monte Carlo Methods

Ed Harstead and Randy Sharpe

Abstract

In the face of relentless growth in traffic, network operators must continuously forecast bandwidth demand to properly dimension their networks and make the correct investments for the future. The consequences of under-investing are poor network performance and dissatisfied subscribers, while over-investing ties up capital that could have been better spent elsewhere. In this article we describe a model that forecasts bandwidth demands of aggregated subscribers on residential fixed access networks. For sustained bandwidth demand, we use statistical techniques to quantify the number of concurrent video streams, the shifting mix of standard definition, high definition, and ultra high definition resolutions, multicast gain, and the trend from multicast to unicast delivery of these streams. Mechanisms to cope with bursty bandwidth demand are included. The results we report can guide operators in making technology and network design choices for future FTTx deployments.

Introduction

Purpose

In the face of relentless growth in traffic, operators must continuously forecast bandwidth demand growth to properly dimension their networks and make the correct investments for the future. The consequences of under-investing are poor network performance and dissatisfied subscribers, while over-investing ties up capital that could have been better spent elsewhere. The purpose of this article is to forecast future bandwidth demands on residential access networks and guide network planners in making access technology choices for the future.

Scope

In this article we describe a model that forecasts future bandwidth demands, specifically the bandwidth demands of aggregated users on residential fixed access networks, in the downstream (toward the subscriber) direction. "Telco" operators with legacy copper plant need to correctly dimension their access network: how deeply to drive fiber to reduce the length of the copper loop, or whether to go all the way to fiber-to-the-home (FTTH). Similarly, cable operators with legacy hybrid fiber coax (HFC) networks need to evolve them to support higher Internet bandwidth demands, as well as an eventual evolution away from broadcast pay-TV video toward all-IP video. This may also require deeper fiber and possibly FTTH.

Prior Art

There are bandwidth forecasts familiar to many, but none of these attempt to quantify and forecast the total aggregated bandwidth demands in the access network. Jakob Nielsen's "Law of Internet Bandwidth" [1] claims a 50 percent year-over-year (YoY) growth in a high-end user's connection speed (specifically his own), but it cannot be applied for a few reasons. Nielsen's Law does not describe bandwidth demanded, but peak bandwidth offered. The data points in recent years correspond to HFC networks, and so the "Law" has already failed to account for the premium bandwidths offered on fiber networks. And finally, it only applies to Internet connections, excluding the connectivity for managed linear pay-TV, which represents a much larger component of traffic on all-IP video networks. Accordingly, applying Nielsen's Law to bandwidth demands, particularly on all-IP video access networks, and then extrapolating it to predict future demands, will be dubious.

Another familiar source is the annual Cisco Visual Networking Index (VNI) [2], which forecasts Internet traffic. It is useful for traffic growth rates, but is also inadequate to this task because it only measures regional demand, not at the subscriber or access network level. Further, it does not distinguish between downstream and upstream traffic.

The authors previously published a bandwidth demand forecast model [3] that included both Internet-based and managed pay-TV video traffic, which was used to gauge the suitability of TDM PON and WDM PON technologies for future FTTH networks. For that purpose, the scope of that model was restricted to video

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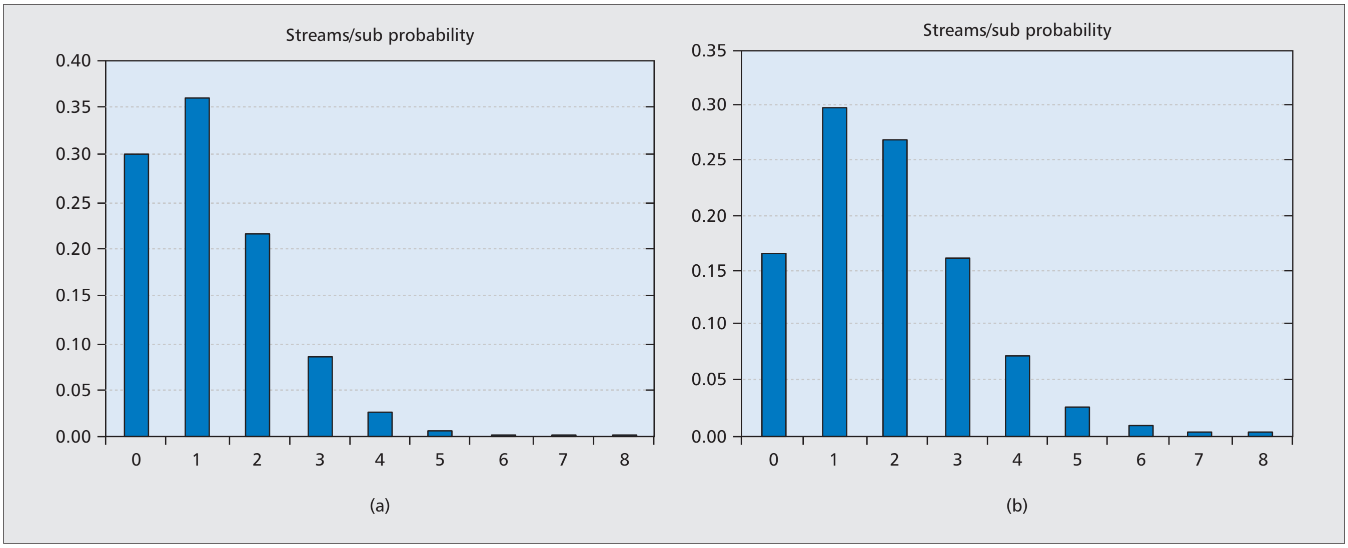

Figure 1. Probability distribution of number of streams per subscriber: (a) moderate scenario; (b) heavy scenario.

bandwidth demand, and rather than quantifying actual demand, it set an upper bound forecast for video bandwidth demand. However, that model does not address the intentions of this article, as we aim for actual, not upper bound, forecasts of demand, and we want to include bursty traffic demand.

Also worth noting is [4], which models and forecasts many kinds of sustained and bursty traffic, but only for a single subscriber. This is a valid method for sizing point-to-point subscriber links, like an xDSL connection, but it does not address the larger problem of aggregate demands on PON and HFC networks, and on the point-to-point backhaul of remote access nodes.

Aspects of the New Model

To accomplish our stated purposes, we need a new model that rather than setting an upper bound, attempts to estimate actual sustained bandwidth demands using statistical methods to model viewing behavior. This new model also includes bursty traffic bandwidth demands, and aggregates the demands of a wide range of subscribers.

Components of Bandwidth Demand

Sustained and burst bandwidth demands are modeled independently. In this section we describe each element of the model and its input parameters. Each of these parameters could take on a range of values. We can test the sensitivity of the model to each of these inputs individually, but this leads to volumes of results. Instead, we define a set of likely input values to represent a "moderate" bandwidth demand scenario, and a set of higher, more aggressive input values to represent a "heavy" bandwidth demand scenario. This allows varying the inputs as an ensemble. Many of the heavy input parameters are derived from the moderate parameters by multiplying by an (admittedly arbitrary) sensitivity factor of 1.5. We do not choose best-case and worst-case values for these input parameter ranges, because cascading such values would produce unrealistic results.

Sustained Bandwidth Demand

Video traffic is by far the largest component of sustained bandwidth. Voice and other primarily audio traffic (e.g. streaming music) are comparatively insignificant, and are ignored. For video we include all managed linear pay-TV, managed video-on-demand (VoD), and over-the-top (OTT) Internet video (e.g. Netflix, YouTube). Managed VoD and OTT video are modeled as unicast traffic. Linear pay-TV traffic is modeled as multicast traffic, in which case the multicasting gain is accounted for.

Our sustained bandwidth demand model applies Monte Carlo methods to statistical distributions that determine the volume of traffic demanded. Each element of the model is described below.

Statistics on Simultaneous Streams per Home — Being a discrete distribution, we have modeled the number of simultaneous streams to the home with the Poisson distribution. This allows us to adjust the distribution with a single knob, λ, the average number of streams per home. There is little public data available to indicate how to set this knob. However, [5] exhibits data from Nielsen Media Research indicating that the "houses using TV" (HUT) linear TV metric, during prime time, has shown stability at around 60 percent in the U.S. from the years 2000–2007. (These were the days before widespread Internet video. Therefore, to the extent that Internet video streams only replaced linear TV streams, but have not actually increased the number of simultaneous streams per home, this 60 percent number is still relevant.) Further, we can infer from that source a peak HUT (Sunday evenings in the colder months) at around 70 percent. Accordingly, we turn the Poisson knob such that the probability of one or more streams equals 0.7. This occurs at an average streams per home of $\lambda = 1.2$. We will use this value for our moderate demand scenario. For our heavy demand scenario, we multiply by the sensitivity factor of 1.5 to obtain $\lambda = 1.8$. This is intended to account for simultaneous home DVR recording (which is, however, expected to decline over time with the uptake in the usage of cloud-based time-shift TV and the roll-out of cloud-based DVR) and possible increased viewing over time due to other factors. The resulting probability distributions are shown in Fig. 1.

Forecast for Probabilistic Multicast/Unicast Ratio — A conventional IPTV service delivers linear channels to the home using multicast to eliminate duplicate delivery of streams of the same channel being watched or recorded by multiple viewers. Unicast streams always require a separate stream to each viewer, and so consequently place a higher burden on the network, particularly on links carrying traffic to multiple subscribers. The model accounts for this difference, but must be able to decide which streams are multicast and which are unicast.

Unicasted video streams today include Internet video streams and managed VoD services, which make up a relatively small proportion of the total viewing time relative to viewing live linear TV. But unicast video is growing with the increasing popularity of OTT video content, and the decreasing reliance on home DVRs for time-shifted viewing of linear TV (as mentioned above). We model unicast adoption with the standard "S-curve," with an asymptote less than 100 percent, because most viewers will probably continue to at least watch live sports and live news channels, which can be effectively multicasted. For the moderate scenario, we start with 13 percent unicast in 2014, approaching an asymptote of 50 percent in 10 years. For the heavy scenario we use the 1.5× sensitivity factor to obtain 20 percent and 75 percent, respectively. These adoption curves are shown in Fig. 2.

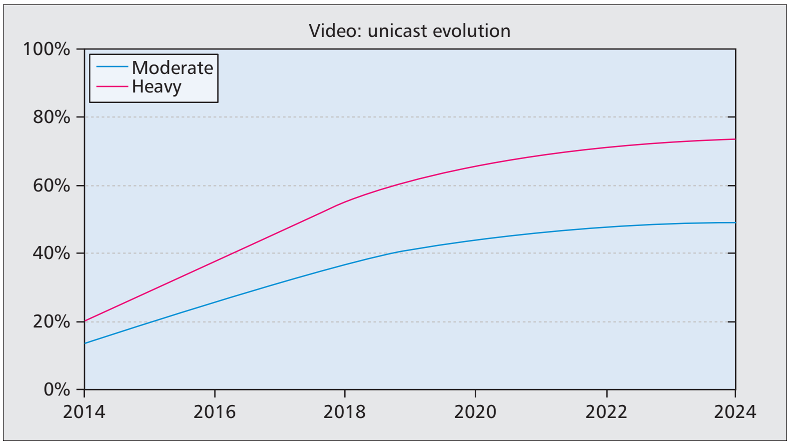

Figure 2. Evolution of unicast vs. multicast video.

Probabilistic Multicast Gain — Multicasting gain is determined by the number of unique multicast channels in an aggregate number of multicast streams. We use the same Monte Carlo model as described in [6], which assumes that channel popularity is described by the Zipf distribution, with the Zipf exponent, α, set to 0.95 to achieve the best fit to the available data.

The more channels that are offered, the lower the multicasting gain will be. Globally there is wide variability in the number of offered pay-TV channels. We assume 100 TV channels for the moderate demand scenario and 500 TV channels for the heavy demand scenario.

Probabilistic Video Resolution Forecast — For each stream transported, the resolution is probabilistically determined. We forecast a distribution of current standard definition (SD), current 720p60 high definition (HD), 1080p60 HD, and also future ultra-high definition (UHD) formats: 2160p60 UHD-1 (also known as "4k" TV) and 4320p120 UHD-2 (also known as "8k" TV) (the p stands for "progressive" scan and the number following it is the frame rate in frames per second (fps)). For the adoption of the last 3, we again use standard S-curves. Consistent with our previous forecast in [3], we expect the first significant streaming of 4k over fixed access networks occurring no earlier than 2015, with 8k becoming significant no earlier than 2020. Fig. 3 shows this stream resolution probability distribution as we forecast it to evolve over time, for both the moderate and heavy demand scenarios. We believe the streaming of the UHD formats will be constrained by the physical size of living units to accommodate multiple large screens and the limited utility of UHD

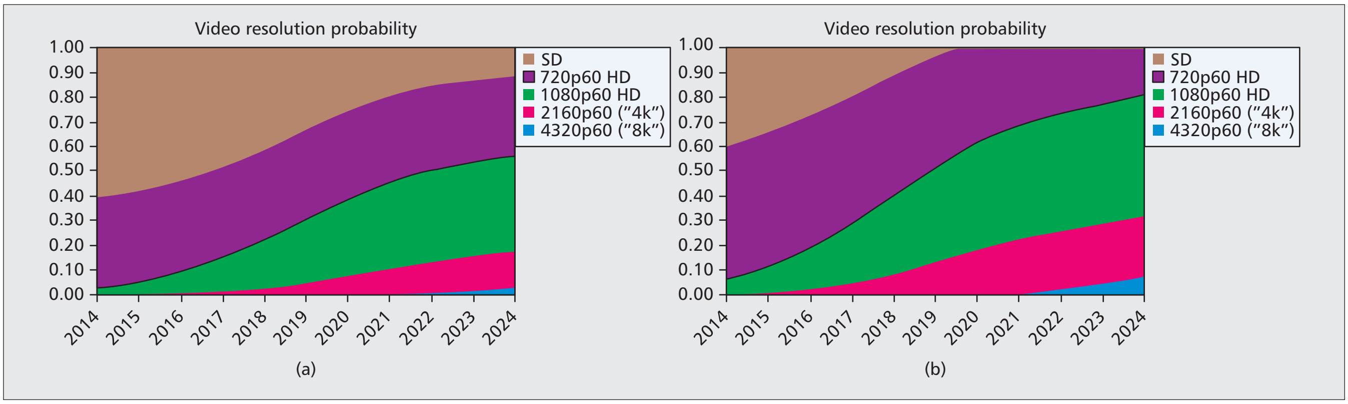

Figure 3. Stream resolution probability distribution forecast: (a) moderate scenario, slower adoption of HD; (b) heavy scenario, faster adoption of HD.

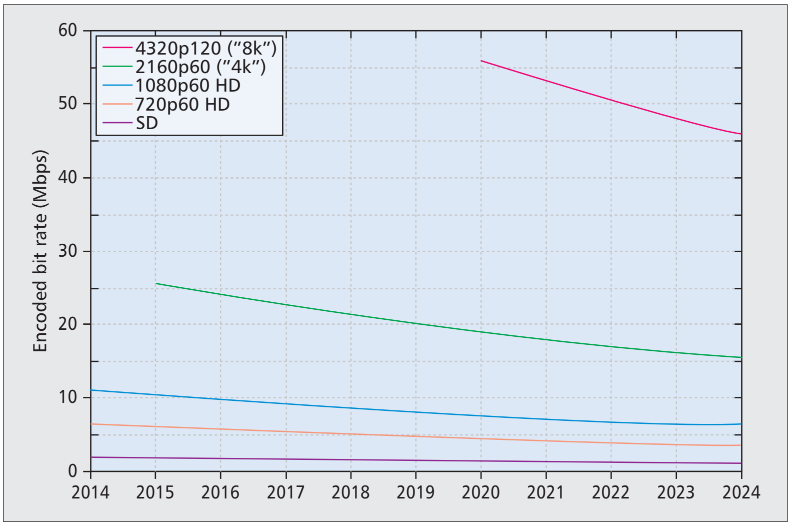

Figure 4. Encoded video bit rate forecasts.

resolution in typical viewing environments. This is explained quantitatively in [3].

Resolution Bit Rate Forecast and Quality Enhancement Factor — Previously in [3] we provided a slightly conservative (i.e. tending toward overestimation for "good" quality video) forecast of the encoded video bit rate for each of the video resolutions and frame rates used. In the meantime, the likely frame rate of 8k has been increased from 60 to 120 fps with a consequent increase in bit rate. Mitigating this increase, recent evaluations of high frame rate HEVC encoding has determined that a doubling of frame rate requires less than a 10 percent increase in the encoded bit rate for high resolution video [7]. We used 10 percent.

Accordingly, our updated, slightly conservative video resolution forecast is displayed in Fig. 4 (as before, these numbers include both video and associated audio components). In practice, some video service providers will compress more heavily than others, depending on the network capacity available and video quality objectives. This can make a significant bandwidth difference, particularly for high resolution content. To account for this, we simply multiply the HD and UHD forecasts by a "quality" factor.

The encoded HD video bit rates provided by many large IPTV operators and OTT content providers are actually lower than our forecast. Therefore, to match our forecasts to these lower, more typical numbers, in the moderate demand scenario we use a quality factor of 0.85. On the other hand, some FTTH operators with "bandwidth to burn," have elected to provide more lightly compressed video to provide perceptibly better quality for difficult scenes. To capture this case, we multiply 0.85 by the 1.5× sensitivity factor to obtain a quality factor of 1.275 for the heavy scenario.

Finally, we need to account for 3D video content. As explained in [3], we assign to 3D streams a transport bit rate that is 40 percent higher than 2D streams. We model 3D adoption with the standard S-curve. Acknowledging that 3D content has yet to gain traction outside of movie theaters, we start with virtually zero streaming 3D content today, and approach low asymptotes of 5 percent of HD streams in 10 years for the moderate scenario, and 1.5× higher (7.5 percent) in the heavy scenario. These adoption curves are shown in Fig. 5; at such modest adoption levels they have only a very small effect on the results.

The Monte Carlo Computation for Aggregated Subscribers — How the sustained bandwidth demand model combines the above elements to predict peak hour aggregated demand in a particular year is explained below. Refer to the flow chart in Fig. 6.

1) For a single subscriber, the number of simultaneous video streams delivered is randomly selected from the probability distribution in Fig. 1.

2) For each stream, whether it is multicasted or unicasted is randomly determined from the binary probability in the unicast evolution forecast in Fig. 2.

3) Steps 1 and 2 are repeated for all subscribers in the aggregate. We now have an aggregate number of multicast and unicast video streams.

4) From the aggregate number of multicast streams, the Monte Carlo multicast model probabilistically determines the number of unique multicasted channels streamed.

5) For each unicast and each unique multicast stream in the aggregate, the model randomly determines the video resolution from the probability distribution in Fig. 3.

6) For each stream, depending on the resolution, the encoded bit rate is looked up from the video resolution forecast in Fig. 4. It is then multiplied by the appropriate quality factor.

7) For HD and UHD streams, a certain percentage of the streams are 3D content, as indicated in Fig. 5. The encoded bit rates of these streams are adjusted. We now have the encoded bit rate of every video stream to the aggregated subscribers.

8) The total aggregate streaming video demand is obtained by summing the bit rates of all the streams.

9) Steps 1–9 are repeated 1000 times, yielding 1000 iterations of aggregate streaming video demand.

10) To determine peak hour demand on the access network, we take the 99th percentile of that distribution. (In principle, if the access network was sized to be exactly equal to this 99th percentile value, then in 1 percent of peak hours some video content would need to be adapted to a lower rate or would consume some fraction of the bandwidth capacity allocated to burst bandwidth demand).

This process is repeated for the various years of interest. In this article we start with the year 2014 and look forward to the year 2024.

Burst Bandwidth Demand

The second component of traffic is bursty traffic. This is more difficult to get a handle on, so we model bursty traffic in two ways:

- We look at averages over large numbers of subscribers.

- We consider the maximum burst demand that a single subscriber could place on the network.

Average Burst Bandwidth — There are no published numbers on bursty traffic per subscriber, but we can infer values from some publicly available information. A North American cable operator reported an average of 400 kb/s of downstream DOCSIS traffic per subscriber during peak hour in 2014 [8]. (Although that may contain some small amount of managed VoD and voice traffic, we will assume that to first order it is all Internet traffic.) We can attest to this number's consistency with private data from other operators that we cannot disclose. Next we need to subtract out sustained bandwidth, predominantly from OTT video, which we have already accounted for above. Sandvine [9] measurements indicate that 64 percent of peak hour downstream traffic in North America (2014) is "real time entertainment," leaving 36 percent for web browsing, file sharing, "marketplace," and other (presumably mostly bursty) traffic. Then, 36 percent of 400 kb/s gives us 144 kb/s average peak hour bursty traffic per subscriber on a DOCSIS network. We'll use this number for the moderate demand scenario. FTTH networks might encourage higher usage (especially more file transfers). Using the 1.5× sensitivity factor, we derive 216 kb/s for the heavy demand scenario.

Next we need to forecast a growth rate for burst bandwidth demand. Here we consult the VNI, which forecasts Internet traffic by application through the year 2018. For bursty traffic we add the global consumer segment traffic for web, email, data, gaming, and file sharing over fixed networks. We extract the downstream portion of this traffic assuming 50 percent of file sharing traffic is downstream, and 90 percent of all other burst traffic is downstream. In the VNI, this (downstream) application conglomerate is forecasted to grow globally at 7 percent YoY. However, part of this 7 percent growth comes from the global increase in subscribers. From the VNI interactive tool we can infer a global subscriber growth rate of 5 percent YoY, and factor that out of the traffic growth. After making that adjustment, we arrive at a global per-subscriber burst traffic growth rate of 2 percent YoY. This may sound low, but most of today's Internet growth is driven by OTT video, and a large component of burst traffic is peer-to-peer file sharing, which is forecasted to decline on a per-subscriber basis.

In our model, we'll use this global average of 2 percent for our moderate scenario. We'll make the same argument as above for the heavy scenario: that the deployment of more fiber access networks (and faster networks in general) may lead to accelerating use of applications associated with high burst bandwidth (for example, movie downloads). With no data to guide us, we will (admittedly arbitrarily) assume 10 percent YoY growth for the heavy scenario.

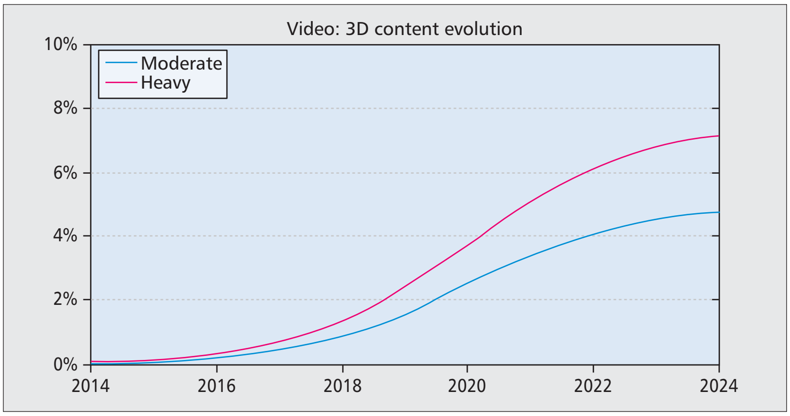

Figure 5. Evolution of 3D content as a percent of all video streams.

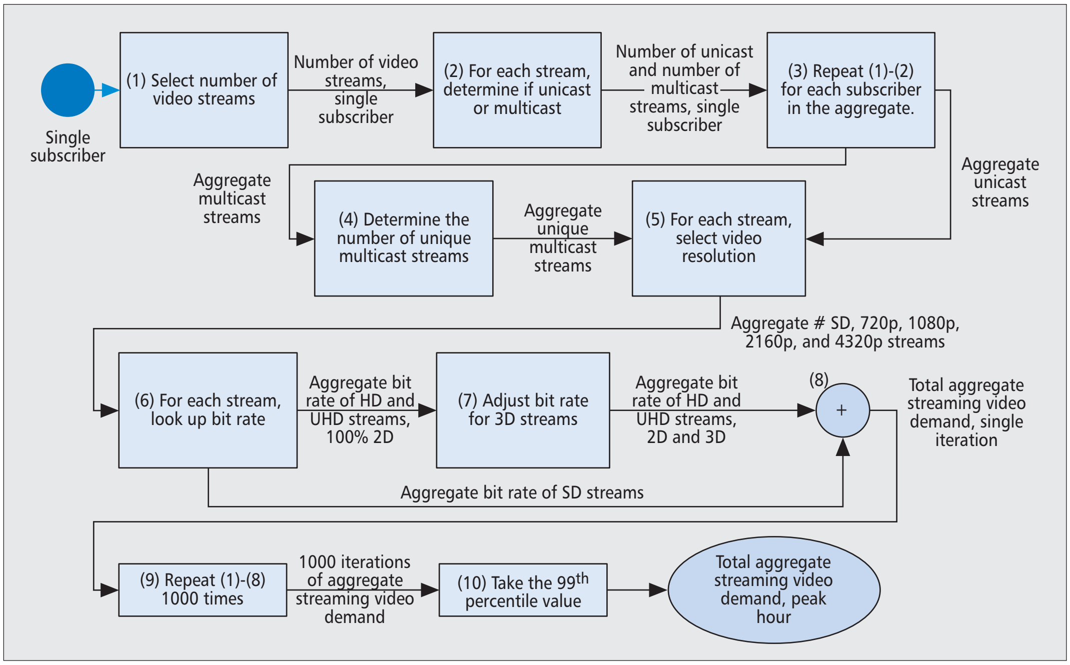

Figure 6. Flow of Monte Carlo sustained bandwidth demand model.

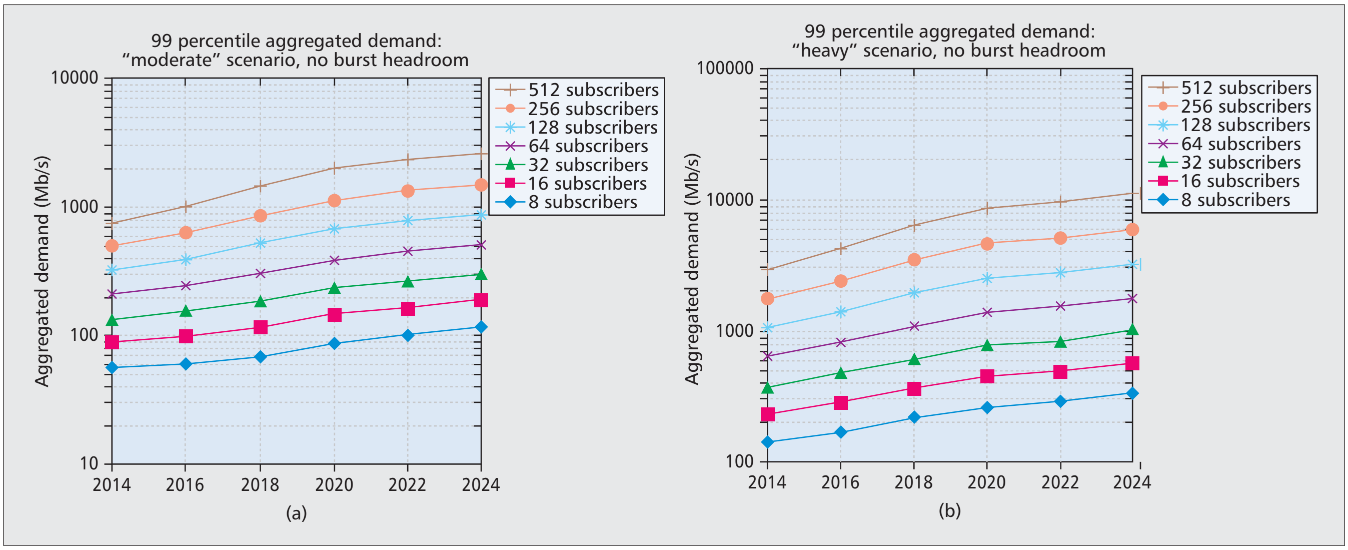

Figure 7. Forecasted demand over time for different number of aggregated subscribers (n); no headroom.

These average per-subscriber traffic numbers are small. We need another mechanism to quantify peak bursts, to be discussed next.

Maximum Single Subscriber Burst Demand — The access network must accommodate a burst rate that corresponds to the highest offered service tier. More concretely, a single subscriber should be able to run a speed test on his broadband connection, while all other subscribers on the network are receiving traffic at peak hour demand, and achieve a speed test result equivalent to his advertised service. To accommodate this in our model, we add burst bandwidth headroom that is equivalent to the maximum service bandwidth offered on the network. (While the authors of [4] consider the peak demand of many bursty applications in a sophisticated bottoms-up method, they don't consider this one application, the speed test, which will produce the highest peak, and the most dissatisfaction if the network fails to deliver on its promise. One could argue whether the speed test should be considered an actual demand or not, but in a world of competing networks we think it is the reality of how networks are being judged. It is in fact the first "killer app" for Gigabit speeds.)

This headroom will, correctly, dominate bandwidth demand for small aggregates of subscribers, and will become relatively less important for larger aggregates, as the video demands scale with the number of subscribers. Below we will suggest headroom values that correspond to premium service tiers that are appropriate for each of the access network architectures under consideration.

(Even in a small aggregate, it is possible that two subscribers push the speed test button at the same time, and may therefore see diminished test results, but we consider this case to be the exception and below we assume networks will not be designed to support this scenario. To accommodate simultaneous speed tests, the bandwidth headroom will just need to be increased accordingly).

Total Burst Bandwidth Demand — To arrive at a total burst bandwidth demand, we achieve an approximation by simply multiplying the average burst demand per subscriber by the number of subscribers in the aggregate, and then add to that the headroom corresponding to a single subscriber maximum burst demand.

Total Bandwidth Demand

In the final step of the model, for a given number of aggregated subscribers in a given year, the 99th percentile of forecasted sustained bandwidth demand is added to the total forecasted burst bandwidth demand, to arrive at a total aggregate bandwidth demand.

(It is worth explaining why there is no separate bandwidth demand category for cloud services, which are a class of services where a client is using remote instead of local compute and storage resources. Cloud services do not generate new kinds of traffic. For example, traffic generated by a client creating and editing a document on a remote server, for example, Google Docs, will generate short bursts of conventional TCP/IP traffic. Remote storage, whether general files, for example, Google Drive, or photo storage, for example, Flickr, will generate large file transfers over conventional TCP/IP. Remotely delivered streaming video, for example, Netflix, will generate sustained TCP/IP traffic. All of these traffic types are already covered in the model above.)

Results

We forecast aggregated demand over the years 2014–2024. We consider the size of the aggregate, n, over the range of 8–512 subscribers. Forecasted demand vs. n over time, for the moderate and heavy scenarios, is shown in Fig. 7. In this figure, *we do not yet include the burst bandwidth headroom*, as the size of the headroom depends on the capability of the particular last mile access technology. We make several observations:

- Increasing aggregate bandwidth demand vs.

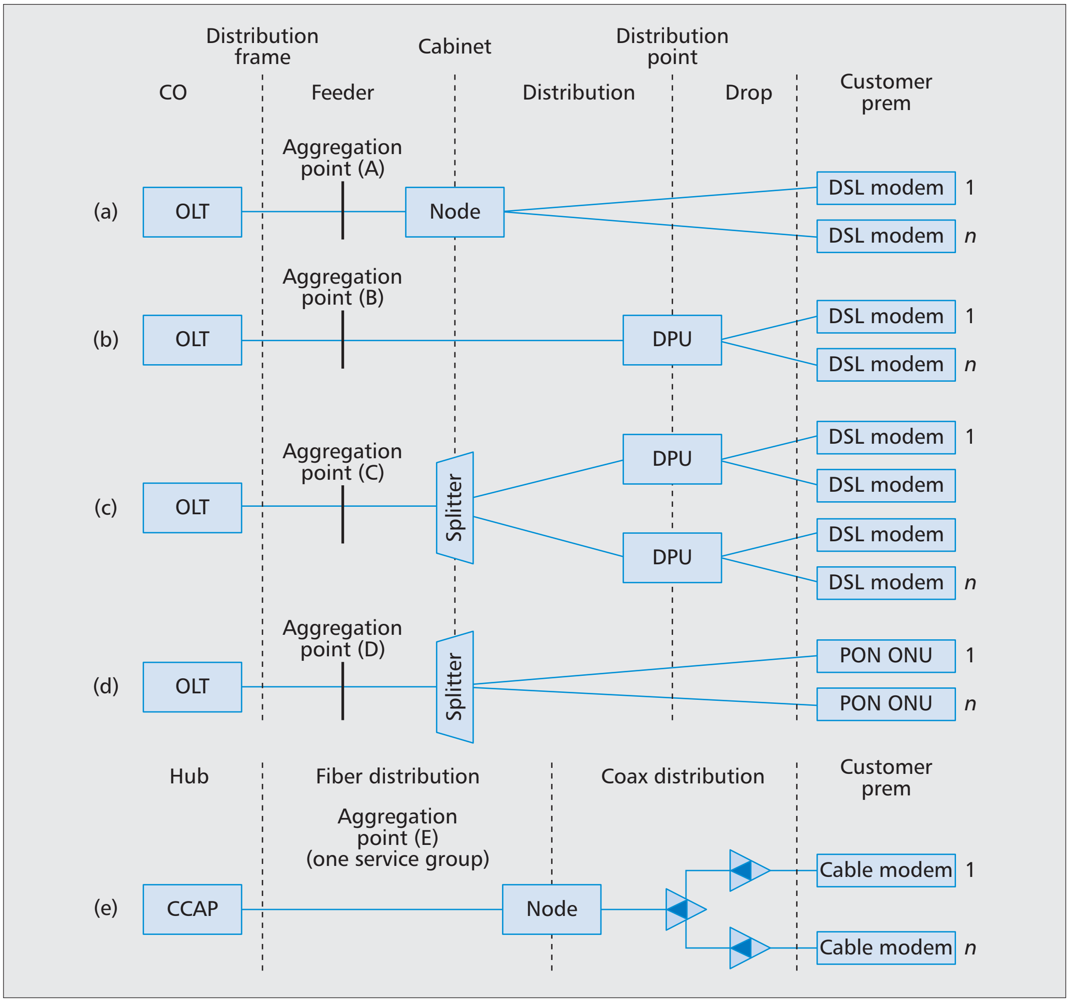

Figure 8. Aggregation points on fixed access networks: (a) PTP backhaul of FTTN; (b) PTP backhaul of FTTdp/FTTB; (c) PON backhaul of FTTdp/FTTB; (d) PON FTTH; (e) HFC network.

n. This relationship is not linear, due to two phenomena. First, multicasting becomes more effective and greatly compresses the demand for higher *n*. Second, for small aggregates (small sample sizes), there is a larger statistical relative variance in demand, which must be accounted for when designing for 99th percentile demand.

• Increase in aggregate bandwidth demand over time. This growth is 8–11 percent YoY for small aggregates and 10–14 percent for large aggregates. The moderate demand scenario is on the low end of those ranges and the heavy demand scenario on the high end. This is again explained by the role of multicast gain in saving bandwidth, which has its largest effect on large aggregates, but whose role diminishes as the proportion of unicasted channels increases over time (and more quickly in the heavy scenario).

• Heavy demand vs. moderate demand. The ratio of heavy demand to moderate demand is roughly 3.5, and is, of course, simply a product of all the more aggressive input assumptions made for the heavy scenario. Slightly more interesting is that the ratio ranges from 2.8 for smaller *n* to 4.3 for larger *n*. Once again, this is caused by the diminished role of multicasting in the heavy scenario (higher proportion of unicasted streams, more channels offered), which has a strong effect for large *n* but little effect for small *n*.

• Although not shown explicitly in the figure, burst traffic only accounts for a few percent of overall traffic. To even get within the 10–20 percent range of overall traffic by year 2024, burst traffic would have to grow about 50 percent annually in the moderate scenario and 80 percent in the heavy scenario. Neither of these extreme growth rates are expected. Therefore it is possible to dimension access networks solely on video traffic forecasting plus headroom to accommodate subscriber speed tests.

Applying Aggregate Demand Forecasts to Network Planning.

"Telco" Access Network Architectures

The sizes of the aggregates correspond to aggregation points in different access network architectures. The "telco" access network architectures and the corresponding aggregation points (A, B, C, and D) considered in this article are indicated in Figs. 8a–8d. (We note that cable operators

Fixed access architecture	Range. of aggregated subscribers (n)	Example top service level (= burst headroom)
PTP backhaulof FTTN	32–256	100 Mb/s
PTP backhaul of FTTdp/FTTB	8–32	500 Mb/s
PON backhaul of FTTdp/FTTB	64–256	500 Mb/s
PON FTTH	16–64	1 Gb/s
HFC	64–512	Variable

Table 1. Fixed access network dimensions.

have begun to roll out PON residential networks, so PONs are actually not for telcos only).

In Table 1 we indicate, for each architecture, the likely range for the size of the aggregate, n, and suggest a top service level that might be offered for each architecture, commensurate with the bandwidth capabilities of the last mile access technology.

In order to plan for an access network to be able to deliver the bandwidth to meet the forecasted demand, these steps are followed:

1) For a given aggregate (n) and planning horizon (range of years), the forecasted demand can be read from the graphs in Fig. 7, depending on whether the moderate or heavy scenario is most likely to represent the behavior of the operator's subscribers.

2) To that demand, the burst headroom (for instance, the applicable value in Table 1) is added. This is the total demand that must be planned for.

3) The network planner must either build the network to support the demand at the end of the planning horizon, or to at least support initial demand and have the network scale to support the ultimate demand.

For architectures 8a and 8b employing PTP backhaul of remote access nodes (e.g. a VDSL2 DSLAM) and G.fast distribution point units (DPUs), the basic PTP facilities available to network planners are Gigabit Ethernet and 10 Gigabit Ethernet. In most cases, a single Gigabit Ethernet will suffice in the near future, but will need to be reinforced with additional Gigabit Ethernet links as demand grows. Alternatively, a single 10 Gigabit Ethernet backhaul link could be deployed on day one, with no need to reinforce it for many years. For the larger aggregates in the heavy scenario, it may be best to start with a 10 Gigabit Ethernet on day one and be done with it.

For PON backhaul facilities, the basic choices are GPON (2.5 Gb/s down and 1.25 Gb/s up) and various 10 Gb/s PON technologies (XG-PON and 10G EPON). For PON backhaul of DPUs in architecture 8c, GPON should be adequate through 2024 in the moderate demand scenario. For most cases in the heavy demand scenario, at some point in the next 10 years the bandwidth of a GPON would be exhausted and a 10 Gb/s PON required. However, when deploying FTTH (8d), GPON should be more than adequate for the next 10 years, even in the heavy demand scenario if the number of subscribers on a PON is limited to about 32. On the other hand, a more future-proof 10 Gb/s PON may well be deployed when its price premium over GPON becomes small enough.

Cable Network Architecture (HFC)

The aggregation point (E) for an HFC cable network is shown in Fig. 8e. The size of the subscriber aggregate n is known in DOCSIS terms as the service group (SG) size. As cable operators split their nodes and go fiber-deep, the value of n, starting at about 256 to 512, may decrease to about 64.

In the vast majority of today's HFC networks, linear pay-TV is broadcasted over MPEG transport streams, while unicast services are delivered by DOCSIS facilities to a cable modem in the home. However, in the future all pay-TV channels are expected to be transported over IP over DOCSIS. It is this latter scenario that corresponds to our aggregate bandwidth demand model.

The top service level offered on HFC networks will increase over time as DOCSIS downstream bandwidth is scaled from relatively small amounts (on the order of 100 Mb/s) to relatively large amounts in the future (on the order of 1 Gb/s and perhaps beyond). Therefore the required burst headroom will likely increase over time. Network capacity planning on the HFC network is done according to the same steps as previously described, but accounting for this variable burst headroom.

For a cable operator, the required downstream DOCSIS bandwidth is a function of the SG size. For example, to provide all-IP video and a 1 Gb/s service tier in the year 2020, a combination of 1.7 Gb/s of DOCSIS bandwidth serving a 128 subscriber SG will suffice, in the moderate demand scenario. In the heavy demand scenario, 2.4 Gb/s serving a 64 subscriber SG will suffice. Satisfying these bandwidths and SG sizes will require more downstream DOCSIS bandwidth and spectrum allocation, and some combination of fiber node splitting and deeper fiber. On brownfield HFC networks with a legacy base of cable modems and set-top boxes, evolving to these points will require careful planning.

(Although out of the scope of this article, DOCSIS upstream bandwidth is constrained and may prove to be problematic in markets where cable operators compete with FTTH operators.)

All-Unicast Video Case

Until now we have assumed that live linear pay-TV is always multicasted, which is typical for today's managed IPTV networks using IP multicasting. However, it is possible that in the future an operator might deploy live linear pay-TV service using emerging HTTP-based unicast delivery. We now model this future all-unicast video scenario. With no multicasting gain available, the demand on the network will be higher.

Figure 9 is a reprise of Fig. 7 for both the moderate and heavy scenarios, but with 100 percent unicast video traffic from day one. Without multicasting gain, the aggregate demands are higher, especially in the earlier years for which we previously assumed a relatively high percent-

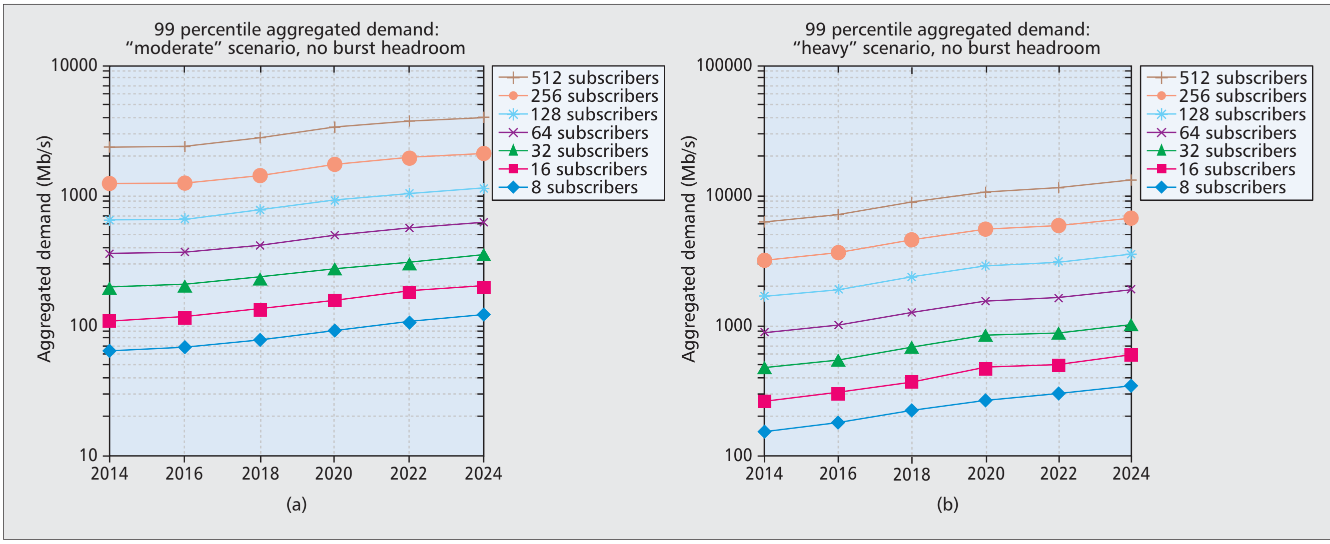

Figure 9. Forecasted demand vs. the number of aggregated subscribers: 100 percent unicast video case; no headroom.

age of multicast video. Even with the higher demands, the same PTP and PON technologies considered above will serve throughout the forecast period.

Another outcome is that the growth rates are lower, now that the decrease in multicast gain over time no longer applies. We see demand growing at about 6 percent YoY for the moderate demand scenario, and at about 8 percent YoY for the heavy demand scenario over all the aggregate sizes. These growth rates fit easily under the 15 percent YoY upper bound growth rate we forecasted in [3].

Conclusion

There are many forecasts in the telecommunications industry about the future of bandwidth demand. None of these forecasts adequately address the realistic aggregate demands on future fixed access networks, a critical consideration for operators making investment decisions on their network infrastructure. Filling that need is the purpose of the new statistical model developed and described in this article. The forecast results published here, bounded by "moderate" and "heavy" bandwidth demand scenarios, should provide operators with guidance on how to choose and size various kinds of access network topologies and technologies.

In general, our downstream residential demand model predicts that for the next 10 years, video traffic will continue to dominate, overall traffic will grow between 8 percent and 14 percent YoY, existing Gigabit and 10 Gigabit Ethernet and GPON and 10G PON technologies will provide enough bandwidth for both FTTH and residential backhaul applications, and HFC networks will need to evolve quickly and significantly if required to provide all-IP video and 1 Gb/s service.

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Biographies

ED HARSTEAD joined Bell Laboratories in 1983. During the 1990s, as a systems engineer and systems engineering manager, Ed led R&D efforts in first generation BPON and WDM PON technologies, while taking a lead role in the FSAN standardization forum. During the 2000s, Ed was a senior product line manager for both optical transport and broadband access products. His current role at Alcatel-Lucent is lead technologist for the Chief Technology Office of the Fixed Networks Division. Ed received the MS degree in mechanical engineering and the MS degree in electrical engineering from Columbia University, New York City.

RANDY SHARPE received a B.S. degree in electrical engineering from the University of Michigan in 1978 and an M.S. degree in electrical engineering from the Massachusetts Institute of Technology in 1979. He joined Alcatel in 2001 and is currently a distinguished member of technical staff in Alcatel-Lucent's Fixed Networks CTO organization with a focus on broadband access networks, IPTV, and SDN. Previously, he was a member of technical staff at AT&T Bell Labs working on digital television compression and transmission products, and later was a founder and architect at BroadBand Technologies. Randy Sharpe is a member of the Alcatel-Lucent Technical Academy.

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STANDARDS FOR INDOOR OPTICAL WIRELESS COMMUNICATIONS

CLOUD ARCHITECTURE FOR SENSOR CONTROLLING AND MONITORING

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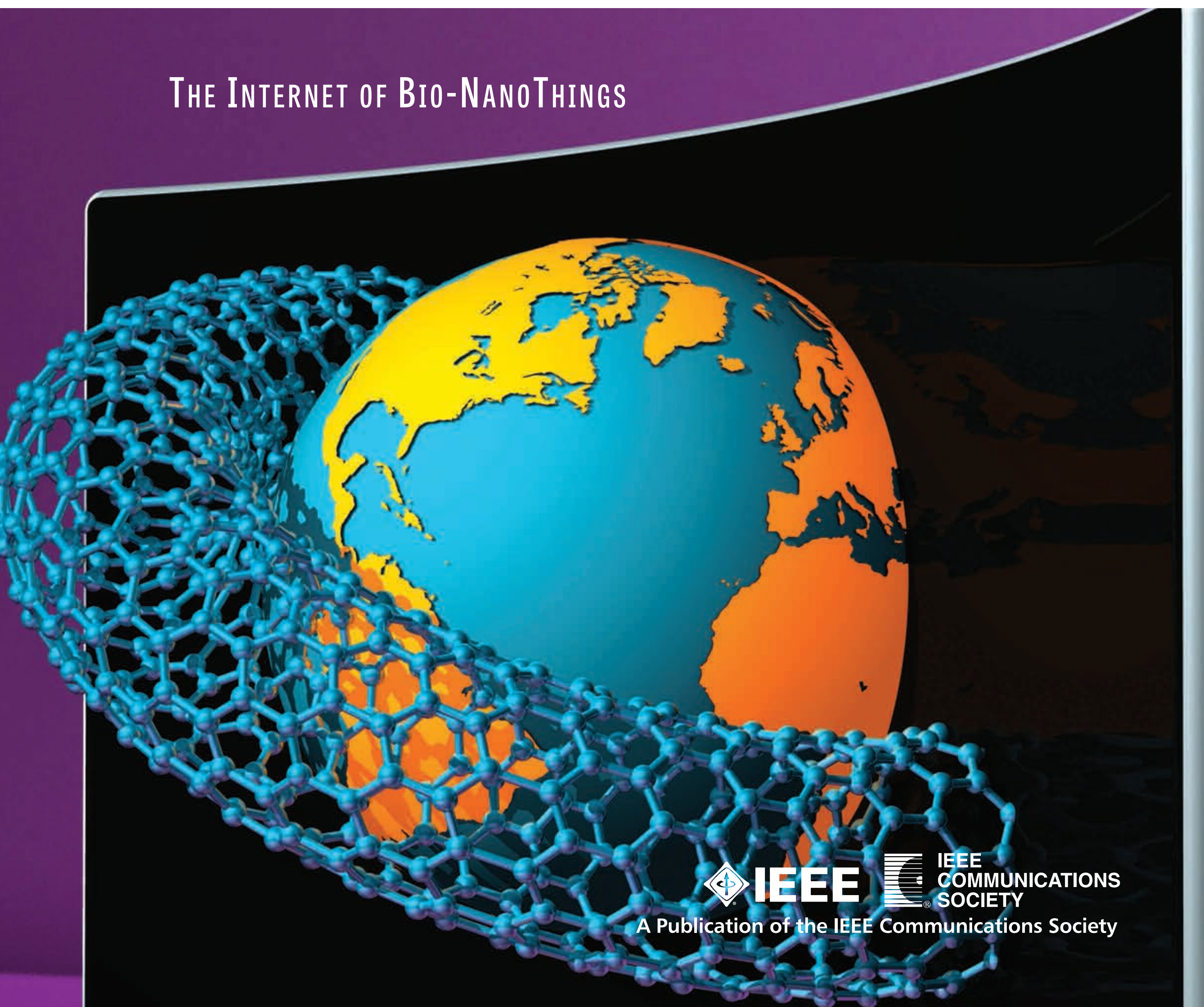
ADAPTABILITY BETWEEN PAYMENT CARD PROCESSING NETWORKS



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MARCH 2015

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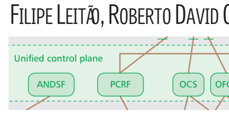

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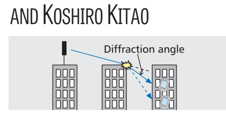

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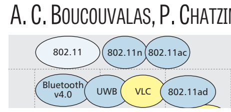

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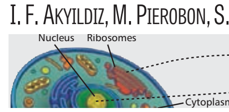

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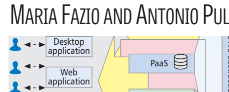

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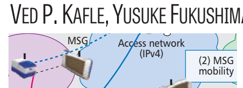

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PAPERS FROM THE JUNE 2014 ITU KALEIDOSCOPE CONFERENCE

Glenn Parsons

Previously, companies could publish the interfaces or other aspects of their products and expect that they would become a standard. In fact, there are more examples of this failing, than examples where a company's market position made its product the de facto standard. Today, consumers and customers expect multi-vendor interoperability and will not tolerate proprietary interfaces. There is a need for market driven standards.

Standards must be marketed and sold to the industry like any other product. This requires an investment of people and time to document the standard, articulate its benefits, and to bring it into suitable existing standards development organizations (SDOs) or perhaps initiate a new forum. At SDOs, like IEEE-SA, proponents from multiple companies advocate the standard to customers, allies, and competitors alike, negotiate the details, work the SDO/forum politics, and hopefully at the end of the day, gain industry concurrence on something close to what we originally proposed. To do this successfully requires a sustained effort over time by a set of experienced standards professionals.

The importance of standards to the work and careers of communications practitioners is the basis of this publication. It is a platform for presenting and discussing standards-related topics in the areas of communications, networking, and related disciplines. The very successful launch, in December 2014, of the *Communication Standards Supplement* in *IEEE Communications Magazine* brings high expectations as we start the new year.

For this issue, a number of papers are based on those accepted at the June 2014 ITU Kaleidoscope conference "Living in a converged world: impossible without standards?" This peer-reviewed academic conference, organized by the ITU, brought together a wide range of views from universities, industry, and research institutions. The aim of the Kaleidoscope conferences is to identify emerging developments in information and communication technologies (ICTs) and, in particular, areas in need of international standards to support the development of successful products and services. The peer review for publication in this supplement was led by Mostafa Hashem Sherif and Yoichi Maeda.

Going forward, future issues of the Standards Supplement will be "anchored" around a topic of current market relevance to drive focus, which is similar to the feature topic model of *IEEE Communications Magazine*. Feature topics this year include "Cloud and Virtualization for 5G" and "Internet of Things (IOT)." Proposals for future standards feature topics are welcome. In this issue, readers will notice a commentary section from leaders of IEEE-SA and ITU-T. Also, Standards News continues with several SDOs offering current status and pointers to SDO material.

Despite the broad call for papers, the articles in this issue of the *Communications Standards Supplement* provide an update on many topical standards technologies and issues. I trust that the reader will find these informative and illustrative of the fundamental role standards play in the communications networking ecosystem.

The first article introduces a unified control plane for subscriber traffic policy and charging. This fixed mobile policy convergence is being discussed in 3GPP and BBF. Letaio *et al.* thoroughly review the use cases of service provider offerings like shared data, parental control, and voice (over IP, LTE, and Wi-Fi) that are driving the requirement. 3GPP and BBF have been collaborating on aspects of fixed mobile convergence that targets operators with both fixed and mobile networks. Finally, the article highlights the different aspects of the unified control plane standards solution that consists of a convergent policy and charging control architecture.

Following on with the 3GPP standards activities, Mondal *et al.* provide an introduction to the standardization proposals for a 3-dimensional (3D) channel model for LTE. The existing 2-dimensional (2D) channel models are insufficient to improve the spectral efficiency and reliability of a radio-link, especially with new multi-antenna transmission techniques capable of exploiting the elevation dimension. Currently, 2D models standardized in ITU-R are being augmented with 3D models, to allow for appropriate modelling of LOS and NLOS use cases in 3GPP. The article describes the use of three 3D model proposals and their applicability to line-of-sight probability, path loss, and fast fading for LTE end user devices (UEs). The models are key to understanding elevation beamforming and FD-MIMO performance.

Optical wireless communications (OWC) has recently been attracting much attention, as it combines the best of wireless and optical standards and technologies. The article by Chatzimisios *et al.* [5] considers three standards activities for OWC and suggests that the industry would benefit from common standards. The IEEE 802.11 infrared standard is described, giving insight into why it was discontinued. The standards of IrDA are then summarized, describing them as targeted at "point and shoot" applications optimized for short interaction times and very high capacities. The IEEE 802.15.7 standard for visible light communications focuses mainly on end-user access, providing permanent connections, moderate capacity, and acceptable user mobility. Overall, the article gives a good summary of the current status of OWC standards and hints at future standardization endeavours, such as the migration to link speeds of over 10 Gb/s.

The article by Balasubramaniam *et al.* is based on a keynote speech at ITU Kaleidoscope and extends the current IOT standardization topic toward an Internet of Bio Nano Things (IoBNT). IoBNT stands as a paradigm-shifting concept for communication and network engineering. This would involve biological cells' functionalities, such as sensing, actuation, processing, and communication, being networked together to allow an unprecedented intra body information exchange (not to mention the benefits for environmental management). The article then describes that IoBNT devices stem from biological cells, and are enabled by synthetic biology and nanotechnology. As a result, communication must be molecular between IoBNT devices and several examples are explored. Finally, the article examines connectivity between these devices and the Internet, and the numerous challenges that remain.

Following on with sensing activities, the article by Fazio *et al.* is based on another keynote speech at the ITU Kaleidoscope conference. The article describes two strategies for managing sensing resources in the cloud and providing them as a service. A solution is presented based on standards from the Open Geospatial Consortium called Cloud4Sens, and provides sensing by combining two models: a data-centric model, where the cloud offers environmental data to its clients as a service, without any knowledge of how data are measured and processed, and a device-centric model, which enables the cloud clients to use a virtual sensing infrastructure. The article concludes with a description of the Cloud4Sens architecture as well as two potential use cases for the solution.

The article by Kafle *et al.*, also part of ITU Kaleidoscope, presents a dynamic mobile sensor network platform that is based on future network standards developed by ITU-T SG13. It uses ID-based communication to identify the nodes. The mobile sensor network architecture is described, and the technical feasibility of the proposed platform has been demonstrated by implementing a prototype. The article concludes with several use cases and suggestions for future standardization.

The final article illustrates how standardization does not necessarily mean interoperability. Mian *et al.* explore the widely deployed ISO 8583 and its derivatives, which describes the request and response cycle of credit/debit card originated transactions. As a result of the flexibility of the standard, its implementations vary slightly vary. The article describes relevant aspects of the ISO standard and several of the varying implementations. The article proposes a meta-data sharing method that would allow the flexibility provided by ISO 8583, but also ensure that vendor implementations of the standard are interoperable.

BIOGRAPHY

GLENN PARSONS [SM] (glenn.parsons@ericsson.com) is an internationally known expert in mobile backhaul and Ethernet technology. He is a standards advisor with Ericsson Canada, where he coordinates standards strategy and policy for Ericsson, including network architecture for LTE mobile backhaul. Previously, he has held positions in development, product management, and standards architecture in the ICT industry. Over the past number of years he has held several management and editor positions in various standards activities, including IETF, IEEE, and ITU-T. He has been an active participant in the IEEE-SA Board of Governors, Standards Board and its Committees since 2004. He is currently involved with mobile backhaul standardization in MEF, IEEE, and ITU-T, and is chair of IEEE 802.1. He is a technical editor for *IEEE Communications Magazine* and has been co-editor of several IEEE Communications Society magazine feature topics. He graduated in 1992 with a B.Eng. degree in electrical engineering from Memorial University of Newfoundland.

Brief Notes on the Evolution of the IEEE Standards Association

By Bruce Kraemer, President, IEEE Standards Association

It is my pleasure to offer some comments on the IEEE Standards Association (IEEE-SA) for this second *Communication Standards Supplement* to *IEEE Communications Magazine*.

First a few bits of ancient history. The IEEE-SA structure of today was established when it became a formal IEEE business unit in 1998. But IEEE development of standards can be traced much further back — to 1890 and one of the IEEE's predecessor organizations, the American Institute of Electrical Engineers (AIEE) which created a recommendation for the practical unit of self-induction.[1] We now have a portfolio of more than 1,100 active standards published under the IEEE brand name. There are currently over 500 standards being actively developed. I expect to see over 100 of those completed and published this year, and another 125 new ones started during 2015.

Having officially been IEEE-SA President for five weeks at the time of this writing, I can't offer a great deal of first-hand history. That would be a task better handled by my recent predecessors, Chuck Adams, Steve Mills, Karen Bartleson, and those before them. I can, however, provide some news on recent events, an assessment of where we are, and with 99 weeks left to serve in this role, I will dare offer some thoughts about the future in this and subsequent columns.

Creation of standards is normally the responsibility of IEEE Societies and Councils, who sponsor and administer 99 percent of IEEE standards-development projects. Only the remaining one percent is sponsored by the IEEE-SA itself. In this way, the IEEE-SA's relationship with IEEE Societies and Councils is synergistic: IEEE Societies and Councils provide technical oversight in development of standards; the IEEE-SA provides process oversight.

The IEEE-SA's primary role is to ensure that an optimal environment is provided for the development of standards. In this capacity, the IEEE-SA manages the process rules associated with initiating, developing, and completing standards projects.

One component of the development process that has recently attracted attention is the IEEE-SA Patent Policy (contained in Clause 6 of the IEEE-SA Standards Board Bylaws). The IEEE-SA Standards Board has a Patent Committee (PatCom) that reviews opportunities to improve its Patent Policy and related procedures, and the text in Clause 6 has been under review for about two years. Updates to Clause 6 were recently approved by the IEEE-SA Standards Board, the IEEE-SA Board of Governors, and the IEEE Board of Directors. Information about the IEEE-SA Patent Policy can be found at: **http://www.ieee.org/about/news/2015/8_february_2015.html**. Prior to the updated Policy going into effect, IEEE-SA will distribute additional information.

But the IEEE-SA does more than just administer the standards development procedures. It also offers increasingly important services in pre- and post-standards development.

In the pre-standards area, the Industry Connections program (**http://standards.ieee.org/develop/indconn/**), active since 2010, helps incubate new standards and related products and services by facilitating collaboration among organizations and individuals as they hone their thinking on rapidly changing technologies. Industry Connections programs produce a variety of results, such as proposals for standards, white papers, position papers, conferences, workshops, databases and registration services, software, tools, and web services. Seventeen Industry Connections programs involve well over 100 participating companies and several hundred individual participants.

In the post-standards-development area, there is a relatively new activity, the IEEE-SA Conformity Assessment Program (ICAP) that offers a convenient way for adding an IEEE-certified test procedure to IEEE published standards. ICAP (**http://standards.ieee.org/about/icap/**) certification extends the IEEE standards-development value chain by demonstrating that a product is compliant with requirements as agreed by ICAP, industry, and its authorized test laboratories. The certification process references test specifications developed by subject-matter experts. Two certification programs have currently launched, and several more are expected to go live during 2015.

By offering development support services across the standards lifecycle, the IEEE-SA nurtures the global development of advanced technologies. In such ways, the IEEE-SA builds on the early-phase technology exploration of IEEE Societies and Councils and helps put innovation to work in the marketplace for the benefit of humanity.

I look forward to continuing this conversation in future issues of this magazine. The IEEE-SA is working to increase collaboration with IEEE Societies and Councils generally, and the IEEE Communications Society specifically, and I hope to share more details as they develop in the coming months.

[1] http://www.ieeeghn.org/wiki/index.php/IEEE_Standards_Association_History

ICT Standards: Building a Trusted Information Society

By Chaesub Lee, Director of the Telecommunication Standardization Bureau, International Telecommunication Union (ITU)

Standards provide a common language, and standards development is a process that builds commitment to consistent use of that language. Benefitting from incredible amounts of standards developed by various communities, society has been able to evolve and progress continuously, moving from an Agricultural Society to an Industrial Society to today's Information Society. Standards have provided the building blocks of the Information Society, with ICT devices generally adhering to an enormous collection of standards developed by a wide range of communities.

Technical standardization has been at the core of ITU's mandate since the organization's inception in 1865, providing stimulus to communication-oriented innovation from the day of the telegraph to today's converged ICT ecosystem. It was in 1994 that the proposal to create a Global Information Infrastructure (GII) for building the Information Society was presented at an ITU governing conference. Today's Information Society is a result of the tireless efforts by ITU members and partners since 1994 to encourage the development and deployment of advanced broadband and mobile networks, designed to provide users across the globe with access to information anywhere, anytime, anyhow.

This year ITU is celebrating its 150th anniversary in tribute to the extraordinary innovation of the global ICT community and the crucial role that ITU plays as a platform to bring cohesion to this innovation. Our 150th anniversary will be both a celebration of ITU's past achievements as well as an analysis of how ITU can best position itself to serve the future development of the Information Society.

Embedding trust into the Information Society will provide a greater level of certainty, confidence, and predictability in network interactions, expanding their scope and benefits.

To achieve this, ITU will work in harmony with other standards bodies to enhance the level of trust in the current digital ecosystem while also considering the impacts of possible future developments. ITU will continue to support smart, context/content-aware, user-centric technologies, while being cognizant of, and in turn informing, the relevant policy and regulatory debates and frameworks.

The value of ITU-T standards stems from the value of their development process. As a United Nations specialized agency, our standardization process must enable peer-learning and knowledge-exchange to assist developing countries in advancing their ICT infrastructure and encouraging economic development. The benefits of international standards should be open to all, as should the global meeting of expertise that drives their development.

Our success in cooperating and collaborating with other bodies such as IEEE-SA is another key measure of the value of ITU-T standards. It is crucial that we take an 'ecosystem' view of ICT standardization. This is gaining in importance every day, driven by the accelerating convergence of technologies and industry sectors.

Standardization experts — asked to consider the implications of 'trust' in the digital ecosystem as well as value chains, taking into account the communications of cyber-physical systems such as billions of networked devices, things, and objects — must preempt relevant technologies' requirements on supporting trustworthy ICT infrastructures. How will we provide the required level of trust for such a ubiquity of connected devices, things, and objects? And in the presence of data collection, storage, processing, analysis, and sharing on such a massive scale, how will we build ICT infrastructures worthy of our trust?

These questions are extremely multifaceted. There are a myriad of perspectives to be considered, and ITU-T will work to provide the global standardization community with an open, neutral platform to bring cohesion to ICT innovation through year 2020 and beyond.

Wireless Access Standards and Spectrum in ITU-R

Jose Costa, Ericsson, Ottawa, Canada

The ITU-R work on Wireless Access Standards is conducted in Working Party 5A (WP 5A), which is responsible for studies related to the land mobile service, including wireless access in the fixed service, but excluding International Mobile Telecommunications (IMT) that is the responsibility of Working Party 5D (WP 5D), comprising IMT-2000 systems, IMT-Advanced systems, and the future development of IMT. Radio interface standards for broadband wireless access systems, for both mobile and nomadic applications, are covered in Recommendation ITU-R M.1801, which includes by reference the RLAN standards (Rec. ITU-R M.1450) and the IMT standards (Recs. M.1457 for IMT-2000 and M.2012 for IMT-Advanced). For further information refer to the guide[1] for the use of ITU-R texts related to the land mobile service, including wireless access in the fixed service that is available on the WP 5A webpage.

Besides the ongoing work on IMT standardization and spectrum harmonization, a major project has been initiated in WP 5D toward the definition of requirements and standards for the next generation mobile networks, "IMT for 2020 and beyond". WP 5D is studying the definition of a work plan, timeline, process, requirements, and deliverables for the future development of IMT, necessary to provide by the 2020 timeframe the expected ITU-R outcome for the evolution of IMT in support of the next generation of mobile broadband communications systems. WP 5D is using the moniker "IMT-2020" as a placeholder terminology, and the specific nomenclature to be adopted for the future development of IMT is expected to be determined at the Radiocommunication Assembly 2015 (RA-15), which will be held in Geneva, Switzerland, 26–30 October 2015.

In addition, efforts are ongoing in ITU-R to consider additional spectrum allocations to the mobile service on a primary basis and identification of additional frequency bands for IMT and related regulatory provisions, to facilitate the development of terrestrial mobile broadband applications, under agenda item 1.1 of the World Radiocommunication Conference 2015 (WRC-15), which will be held in Geneva, from 2–27 November 2015. Furthermore, due to the unprecedented growth of mobile broadband in recent years, in terms of both number of subscribers and bandwidth-rich applications, discussions are underway about a possible new agenda item for a future WRC, such as WRC-19, to consider identification of certain bands for IMT in higher frequency ranges, say between 6 GHz and 100 GHz, including possible additional allocations to the mobile service on a primary basis if required, that would be capable of supporting much wider contiguous bandwidths, such as at least 500 MHz. This is being supported by research on IMT above 6 GHz that is being carried out by various projects and organizations on a global scale as well as by WP 5D. In particular, several presentations on this were made at the "workshop on research views of IMT beyond 2020"[2] hosted by ITU during the 18th meeting of WP 5D in February 2014.

[1] http://www.itu.int/oth/R0A06000001/en

[2] http://www.itu.int/dms_pub/itu-r/oth/0a/06/R0A060000630001MSWE.docx

ITU-T Study Group 15 at a Glance

Stephen Trowbridge, Alcatel-Lucent, SG15 Chairman

ITU-T Study Group 15, Networks, Technologies and Infrastructures for Transport, Access and Home, is the largest and most prolific of the ten ITU-T Study Groups. SG15 standardizes architectures of optical transport networks as well as physical and operational characteristics of their constituent technologies.

SG15 produces the digital subscriber line (DSL) standards that provide broadband Internet connections to over 600 million households around the world. The group continues to challenge the existence of a ceiling to network capacity in the predominantly copper "last mile" (between the exchange and the customer premises). VDSL2 vectoring achieves access speeds of 250 Mbit/s.

The recently approved G.fast (ITU-T Recs G.9700 and G.9701) provides high bit-rate access over copper telephone lines from a fiber-fed network distribution point, over existing telephone wires to the customer premises and inside the customer premises for a total distance up to 400 meters. On short lines, aggregate net data rates up to 1 Gbps are supported. Since G.fast operates over the existing telephone wires inside the home, it can facilitate installation of the service by the customer with no need for a network technician visit inside the home. Even higher performance can be achieved if CAT-5 (or better) wire is used from the Residential Gateway to the point-of-entry to the home (typically installed by a technician). G.fast provides a comprehensive set of spectrum management tools to adjust the range of transmitted frequencies (from 2 MHz up to 106 MHz), RF band notching, and adjustable PSD (power spectral density) limits. The ratio of downstream to upstream data rate can be set by the service provider. Vectoring is included to cancel the far-end crosstalk between G.fast lines. The reverse power feed RPF) option allows the network DPU (distribution point unit) to be powered via DC current fed from the customer's residential gateway over the copper drop wire. G.fast silicon is available now from multiple vendors.

SG15's experience in optimizing the communication capabilities of wired infrastructure makes it the natural home of ITU's work on Smart Grid. SG15 has produced a family of orthogonal frequency-division multiplexing (OFDM)-based narrowband powerline communication (NB-PLC) standards that reuse the electric grid as a telecommunication medium, primarily to monitor, analyze, and control power supply/usage. This work builds on G.hn (ITU-T G.996x-series) which provides broadband home-networking over telephone wiring, coaxial cable and power-line wiring. The recently approved ITU-T Rec. G.9979 employs codepoints assigned by the IEEE to add G.hn (G.9960, G.9961) and HPNA (G.9954) technology to the IEEE Std 1905.1a (IEEE Standard for a Convergent Digital Home Network for Heterogeneous Technologies).

The Optical Transport Network (OTN), which provides a terabit-capable framework equipped to carry ever-rising volumes of data and video traffic, is rapidly supplanting its predecessor, Synchronous Digital Hierarchy (SDH), which has been the dominant transport protocol for the previous 20 years. OTN's support for both optical (wavelength division or WDM) and digital multiplexing techniques improves network efficiency. The WDM aspects of OTN increase the traffic-carrying capacity of optical fibres by allowing simultaneous operation over multiple wavelengths. The digital hierarchy and mappings provide transport of new packet, data-center and video protocols (for example, IP/MPLS, Ethernet, Fibre Channel, SDI, DVB_ASI), in addition to legacy protocols (such as SDH) which allows for the seamless convergence of operators' networks. OTN also offers the flexibility required to support future protocols as they emerge.

In its last plenary meeting in December 2014, SG15 made considerable progress on a number of transport network technology fronts, including evolution of OTN beyond 100G, SDN in transport networks, OTN protection, Ethernet OAM, and equipment specifications for Packet based Transport Networks, transport network architecture, timing and frequency transport, generic and technology-specific transport equipment management requirements, automatic discovery, information models and Transport SDN.

SG15 has agreed to revise the G.8011 series of recommendations (Ethernet service characteristics) and aligned these recommendations with the core Metro Ethernet Forum (MEF) specifications. Additionally, we agreed on a supplement that provides guidance on Ethernet OAM performance monitoring.

In the area of Optical Transport Network (OTN), progress was made on Beyond100G where working assumptions were updated, taking into consideration the work on 400G and 25G in the IEEE802.3.

Considerable progress has been made on recommendations defining transport of frequency, time and phase over packet networks.

In the area of management and control of transport systems and equipment, work is progressing well on the generic and technology-specific transport equipment management requirements and information models, DCN, SDN, and ASON.

Activity was initiated on three new Recommendations addressing the common control aspects of the use of SDN in the transport network, the timing characteristics of primary reference clocks, and the timing characteristics of slave reference clocks.

Standards Community Preparing for 5G

Karri Ranta-aho and Athul Prasad, Nokia, Finland

During the course of 2013 and 2014, regional research initiatives were being set up to kick-start the fifth generation (5G) race, and to provide platforms for academia and the industry to join forces and exchange ideas about the potential components in 5G networks. At the same time many organizations such as the European 5G Infrastructure Public Private Partnership (5G-PPP), Japanese 5G Promotion Forum, Chinese IMT-2020 Promotion Association and Future Forum, Korean 5G Forum, the next generation mobile networks (NGMN) alliance, International Telecommunication Union-Radio Communication Sector (ITU-R) and 4G Americas, as well as telecom companies, were working individually on the future visions of wireless communication for the next decade. Further, at the Mobile World Congress of 2015 (to be held in the first week of March 2015) many 5G related presentations and technology and use case demonstrations will be seen. All this is a pretext for global standardization of 5G. ITU-R working party 5D (WP5D), the group nominally holding the keys on what qualifies as a 3G, 4G and now 5G (or more specifically IMT-2000, IMT-Advanced, and the yet-to-be named IMT, respectively), agreed on the 5G definition timeline in October 2014, and is expected to finalize its 5G vision, including the key performance indicators needed for a technology to qualify as 5G, in June 2015, and continue with detailed 5G requirement and evaluation scenario definitions in 2016. It is evident that in order to meet the capacity and performance needs of the next decade, new bands for 5G need to be considered. Especially the bands 6-100 GHz are expected to be addressed in WRC-19 (World Radio-communication Conference).

With all the visions and research on 5G, some patterns are emerging on what 5G might look like. This is the triggering point for the standards organizations to start planning how the faintly emerging consensus on 5G can be turned into standard specifications. The 3rd generation partnership project (3GPP) system architecture working group-1 (SA1) initiated a study on New Services and Markets Technology Enablers, carefully avoiding the term 5G in its name, but not in its justification text. 3GPP radio access network (RAN) and SA plenaries in March 2015 are expected to discuss when and how the technical standardization work in 3GPP should be conducted. The 5G standardization in 3GPP will need to happen in multiple phases. Initially, the work is required to address detailed 5G requirements and develop the radio propagation channel models for the bands above 6 GHz. At the same time the IEEE Computer Society is looking at new developments such as. 802.11ax and the NG60 Study Group that could qualify as 5G, and the IEEE Communications Society is planning on a session to discuss standardization opportunities in 5G in April 2015. The standard definitions fully meeting the ITU-R 5G requirements are expected around 2019, with the first phase commercial 5G network deployments in the 2020 timeframe.

The research organizations are only starting to produce results and the process will be ongoing for several years. According to the ITU-R agreed timeline, the requirements that the new 5G system must fulfill will only be detailed during the 2016-2017 timeframe. As such, there cannot yet be a clear consensus of what 5G is or when, how or where it is supposed to be standardized, even if the 5G vision of vendors, researchers, and operators may not be very different. Nevertheless, the time for the standards organization to actively start working on 5G is near, and when the potential of the technology options is better understood, the standards defining organizations will be ready to evaluate them and work toward detailed specifications, finally finally giving the world the 5G system.

Acknowledgment

The authors would like to thank their colleagues from Nokia for contributing to this article.

3GPP System Architecture News

Andreas Kunz, NEC Europe Ltd.

The 3GPP services and requirements working group SA1 agreed to study items on the next generation of the 3GPP system as well as different enhancements to the current Evolved Packet System (EPS). The new study item on new Services and Markets Technology Enablers (SMARTER) is already pointing toward "5G" and aims to develop high-level use cases and to identify the related high-level potential requirements to enable 3GPP network operators to support the needs of new services and markets. Public safety communications is one ongoing hot topic, now extending the current Release 13 with use cases for mission critical group based video communications. This new study targets the definition of real-time group-based video services, reusing developments already done in the Mission Critical Push To Talk (MCPTT) work in Release 13.

Another new agreed study is exploring the opportunities of the deployed LTE based infrastructure for the vehicle industry in order to realize the concept of a "connected car". The study proposes to reuse the Release 13 Proximity Services (ProSe) feature for device to device (D2D) communication to provide functionalities to implement vehicle to vehicle (V2V) services. Other use cases and requirements covering vehicle to passenger (V2P) and vehicle to infrastructure (V2I), a roadside unit, are studied also considering work from other SDOs (e.g. GSMA Connected Car, ETSI ITS (Intelligent Transportation System), US SAE) or related governmental agency (e.g. C-ITS project in Korean Ministry of Land, Transport and Maritime Affairs).

The 3GPP System Architecture working group (SA2) finished study items during the last meeting. The current MCPTT technical report (TR) from SA2 was transferred to SA6, and SA2 may decide to create a new TR for their further work or may directly respond to questions and requests from SA6. SA2 discussed for the first time a study targeting Release 13 on Isolated E-UTRAN Operations for Public Safety (IOPS), where two scenarios on no backhaul to the Evolved Packet Core (EPC) and limited backhaul to the EPC are addressed, including the formation and break-up of an Isolated E-UTRAN by multiple eNBs.

The newly established SA6 working group for mission-critical applications met for the first meeting and attracted many delegates from various standardization bodies, e.g. ETSI TETRA and Critical Communications Evolution (TCCE), Open Mobile Alliance (OMA), TETRA and Critical Communications Association (TCCA), 3GPP and governmental organizations, resulting in a good representation from all the stakeholders interested in critical communications for public safety. One major topic was the discussion of the MCPTT work plan, especially the coordination with SA2 for the individual steps. Key issues on charging, codec selection, service continuity as well as the architecture based on the 3GPP IP Multimedia Subsystem (IMS) or a non-IMS approach will be further discussed in the following meetings.

The Telecom Management group SA5 is currently studying the use cases and concepts for the network management of virtualized networks, taking into account the relevant use cases from ETSI Network Functions Virtualization Industry Specification Group (NFV ISG). The main objective is to analyze and classify the scenarios when all or some instances of 3GPP-defined network elements in a subsystem/domain are virtualized and to create the requirements for potential solutions to the scenarios. A joint workshop was held in February in Prague with all involved organizations ETSI NFV ISG, Open Platform for NFV (OPNFV), 3GPP SA5 and the Telecom Management (TM) Forum to coordinate the work.

What is New at the MEF?

Scott Mansfield, Ericsson, Member of MEF Board

The MEF has a long history of providing the industry with new ways of looking at telecommunication services. The MEF's original scope was to provide service definitions to help service provides and network operators offer standardized access to metro Ethernet networks. By 2012, the success of the MEF service definitions led to the creation of a new market segment called Carrier Ethernet 2.0 (CE 2.0). In CE 2.0, the MEF continued enhancing the Ethernet services to support multiple classes of service, to be more interconnected, and to support fault management. Starting in 2014 and looking forward to 2015, the MEF is embarking on another redefinition of the industry landscape by launching the "MEF's Third Network Vision".

The Third Network Vision continues the MEF's strategy of providing service definitions to aid the creation of new market segments. The vision points to the importance of software-definition and virtualization as enablers of new services. The MEF is concentrating on the Service Orchestration aspect within networks and between networks. Service Orchestration consists of a number of capabilities including: order fulfillment, agile and elastic control of connection characteristics, performance monitoring to ensure the service meets the service level agreement, assurance which ensures fault identification and recovery, usage monitoring to enable billing for services and analytics to leverage trends and support network load predictions and planning. The MEF is creating a framework that will promote the definition of service specifications that will support the orchestration of end-to-end connectivity-as-a-service products.

The Third Network Vision is all about building agility, assurance and orchestration into the network. A follow-up paper is being written to describe the lifecycle service orchestration framework that describes how management models and application program interfaces (APIs) are leveraged to enable the automation necessary to allow the network the agility needed to provide the end-users with an end-to-end secure service experience.

The MEF has also formalized a cross-committee group to provide coordination and collaboration within the MEF and between the MEF and other players in the evolving ecosystem needed for connectivity as a service deployments. The program, called UNITE, is an extension of the MEF's liaison program and allows the MEF domain experts to concentrate on the technology while the MEF staff worries about the external cooperation agreements and tooling to support collaboration.

The Technical Committee (TC) has launched a new ad hoc group to consider a project to create a Generalized Service Constructs document. The work will start from the existing MEF 29 Ethernet Services Constructs. The TC has started a new project to create YANG data models for Ethernet Services. The YANG work is being coordinated with the existing Information Modeling effort MEF 7.3.

The Service Operations Committee (SoC) continues to be very busy. The SoC completed its first MEF Guidelines document (MEF 50) called "Carrier Ethernet Service Lifecycle Process Model". MEF 50 provides a process model for the generic Carrier Ethernet service lifecycle, including Service Operations Lifecycle management and Product Lifecycle management. The Lifecycle Service Orchestration (LSO) Reference Architecture (RA) work continues. The LSO RA provides a layered architecture that characterizes the management and control domains, the artefacts, the use cases, and the engineering methodology for end-to-end connectivity services. Two of the most active areas in the SoC are related to Carrier Ethernet Billing and the Ethernet Product catalog.

With its focus on Lifecycle and Service Operations, The MEF is attracting new members from network systems integrator and software segments of the industry. This is providing the MEF with the opportunity to reach new audiences and continue to play a major role in the definition of an Orchestrated, Software-Defined and Virtualized network ecosystem.

ETSI Mobile-edge Computing ISG

Konstantinos Samdanis, NEC Europe Ltd.

ETSI recently launched a new Industry Specification Group (ISG) for Mobile-edge Computing (MEC) to create a standardized, open environment that will enable an efficient and seamless integration of applications from vendors, service providers, and third-parties across multi-vendor cloud platforms located at the base stations or close to the Radio Access Network (RAN). MEC will allow mobile operators to open their RAN edge to authorized application developers, content providers and OTT players, enabling flexibility and higher cooperation with mobile operators and vendors, enhancing in this way service innovation. The first meeting of the MEC group took place in December 2014 in Munich, Germany with the group aiming to deliver the first phase of specifications by the end of 2016.

The MEC initiative brings together the telecommunication industry with the Information Technology (IT) world, facilitating cloud-computing capabilities at or above the RAN in a close proximity to mobile subscribers. MEC's objective is to enable software applications and services to exploit real-time radio information, i.e. RAN analytics, and subscriber location offering efficient context-related services. In addition, MEC permits content and services to be stored close to the subscriber, allowing content, services and applications to be accelerated, securing in this way ultra-low latency and high-bandwidth, being able to react rapidly to different radio and network conditions. MEC alleviates potential congestion in the mobile backhaul, aggregation and core networks since selected, popular content can efficiently serve local goals, while reducing significantly energy consumption related with retreating it from the core network or the Internet.

MEC consists of a natural evolution of RANs, enhancing the performance of current applications and empowering new types of services for consumers and enterprises. In particular, MEC is considering the following uses cases including radio-aware application optimization, RAN and video analytics, big data processing, Machine Type Communications (MTC) and Internet of Things (IoT), location services, augmented reality, optimized local content distribution and data caching.

The MEC industry standards and the adoption of the MEC cloud platform will act as a main driver for new revenue streams among operators, vendors and third-parties. The ETSI MEC ISG will concentrate on the framework and architecture specifying business and service requirements that facilitate the development of edge applications and specify a MEC application-platform Application Programming Interfaces (APIs), which will give the players in the value chain the opportunity to revolutionize, differentiate and create value in a multi-vendor mobile environment.

IEEE ComSoc Standardization Research Groups on IOT and SDN/NV

Periklis Chatzimisios, Alexander TEI of Thessaloniki, Greece

The IEEE Communications Society (ComSoc) has strong competencies in current trends in communication-related areas and is looking for opportunities to contribute to standardization on hot and emerging topics. Recently, the IEEE ComSoc Standards Activities Council has organized Rapid Standardization Activity Working Meetings (at IEEE SA Headquarters in Piscataway, NJ USA) with the purpose to identify potential standardization opportunities in the framework of Internet of Things (IoT), Software Defined Networking (SDN)/Network Virtualization and related areas.

Among others, two very promising research groups (RGs) were formed: Internet of Things Communications & Networking Infrastructure; and Software Defined and Virtualized Wireless Access, chaired by Prof. Stefano Giordano (University of Pisa) and Prof. Fabrizio Granelli (University of Trento), respectively.

The purpose of the Standardization RG on "Internet of Things Communications and Networking Infrastructure" is to discuss specific ideas for standardization by IEEE in related IoT areas and to collaborate with other associated research and study groups. The very first action of the RG is to organize a special session on "Standardization Opportunities for Software Defined Sensors and Actuators Networks" at the IEEE Conference on Standards for Communications & Networking (CSCN), which will take place from 28–30 October 2015 in Tokyo, Japan.

Future activities include the following:

- Generate a White Paper that will investigate new solutions (suitable for standardization) for the most effective integration of wireless sensor and actuator networks, carrier networks and data center networks that together will compose the information infrastructure to enable new services in this field.
- Organize related events (conferences/workshops/panels/training) where researchers from industry (hardware manufacturers, operators, service and cloud providers) and academia present their original visions on this sector, stimulating new standardization activities.
- Organize special issues in IEEE ComSoc magazines and journals on the topics of the RG.

The Standardization RG on "Software Defined and Virtualized Wireless Access" works on identifying and addressing the research issues that need to be solved and assess the feasibility of launching an IEEE standardization effort on software defined and virtualized wireless access, with specific focus on different future scenarios. The purpose of this RG is to identify possible standardization opportunities in the framework of its scope. Currently the members are involved in the preparation of a White Paper entitled "Future Directions and Standardization Opportunities" that will analyze the research and existing standards in the following topics:

- Enabling SDN over wireless (e.g. interoperability among layers 1, 2, 3; definition and extraction of information and primitives to implement SDN over wireless).
- SDN and virtualization at the wireless access (e.g. from SON to SDN in wireless, heterogeneous multi-owner RAN management).
- End-to-end SDN (e.g. extending SDN to mobile devices, joint optimization of e2e transport and RAN).

For more information or to contribute to the activities/efforts of the two research groups, please visit [1] and [2].

Acknowledgment

The author would like to thank Prof. Giordano and Prof. Granelli for providing information and details about the two recently formed research groups.

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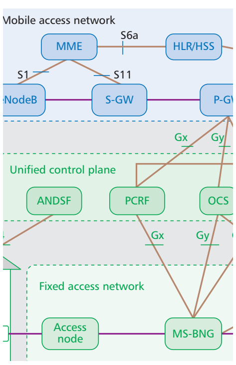
[1] http://community.comsoc.org/ groups/research-group-iot-communications-and-networking-infrustructure
[2] http://community.comsoc.org/groups/rg-software-defined-and-virtualized-wireless-access

Unified Control Plane: Converged Policy and Charging Control

The authors describe the existing framework for converged PCC. They highlight how converged policy control architectures allow the carriers to deliver differentiated offerings, while dynamically controlling network resources and allowing the different operational domains to charge accordingly.

Filipe Leitão, Roberto David Carnero Ros and Susana Fernandez, Jaume Rius I Riu, and Stefan Rommer



Filipe Leitão, Roberto David Carnero Ros and Susana Fernandez are with Ericsson, Spain.

Jaume Rius I Riu and Stefan Rommer are with Ericsson, Sweden.

Abstract

This article describes a standardized architectural framework that enables network resources and policy control, and charging enforcement for both mobile and fixed networks. These capabilities together facilitate the realization of a unified control plane. The most relevant use cases supported by such a unified control plane are presented and described.

Introduction

Subscribers are getting used to accessing all their services on an ever increasing variety of devices and in all types of access networks (including cellular and Wi-Fi). Furthermore, seamless quality of experience is now expected by the subscribers, for all the terminals being used, especially when the users are paying for premium *All-in-One* bundled plans. Facing this situation, carriers should be capable of delivering the same services, agnostic to the type of access network, to any type of devices, with the required quality of service. Most importantly, carriers need to be able to charge for the service delivery and the network usage in all these situations and in appropriate ways.

The Converged Policy and Charging Control (PCC) architecture provides a framework for carriers to apply common policies to subscribers' traffic and to control the subscriber account when the subscriber accesses services from both fixed and mobile networks. Further benefits of a converged policy control approach include the enablement of a unified control plane, which might serve the carrier not only to differentiate subscriber services agnostic to the access type used, but also to plan service delivery, use the network resources more efficiently, and charge for all of these events appropriately.

This article describes the existing framework for converged PCC. It highlights how converged policy control architectures allow the carriers to deliver differentiated offerings, while dynamically controlling network resources and allowing the different operational domains to charge accordingly. Finally, it describes how this framework facilitates delivering the expected Quality of Experience over both fixed and mobile access networks. The article provides an overview of the standardization activities for fixed mobile policy convergence that are ongoing in the 3rd Generation Partnership Project (3GPP) and the Broadband Forum (BBF).

The article is organized as follows. First the main use cases are presented as motivation for the work, followed by the description of the key standardization activities ongoing in 3GPP and BBF. Then the overall solution is presented, including a detailed description of the architecture and an example where the solution is applied to one of the use case's workflows. Finally, thoughts about the benefits provided by the unified control plane, as described in this article, are discussed in the conclusion.

Use Cases

For a carrier network offering services on both fixed broadband access and mobile access, having the same service deployed in both accesses usually involves highly complex systems to manage user resources differently in each access. In order to reduce the service's time-to-market, reduce Capital/Operational expenditure (CAPEX/OPEX), and enhance the user experience, an operator may follow a convergent policy and charging control paradigm, as presented in Fig. 1. The fixed/mobile operator would be able to extend the offer of some of the most popular services in the mobile domain (or vice-versa) e.g. shared data plans, parental control, or multimedia delivery, to both access types and reduce deployment costs by having the same policy and charging decision point for both.

Shared Data Plans

Shared data plans are very common in the mobile access context where resources are scarce and operators tend to combine bandwidth/time consumption with usage limits. In convergent operators, this use case implies a step beyond: making use of the same quota regardless of the kind of access (fixed or mobile) the subscriber is making use of separately. This means that, in addition to the subscription models where users share the same data allowance by the 3GPP UE devices, the same subscription may also allow users to include the fixed broadband service allowance as part of the shared plan [1].

Security and Parental Control

With increasing concern about Internet security and web content access restrictions, many families are looking for mobile subscriptions for their children that include easy to manage parental control. However, to be efficient, security aspects and parental control restrictions should be the same for every access available to the user (e.g. LTE, Wi-Fi). With a converged policy control, the carrier with both mobile access and fixed broadband access can provide subscribers a unique management system for both access types.

This approach also works for the cases where the 3GPP and Wi-Fi access are operated by different network providers (e.g. when the subscriber is using his favorite cafeteria Wi-Fi service). In this situation the fixed broadband

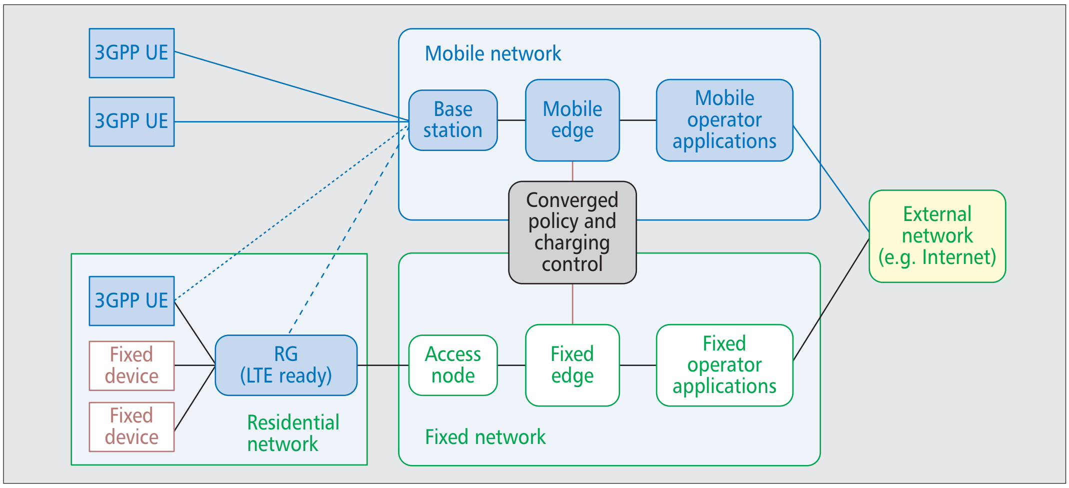


Figure 1. Conceptual architecture for convergent PCC and Residential Gateway with dual connectivity.

> Soon voice over IP and voice over LTE (VoLTE) data plans will be part of the mobile portfolio and carriers are preparing their core network to assure the required QoS and user experience. If the operator is also offering Wi-Fi connectivity, the converged policy and charging control framework allows for combined cellular and Wi-Fi control solutions management.

access that provides the Wi-Fi access will establish a tunnel from the fixed edge toward the mobile edge and route the subscriber traffic to the mobile operator core where service management is being offered [2, 3].

Seamless Multimedia Policy and Charging

Soon voice over IP (VoIP) and voice over LTE (VoLTE) data plans will be part of the mobile portfolio, and carriers are preparing their core network to assure the required QoS and user experience.

If the operator is also offering Wi-Fi connectivity, the converged policy and charging control framework allows for combined cellular and Wi-Fi control solutions management. This way, policy characteristics associated with multimedia applications can be enforced in the fixed access broadband providing the same end user experience to the subscriber. This is also useful for enterprise engagement where different charging schemes may apply to differentiate the use of the Wi-Fi infrastructure when at the office, from the use of cellular when moving outdoors.

Dual Connectivity

It is becoming common for operators to offer LTE in the residential market as a fixed broadband access solution by placing LTE residential gateways in their subscribers' home.

In these deployments operators might take advantage of the dual connectivity provided by the residential gateway (Fig. 1), for instance by enabling some load balancing in the outgoing traffic for particular applications (e.g. based on the congestion situation of the access networks), or by facilitating a fallback plan in malfunction situations of a given access. In both cases, a convergent policy and charging framework could assure the enforcement of the same traffic treatment for both access types seamlessly to both the operator and end user.

Key Standardization Activities Overview

Different aspects of fixed mobile convergence (FMC) have been discussed by standardization organizations in recent years. One example is the common IMS solution for both fixed and mobile access types [4]. The discussion around common policy control for fixed and mobile started in 2008 with the collaboration between 3GPP and the BBF. By that time, the evolved packet core (EPC) defined for the 2G/3G and 4G (LTE) radio accesses had reached maturity in 3GPP. Then began the interest to investigate using 3GPP EPC for a convergent control plane, also supporting fixed accesses. Therefore, initiatives were taken in 2008 to initiate discussions between 3GPP and BBF about this topic. This resulted in collaboration on interworking and convergence topics where policy control aspects turned out to be the main component. This collaboration has successfully resulted in standardized solutions (further detailed below) and guidance for the industry for policy control in a FMC context [1, 5, 6].

In the 3GPP-BBF collaboration, 3GPP has worked on aspects related to the 3GPP domain (e.g. the policy control function) while BBF has worked on the aspects related to the BBF domain (e.g. the fixed access network, including the policy enforcement point). The work has mostly been done as part of the normal standardization work in each group, but several joint workshops have also taken place.

The work was from the start divided into two phases. In the first phase, policy interworking between 3GPP and BBF domains was specified. The interworking architecture targets scenarios where a mobile operator collaborates with a fixed operator. A new policy peering interface between the two operator domains was defined for this purpose (the S9a reference point). The main part of the interworking solution has been specified in 3GPP TS 23.139 [7] starting from 3GPP release 11 and in the BBF TR-203 [2] and TR-291 [8]. This interworking solution is not described further in this article but has been described in [3, 7, 8].

In the second phase, 3GPP and BBF have been working on a policy convergence solution (which is the topic of this article). This solution targets converged operators that own both fixed and mobile access networks. The work is mainly captured in 3GPP TS 23.203 [9] starting from release 12 and BBF TR-300 [1].

In the 3GPP-BBF collaboration, 3GPP has worked on aspects related to the 3GPP domain while BBF has worked on the aspects related to the BBF domain. The work has mostly been done as part of the normal standardization work in each group, but several joint workshops have also taken place.

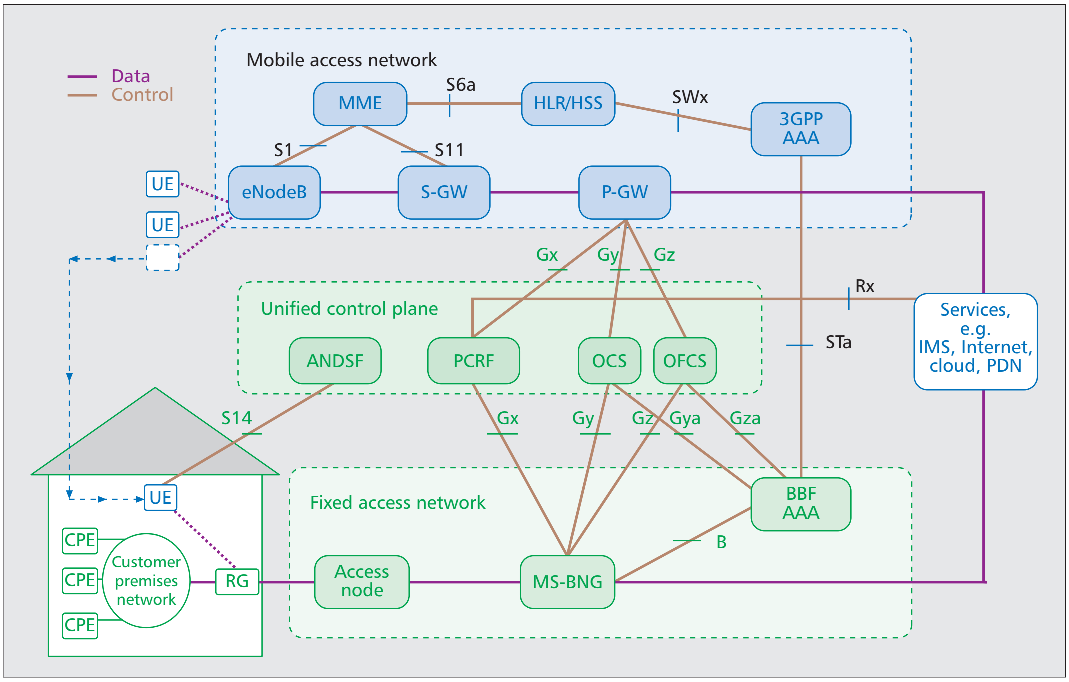


Figure 2. Unified control plane for the fixed and mobile architectures.

Overall Solution Description

This article has previously presented a set of use cases that may imply the introduction of a unified control plane consisting of a convergent policy and charging control architecture. Two different network domains are affected by this paradigm: the mobile access domain components, covered by 3GPP standardization, and the fixed access domain components, covered by the BBF standardization.

In this section the overall architecture is described with special focus on the different roles performed by each domain component in order to achieve the fixed-mobile convergent vision.

Architecture

Figure 2 presents a diagram detailing the two different data path alternatives for a 3GPP UE: using the 3GPP mobile access and having the data traffic handled by the evolved packet core; or using a fixed access provided by the residential gateway access (e.g. Wi-Fi) and having the data handled by the carrier fixed network. This diagram depicts the network entities involved in the Policy and Charging Control (PCC) architecture for both the 3GPP mobile network and the BBF fixed network in order to provide a seamless experience for the end user. The diagram also includes the access network discovery and selection function (ANDSF) as part of the Unified Control Plane.

Functions

Policy: In the 3GPP's PCC architecture, the policy and charging rules function (PCRF) is the central node responsible for the policy control of every procedure related to the packet data network (PDN) session: initiation, modification, and termination [9]. In the Broadband Forum (BBF), the logical entity that performs policy control is the policy decision point (PDP). The PDP is responsible for sending policy and charging information through the R reference point to the policy enforcement point (PEP). This is part of the broadband policy control framework (BPCF) defined by [10]. In the architecture presented by Fig. 2, the PDP's role is played by the PCRF [1]. This means that BBF reference point R between the PDP and the PEP, which will be the multi service broadband network gateway (MS-BNG) in this architecture, is instantiated by the 3GPP diameter Gx interface [1, 9]. The diameter Gx reference point makes this possible by covering the requirements for the policy information model as defined by the BPCF [1, 2]. To support this functionality, the MS-BNG has to implement the policy and charging enforcement function (PCEF) logical entity as defined by 3GPP [1, 9].

From a technological point of view this approach has clear advantages: only one PDP for both domains is deployed; and subscriber information models already existing for policy control can be re-used with the addition of fixed access subscription information (e.g. fixed access identifiers).

AAA: In a BBF-3GPP convergent architecture, as the one presented in Fig. 2, authentication authorization and accounting (AAA) for the fixed network domain is still being done by the BBF AAA server aided by the B reference point. In order to support the authentication of 3GPP UE devices for non-seamless WLAN offload (NSWO), the BBF AAA server implements the diameter STa reference point toward the AAA server that serves the 3GPP domain. With this interface the BBF AAA server is able to request the 3GPP AAA proxy/server for authentication, authorization, subscription parameters, and accounting related information in a secure manner [1].

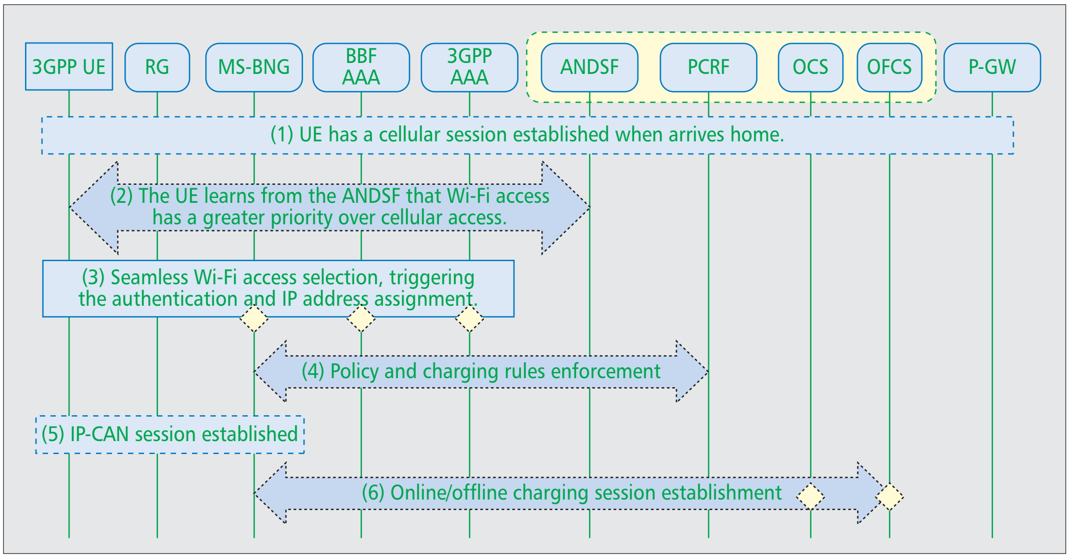


Figure 3. Wi-Fi offload procedure handled by the unified control plane.

By deploying this network element the carrier operator could setup complex use cases by providing not only seamless policy and charging mechanisms to the 3GPP UE agnostic to the access network, but also by having control of the network selection decision.

Charging: The convergent architecture presented in Fig. 2 also provides mechanisms to perform charging of fixed and 3GPP UE devices performing NSWO. In fact, this framework provides two different alternatives to be deployed depending on the particular scenario: diameter Gy/Gz based charging, and AAA based charging.

The first charging alternative, i.e. relying on the diameter Gy/Gz online/offline charging procedures, is suitable in scenarios where the MS-BNG is supporting the PCEF capabilities required to setup a diameter Gy interface toward the 3GPP online charging system (OCS). This interface allows online credit control for service data flow based charging for both fixed and 3GPP UE devices [1]. This technical framework supports, for example, the implementation of pre-paid use cases in a seamless and access type agnostic way.

The same happens to offline charging scenarios where the support of diameter Gz interface toward the 3GPP offline charging system (OCFS) is required for specific postpaid use cases. In that case the MS-BNG should support PCEF capabilities that allow the setup of the diameter Gz interface, enabling transport of service data flow based offline charging information [1, 11, 12].

The second charging alternative, AAA based charging, depends on the BBF AAA support for the accounting interworking function (AIF). In this alternative, charging control relies on the accounting achieved by the BBF AAA server and the MS-BNG via the B reference point. For real time charging use cases, the AIF provides the BBF AAA server the ability of establishing a diameter Gya interface toward the OCS. For offline charging purposes, the AIF provides the BBF AAA server the ability to establish a diameter Gza interface toward the OFCS [1, 9]. At the time of this writing, this solution is still under development."

Figure 2 shows how this functionality could be represented in a convergent scenario. This charging alternative is oriented to legacy deployments with BNGs that do not support the diameter Gy/Gz setup. However, in order to implement an AIF, the BBF AAA needs to support RADIUS–Diameter protocol translation capabilities.

Network Selection: In a convergent scenario as the one represented by the diagram in Fig. 2, policy and charging use cases can be enhanced with an access network discovery and selection function (ANDSF) [13]. The purpose of this network element is to assist the 3GPP UE on the network selection when this device is placed somewhere covered by the 3GPP access network (e.g. LTE) and by a non-3GPP access network (e.g. Wi-Fi). In this situation, the ANDSF would be providing network selection policies to the 3GPP UE by the 3GPP S14 reference point [14].

By deploying this network element, the carrier operator could set up complex use cases by providing not only seamless policy and charging mechanisms to the 3GPP UE agnostic to the access network, but also by having control of the network selection decision.

Procedures for Seamless Multimedia Policy and Charging

In the Use Cases section, seamless multimedia policy and charging is described as one of the drivers for a unified control plane. An example of such a use case can be the situation where a user arrives home while talking through a Voice application supported by the cellular data network and switches to the Wi-Fi network at home. Figure 3 provides an example of a Wi-Fi offload procedure handled by that framework that serves both the mobile and fixed access. To simplify the explanation, some minor steps will be excused.

In the process it is assumed that the user has a 3GPP session ongoing, e.g. LTE, when they arrive home (Step 1). The terminal learned from the ANDSF server that the Wi-Fi network served by his RG and identified by his operator's community Wi-Fi SSID has priority over the cellular access (Step 2). By detecting that the known SSID is being broadcasted by the RG, the UE will start the establishment of the Wi-Fi session.

This unified control plane provides a foundation that would facilitate, for instance, the introduction of automated and programmable control of the carrier core network that provides both fixed and mobile access.

The authentication procedure is triggered by the UE toward the BBF AAA that proxies the authentication request to the 3GPP AAA (Step 3).

Having an IP address assigned will trigger the MS-BNG to send a request for the policy and charging control rules associated with the user subscription. In the policy decision, the PCRF can take into account the type of access, for example to install a PCC rule that relaxes the consumption by skipping charging monitoring for certain applications over Wi-Fi (Step 4).

With the installation of the corresponding PCC rules the IP-CAN session is established, or updated in case of session continuity, and the UE is ready to use the Wi-Fi/fixed access under the control of the unified control plane (Step 5). By the moment the user starts to send data packets to the network, the charging session will be established accordingly to the previous installed PCC rules in Step 4 (Step 6).

Conclusion

This article provides a description of how converged policy and charging control architectures provide a framework for carriers to:

- Apply common policies to subscribers' traffic, independently of the access network the traffic is generated in.
- Deliver differentiated offerings, while dynamically controlling network resources and at the expected quality by the end user over fixed and mobile access networks.
- Charge for service delivery and network usage accordingly, depending on the combination of network resources and services consumed.

These three features facilitated by the converged PCC framework enable a unified control plane, which enables the carrier not only to differentiate subscriber service delivery but to optimally plan and use their network resources. This unified control plane provides a foundation that would facilitate, for instance, the introduction of automated and programmable control of the carrier core network that provides both fixed and mobile access.

The article also describes the standardization activities that are the foundation of the converged policy and charging control architectures. These activities illustrate the consensus and acceptance in the industry of the converged policy and charging control architectures. These standardization activities describe a flexible and reliable framework that can be successfully deployed in a number of largely different scenarios.

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Biographies

Filipe Leitão (filipe.leitao@ericsson.com) received his MSc. in informatics engineering specialized in computer networks and services engineering from the University of Minho in Braga, Portugal. During this period he was co-author of several papers. He has been working in the telecommunications industry since 2008 and joined Ericsson in 2012. He is a system manager for policy control products and is involved in fixed mobile convergence standardization activities, as a delegate to the Broadband Forum and the Wireless Broadband Alliance.

Jaume Rius i Riu (jaume.rius.i.riu@ericsson.com) received an MSc. in physics from Autonomic University of Barcelona in 1996, a master degree in teaching and pedagogy from the University of Lleida in 1997, researcher qualifying degrees from the University of Barcelona in 1998 and the Royal Institute of Technology in 2001, and a Ph.D. in experimental physics from the Royal Institute of Technology in 2002. Since 2004 he has been with Ericsson as a senior specialist in fixed broadband networks and convergence working on mobile transport, IPv6 migration, and Wi-Fi integration. He is also with the Department of Electrical and Information Technology at Lund University .

Stefan Rommer (stefan.rommer@ericsson.com) has an MSc. in engineering physics and a Ph.D. in theoretical physics, both from Chalmers University of Technology, Gothenburg, Sweden. After joining Ericsson in 2001 he has worked in various projects related to mobile packet core networks, including product development, system integration, and standardization. He is a senior specialist in the area of IP mobile networks. Since 2006 Stefan has participated in packet core standardization in 3GPP SA WG 2, including in the fixed mobile convergence area.

Roberto David Carnero Ros (roberto.david.carnero@ericsson.com) received a MSc. in telecommunications engineering from Polytechnic University of Madrid in 2000, and a master in pedagogy from Complutense University of Madrid in 2003. He joined Ericsson in 1998. Since 2003 he has been working as a product manager in the digital identity and policy management areas. He has been participating in the End-to-End Architecture group of the Broadband Forum (BBF) since 2011. He is BBF editor of technical reports in the fixed mobile convergence area.

Susana Fernandez (susana.fernandez@ericsson.com) graduated from the Polytechnic University of Madrid, Department of Telecommunication Engineering, in 1995. From 1995 until 2008 she held several positions in Ericsson working in areas such as security, addressing, network interworking, and policy control related to different standards and accesses. In 2008 she joined the Ericsson delegation to 3GPP CT3. In 2009 she became the Ericsson main delegate in the packet core networks area. She is currently Vice-Chairman of the 3GPP CT3 Working Group.

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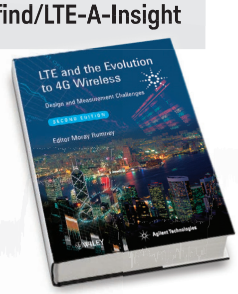

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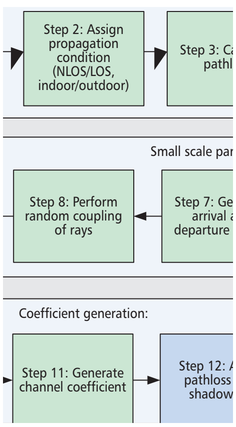
3D Channel Model in 3GPP

Multi-antenna techniques capable of exploiting the elevation dimension are anticipated to be an important air-interface enhancement targeted to handle the expected growth in mobile traffic. In order to enable the development and evaluation of such multi-antenna techniques, 3GPP has recently developed a 3-dimensional (3D) channel model.

Bishwarup Mondal, Timothy A. Thomas, Eugene Visotsky, Frederick W. Vook, Amitava Ghosh, Young-Han Nam, Yang Li, Jianzhong (Charlie) Zhang, Min Zhang, Qinglin Luo, Yuichi Kakishima, and Koshiro Kitao



Abstract

Multi-antenna techniques capable of exploiting the elevation dimension are anticipated to be an important air-interface enhancement targeted to handle the expected growth in mobile traffic. In order to enable the development and evaluation of such multi-antenna techniques, the 3rd Generation Partnership Project (3GPP) has recently developed a three-dimensional (3D) channel model. The existing two-dimensional (2D) channel models do not capture the elevation channel characteristics, making them insufficient for such studies. This article describes the main components of the newly developed 3D channel model and the motivations behind introducing them. One key factor is the ability to model channels for users located on different floors of a building (at different heights). This is achieved by capturing a user height dependency in modelling some channel characteristics including pathloss, line-of-sight (LOS) probability, etc. In general, this 3D channel model follows the framework of WINNERII/WINNER+ while also extending the applicability and the accuracy of the model by introducing some height dependent and distance dependent elevation related parameters.

Introduction

It is predicted that the traffic carried by wireless and mobile communication systems will increase approximately 1000-fold between 2010 and 2020. In addition, there will be a proliferation of the number of connected devices, and the communication systems will experience a coexistence of a diverse range of communication links ranging from very high data-rate mobile multimedia services to low data-rate machine-to-machine type communication [1]. In view of this, the International Telecommunications Union (ITU) Working Party 5D is developing a recommendation on the framework and objectives of the future development of International Mobile Telecommunications (IMT) for 2020 and beyond.

Multi-antenna transmission techniques using massive antenna configurations (such as more than eight transmit antenna ports) and exploiting 3D spatial dimensions (azimuth and elevation) of a multiple-input multiple-output (MIMO) channel are critical in improving the spectral efficiency and reliability of a radio-link. As a result of the significant interest shown in 3GPP in supporting advanced MIMO transmission techniques exploiting both azimuth and elevation dimensions of a wireless channel, a study was started in January 2013 targeting the incorporation of a 3D channel model into the 3GPP evaluation methodology [2]. In this article we describe the main elements of the finalized 3D channel model and also provide the motivations for the various decisions made during the channel model development in 3GPP RAN1 which may not be obvious and may not be documented in the official report [3].

So far, the evaluation and standardization of MIMO techniques in 3GPP has been primarily based on 2D channel models from SCM, ITU, and WINNERII [4–6]. These models assume a 2D plane for the location of scatterers, reflectors, transmit/receive antennas, etc. This 2D assumption limits the MIMO transmission techniques (beamforming, precoding, spatial multiplexing, multi-user MIMO (MU-MIMO), etc.) to the azimuth dimension. In order to evaluate techniques such as user equipment (UE, aka mobile) specific elevation beamforming and full-dimension MIMO (FD-MIMO) [8], where the transmission is adapted efficiently in both elevation and azimuth to a particular UE, or vertical sectorization, where a narrow elevation beam is tailored to each vertical sector, a 3D channel model is necessary. The 3D channel model described here is a geometry-based stochastic (GSCM) model (following the cluster-based approach common to COST-259, 273, and 2100, as well as SCM, WINNERII models) and naturally extends the 2D channel models from ITU/WINNERII. It is also inspired by the extension from the 2D channel model to the 3D channel model published as part of WINNERII/WINNER+ [6, 7].

This article is organized as follows. We first describe the target environments and the scope of the 3D channel model. Then we address the antenna model for a 2D polarized antenna array that can be used for simulations using the 3D channel model. The following sections describe two key aspects of the 3D channel model: pathloss and LOS probability, considering UE locations in high floors and modelling of a multipath component in 3D. A summary of 3D channel model features from WINNERII, WINNER+, and 3GPP is provided later.

Applicability of the 3D Channel Model

The first step in 3D channel modelling is to identify application environments. Urban macro (3D-UMa) and urban micro (3D-UMi) with enhanced node Bs (eNBs, aka base stations) located outdoors are considered to be typical usage scenarios for elevation beamforming and FD-MIMO. Using active antenna systems (AAS) in outdoor eNBs can provide better services and a better user experience for indoor as well as outdoor UEs. The use of elevation beamforming and FD-MIMO for indoor eNBs (or distributed antenna systems (DAS)) was not prioritized due to the cost of AAS and the reduced scope of adaptability in the elevation dimension in some

Bishwarup Mondal, Timothy A. Thomas, Eugene Visotsky, Frederick W. Vook, and Amitava Ghosh are with Nokia Networks.

Young-Han Nam, Yang Li, and Jianzhong (Charlie) Zhang are with Samsung Research America.

Min Zhang and Qinglin Luo are with Alcatel-Lucent, Alcatel-Lucent Shanghai Bell.

Yuichi Kakishima and Koshiro Kitao are with NTT DOCOMO, INC.

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indoor environments. The 3D-UMa and the 3D-UMi scenarios follow the conventional 2D-UMa and 2D-UMi scenarios as determined in ITU-R [5]. Both scenarios are considered to be densely populated by buildings and homogeneous in nature, in terms of building height and building density. It may be assumed that the building blocks (in the 3D-UMa and 3D-UMi scenarios) form a regular Manhattan type grid or can have a more irregular distribution while the building heights are typically distributed between four and eight floors. In the case of 3D-UMa, it is assumed that the eNB height (25m) is well above the heights of the surrounding buildings so that over-the-rooftop diffractions form the dominant propagation mechanism for both indoor and outdoor UEs. In the case of 3D-UMi, it is assumed that the eNB height (10m) is well below the heights of the surrounding buildings and therefore the received signal strength at the UE include contributions from both over-the-rooftop and around-building propagation mechanisms. Note that the building density and building heights as well as the street layout described above were considered for both ray-tracing simulations and field measurements in order to derive the 3D channel models presented in this article.

The 3D channel model is applicable to UE heights ranging from 1.5m (street level) to 22.5m. A statistical approach for placing UEs is suggested for 3D-UMa and 3D-UMi that does not require modelling building dimensions explicitly. An indoor UE can be associated with a height given by $h_{UE} = 3(n_{fl} - 1) + 1.5$, where n_{fl} is the floor number uniformly distributed between 1 and N_{fl}. N_{fl} denotes the building height measured in floors and is uniformly distributed between 4 and 8. Outdoor UEs can be assumed to be at 1.5m height. This is a key aspect that extends the existing 3GPP [4] and ITU-R [5] methodologies where a UE is always modelled at the street level.

The 3D channel model is applicable to carrier frequencies between 2 GHz and 6 GHz with up to 100 MHz of bandwidth. Higher carrier frequencies, e.g. up to 300 GHz, are of interest in 5G wireless communications and present a different set of challenges that are outside the scope of this 3D channel model. It also assumes that the size of the antenna array is negligible compared to the correlation distances of the large scale parameters such as shadow fading, delay spread, angle spread, and Rician factor. These assumptions follow the practices of other existing channel models [5–7].

Antenna Modelling

Now that the simulation environments have been described, specific details of the 3D channel can be explained with a logical starting point being the eNB antennas themselves. A conventional deployment of a multi-antenna array at a macro eNB may use one or more cross-polarized antenna panels with +/–45 degree polarizations. Within each antenna panel multiple antenna elements per polarization are arranged in the vertical dimension, e.g. eight elements, to concentrate the transmission within a narrow beamwidth in the vertical dimension (a half-power beamwidth on the order of 10 degrees). All of the antenna elements in the panel are used for transmission and reception but only one (logical) antenna port has been typically modelled along with a 2D channel model for system design and evaluation. As an example, a four-port cross-polarized multiple-antenna array has been modelled as a set of two pairs of antenna ports arranged in the azimuth dimension, each pair comprising of a +45 and a –45 degree polarized co-located antenna port. This arrangement provides an azimuth-only logical representation of a 2D antenna array comprised of 32 antenna elements (four elements per row and eight elements per column). Thus the study of MIMO techniques such as UE-specific beamforming, MU-MIMO, etc. was limited to spatial adaptation in the azimuth dimension only.

The 3D channel model allows us to overcome this limitation and can be used for generating channel responses to each of the 32 antenna elements in the 2D array. Toward this end, a model for an individual antenna element is required for the 3D channel model.

Polarized Antenna Element Modelling

In the interest of simplicity, an antenna element at an eNB is characterized by an idealized parabolic antenna pattern with 65° half-power beamwidth (in both azimuth and elevation) with 8 dBi antenna element gain. Polarization can be simulated by two models: a constant polarization model and a slanted dipole polarization model. A constant polarization model assumes that the polarization power split is independent of UE location (i.e. UE azimuth and elevation angles relative to broadside to the eNB array). A slanted dipole polarization model is based on the idea that a polarization slant can be modelled as a mechanical tilt. Considering a +45/–45 degree cross-polarized transmit antenna pair, the constant polarization model assumption leads to an equal power split in vertical and horizontal directions for all UE locations. The slanted dipole polarization model achieves equal power split in vertical and horizontal directions at the antenna boresight, but the power split ratio depends on the UE location in both azimuth and elevation dimensions. The choice of which polarization model to use in a particular simulation would be based on the expected antenna patterns of the eNBs being simulated. However, it is worth noting that in detailed system simulations little difference is seen in the performance between the two polarization models.

LOS Probability and Pathloss

With the basic antenna modelling done the next consideration before fast fading modelling is LOS/NLOS state and pathloss determination. In the existing 3GPP [4] or ITU methodologies [5] the LOS probability model considers mainly street level UEs and the dependency on UE height is not explicitly considered. The impact of varying UE heights from 1.5m to 22.5m on LOS probability and pathloss modelling was studied for both the 3D-UMa and 3D-UMi scenarios.

It may be recognized that one approach to

The choice of which polarization model to use in a particular simulation would be based on the expected antenna patterns of the eNBs being simulated. However it is worth noting that in detailed system simulations little difference is seen in the performance between the two polarization models.

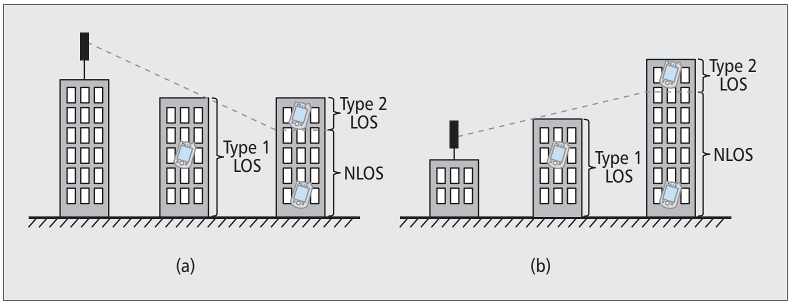

Figure 1. Type 1 and type 2 LOS probabilities: a) 3D-UMa scenario; b) 3D-UMi scenario.

modelling LOS probability, pathloss, and fast-fading channel characteristics is to incorporate explicit building dimensions (a simplified ray-tracing approach) in the model. However, in 3GPP it was decided to retain the fully stochastic approach of modelling as used in SCM [4] and in WINNER II [6] that does not depend on building/street dimensions explicitly. This enables reusing partly the existing modelling parameters from 2D stochastic models and helps incorporating measurement results from multiple sources while also reducing the complexity and processing time for system-level simulations. Also note that for UEs located indoors, the pathloss is simply determined as a sum of an outdoor pathloss component, wall penetration loss, and an indoor pathloss component. Following this methodology, a LOS/NLOS state is associated with an indoor UE, which means that the LOS/NLOS state is actually applicable to the outdoor pathloss component for that UE.

LOS Probability

The LOS probability for the 3D-UMa scenario is modelled as a sum of two probabilities: type-1 and type-2 LOS probabilities [11]. The decomposition of the overall LOS probability into type-1 and type-2 components is for modelling simplification. As shown in Fig. 1a, a UE is considered to be in type-1 LOS state if a UE on the first floor of the same building is also in a LOS state (UEs on the first floor can only be in type-1 LOS state). The type-1 LOS probability depends only on the horizontal distance between the eNB and the UE and follows the formula defined in the ITU model [5]. A LOS UE on a high floor of a building is considered to be in type-2 LOS state if a UE on the first floor of the same building can never achieve a LOS state. Note that the buildings are assumed to be at least four stories high in 3D-UMa, and hence type-2 LOS cases occur only when the UE is located on a floor higher than a four-story building, i.e. 12 m. Beyond 12m the probability to be in type-2 LOS state progressively increases with UE height, and so is the overall LOS probability. The LOS probability model, which is the sum of the type-1 and type-2 LOS probabilities, for 3D-UMa as a function of UE height and distance is illustrated in Fig. 3.

In a 3D-UMi scenario, where the eNB antenna is lower than the surrounding buildings, type 1 and type 2 LOS states can be similarly defined as shown in Fig. 1b. However, it was found by ray-tracing simulations that the UE height is not likely to affect LOS probability in the 3D-UMi scenario since the type 2 LOS condition is limited to situations where the UE height is significantly higher than that of the blocking buildings and hence it can rarely occur (mainly due to the low eNB height). Therefore, the LOS probability model from ITU 2D-UMi [5] is reused for the 3D-UMi scenario.

Pathloss Modelling (LOS)

In [12] it is shown that a height-dependent pathloss for an indoor UE associated with a LOS state can be modelled considering the dimensions of the building and the location of the UE inside the building. In 3GPP it was agreed to model LOS pathloss by using the 3D distance between the eNB and the UE (d_{3D}) along with the coefficients given by the ITU LOS pathloss equations for both 3D-UMa and 3D-UMi. This provides a reasonable approximation to the more accurate model in [12] and can be determined without modelling building dimensions explicitly [13]. The ITU LOS pathloss model assumes a two-ray model resulting in a pathloss equation transitioning from a 22dB/decade slope to a steeper slope at a breakpoint depending on an environmental height. The environmental height represents the height of a dominant reflection from the ground (or a car) that can add constructively or destructively to the direct ray received at a UE located at street level. In the 3D-UMa scenario, it is likely that such a dominant reflection path may come from the street level for indoor UEs associated with a type-1 LOS condition. Therefore, the environmental height is fixed at 1m for a UE associated with a type-1 LOS condition. In the case of a UE associated with a type-2 LOS condition, a dominant reflection can likely bounce of the rooftop of a neighbouring building. Noting that a rooftop is at least 12m in height, in this case the environmental height is randomly determined from a discrete uniform distribution of (12m, 15m,…, (h-1.5)m) where h is the UE height in meters.

Pathloss Modelling (NLOS)

For NLOS pathloss modelling, Fig. 3a shows the primary radio propagation mechanism in a 3D-UMa environment where the dominant propagation paths travel via multiple-diffraction over rooftops followed by diffraction at the edge of a building. The pathloss attenuation increases with the diffraction angle as a UE transitions from a high floor to a low floor. In order to model this phenomenon, a linear height gain term given by $-\alpha(h - 1.5)$ is introduced, where α (in dB/m) is the gain coefficient. A range of values between 0.6 and 1.5 were observed in different results based on field measurements and ray tracing simulations, and eventually a value of 0.6 dB/m was agreed on (see section 7.2.7.1 in [16] for references). Figure 3b shows the principle of radio propagation for a 3D-UMi environment where the dominant propagation paths travel through and around buildings. The UE may also receive small energy from propagation above rooftops. Considering the simplicity of modelling, a linear height gain is also applied to the 3D-UMi NLOS pathloss with a 0.3 dB/m gain coefficient based on the results from multiple sources. In addition, for both the 3D-UMa and 3D-UMi scenarios, the NLOS pathloss is lower-bounded by the corresponding LOS pathloss because the pathloss in a NLOS environment is in principle larger than that in a LOS environment.

Fast Fading Model

Now that antenna modelling, LOS probability, and pathloss have been described, the fast fading model can be readily presented in this section. In the following, a cellular downlink is assumed for describing the fast fading model and hence the departure angles are defined at the eNB side and the arrival angles are defined at the UE side. The fast fading channel coefficients model the time-varying fluctuations of wireless channels that are caused by the combination of multipath and UE movement. The channel coefficients of a link between a transmitter and a receiver are determined by the composite channel impulse responses of the multiple path components (MPCs). Each MPC is characterized by a path delay, a path power, and random phases introduced during the propagation, as well as the incident path angles, i.e. azimuth angles of departure and arrival (AOD and AOA) and zenith angles of departure and arrival (ZOD and ZOA).

Figure 4a shows the steps to generate fast fading channel coefficients in the 3GPP 3D channel model. The high-level procedure shown in Fig. 4a is the same as its precedent 2D channel model in TR 36.814 [9], but a few steps have been revised to take into account the channel characteristics in the elevation domain. As explained in the previous section, a LOS or NLOS state associated with a UE is jointly determined in step 2 by both UE horizontal distance and UE height. The pathloss in step 3 is determined based upon a LOS probability and a height-gain term depending on the UE height and distance from the eNB. In step 4 and step 7 zenith angle spreads at departure and at arrival (ZSD and ZSA) and ZOD and ZOA are generated in addition to the azimuth spread values and angles ASD, ASA, AOD, and AOA. In step 8, for each path, the AOA sub-paths are randomly coupled with the AOD sub-paths, the ZOA sub-paths, and the ZOD sub-paths. Finally, in step 11 the channel generation equation is modified to take into account ZOAs and ZODs. The rest of this section explains in more detail the newly-introduced elevation parameters and procedures in these revised steps in the 3GPP 3D channel models.

Power Angular Spectrum in Zenith (PAS-Z)

Measurement results and ray tracing data have indicated that the marginal distribution of the composite power angular spectrum in zenith (PAS-Z) is Laplacian, and its conditional distribution given a certain link distance and UE height can also be approximated by Laplacian functions [14, 15]. To incorporate these observations, ZOD and ZOA are modeled by inverse Laplacian functions.

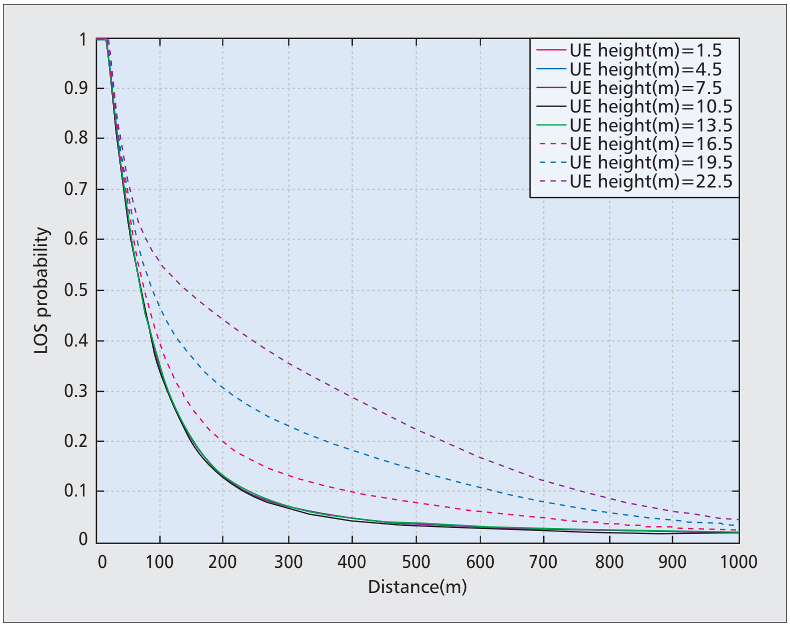

Figure 2. Distance and height dependent LOS probability model for 3D-UMa.

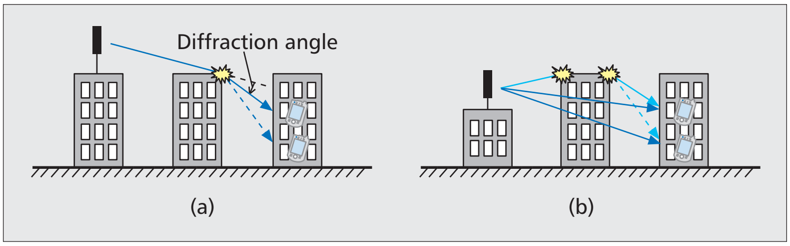

Figure 3. Principle of the radio propagation for NLOS environment: a) 3D-UMa scenario; b) 3D-UMi scenario.

Composite ZSD

It is also observed that the ZSD decreases significantly as a UE moves further away from the eNB [14]. An intuitive explanation is provided by the fact that the angle subtended by a fixed local ring of scatterers at the UE to the eNB decreases as the UE moves away from the eNB. The ZSD is also observed to change to a smaller extent as an indoor UE moves up to higher floors [14, 15]. The ZSD model incorporated in 3GPP is a function of UE height and distance from the eNB as shown in Fig. 5a and Fig. 5b. Applying these ZSD models to 3D-UMa with 500m inter site distance (ISD) and 3D-UMi with 200m ISD described in the second section with UE height distributed between 1.5 and 22.5m, we can observe the overall distribution of ZSD in a wireless network deployment. In Fig. 5e we show the empirical probability density of ZSD for the serving cell links as observed in the 3D-UMa and 3D-UMi scenarios. Note that the ZSD tends to be a little smaller in the case of 3D-UMa reflecting a clear dominance of above-rooftop propagation mechanism.

ZOD Offset

Note that in the azimuth dimension the AODs (corresponding to different MPCs) for a given link are centered at the azimuth LOS angle

Note that in the azimuth dimension the AODs (corresponding to different MPCs) for a given link are centered at the azimuth LOS angle between the UE and the eNB. In the elevation dimension, however, the ZODs are not centered at the LOS zenith angle for NLOS cases.

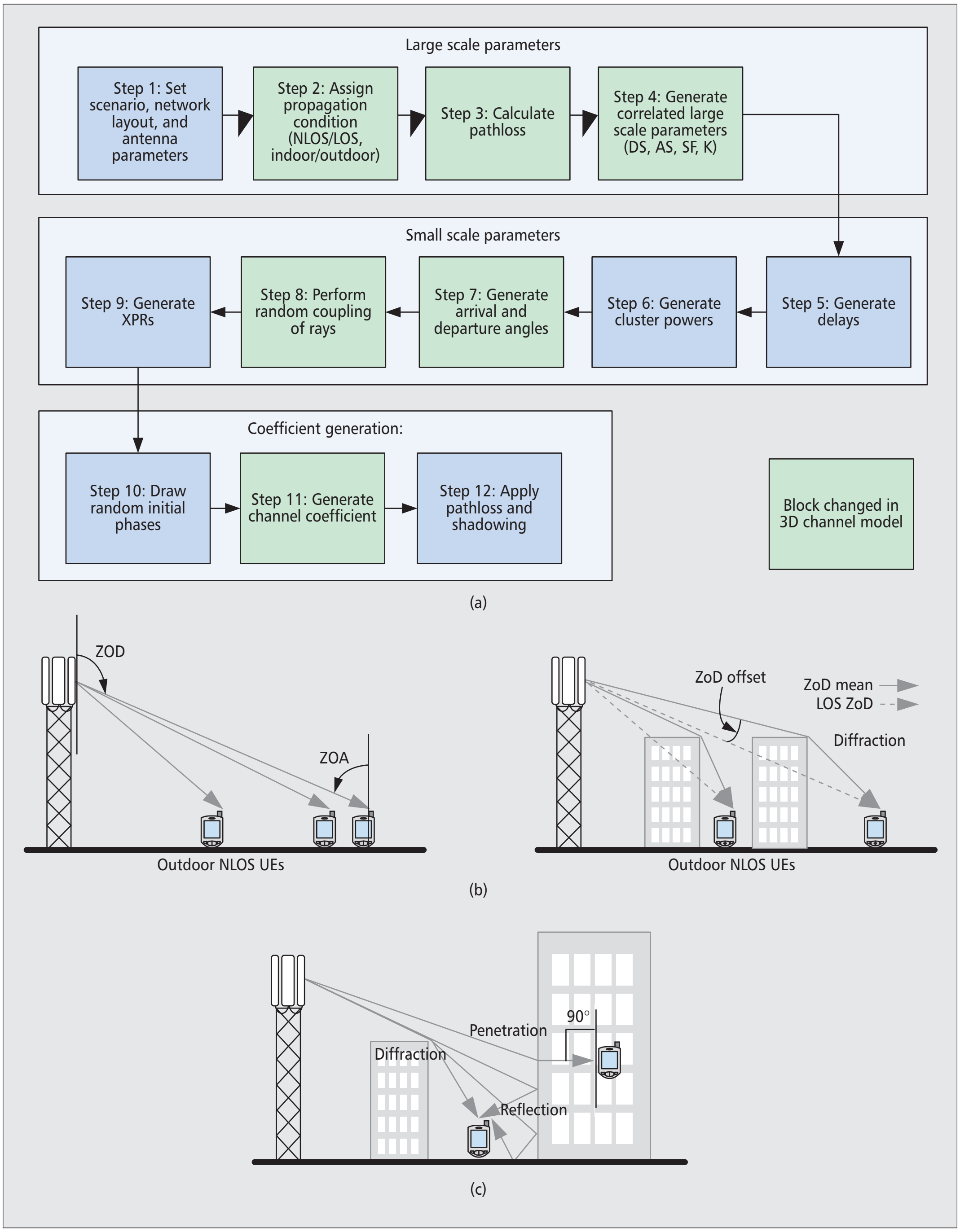

Figure 4. Fast-fading channel generation steps and propagation mechanisms: a) Channel generation steps in the 3GPP 3D channel model [3]; b) ZOD in outdoor LOS and NLOS conditions; and c) ZOA in indoor and outdoor NLOS conditions.

between the UE and the eNB. However, in the elevation dimension the ZODs are not centered at the LOS zenith angle for NLOS cases. This can be observed from the Fig. 4b NLOS case where it is shown that the mean ZOD is shifted from the LOS ZOD because the bulk of the energy is received via diffraction from the building edges (rooftops) blocking the direct path between the UE and the eNB. The magnitude of the ZOD offset becomes smaller as the distance of the link increases, partly because the angle subtended by a given building to an eNB decreases with distance. It is also clear from the Fig. 4b NLOS case that the ZOD offset decreases with an increase in UE height. These observations were captured in the ZOD offset model shown in Fig. 5c and Fig. 5d. In the case of 3D-UMi the dependence of ZOD offset on UE height was not modeled because the results showed some variation across different sources (Table 7.3-8 in [3] for references). On the other hand, it may be noted that when an indoor UE is associated with a LOS state (denoted as LOS outdoor to indoor, or LOS O-to-I), the effect of diffraction paths does not dominate and hence the ZOD offset is modeled to be zero. The application of the ZOD

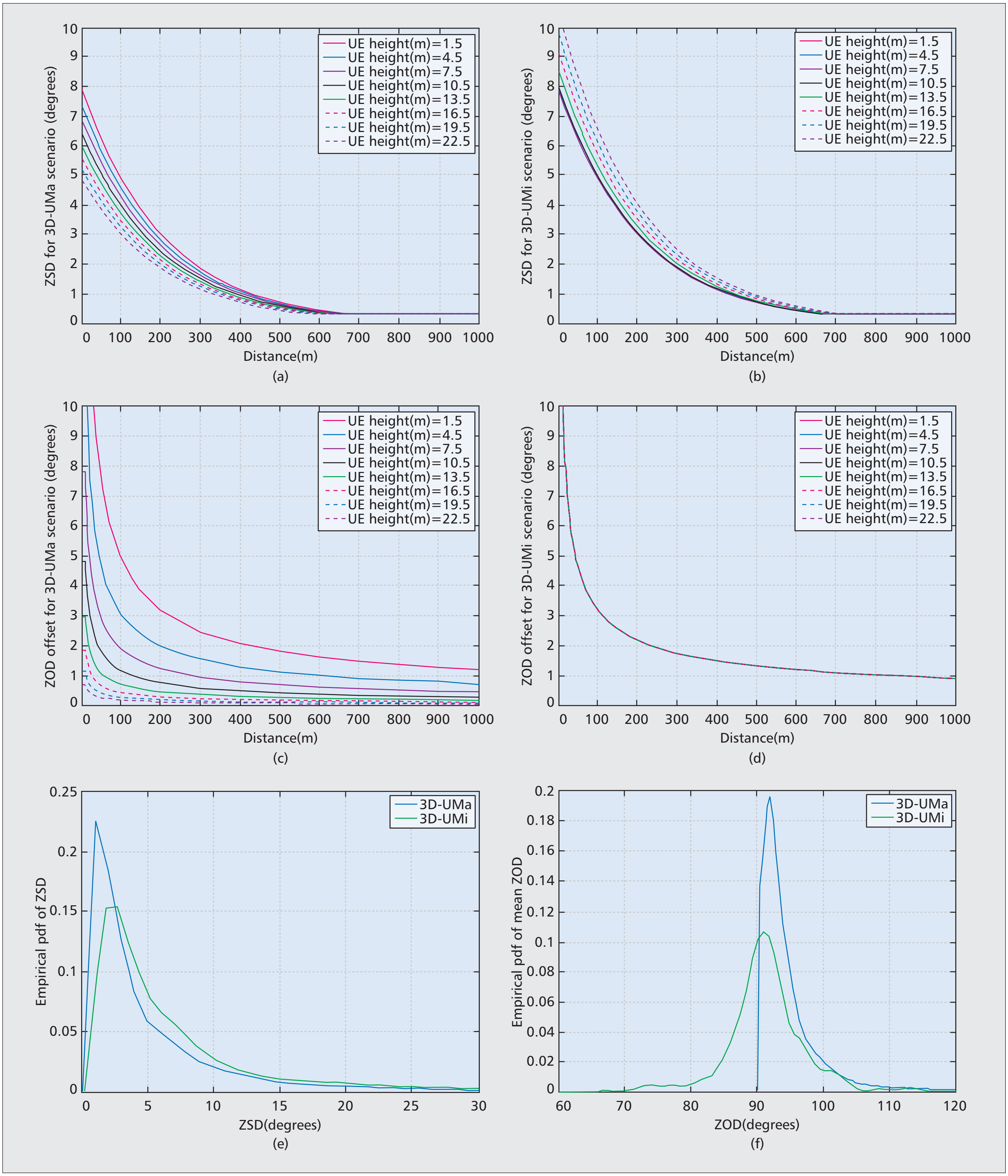

Figure 5. a) ZSD model for 3D-UMa; b) ZSD model for 3D-UMi; c) model for the magnitude of ZOD offset for 3D-UMa; d) model for the magnitude of ZOD offset for 3D-UMi; e) empirical pdf of ZSD (for the serving cell links) as observed in 3D-UMa and 3D-UMi scenarios; f) empirical pdf of mean ZOD (for the serving cell links) as observed in 3D-UMa and 3DUMi scenarios (90° is horizon).

offset models from Fig. 5c and Fig. 5d to the 3D-UMa and 3D-UMi scenarios with UE height ranging from 1.5m to 22.5m results in mean ZOD distributions as shown in Fig. 5f. It may be noted that for 3D-UMi the eNB height is below the surrounding buildings leading to mean ZOD distributed on both sides of the horizon (90 degrees), while for 3D-UMa the eNB height is above the surrounding buildings leading to mean ZOD distributed on only one side of the horizon.

In general this 3D channel model follows the framework of WINNERII/WINNER+ while also extending the applicability and the accuracy of the model by introducing some height and distance dependent elevation related parameters.

$$H_{u,s,n}(t) = \sqrt{P_n/M} \sum_{m=1}^{M} \begin{bmatrix} F_{rx,u,\theta}\left(\theta_{n,m,ZOA}, \phi_{n,m,AOA}\right) \\ F_{rx,u,\phi}\left(\theta_{n,m,ZOA}, \phi_{n,m,AOA}\right) \end{bmatrix}^T \begin{bmatrix} \exp\left(j\Phi_{n,m}^{\theta\theta}\right) & \sqrt{\kappa_{n,m}^{-1}}\exp\left(j\Phi_{n,m}^{\theta\phi}\right) \\ \sqrt{\kappa_{n,m}^{-1}}\exp\left(j\Phi_{n,m}^{\phi\theta}\right) & \exp\left(j\Phi_{n,m}^{\phi\phi}\right) \end{bmatrix}$$

$$\begin{bmatrix} F_{tx,s,\theta}\left(\theta_{n,m,ZOD}, \phi_{n,m,AOD}\right) \\ F_{tx,s,\phi}\left(\theta_{n,m,ZOD}, \phi_{n,m,AOD}\right) \end{bmatrix} \exp\left(j2\pi\lambda_0^{-1}\left(\hat{r}_{rx,n,m}^T \bar{d}_{rx,u}\right)\right) \exp\left(j2\pi\lambda_0^{-1}\left(\hat{r}_{tx,n,m}^T \bar{d}_{tx,s}\right)\right) \exp\left(j2\pi v_{n,m} t\right)$$

The ZOAs (corresponding to different MPCs) for UEs located indoors is modeled to be centered at 90 degrees. This is justified especially when the UE is deep inside the building where the indoor paths are guided by the floor and the ceiling as Fig. 4c illustrates. The ZOA center angle for outdoor UEs is modeled as the LOS ZOD angle for simplicity.

Fast Fading Channel Generation Equation

After characterizing all azimuth and zenith angles of MPCs and related parameters, the channel of the n-th path between u-th UE antenna and s-th eNB antenna is given by the above equation.

Here, $F_{rx,u,\theta}$ and $F_{rx,u,\phi}$ are field patterns of the receive antenna element u in the direction of the spherical basis vectors, a zenith basis vector $\hat{\theta}$, and an azimuth basis vector $\hat{\phi}$, respectively. $F_{tx,s,\theta}$ and $F_{tx,s,\phi}$ are field patterns of the transmit antenna element s in the direction of $\hat{\theta}$ and $\hat{\phi}$, respectively. The 2×2 matrix between the arrival and departure field pattern vectors is the depolarization matrix, of which two diagonal elements characterize the co-polarized phase response, and the two off-diagonal elements model the cross-polarized phase response and power attenuation

$$\sqrt{\kappa_{n,m}{}^{-1}}.$$

If polarization is not considered, the 2×2 polarization matrix can be replaced by the scalar $\exp(j\Phi_{n,m})$ and only vertically polarized field patterns are applied. $\hat{r}_{x,n,m}$ is the spherical unit vector with azimuth arrival angle $\phi_{n,m,AOA}$ and zenith arrival angle $\theta_{n,m,ZOA}$, given by:

$$\hat{r}_{rx,n,m} = \begin{bmatrix} \sin\theta_{n,m,ZOA}\cos\phi_{n,m,AOA} \\ \sin\theta_{n,m,ZOA}\sin\phi_{n,m,AOA} \\ \cos\theta_{n,m,ZOA} \end{bmatrix};$$

and $\hat{r}_{x,n,m}$ is the spherical unit vector with azimuth departure angle $\phi_{n,m,AOD}$ and zenith departure angle $\theta_{n,m,ZOD}$, given by

$$\hat{r}_{tx,n,m} = \begin{bmatrix} \sin\theta_{n,m,ZOD}\cos\phi_{n,m,AOD} \\ \sin\theta_{n,m,ZOD}\sin\phi_{n,m,AOD} \\ \cos\theta_{n,m,ZOD} \end{bmatrix}.$$

$\bar{d}_{rx,u}$ and $\bar{d}_{tx,s}$ are, respectively, the location vectors of receive antenna element u and transmit antenna element s. λ_0 is the wavelength of the carrier frequency.

Summary/Conclusions

A 3D channel model is an integral part of the evaluation of multi-antenna techniques exploiting elevation domain channel characteristics for supporting higher spectral efficiency. In this article we provided a summary and insight into the 3D channel model development in 3GPP spanning approximately a year starting in January 2013. We described the modelling of UEs at high-floors that is necessary to understand elevation beamforming and FD-MIMO performance. The height-gain model used to extend the 2D pathloss models to 3D was described, as was the height dependence of LOS probability. The methodology of extending the 2D fast-fading model from WINNER-II/ITU to 3D was described and the dependence of the new elevation parameters on UE height and distance from the eNB was illustrated. A comparison of the 3D channel model features from WINNERII, WINNER+, and 3GPP is provided in Table 1.

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	WINNERII [6]	WINNER+ [7]	3GPP TR36.873 [3]
Scenarios	Indoor, indoor to outdoor, outdoor to indoor (macro, micro), urban	Indoor, outdoor, outdoor to indoor (macro, micro), urban, suburban	Outdoor, outdoor to indoor (macro, micro), urban
eNB height (outdoor)	25m (macro), 10m (micro)	25m (macro), 10m (micro)	25m (macro), 10m (micro)
UE height (indoor)	1.5m – 7.5m (3 floors)	1.5m – 7.5m (3 floors)	1.5 m – 22.5m (8 floors)
LOS probability	Not height dependent	Not height dependent	Height dependent
Pathloss	No height-gain	Height-gain modeled	Height-gain modeled
Power angular spectrum (zenith)	Gaussian	Laplacian	Laplacian
Mean ZSD	Constant	Constant	Distance and height dependent
ZOD offset	Not modeled	Constant	Distance and height dependent

Table 1. High-level comparison of 3D channel models from WINNERII, WINNER+ and 3GPP.

Note: 3GPP documents may be downloaded from ftp://ftp.3gpp.org. WINNER II, WINNER+ documents may be downloaded from http://projects.celtic-initiative.org/winner+/index.html.

Biographies

BISHWARUP MONDAL (bishwarup@gmail.com) received the B.E. and M.E. degrees from Jadavpur University and the Indian Institute of Science in 1997 and 2000, respectively, and the Ph.D. degree from the University of Texas at Austin in 2006. He is presently with Nokia, Arlington Heights, IL. He is the recipient of the 2005 IEEE Vehicular Technology Society Daniel E. Noble Fellowship and a co-author of the best student paper award at IEEE Globecom 2006.

EUGENE VISOTSKY received a B.S., an M.S., and a Ph.D. in electrical engineering in 1996, 1998, and 2000, respectively, from the University of Illinois at Urbana-Champaign. He is currently with Nokia, Arlington Heights, IL. His current areas of interest are in advanced inter-cell interference coordination, cooperative transmission algorithms, and 3D MIMO techniques.

FREDERICK W. VOOK (SM'04) received the B.S. degree from Syracuse University in 1987 and the M.S. and Ph.D. degrees from The Ohio State University in 1989 and 1992, respectively, all in electrical engineering. From 1992 to 2011 he was with Motorola, Schaumburg, IL. Since 2011 he has been with Nokia, Arlington Heights, IL, where his current work involves advanced antenna array solutions for LTE and 5G cellular systems.

TIMOTHY A. THOMAS received the B.S. degree from the University of Illinois at Urbana-Champaign in 1989, the M.S. degree from the University of Michigan in 1990, and the Ph.D. degree from Purdue University in 1997. From 1997 to 2011 he was with Motorola, Schaumburg, IL. Since 2011 he has been with the North American Radio Systems Research Group, Technology and Innovation Office, Nokia, Arlington Heights, IL.

AMITABHA (AMITAVA) GHOSH joined Motorola in 1990 after receiving his Ph.D in electrical engineering from Southern Methodist University, Dallas. Since joining Motorola he has worked on multiple wireless technologies starting from IS-95, cdma-2000, 1xEV-DV/1XTREME, 1xEV-DO, UMTS, HSPA, 802.16e/WiMAX/802.16m, Enhanced EDGE, and 3GPP LTE. Dr. Ghosh has been issued 60 patents, and he has written numerous external and internal technical papers. Currently he is head of North America Radio Systems Research within the Technology and Innovation office of Nokia Networks. He is currently working on 3GPP LTE-Advanced and 5G technologies. His research interests are in the area of digital communications, signal processing, and wireless communications. He is a senior member of IEEE and co-author of the book titled *Essentials of LTE and LTE-A*.

YOUNG-HAN NAM received his B.S. and M.S. degrees from Seoul National University, Korea, in 1998 and 2002, respectively. He received a Ph.D. in electrical engineering from The Ohio State University, Columbus, in 2008. Since February 2008 he has been working at Samsung Research America, Dallas. He has been engaged in standardization, design, and analysis of the 3GPP LTE and LTE-Advanced from Release 8 through 12. His research interests include MIMO/multi-user/cooperative wireless communications, cross-layer design, and information theory.

YANG LI is a senior research engineer with Samsung Research America, Dallas. He received B.S. and M.S. degrees in 2005 and 2008, respectively, in electronic engineering from Shanghai Jiao Tong University, Shanghai, China, and a Ph.D. degree in 2012 in electrical engineering from the University of Texas at Dallas, Richardson, TX, USA. His research interests include MIMO, interference management, cooperative communication, and cognitive radio.

JIANZHONG (CHARLIE) ZHANG [S'96, M'02, SM'09] is currently senior director and head of the Wireless Communications Lab at Samsung Research America, Dallas, where he leads technology development, prototyping, and standardization for Beyond 4G and 5G wireless systems. From August 2009 to August 2013 he served as the vice chairman of the 3GPP RAN1 working group and led development of LTE and LTE-Advanced technologies such as 3D channel modeling, UL-MIMO and CoMP, carrier aggregation for TD-LTE, etc. Before joining Samsung he was with Motorola from 2006 to 2007, working on 3GPP HSPA standards, and with Nokia Research Center from 2001 to 2006, working on IEEE 802.16e (WiMAX) standard and EDGE/CDMA receiver algorithms. He received his Ph.D. degree from the University of Wisconsin, Madison.

MIN ZHANG received his M.S. degree with distinction in telecommunication from the University of Canterbury, New Zealand in 2005, and the Ph.D. degree in telecommunication from the Australian National University in 2009. He has worked for Alcatel-Lucent since 2009 as a wireless researcher and senior standards engineer, and is currently working on 3GPP LTE/LTE-A standardization. His interests include wireless channel modelling, communication and information theory, MIMO-OFDM system design, and performance optimization of standards.

QINGLIN LUO is a leading research scientist with Bell Labs Shanghai. His primary responsibilities include conducting and leading research projects on air-interface technologies for next generation communications. Previously he was with Motorola, UK as a base station system architect, and with the China Academy of Telecommunication Technologies (CATT) as a standards engineer. He holds a Ph.D. degree in electrical engineering from the University of Surrey, UK. His current research interests include advanced antenna array techniques, massive MIMO, error control coding, wireless cloud, etc.

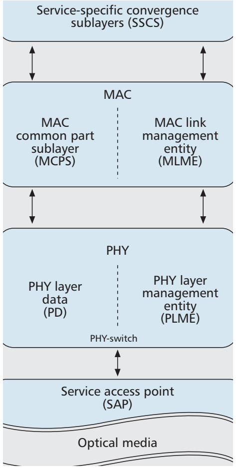
YUICHI KAKISHIMA received the B.S. and M.S. degrees from the Tokyo Institute of Technology, Tokyo, Japan, in 2005 and 2007, respectively. In 2007 he joined NTT DOCOMO, Inc., where he has been engaged in the research and development of wireless access technologies, including multiple-antenna transmission techniques for LTE and LTE-Advanced systems.

KOSHIRO KITAO received B.S., M.S., and Ph. D. degrees from Tottori University, Tottori, Japan in 1994, 1996, and 2009, respectively. He joined the Wireless Systems Laboratories, Nippon Telegraph and Telephone Corporation (NTT), Kanagawa, Japan, in 1996. Since then he has been engaged in the research of radio propagation for mobile communications. He is now a research engineer in the Research Laboratories, NTT DOCOMO, INC., Kanagawa, Japan.

Standards for Indoor Optical Wireless Communications

The application of Optical Wireless Communications (OWC) has grown so much in recent years that the whole industry would benefit from common standards to which competitive products comply. Standards are essential, particularly when the market expands into high-volume products like home appliances and other consumer goods. This article reviews the existing OWC standards and points toward future directions

A. C. Boucouvalas, Periklis Chatzimisios, Zabih Ghassemlooy, Murat Uysal, and Konstantinos Yiannopoulos



A. C. Boucouvalas and Konstantinos Yiannopoulos are with the University of Peloponnese.

Periklis Chatzimisios is with Alexander TEI of Thessaloniki.

Zabih Ghassemlooy is with the University of Northumbria at Newcastle.

Murat Uysal is with the Ozyegin University.

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Abstract

The application of Optical Wireless Communications (OWC) has grown in recent years that the whole industry would benefit from common standards to which competitive products comply. Standards are essential, particularly when the market expands into high-volume products like home appliances and other consumer goods. This article offers a timely review of standards-writing activity, as OWCs find their way into diverse products varying from TV remote controls to satellite links. This article discusses the most popular standards for optical wireless communications. We outline the IEEE 802.11 standard for optical wireless local area networks, and the ongoing standardization effort by IrDA on personal optical wireless systems. The article concludes with a discussion of the recently announced IEEE 802.15.7 standard on visible light communications.

Introduction

Optical Wireless Communications (OWC) is a niche technology combining the best of both the wireless and optical worlds, enabling the development of ultra-broadband, agile and non-obtrusive communication systems [1, 2]. OWC is being applied in numerous commercial indoor and outdoor systems that range in scale from short-range/personal-area to access/metro area networks. In particular, OWC is compatible with other well-established technologies like optical fiber and interior lighting, and can benefit significantly from advances in these industries. As a result, OWC during the last decade has seen a significant increase in the attained bandwidth and an equally important cost decrease. OWC is expected to keep up these trends, closely following the advance of fiber optics, and provide even more broadband and cost-effective systems in the future.

The potential of OWC for broadband, cost-effective, safe and non-obtrusive communication systems has attracted the interest of two major standardization bodies, IEEE and IrDA (Fig. 1). IEEE included an OWC implementation scenario of the original 802.11 standard, which has been discontinued.

Instead, IEEE continued to pursue OWC standardization. The result is the state-of-the-art 802.15 standard on Visible Light Communications (VLC) that use commercial LEDs to provide broadband indoor communications. IrDA, on the other hand, has maintained a comprehensive list of standards that focus mainly on the implementation of OW links for Personal Communication Systems (PCS). OW-based personal communications at 1 Gb/s have been demonstrated and a ten-fold increase in bandwidth is expected in the near future.

This article is organized as follows: we offer a short overview of the IEEE 802.11 OW standard and the challenges it faced. We discuss the IrDA protocol stack and unique solutions IrDA adopted for implementing a low-cost, high-bandwidth PCS. We present the IEEE 802.15 VLC standard, focusing on the key requirements of the physical and link layers. Finally, we summarize ongoing efforts for the development of future standards.

Optical Wireless Local Area Networks

Following the introduction of Wireless Local Area Networks (WLANs), in 1997, IEEE released the initial IEEE 802.11 standard for the physical (PHY) and Medium Access Control (MAC) layers [1]. Three choices are standardized in the PHY layer; two utilize radio (Direct Sequence and Frequency Hopping Spread Spectrum modulation) and one utilizes IR signals, although a single product can support only one of them. All three transmission methods provide a basic data rate (R) of 1 Mb/s and an optional rate of 2 Mb/s. The PHY of 802.11 is split into the Physical Layer Convergence Protocol (PLCP) and the Physical Medium Dependent (PMD) sublayers. PLCP prepares (parses) data units that are transmitted (received), whereas PMD performs the data transmission (reception) and modulation (demodulation) under the guidance of PLCP. The IR PHY supports two different Pulse Position Modulation (PPM) schemes, 16 PPM for 1 Mb/s and 4 PPM for 2 Mb/s. The peak-power wavelength of the transmitter (Tx) is between 850 and 950 nm, while a typical link length L is limited to 10 m. Finally, two main transmission techniques are used: point-to-point and diffused.

The MAC layer is based on the CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) technique, where the station that has a frame ready for transmission first senses the channel. If the channel is free for a period longer than the DIFS (Distributed Inter-Frame Spacing) duration, the station transmits. If the channel is busy, the station continues sensing until the channel is free for a period longer than DIFS. Afterward, the station employs a contention-resolution method prior to sending, namely Binary Exponential Back-off, so as to minimize the probability of frame collisions due to multiple simultaneous transmissions.

IR transmission was initially employed in IEEE 802.11 because it provides a number of significant advantages over radio approaches such as unregulated broadband spectrum. It is low cost, offers high security against eavesdropping and a large LAN can be established over separate installations without interference. However, Infrared Links (IRL) exhibit significant drawbacks, including multipath-induced inter-symbol interference, difficulties in detecting effectively optical collisions, interference from ambient lights and shadowing. Additional drawbacks are limited device mobility, exclusively indoor operation (meaning a limited coverage area), as well as dangers to the eye and relatively high power consumption. Because of drawbacks, IR 802.11 was never successfully implemented commercially. As a result, IEEE discontinued the IR version of 802.11 and since 1999 it has released WLAN standards that utilize only radio waves.

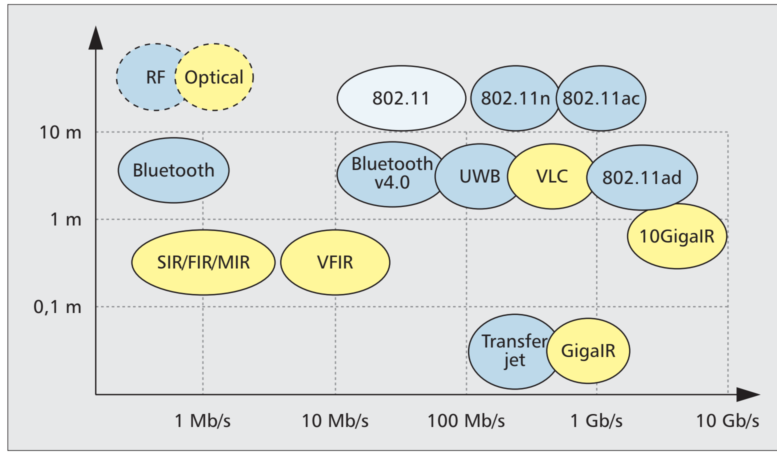

Figure 1. Popular OW/ RF standards for indoor and personal communications.

Personal Area Communications

Infrared Data Association (IrDA) produced a set of OWC standards focusing on low cost, short-range, point-to-point links mainly for handheld devices and maintains a comprehensive list of technical documents that define the operation of short-range PCS over IRLs (Fig. 2). Similar to the OSI reference model, IrDA has adopted a layered approach to networking, although there is no exact one-to-one mapping between OSI and IrDA layers. Following the layered approach, different protocols are typically implemented on each layer, especially the upper ones, with each protocol engineered to meet the needs of one or more application profiles. Three application profiles are currently in use, ranging from contact information exchanges to ultra-fast file transfers.

The IrDA Physical Layer (IrPHY)

IrPHY *provides high-speed* and almost *error-free connections* between mobile devices or mobile devices and fixed stations by means of *half-duplex*, *narrow-cone point-to-point IRLs* at wavelengths of 850-900 nm. PHY standards include 2.4–115.2 kb/s Serial Infrared (SIR), 0.576 and 1.152 Mb/s Medium Infrared (MIR), 4 Mb/s Fast Infrared (FIR), 16 Mb/s Very Fast Infrared (VFIR) and 96 Mb/s Ultra Fast Infrared (UFIR). The Gigabit Infrared (Giga-IR) standard currently supports 512 Mb/s and 1.024 Gb/s [4], while a data rate up to 10 Gb/s is being standardized for the next version of Giga-IR.

IrPHY provides recommendations for a number of IRL (operating over limited distances and narrow emission and reception angles) parameters to enhance link security, unobtrusive operation, conformance to the eye/skin safety regulations and immunity to background radiation. In SIR-to-UFIR speeds the maximum link length L_{max} between devices may not exceed 1 m (or 20 cm for low-power implementations), while the 3-dB half-width angle ranges between 15°–30° and >15° at Tx and receiver (Rx), respectively. Beam alignment is more crucial in Giga-IR and a docking station is typically used. Although beaming techniques are expected to allow for $L > 1$ m in future standards, the docking station currently limits L_{max} to 6 cm, while the transceiver angles are limited to 5°.

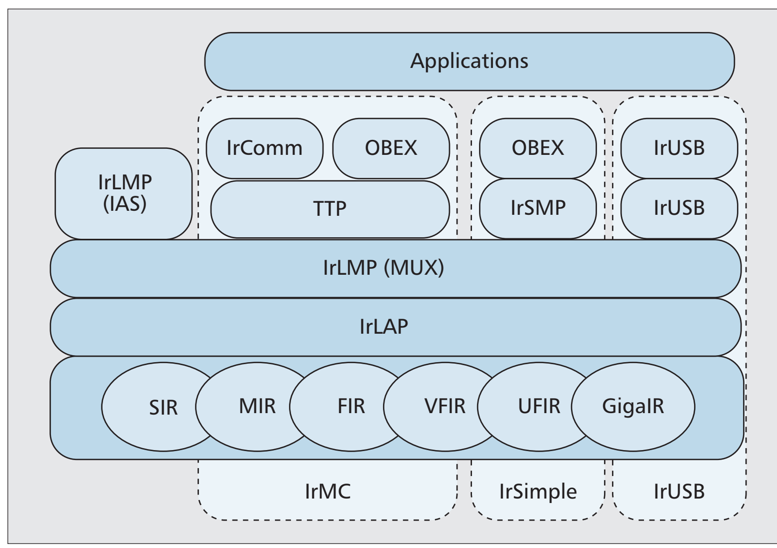

Figure 2. The IrDA protocol stack and application profiles.

With respect to link reliability, the achievable Bit-Error-Rate (BER) for all IRLs should be better than 10^{-8} (SIR-VFIR) and 10^{-10} (Giga-IR), which is comparable to BERs of optical fiber networks. IrPHY is responsible for physical-layer framing and appends headers and trailers to the transmitted higher-layer frames. The IrPHY header consists of a repeated padding sequence followed by a start character to indicate the arrival of a new frame and ensure that Rx electronics have adequate stabilization time. The trailer contains a Cyclic Redundancy Check (CRC) field that protects frames against errors plus a stop character to indicate the end of frame transmission.

The IrDA Link Access Protocol

The key functions of the IrDA link-layer are to *establish*, *maintain* and *terminate* connections over IrPHY [5]. This is accomplished by the IrDA Link Access Protocol (IrLAP), an HDLC variant augmented with features that compensate for the volatile nature of IrPHY. IrLAP operates in two distinct modes:

- Normal Disconnected Mode (NDM) when no link-layer connection is present — for

Whenever set to exclusive mode, LM-MUX does not provide multiplexing but is fully commanded by a single upper-layer application. Exclusive mode is primarily aimed for low-latency applications or ones that require self-control over the link.

instance when a device boots or after a previously established connection has been terminated

- Normal Response Mode (NRM), which provides all the procedures for establishing a new connection, including *device initialization*, *device discovery* and *link negotiation*.

During *initialization*, IrLAP generates a random 32-bit device address (similar to the Ethernet MAC address) and initializes the link-layer parameters to default values, common to all IrDA-compliant devices. Default values must be used in NDM procedures so that all devices may participate in the subsequent device discovery and link negotiation/establishment processes regardless of their capabilities. Device discovery is initiated when no other IR communication takes place. To this end, the device that initiates the discovery process monitors the link for 500 msec. If no ongoing communications are perceived, the initiator broadcasts an unnumbered U-frame to notify nearby peers of its availability. Nearby devices respond to the U-frame and if more than one device is present users are prompted to select the device they wish to connect to. Finally, link negotiation follows a successful device discovery, where the initiator and responder devices agree upon link parameter values that will be used during NRM. Negotiated link parameters include the transmission speed, the frame payload size, the protocol window size, the minimum and maximum link turnaround times (TLT), and the the link disconnect threshold (3–40 sec).

Following a successful link negotiation IrLAP switches to NRM, which revolves around the communication of information I-frames that carry the application data. The payload size of I-frames ranges between 64 and 2048 bytes at SIR-VFIR, and is extended to 32 kbytes (UFIR) or 64 kbytes (Giga-IR). Due to the half-duplex link (HDL) nature of IR, *a key responsibility of IrLAP is to supervise the link operation* and ensure that both communicating devices take their fair share of time transmitting frames. Thus, each device is allowed to transmit its own I-frames until a predefined amount of time (up to 500 msec) has elapsed. After a full transmission or if the device has no more frames to send, it informs the remote side about the imminent link turnaround time using a special field on the header of the last I-frame, called the Poll/Final bit. The link direction is reversed after a short pause (0.01–10 msec) required for transceivers to stabilize for reverse operation.

Apart from supervising HDL operation, *IrLAP is also responsible for the correct and in-order-delivery of I-frames to upper layers*. To this end, IrLAP sequences I-frames and utilizes the error-detection mechanism provided by IrPHY CRC to isolate transmission errors. When an I-frame error is detected, Rx provides negative feedback and requests retransmission(s). At SIR-UFIR speeds, the retransmissions take place in a Go-Back-N fashion so as to simplify Rx; a higher throughput Selective-Repeat mode of operation is standardized for contemporary Giga-IR devices with extended reception capabilities [6]. In both cases, a protocol window of up to 7 (SIR-MIR) or 127 (VFIR and higher) frames is utilized so as to limit the number of pending frames at Tx, as well.

The IrDA Link Management Protocol (IrLMP)

IrLMP is responsible for providing upper layers with *a multiplexing scheme* over IrLAP, a Link Control (LCon) mechanism and *a service discovery mechanism*. IrLMP is separated in two logical entities: the Link Management Multiplexer (LM-MUX) and the Information Access Service (IAS) [7]. The LM-MUX entity implements *frame multiplexing* by means of Logical Service Access Points (LSAPs), which operate similarly to TCP ports. Each upper layer application is assigned a unique LSAP identifier (i.e., a LSAP-selector, or LSAP-sel). Applications utilize their LSAP-sels to establish IrLMP connections. A source-destination LSAP-sel pair defines a unique IrLMP connection and creates communication endpoints required for demultiplexing frames inside LSAPs. IrLMP frames carry the source-destination LSAP-sels on their headers to indicate the IrLMP connection they belong to. LM-MUX also supports an exclusive mode of operation to facilitate *LCon*. Whenever set to exclusive mode, LM-MUX does not provide multiplexing but is fully commanded by a single upper-layer application. Exclusive mode is primarily aimed for low-latency applications or ones that require self-control over the link.

Service discovery is performed in the IAS entity. This consists of two separate components: a local database that stores the associations between applications with LSAP-sels, and an Information Access Protocol (IAP) that communicates the database entries between IrLMP peers. The local database stores registered applications in the form of objects associated with a class (application) name, a unique identifier and several attributes, including LSAP-sel. Remote applications may access the database and retrieve these objects through the command/response capabilities provided by IAP. IAP utilizes its own operation frames encapsulated in the payloads of IrLMP frames, to communicate queries and responses. For implementation purposes, LM-MUX regards IAS as an upper-layer application with a "well-known" LSAP-sel value of (0×00), so that all remote connections have a common access reference.

Application Profiles

The first application profile standardized by IrDA was the Infrared Mobile Communications (IrMC) [8]. IrMC *defines methods for exchanging objects typically found in mobile phone*s (like contact cards, phonebooks, text messages and call logs). IrMC-oriented applications utilize two core protocols, the Infrared Object Exchange (OBEX) [9] and the Tiny Transport Protocol (TTP). OBEX is a session layer protocol that *provides an API for transferring application objects*. OBEX follows a client/server request-response model (similar to HTTP) and applications may request the transfer of objects using PUT/GET commands. Requests and responses are transferred over OBEX frames in the form of headers, which typically include information about the communicated objects such as the size, name or the object itself. TTP, on the other hand, *performs higher level flow control* as well as *segmentation and re-assembly*. *Flow control*, required to avoid flooding the OBEX server with an excessive number of requests, is implemented using a credit-based mechanism. TTP peers use up a credit upon sending a TTP frame and receive a new credit with each TTP frame. Peers reaching zero credits must wait for credits to be issued from the other side before sending additional frames. Segmentation and re-assembly is also a critical TTP function because the maximum size of OBEX frames is 64 kbytes and cannot be accommodated by the standard IrLAP payload in SIR-UFIR speeds.

IrMC provides a generic framework for transferring data over IRLs. But it has been enhanced to better suit "point-and-shoot"(P&S) applications that require short-lived connections — for example, a camera sending a digital picture to a printer. The key requirement in P&S applications is *the minimization of the interaction time between devices* and the IrSimple application profile achieves this by modifying time-consuming procedures in IrLAP and TTP.

The three primary reasons for delay in IrLAP are device discovery, the TLT, and automatic-repeat-requests, especially those taking place in a Go-back-N fashion. IrSimple-compliant devices completely ignore the 500 msec listening period during device discovery, assuming that no IRL is present. Moreover, link turnarounds occur less often in IrSimple, with the maximum turnaround time raised to 1 sec from 500 msec. Data transfers are also considered uni-directional and the receiving side immediately reverses the link direction without sending any data of its own. Finally, application data are sent over unnumbered U-frames instead of information I-frames, alleviating the requirements for frame sequencing, conforming to a transmission window and providing feedback about erroneous frames. The Infrared Sequence Management Protocol (IrSMP), which completely ignores flow control, assuming that the receiving side may not be overwhelmed by a simple P&S application, replaces TTP. IrSMP maintains the segmentation and re-assembly function of TTP and also provides a frame-sequencing mechanism for Automatic Repeat-reQuest (ARQ) purposes, since ARQ is not implemented in IrLAP. To these ends, IrSMP transfers data blocks segmented into frames before they are passed to IrLMP and IrLAP. Erroneous frames inside a block are re-transmitted by IrSMP; the transfer of a new block begins only after all frames in a preceding block have been delivered correctly.

Another application framework provided by IrDA is Infrared Financial Messaging (IrFM) that supports wireless digital financial transactions (known as Point-and-Pay) [10]. In IrFM, a user who wants to pay for services and merchandise, utilizes a device (also called a Personal Trusted Device, or PTD) and interacts with a point of sale terminal. IrFM addresses many aspects of the requirements of a desirable digital payment system by achieving a fast transaction due to the P&S nature of IrDA. Moreover, IrFM maintains simplicity, security (it's difficult to eavesdrop), interoperability and reliability for IR financial transactions.

Within the same rationale of *minimization of the interaction time between devices*, the IrBurst profile replaces OBEX with a more efficient session protocol, without altering the operation of TTP and IrLAP [11]. The main drawbacks of OBEX are

- Its *stop-and-wait* method for transferring objects
- A *limited frame* size of 64 kbytes

To transfer a large-size object with OBEX requires multiple sequential GET commands; none of them, however, has sufficient data to fully occupy an IrLAP window at high R (UFIR or higher), while the response to each command is additionally delayed by one t_{lt} due to the HDL operation. As a result, the complete object transmission is delayed by the number of OBEX frames that the object consists of, times the t_{lt}. IrBurst alleviates this delay by organizing data in streams that need not be acknowledged in a stop-and-wait fashion. Moreover, IrBurst can create multiple data streams that run over parallel TTP connections to bypass the TTP flow-control mechanism. The net result is a fully utilized IrLAP transmission window and a drastic decrease in the number of turnarounds required to transmit the object.

Finally, IrDA provides standards for using IRLs as "wire-replacements." The initial "wire-replacement" standard, known as IrComm, is part of the IrMC profile [12]. IrComm *provides an interface that emulates the operation of legacy serial and parallel ports over IRLs* and takes into account three things: that these ports are bidirectional, typically involve data transmission over more than one wire and are sensitive to timing. To this end, IrComm creates multiple data and control channels that emulate the electrical operation of wires, and utilizes the flow control mechanisms provided by IrLAP and TTP to emulate the software and hardware flow control of serial and parallel ports. A more recent application profile for "wire-replacement" is IrUSB now under development and not officially accepted by IrDA [13]. IrUSB aims to *provide applications with a USB 2.0 emulation interface* that runs solely over Giga-IR, since USB 2.0 supports R of up to 480 Mb/s.

Current OWC systems mainly use the IR band due to the availability of Tx and Rx devices originally developed for fiber-optic communication systems. However, recently, a growing number of academic and industrial interests are looking toward the potential deployment of the visible band for various OWC applications.

Visible Light Communications

Current OWC systems mainly use the IR band due to the availability of Tx and Rx devices originally developed for fiber-optic communication systems. However, recently, a growing number of academic and industrial interests are looking toward the potential deployment of the visible band (380–780 nm) for various OWC applications. This has been triggered by recent advances in solid-state technologies that have led to the development of highly-efficient (up to 60% energy efficiency) light emitting diodes (LEDs) [14]. In line with government plans worldwide to phase out incandescent light bulbs in favor of more energy-efficient lighting, it is predicted that LEDs will gradually replace both incandescent and fluorescent lights and be the ultimate light source in the near future. Visible LEDs are already widely employed for indoor and outdoor illumination.

The main drivers for this emerging green technology include: longer lifetime of high-brightness LEDs compared to other light sources, higher bandwidth/data rates, higher tolerance to humidity, higher data security, mer-

	Infrastructure	Mobile	Vehicle
Fixed coordinator	Yes	No	No
Power supply	Ample	Limited	Moderate
Form factor	Unconstrained	Constrained	Unconstrained
Light source	Intense	Weak	Intense
Physical mobility	No	Yes	Yes
Range	Short/long	Short	Long
Data rates	High/low	High	Low

Table 1. VLC device classification.

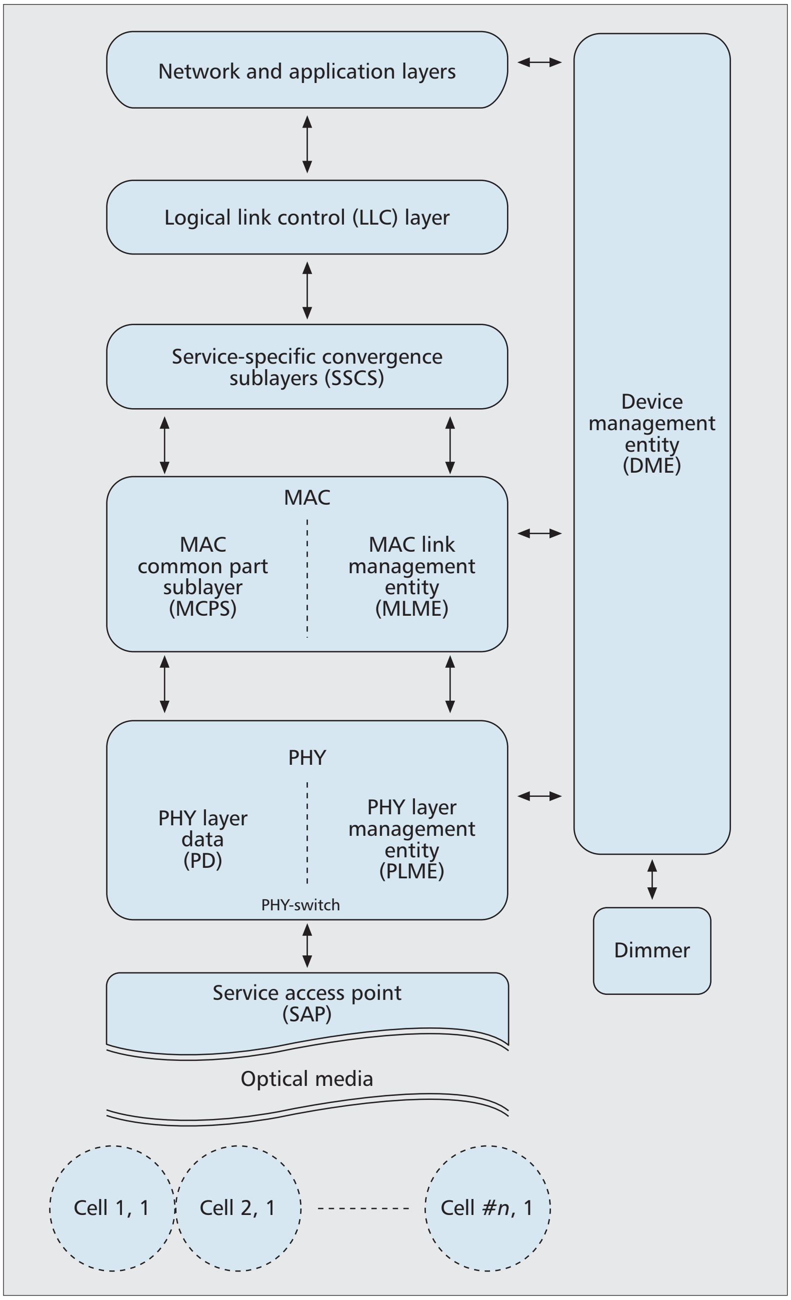

Figure 3. VLC device architecture.

cury-free manufacture, extreme power efficiency, compactness and, most importantly, fast switching characteristics [15]. The expected wide availability of LEDs in the near future will open the door to so-called "Visible Light Communications (VLCs)." Colors of commercially available and highly efficient blue, green and red LEDs can be mixed to produce a white light. These LEDs can be modulated individually to provide high-bandwidth communications. A more conventional way is to directly modulate a single LED at tens or hundreds of megahertz, without noticeable effect on the lighting output and human eyes. The dual functionality (i.e., illumination and data communication) of high-brightness visible LEDs has created a wide range of short- and medium-range communication applications. These include wireless local, personal and body area networks (WLAN, WPAN, and WBANs), vehicular networks and machine-to-machine, airplane cabin, train, ship and traffic-light communications among many others. In comparison to RF technologies, VLC offers inherent advantages, such as immunity to electromagnetic interference, unlicensed bands, physical security and a high degree of spatial confinement that allows for a much higher reuse factor.

Recognizing the potential of this technology, IEEE produced the 802.15.7 standard, approved in 2011 [16, 17]. This standard defines PHY and MAC layers for short-range VLC in an optically transparent media to support audio and video multimedia services. This new standard was required due to the unique challenges evident in VLC usage profiles that render inapplicable the existing 802.11 and IrDA standards .

VLCs, for example, are designed to provide permanent or semi-permanent data connections that also enable non-trivial user mobility. This is in stark contrast to IrDA P&S systems that are optimized for short interaction times and very high speeds at the expense of users remaining practically stationary. In a sense, VLCs are closer to 802.11 in terms of user mobility and connection life-time. Still, VLCs must also provide high-quality illumination that is simply beyond the scope of an evolving 802.11.

VLC Device Classes and Overall Architecture

The IEEE 802.15.7 standard considers three classes of VLC devices, namely "infrastructure," "mobile" and "vehicle," see Table 1 for their main features. The VLC device is comprised of a PHY layer, which contains the transceiver with intensity modulation/direct detection, and a MAC layer that provides access to the physical channel for all types of transfers. A logical LCon layer accesses the MAC layer through the service-specific convergence sub-layer. The Device Management Entity (DME) has access to certain dimmer-related attributes and provides dimming information to both MAC and PHY layers. DME also controls the PHY switch for selecting optical sources/photodetectors in devices comprised of multiple transmitters/receivers. The PHY switch interfaces to the optical Service Access Point (SAP) and connects to the optical media, which may consist of one or more transmitter/receivers. The upper layers consist of the network and application layers (Fig. 3).

The Physical Layer

The standard supports the following three PHY types (Table 2 for a complete summary):

- **PHY I** — mainly for lower R (11.6 to 266.6 kb/s) outdoor applications. It employs On-Off Keying (OOK) and Variable PPM, and also supports concatenated coding with Reed-Solomon (RS) and Convolutional Codes.
- **PHY II** — for higher R outdoor/indoor applications (1.25 to 96 Mb/s). It also employs OOK and VPPM, but supports only RS coding.
- **PHY III** — for applications with multiple light sources/detectors at different frequencies (colors). It uses a particular modulation format, called Color-Shift Keying (CSK), and RS coding to achieve R in the range of 12 to 96 Mb/s.

The above PHYs may co-exist but do not interoperate. An IEEE 802.15.7-compliant device must implement at least one of the PHY I or PHY II types. A device implementing the PHY III type must also implement PHY II mode for co-existence. Note that multiple optical rates are provided for all PHY types in order to support a broad class of LEDs for various applications. The choice of the optical rate for communications is determined by the MAC layer during device discovery.

Other details of the main PHY functionalities include:

Modulation: OOK and VPPM provide a trade-off between R and the dimming range. PHYs I and II also support a run length limited coding to provide a DC balance, clock recovery and flicker mitigation. Type I utilizes Manchester DC-balancing and 4B6B encoding, while PHY II utilizes 8B10B/4B6B encoding. In PHY III, CSK modulation is generated by using 3-color light sources out of the 7-color bands defined in the standard.

Error-correction coding: In PHY I (outdoor applications), path losses due to long link span and the interference due to ambient and artificial light sources require more powerful concatenated coding with a combination of convolutional outer codes and RS inner codes. The RS encoder output is padded with zeros to form an interleaved boundary, which is then punctured and the result sent to the inner convolutional encoder. On the other hand, RS codes are used for indoor applications (PHYs II and III), since the coding requirements are less stringent.

Channel selection: The IEEE 802.15.7 PHY can perform Clear Channel Assessment (CCA) according to one of the following three methods:

- CCA Mode 1 (energy E above threshold E_{th}): In this mode, CCA reports a busy medium if it detects any E above E_{th}.
- CCA Mode-2 (carrier sense only): In this mode, CCA reports a busy medium if it detects a signal with the modulation characteristics of IEEE 802.15.7. This signal may be above or below E_{th}. Finally, there is
- CCA Mode-3 (carrier sense with E above E_{th}): In this mode, a busy medium is reported if a signal with the modulation characteristics of IEEE 802.15.7 exists and its E is above E_{th}.

	Modulation	RLL code	Optical clock rate	FEC		Data rate
				Outer code (RS)	Inner code (CC)	
PHY I	OOK	Manchester	200 kHz	(15,7)	1/4	11.67 kb/s
				(15,11)	1/3	24.44 kb/s
				(15,11)	2/3	48.89 kb/s
				(15,11)	—	73.3 kb/s
				—	—	100 kb/s
	VPPM	4B6B	400 kHz	(15,2)	—	35.56 kb/s
				(15,4)	—	71.11 kb/s
				(15,7)	—	124.4 kb/s
				—	—	266.6 kb/s
PHY II	VPPM	4B6B	3.75 MHz	RS(64,32)		1.25 Mb/s
				RS(160,128)		2 Mb/s
			7.5 MHz	RS(64,32)		2.5 Mb/s
				RS(160,128)		4 Mb/s
				—		5 Mb/s
	OOK	8B10B	15 MHz	RS(64,32)		6 Mb/s
				RS(160,128)		9.6 Mb/s
			30 MHz	RS(64,32)		12 Mb/s
				RS(160,128)		19.2 Mb/s
			60 MHz	RS(64,32)		24 Mb/s
				RS(160,128)		38.4 Mb/s
			120 MHz	RS(64,32)		48 Mb/s
				RS(160,128)		76.8 Mb/s
				—		96 Mb/s
PHY III	4-CSK	—	12 MHz	RS(64,32)		12 Mb/s
	8-CSK			RS(64,32)		18 Mb/s
	4-CSK		24 MHz	RS(64,32)		24 Mb/s
	8-CSK			RS(64,32)		36 Mb/s
	16-CSK			RS(64,32)		48 Mb/s
	8-CSK			—		72 Mb/s
	16-CSK			—		96 Mb/s

Table 1. VLC PHY operating.

The MAC layer provides two services accessed through two SAPs. The MAC management is accessed through the MAC sublayer management entity SAP (MLME-SAP), while MAC data is accessed through the MAC common part sub-layer SAP (MCPS-SAP).

Wavelength Quality Indication (WQI): WQI measurements characterize the strength and/or quality of a received frame, with the result reported to the MAC sublayer. At least seven unique values of WQI are required in a standard-compliant device. Neither its measurement nor its use by the network or application layers are specified in the standard; they are left vendor specific. In practice, WQI may be implemented using an energy detector, a signal-to-noise ratio estimator or a combination of these methods.

Synchronization: A preamble field in the PHY frame is used by the transceiver to synchronize an optical clock with an incoming message. The standard defines one fast locking pattern followed by the choice of four topology-dependent patterns to distinguish between different PHY topologies. The preamble is sent at a clock rate chosen by the Tx and supported by the Rx. A time-domain sequence the preamble does not incorporate any channel coding or line coding.

Dimming mechanism: Light dimming is defined as controlling the perceived brightness of the light source according to the user's requirement, and is a cross layer function between the PHY and MAC layers. There are several schemes for dimming. It can be achieved by either redefining the "on" or "off" levels of the OOK symbol to have a lower intensity or by keeping the same levels and only changing the average duty cycle by inserting a "compensation" time in the modulation symbol [2].

Pulse Width Modulation (PWM): Another option that offers the advantage of linear dimming control. A PWM dimming scheme does not suffer from the wavelength shift caused by the current variation in amplitude-modulated dimming techniques. In PWM, the pulse duration is used to control the light-source drive current, thus adjusting the dimming level (brightness). Frequency of the PWM dimming signal is typically above 100 Hz and human eyes cannot see the current switching [2].

MAC Layer

The MAC layer provides two services accessed through two SAPs. The MAC management is accessed through the MAC sublayer Management Entity SAP (MLME-SAP), while MAC data is accessed through the MAC Common Part Sub-layer SAP (MCPS-SAP). The IEEE 802.15.7 standard supports three MAC topologies:

- Star topology: communication is established between devices and a single central controller, referred to as the coordinator
- Peer-to-peer topology: each device can communicate with any other device within its coverage range, with one of the two devices in an association acting as a coordinator
- Broadcast topology: each device can transmit (in unidirectional mode) a signal to other devices without being associated with any device or having any devices associated with it.

The standard allows the optional use of a Super-Frame (SF) structure with its format defined by the coordinator and its limits bounded by a Beacon Signal (BS) initiated by the coordinator. Beacons are used in networks that either require synchronization or support for low-latency devices. Networks not requiring synchronization or supporting low-latency devices can elect to turn off BS for normal transfers. However, BS is still required for network discovery.

The standard has adopted four types of channel-access mechanisms, depending on the network configuration. The non-Beacon-Enabled Network (non-BEN) uses an *unslotted random channel-access* mechanism, with or without CSMA/CA. When a device needs to transmit data frames or MAC commands, it first waits for a random back-off period and then transmits its data frame. If the optional carrier sense mechanism is active and the channel is found to be busy, the device waits for another random period before trying again to access the channel. On the other hand, BEN uses a *slotted random channel access* mechanism, with or without CSMA/CA, where the back-off slots are aligned with the start of BS transmission. When a device wishes to start data transmission, it first locates the boundary of the next back-off slot and then waits for a random number of back-off slots. If the channel is idle, the device begins transmitting on the next available back-off slot boundary. A successful reception and validation of a data or MAC command frame can be optionally confirmed with an acknowledgment command. Acknowledgment and beacon frames are sent without using a random-access mechanism.

The MAC layer, which also handles all access to the PHY, is responsible for the following main tasks [16]:

Procedures for initiating/maintaining a VLC network: VLC devices perform channel scanning to assess the current state of a channel (or channels), locate all beacons within its operating space or locate a particular beacon if synchronization has been lost. Results of a channel scan are used to select an appropriate logical channel along with a VLC network identifier not being used by any other VLC networks in the same area.

Association/disassociation procedures: outline the conditions under which a device may join a VLC network and the conditions for a coordinator to permit devices to join in. Similarly, a disassociation procedure exists, which can be initiated by the associated device or its coordinator.

Color-function support mechanism: provides device information to the human eye via color. Various device states such as the device discovery (scan, association and disassociation), file transfer status, wavelength quality indication and acknowledgments can be visually indicated to the user to help with device alignment for communications.

Illumination and dimming support mechanisms: feature extended preamble mode for providing illumination with improved synchronization performance, dimming overrides, adjusting the MAC layer transmission schedule to accommodate dimming, association and link adaptation in the presence of dimming.

Mobility support mechanism: supports mobility of the device under an infrastructure that supports multiple optical elements over a wide coverage area ("cells").

Color stabilization: used to stabilize the optical color emitted by the Tx. The color-visibility-dimming frames are used to estimate the change in color and this information can be provided as feedback to the Tx to stabilize its color.

Conclusions and Future Directions

We have presented the three most important families of standards for OWCs maintained by IEEE and IrDA. IEEE standards focus mainly on the end-user access, providing permanent connections, moderate line data rate and acceptable user mobility. VLCs, in particular, can also provide high-quality illumination without additional equipment. On the other hand, IrDA systems target "P&S" applications and are optimized for short interaction times and very high capacities.

With respect to future standardization endeavors, the main evolutionary drive in IrDA standards at the time of writing this article is the migration to link speeds of over 10 Gb/s. Currently, a Special-Interest-Group (SIG) is working on 5 Gb/s and 10 Gb/s IrPHY and IrLAP standard extensions, aiming to achieve link speeds currently unattainable by competing RF technologies. We must also consider the future evolution of 5G technology offering possibly a cellular optical wireless PHY option. Such a cellular architecture could borrow from the IrDA protocol stack experience, especially for the Device-to-Device mode as well as from VLC technology developments. It is likely therefore that new versions of standards in such cases would emerge. An interesting evolution of IrDA would be the development of 'zero delay' links. Such links would be important in very fast connection and payment situations [18] where large crowds are involved using low cost short-range links, in large volume applications (i.e. ticket payment, shopping) within mobile devices. The SIG is also working on alleviating some of the cumbersome features of Giga-IR, such as docking. To this end, future standards are being designed to support longer L (>1 m), hand-held operation of devices and broader coverage by implementing optical beaming, spotting and Multiple Input Multiple Output techniques. Finally, future IrDA links will be more efficient networking-wise and will support full-duplex mode operation, where devices are able to transmit and receive data simultaneously.

Currently, a Special-Interest-Group (SIG) is working on 5 Gb/s and 10 Gb/s IrPHY and IrLAP standard extensions, aiming to achieve link speeds currently unattainable by competing RF technologies. The SIG is also working on alleviating some of the cumbersome features of Giga-IR, such as docking.

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Biographies

A. C. Boucouvalas (acb@uop.gr) is a professor of communication networks and applications at the University of Peloponnese in Tripoli, Greece. He has been involved with research in various aspects of fiber-optic communications, wireless communications and multimedia, and has accumulated 25 years of experience working in well-known academic and industrial research centers. He was a professor of multimedia communications and director of the Microelectronics and Multimedia Communications Research Centre of Bournemouth University, in the UK. He has worked as a project manager for Hewlett-Packard Laboratories and as group leader and divisional chief scientist at GEC Hirst Research Laboratories, both in the UK. His work, supported by numerous research grants and contracts from European Union and industrial organizations, has been published in over 300 publications. He is a Fellow of the Institution of Electrical Engineers (UK) and of the Institute of Electrical and Electronic Engineers (USA). He has also been a Member of the Architectures Council of the Infrared Data Association (IrDA) since 2003.

Periklis Chatzimisios (peris@it.teithe.gr) is an associate professor with the Computing Systems, Security and Networks (CSSN) Research Lab at Alexander TEI of Thessaloniki, Greece. He is involved in several standardization activities and serves on the Standards Development Board of the IEEE Communication Society (ComSoc) (from 2010 until today). He is acting as the director for the E-Letter of the IEEE Technical Committee on Multimedia Communications (MMTC). He is the author/editor of eight books and more than 80 peer-reviewed papers on the topics of performance evaluation and standardization activities of mobile/wireless communications and networked multimedia. His published research work has received more than 1300 citations.

Zabih Ghassemlooy (z.ghassemlooy@northumbria.ac.uk) is a professor in optical communications on the Faculty of Engineering and Environment, Northumbria University at Newcastle, UK. He heads the Northumbria Communications Research Laboratories within the Faculty. He is the editor-in-chief of the International Journal of Optics and Applications and The Mediterranean Journal of Electronics and Communications. His research interests are on photonics switching, optical wireless and wired communications, free-space optics, visible light communications, radio over FSO/fiber, and mobile communications. He has received a number of research grants from a number of UK Research Councils, the UK government, the European Union and industry and has supervised almost 50 Ph.D. students and published over 520 papers (in 200 journals) and presented several keynote and invited talks. He is a co-author of a CRC book, Optical Wireless Communications — Systems and Channel Modelling with Matlab (2012). From 2004–2006 he was the IEEE UK/IR Communications Chapter secretary and its vice-chairman (2004–2008) and chairman (2008–2011), and chairman of the IET Northumbria Network (Oct 2011–present).

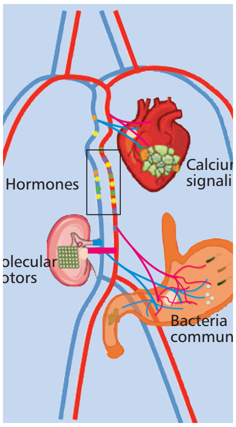
Murat Uysal (murat.uysal@ozyegin.edu.tr) is a professor at Ozyegin University, Istanbul, Turkey, where he leads the Communication Theory and Technologies (CT&T) Research Group. Before joining Ozyegin University, he was a tenured associate professor at the University of Waterloo, Canada, where he still holds an adjunct faculty position. His research interests are in the broad areas of communications theory and signal processing with particular emphasis on the physical layer aspects of wireless communication systems in radio-, acoustic-, and optical-frequency bands. He has authored more than 200 journal and conference papers on these topics and received more than 3400 citations.

Konstantinos Yiannopoulos (kyianno@uop.gr) is a lecturer at the University of Peloponnese, Greece. He was a member of the research teams of the Photonics Communications Research Laboratory at the National Technical University of Athens, Greece (2000–2004), and the Computer Networks Laboratory at the University of Patras, Greece (2005–2010). At present, his research focuses on optical wireless networks (system architectures and protocol analysis). He has more than 40 published papers in international journals and conferences. His research work was granted the IEEE/LEOS Graduate Student Fellowship Program Award for 2004 and he has received more than 600 independent citations.

The Internet of Bio-NanoThings

The IoBNT stands as a paradigm-shifting concept for communication and network engineering, where novel challenges are faced to develop efficient and safe techniques for the exchange of information, interaction, and networking within the biochemical domain, while enabling an interface to the electrical domain of the Internet.

I. F. Akyildiz, M. Pierobon, S. Balasubramaniam, and Y. Koucheryavy



I. F. Akyildiz is with the Georgia Institute of Technology and Tampere University of Technology.

S. Balasubramaniam and Y. Koucheryavy are with Tampere University of Technology.

M. Pierobon is with the University of Nebraska.

Abstract

The Internet of Things (IoT) has become an important research topic in the last decade, where things refer to interconnected machines and objects with embedded computing capabilities employed to extend the Internet to many application domains. While research and development continue for general IoT devices, there are many application domains where very tiny, concealable, and non-intrusive Things are needed. The properties of recently studied nanomaterials, such as graphene, have inspired the concept of Internet of NanoThings (IoNT), based on the interconnection of nanoscale devices. Despite being an enabler for many applications, the artificial nature of IoNT devices can be detrimental where the deployment of NanoThings could result in unwanted effects on health or pollution. The novel paradigm of the Internet of Bio-Nano Things (IoBNT) is introduced in this paper by stemming from synthetic biology and nanotechnology tools that allow the engineering of biological embedded computing devices. Based on biological cells, and their functionalities in the biochemical domain, Bio-NanoThings promise to enable applications such as intra-body sensing and actuation networks, and environmental control of toxic agents and pollution. The IoBNT stands as a paradigm-shifting concept for communication and network engineering, where novel challenges are faced to develop efficient and safe techniques for the exchange of information, interaction, and networking within the biochemical domain, while enabling an interface to the electrical domain of the Internet.

Introduction

The Internet of Things (IoT) defines a cyber physical paradigm, where all types of real-world physical elements (sensors, actuators, personal electronic devices, or home appliances, among others) are connected, and are able to autonomously interact with each other. This new form of seamless connectivity is the enabler for many applications such as machine to machine communication, real time monitoring of industrial processes, smart cities, smart grids for energy management, intelligent transportation, environmental monitoring, infrastructure management, medical and healthcare systems, building and home automation, and large scale deployments. The Internet of Things became a focus for research and development in the last 15 years. A large amount of investments for Internet of Things was and is still being made by government agencies and industry worldwide.

Recently, the concept of IoT has been revised in light of novel research advances made in the field of nanotechnology and communication engineering, which enable the development of networks of embedded computing devices, based on nanomaterials such as graphene or metamaterials, having scales ranging from one to a few hundred nanometers, called nanothings. The Internet of NanoThings (IoNT), introduced for the first time in [1], is proposed as the basis of numerous future applications, such as in the military, healthcare, and security fields, where the nanothings, thanks to their limited size, can be easily concealed, implanted, and scattered in the environment, where they can cooperatively perform sensing, actuation, processing, and networking.

While nanothings can push the engineering of devices and systems to unprecedented environments and scales, similarly to other devices, they have an artificial nature, since they are based on synthesized materials, electronic circuits, and interact through electromagnetic (EM) communications [1]. These characteristics can be detrimental for some application environments, such as inside the body or in natural ecosystems, where the deployment of nanothings and their EM radiation could result in unwanted effects on health or pollution.

A novel research direction in the engineering of nanoscale devices and systems is being pursued in the field of biology, by combining nanotechnology with tools from synthetic biology to control, reuse, modify, and reengineer biological cells [2]. By stemming from an analogy between a biological cell and a typical IoT embedded computing device, a cell can be effectively utilized as a substrate to realize a so-called Bio-NanoThing, through the control, reuse, and reengineering of biological cells' functionalities, such as sensing, actuation, processing, and communication. Since cells are based on biological molecules and biochemical reactions, rather than electronics, the concept of Internet of Bio-NanoThing (IoBNT), introduced in this article, is expected to be paradigm shifting for many related disciplines, such as communication and network engineering, which is the focus of this article. The execution of DNA-based instructions, the biochemical processing of data, the transformation of chemical energy, and the exchange of information through the transmission and reception of molecules, termed molecular communication (MC) [3], are at the basis of a plethora of applications that will be enabled by the IoBNT, such as:
- Intra-body sensing and actuation, where Bio-NanoThings inside the human body would collaboratively collect health-related information, transmit it to an external healthcare provider through the Internet, and execute commands from the same provider such as synthesis and release of drugs.
- Intra-body connectivity control, where Bio-NanoThings would repair or prevent fail-

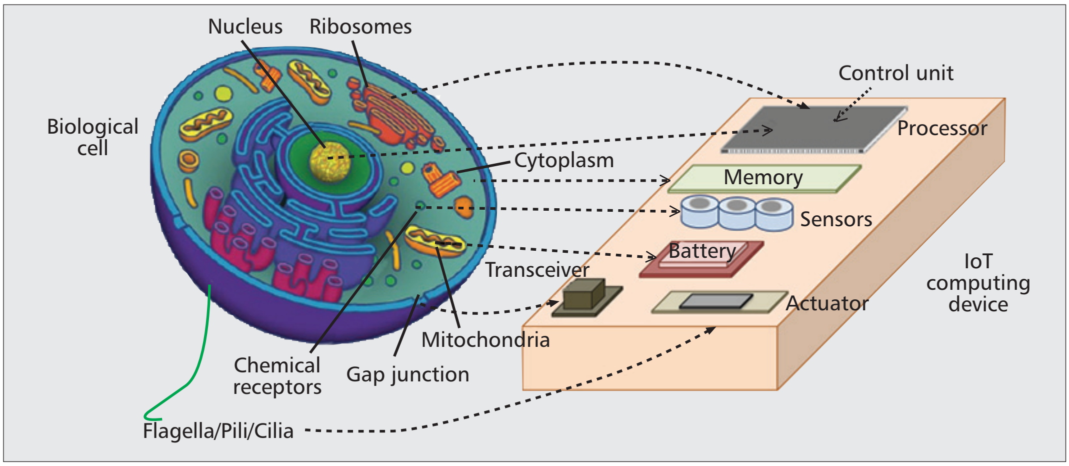


Figure 1. Elements of a biological cell and components of a typical IoT device.

Stemming from biological cells, and enabled by synthetic biology and nanotechnology, Bio-NanoThings are expected to perform tasks and functionalities typical of the embedded computing devices in the IoT, such as sensing, processing, actuation, and interaction with each other.

ures in the communications between our internal organs, such as those based on the endocrine and the nervous systems, which are at the basis of many diseases.

- Environmental control and cleaning, where Bio-NanoThings deployed in the environment, such as a natural ecosystem, would check for toxic and pollutant agents, and collaboratively transform these agents through bioremediation, e.g. bacteria employed to clean oil spills.

This article is organized as follows. First, Bio-Nano-Things are defined in light of the tools available today from synthetic biology and nanotechnology. Second, the application of communication engineering to design Bio-NanoThings telecommuni- cations is detailed, while the challenges to engineer Bio-NanoThings networks and Internet connections are discussed. Third, we describe further research challenges for the realization of IoBNT. Finally, we conclude the article.

Bio-NanoThings

Within the scope of the IoBNT, Bio-NanoThings are defined as uniquely identifiable basic structural and functional units that operate and interact within the biological environment. Stemming from biological cells, and enabled by synthetic biology and nanotechnology, Bio-NanoThings are expected to perform tasks and functionalities typical of the embedded computing devices in the IoT, such as sensing, processing, actuation, and interaction with each other.

Biological Cells as the Substrates of Bio-NanoThings

A biological cell is the basic unit of life, consisting of a membrane that encloses a mixture of highly specialized molecules, with defined chemical composition and function, which may also be organized into functional structures [4]. A mapping between the components of a typical IoT embedded computing device, and the elements of a cell, becomes apparent if we compare electrons' propagation in semiconductors to functionally similar, although much more complex, biochemical reactions. In this context, as illustrated in Fig. 1, some examples are as follows.

The **control unit**, which contains the embedded software of the device, would correspond to the genetic instructions densely packed into the cells' DNA molecules, which encode protein structures, the cell's "data units," and regulatory sequences, similar to the software conditional expressions. The **memory unit**, which contains the values of the embedded system data, would correspond to the chemical content of the cytoplasm, i.e. the interior of the cell, comprised of molecules synthesized by the cell as a result of DNA instructions, and other molecules or structures, e.g. vesicles, exchanged with the external environment. The **processing unit**, which executes the software instructions and manages memory and peripherals, would correspond to the molecular machinery that, from the DNA molecules, through the so-called transcription and translation, generates protein molecules with instruction-dependent types and concentrations. The **power unit**, which supplies the energy to maintain the electrical currents in the embedded system's circuits, would correspond to the reservoir in the cell of the Adenosine TriPhosphate (ATP) molecule, which is synthesized by the cell from energy supplied from the external environment in various forms, and provides the energy necessary for the cell's biochemical reactions to take place. The **transceivers**, which allow the embedded systems to exchange information, would correspond to the specific chains of chemical reactions, i.e. signaling pathways, through which cells exchange information-bearing molecules. **Sensing and actuation**, which allow embedded systems to acquire data and interact with the environment, would correspond to the capability of a cell to chemically recognize external molecules or physical stimuli, e.g. light or mechanical stress, and to change the chemical characteristics of the environment or mechanically interact through moving elements, such as flagella, pili, or cilia.

Enabling Technologies and Challenges

The discipline of synthetic biology is providing tools to control, reuse, modify, and reengineer the cells' structure and function, and it is expected to enable engineers to effectively use the bio-

At the basis of the IoBNT concept there is the need for Bio-NanoThings to communicate with each other, and interact on the basis of the exchanged information. Since Bio-NanoThings stem from the engineering of biological cells, the natural environment is the main inspiration for studying communication techniques for IoBNT.

logical cells as programmable substrates to realize Bio-NanoThings as biological embedded computing devices [2]. DNA sequencing and synthesis technologies, enabling the reading and writing of genetic code information in the DNA molecules of biological cells, are giving engineers an increasingly open access to the set of structural and functional instructions at the basis of life. In particular, the engineering of synthetic biological circuits [5] through genetic code manipulation has enabled the programming of specifically designed functions to be executed by cells. A biological circuit is a set of genes that encode proteins and regulatory sequences, which link together the protein synthesis by mechanisms of mutual activation and repression. The functions today successfully developed via biological circuits range from AND and OR logic gates, to various types of tunable oscillators, toggle switches, and counters. The development of databases with characterized standard biological circuit parts with known functions and behaviors, e.g. BioBricks, and tools to combine them into more complex designs [6], are pushing synthetic biology to a future development similar to that experienced by integrated electrical circuit design in electronics. As a consequence, engineers will be soon able to gain full access to the functionalities of the aforementioned cells' elements, and reuse cells and their features, without requiring an in-depth knowledge of biotechnology.

One of the latest frontiers in synthetic biology is the development of artificial cells, enabled, among others, by tools from nanotechnology. Artificial cells have minimal functionalities and structural components compared to natural cells, and are assembled bottom-up by encapsulating the necessary elements into either biological or fully synthetic enclosing membranes [7]. Artificial cells can therefore contain genetic information, the related molecular machineries for their transcription, translation, and replication, and all the required specialized molecules and structures. Artificial cells are expected to enable a more agile and controllable use of synthetic biological circuits by removing all the additional complexity of natural cells that are not necessary to perform the designed functions. Although still in its infancy, this technology has been successfully applied, e.g. for drug delivery, gene therapy, and artificial blood cell production, and it is expected to deliver ideal substrates for synthetic biology with a more predictable behavior.

Although very promising, the aforementioned technologies have to provide solutions to major research challenges in biotechnology and engineering before being considered as reliable tools for the realization of Bio-NanoThings. Focusing on the engineering design viewpoint, a major challenge is to develop reliable mathematical and physical models, and computer simulation environments, able to capture the peculiar characteristics of the biological processes underlying engineered cells, such as intrinsic non-linear phenomena and processes with noisy outcomes. Moreover, engineered cells, similar to natural cells, reproduce and mutate, i.e. tend to randomly change parts of their genetic programs, and selectively evolve, i.e. tend to maintain the best mutations for their survival while reproducing, adding possible problems but also new degrees of freedom to the biological device designer. Another challenge that needs to be considered is related to bioethics and security, since autonomously evolving engineered organisms could pose a threat to the natural ecosystems, and even become new pathogens. The recent development of "kill" switches into biological circuits, able to stop cell reproduction or trigger cell destruction upon an external command, is only partially addressing these problems.

Bio-NanoThings Communications

At the basis of the IoBNT concept there is the need for Bio-NanoThings to communicate with each other, and interact on the basis of the exchanged information. Since Bio-NanoThings stem from the engineering of biological cells, as detailed above, the natural environment is the main inspiration for studying communication techniques for IoBNT.

Molecular Communication in Nature

In nature, the exchange of information between cells is based on the synthesis, transformation, emission, propagation, and reception of molecules through biochemical and physical processes. This information exchange, recently classified in telecommunications engineering as MC [1], enables cells' interactions and coordination of uni-cellular and multi-cellular organisms, populations, and multi-species consortia, and participates in most of the major cellular functionalities such as cell growth and proliferation.

MC in cells is based on the aforementioned signaling pathways, which are chains of chemical reactions that process information signals modulated into chemical characteristics, such as molecule concentration, type, and energy state, and propagate them from a source, or transmitter, to a destination or receiver [4]. Cell signaling pathways can be classified on the basis of the distance between source and destination into intracrine (source and destination are within the same cell), juxtracrine (source and destination are cells in contact with each other), paracrine (source and destination are in the vicinity of each other, but not in contact), or endocrine (source and destination are distant from each other).

An example of **intracrine** communication is given by the intracellular transport of molecules or molecule structures operated by cytoskeletal molecular motors. Molecular motors are intracellular specialized proteins able to convert the aforementioned ATP molecules into mechanical energy. The cytoskeletal molecular motors are able to bind to a particular cargo, such as vesicles enclosing sets of molecules, or whole cell organelles, attach to the microfilament structures that compose the cell's skeleton, and crawl along them transporting the cargo from the nucleus to the membrane of the cell and vice versa.

The exchange of molecules, such as calcium ions Ca^{2+}, between two cells connected by communicating junctions in their membrane, is an example of juxtacrine communication. Several examples in nature, such as the signaling during a cardiac contraction happening between muscular cells, or myocytes, show how a small quantity of molecules can flow by diffusion between neighboring cells, and be responsible for synchronizing coordinated actions.

Bacteria show several means of communication in nature, such as the **paracrine** communication underlying the emission of signaling molecules called autoinducers by members of a population. In this process, called bacterial quorum sensing, the autoinducers diffuse within the intercellular space and, upon reception, allow the bacteria to estimate the population density, and have a correlated response, such as the production of specific types of proteins. Bacteria can also exchange specific DNA molecules, i.e. plasmids, via direct contact, through a process called conjugation, and carry the plasmids to other distant bacteria within the intercellular space by swimming according to chemical trails, with a process called chemotaxis.

In multicellular organisms, an example of **endocrine** communication is realized through signaling molecules called hormones that are emitted from cells composing glands, propagate through the circulatory system, and are received by the cells of distant organs, where they elicit specific responses, such as increased cell growth and reproduction.

Challenges in Engineering Molecular Communication for IoBNT

Within the IoBNT, Bio-NanoThings are expected to interact with each other by exchanging various types of information, e.g. synchronization signals, values of sensed chemical/physical parameters, results of logical operations, and sets of instructions and commands. The engineering of the communication techniques to support these interactions in the biological environment has to stem from solutions found in nature, such as those described above. One of the major challenges is to understand how these natural solutions can be controlled, modified, or reengineered for the transmission of information that can be different from the natural. By stemming from the aforementioned tools that are being developed in synthetic biology and nanotechnology, engineers have recently started to analyze several different possibilities to realize MC systems, either by genetically reprogramming cells' behaviors within their natural communications [8], or by developing totally new artificial communication systems by assembling natural biological components [9].

Examples of MC systems that have been envisioned so far can be classified on the basis of the distance range that they are expected to cover from transmission to reception. For example, the control of juxtracrine communications through the genetic programming of biological cells can enable the engineering of networks where Bio-NanoThings are in contact with each other, e.g. when organized in a tissue or biofilm [10]. This MC technique, usually referred to the aforementioned Ca^{2+} exchange, shown in Fig. 2a, covers distances proportional to the thickness of cell membranes, and it can be considered as **very short range** (tens to hundreds of nm) MC. The aforementioned cytoskeletal molecular motors can be considered for the realization of MC in the **short range** (nm-μm) [11], as shown in Fig. 2b, to cover intracrine Bio-NanoThings communications. Communication engineers have also combined models of the bacterial conjugation and chemotaxis processes described above to theoretically study a possible artificial MC system, which can be considered, according to the known chemotaxis characteristics, as covering the **medium range** (μm-mm) [9]. In particular, the information is represented into DNA molecules, i.e. plasmids, which are loaded at the transmitter location into bacteria and extracted from the same bacteria at the receiver through a conjugation process. These bacteria are able to swim by chemotaxis toward the receiver, by following the receiver's release of specific molecules, i.e. chemoattractants, as shown in Fig. 2c. An example of **long range** (mm-m) MC system has been envisioned by stemming from the hormonal communication within the human endocrine system [12], as shown in Fig. 2d.

From the telecommunications engineering perspective, one of the main challenges stands in mapping MC into the classical elements of an engineered communication system, and in the use of tools from systems and information theory with the final goals of modeling and analyzing the main telecommunication characteristics and performance, such as range, delay (latency), capacity, throughput, and bit error rate [13]. The knowledge of these characteristics will then allow for the comparison and classification of possible different techniques to realize MC for different IoBNT application scenarios, and the optimization of their design and realization.

Examples of the aforementioned mapping are shown in Fig. 3, where the main processes involved in each MC system described above are divided into communication elements as follows. Encoding and decoding are related to how the information to be transmitted is represented into one or more molecule characteristics, such as sets of particular molecule types and numbers (molecular motors and hormonal communication), composition of biological macromolecules, such as DNA plasmids (bacteria conjugation-chemotaxis), or released molecule concentration (Ca^{2+} exchange). Transmission and reception involve the chemical and physical processes to initiate the propagation of molecules, e.g. the encapsulation into vesicles for molecular motor transportation, the release of molecules in a fluid, such as the blood stream, or through a junction between two adjacent cells, or the release of bacteria upon the presence of chemoattractant molecules in the environment. Finally, the propagation is concerned with the mobilization of the information-bearing molecules from the location of the transmitter to the receiver, such as through molecular motor crawling along microfilament structures, diffusion through membrane junctions, diffusion and advection in the blood stream, and bacterial chemotaxis toward the chemoattractant source (receiver).

While a great amount of the literature within the MC field has been devoted to the modeling and analysis of the aforementioned systems through simplifying assumptions, which increase the mathematical tractability of the underlying physical and chemical phenomena, there is still a long way to go for a communication engineer to

Within the IoBNT, Bio-NanoThings are expected to interact with each other by exchanging various types of information, e.g., synchronization signals, values of sensed chemical/physical parameters, results of logical operations, sets of instructions and commands.

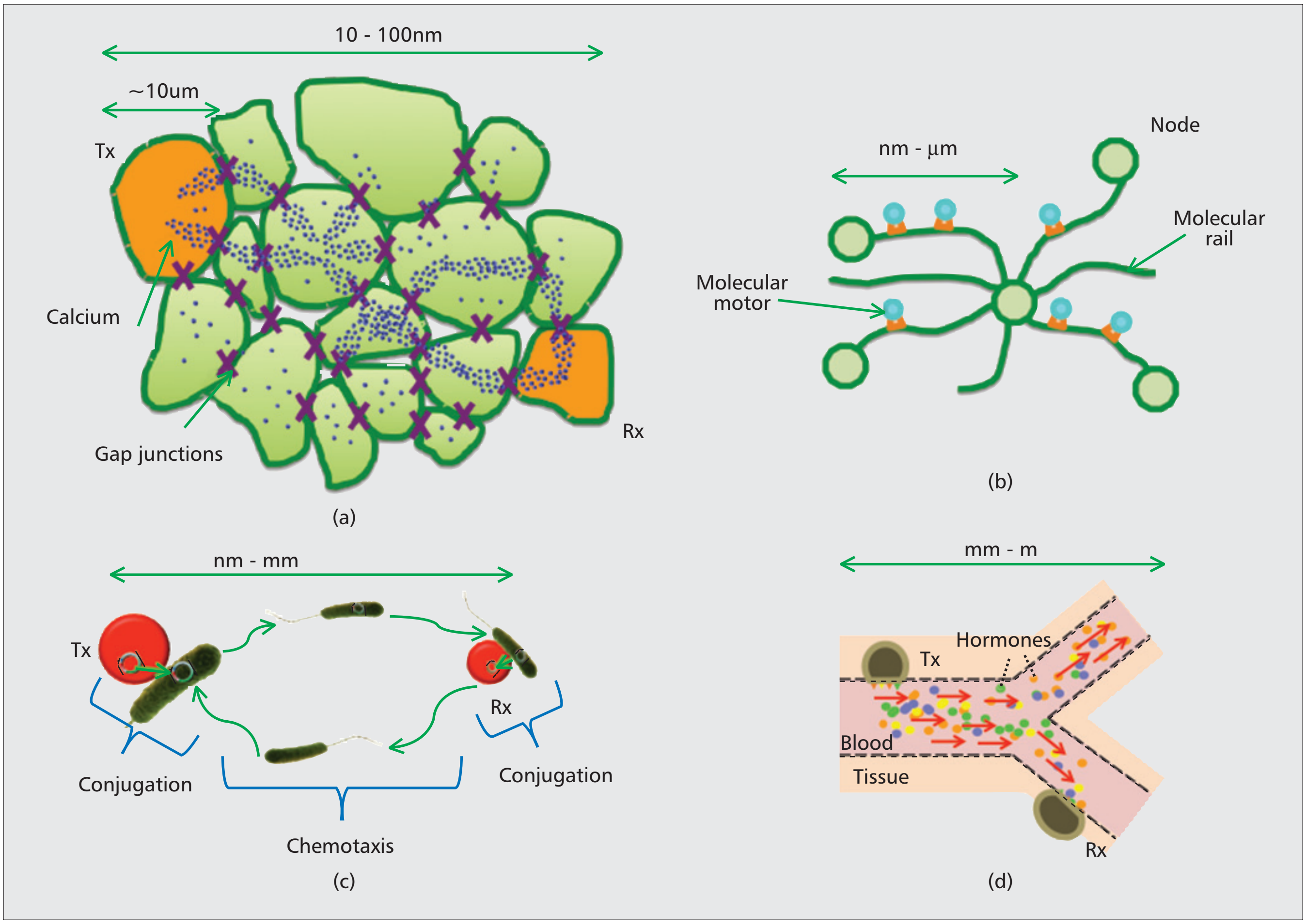


Figure 2. Examples of molecular communication,: a) Ca^{2+} communication; b) molecular motors; c) bacteria conjugation-chemotaxis; and d) hormonal communication.

fully understand how to design realistic MC systems for IoBNT communications. The main challenges are given by the conversion of these simplified models to more realistic scenarios. For example, the free diffusion models considered so far in MC engineering for the propagation and reaction of molecules in the intracellular environment, e.g. in Ca^{2+} communication, have to be revised to include more realistic phenomena, such as the effect of high concentrations of macromolecules, e.g. proteins, called macromolecular crowding. Another example is given by the endocrine propagation, so far considered for a small subset of well-defined blood vessels, where models should take into account not only the whole average physiology of the human cardiovascular system, but also that the specific characteristics of each individual can result in very different propagation dynamics. Also, the models of bacteria chemotaxis used so far in MC engineering are only based on the behavior and properties of single bacteria and in-vitro environments, where in fact more realistic environments, such as within the human body, and the fact that bacteria can replicate and proliferate dynamically and interact within multi-species consortia, should be taken into account. Other challenges for the development of reliable analytical tools for MC engineering are given by the non-linear nature of many biochemical phenomena, and the presence of very different noise sources, such as genetic mutations, compared to classical systems.

Bio-NanoThing Networks and the Internet

Within the IoBNT, Bio-NanoThings are expected to not only communicate with each other, but also interact into networks, which will ultimately interface with the Internet. To this end, the definition of network architectures and protocols on top of the aforementioned MC systems is an essential step for IoBNT development. A further challenge for the IoBNT is the interconnection of heterogeneous networks, i.e. composed of different types of Bio-NanoThings and based on different MC systems. Finally, the realization of interfaces between the electrical domain of the Internet and the biochemical domain of the IoBNT networks will be the ultimate frontier to create a seamless interconnection between today's cyber-world and the biological environment. In Fig. 4 we show a possible scenario where a complete IoBNT, composed of several networks based on different MC systems, is deployed inside the human body, and interfaces through a personal electrical device connected to the Internet to deliver intra-body status parameters (and receive commands and instructions) to (from) a healthcare provider.

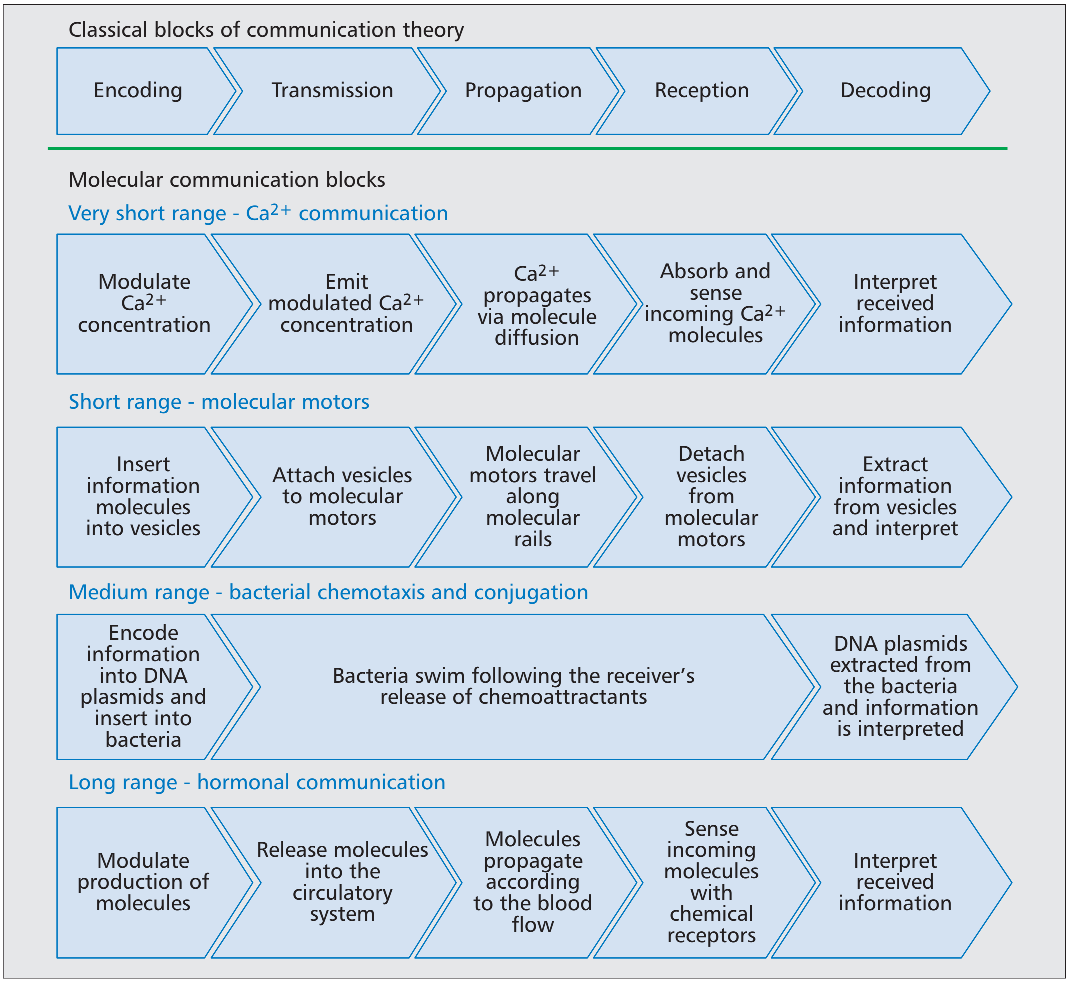


Figure 3. Mapping of classical communication theory to bio-nanothings communication, including: a) very short range — Ca^{2+} signaling: a) short range — molecular motors; c) medium range — bacteria chemotaxis and conjugation; and d) long range — hormonal communication.

> Within the IoBNT, Bio-NanoThings are expected to not only communicate with each other, but also interact into networks, which will ultimately interface with the Internet. To this end, the definition of network architectures and protocols on top of MC systems is an essential step for IoBNT development.

Challenges for Realizing Bio-NanoThing Networks

While the engineering of computer networks is a well-established field, where several different solutions have been provided for many different technologies and application scenarios, the design of networks within the biological environment, and based on the MC paradigm as the physical medium, poses new challenges to the networking community. For example, molecular information generally does not follow predictable and definite propagation directions, as otherwise done by electromagnetic signals in classical communications [13]. The diffusion of molecules, the bacterial chemotaxis, and the filaments supporting molecular motors, tend to cover random patterns between source and destination. This and other peculiarities, such as the non-linear nature of many biochemical phenomena, make it particularly challenging to utilize classical techniques for regulating Bio-NanoThings access to shared media, such as fluids, addressing Bio-NanoThings, and designing information routing mechanisms, which are important basic aspects of computer networks.

As done for the MC systems, one possible solution will be to model, analyze, and reuse the mechanisms of interactions of multiple cells in nature, such as in bacteria populations [14] and multispecies consortia, or within the tissues of multi-cellular organisms, to relay the IoBNT information.

In this direction, a solution for the interconnection of heterogeneous Bio-NanoThing networks, based on different MC systems, might as well come from the natural way our body manages and fuses several types of information to maintain a stable, healthy status, or homeostasis [4]. These intra-body processes allow heterogeneous communications to occur at various scales, translating from intracrine communications within a cell, to juxtacrine communication within tissues, to endocrine communications between different organs. For example, the cells of the pituitary gland perform this type of translation by releasing hormones to body organs to control several processes, such as growth, blood pressure, temperature, and sleeping patterns, as a result of the reception of other hormones from the cells of the adjacent hypothalamic tissue. Biological circuits based on these processes could effectively provide a set of genetic instructions that mimic the classical gateways between different subnets on the Internet. Figure 5a illustrates a general example of an artificial cell that translates the information encoded into molecules emitted

A bio-cyber interface is here defined as the set of processes necessary to translate information from the biochemical domain of Bio-NanoThing networks to the Internet cyber-domain, which is based on electrical circuits and electromagnetic communications, and vice versa.

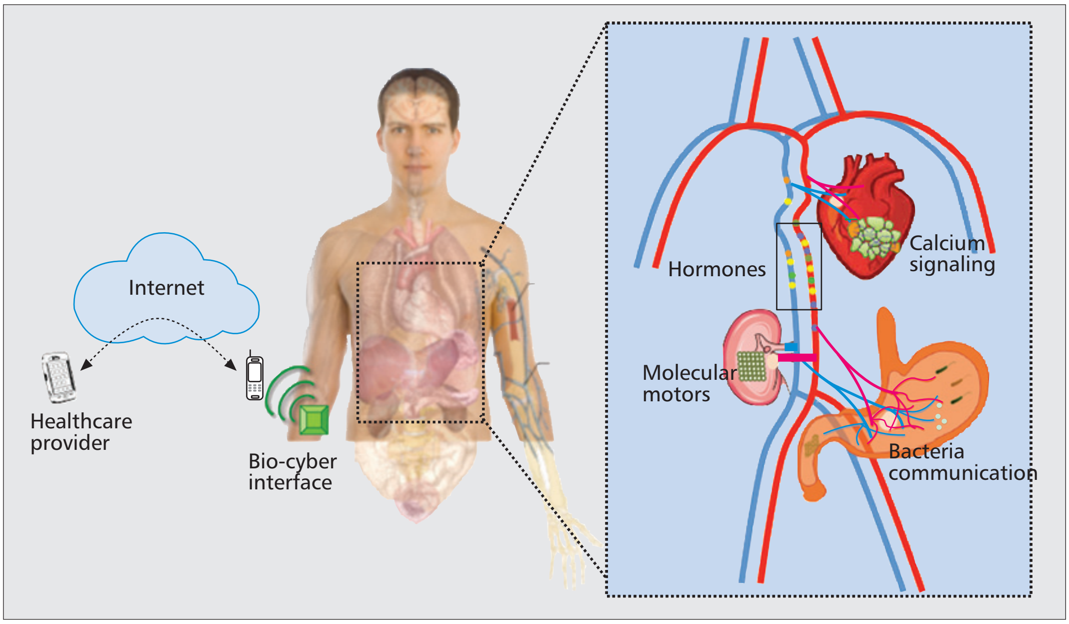


Figure 4. Network architecture for the Internet of Bio-NanoThings for Intra-body applications.

from engineered bacteria into hormones that can be secreted into the circulatory system. In this design, receptors would intercept the incoming molecules that, through a cascade of chemical reactions, would activate a biological circuit, which in turn would synthesize proteins able to trigger the necessary chemical reactions to produce the hormones.

Challenges for Bio-Cyber Interfaces

A bio-cyber interface is here defined as the set of processes necessary to translate information from the biochemical domain of Bio-NanoThing networks to the Internet cyber-domain, which is based on electrical circuits and electromagnetic communications, and *vice versa*.

One of the main challenges for the realization of these interfaces stands in the engineering of chemical and physical processes able to accurately read the molecule characteristics where information is encoded, and translate them into the modulation of electromagnetic parameters. A possible solution in this direction might come from novel chemical and biological sensors enabled by nanotechnology, which promise unprecedented sensing capabilities [15]. These sensors are in general composed of materials characterized by electrical or electromagnetic properties that can be altered by the presence of determinate molecules or molecule complexes, such as biological receptors bound to molecules, and accordingly modulate the current in an electrical circuit. The major problems for using this sensing technology for IoBNT applications stand in their currently high latency, low selectivity, lack of standardized response, and, more importantly, unknown biocompatibility, which is considered next.

Biocompatibility, intended here as the property of an engineered system of limiting its action on the biological environment exclusively to its intended function, without any unwanted alteration of biological parameters, is another challenge for the deployment of bio-cyber interfaces, especially for intra-body IoBNT applications as shown in Fig. 4. Given the limited size of the aforementioned nanosensors, and current promising research results in electromagnetic (EM) nanocommunications, we envision the possibility to develop bio-cyber interfaces by encapsulating biological nanosensors and EM nanocommunication units within the aforementioned artificial cells, as shown in Fig. 5b. In this design, the biological nanosensor would be responsible for interfacing chemical and electrical domains, the EM nano-communication unit would wirelessly communicate with electrical devices outside of the biological environment, and the artificial cell would assure biocompatibility. However, a challenge lies in the ability to produce sufficient power for the wireless transmitter to emit electromagnetic waves that can propagate through the artificial cell membrane. At the same time, approaches are also required to harvest energy for the transmitter unit from within the cell. Another alternative is to push the electrical/EM domain at the physical interface between the biological environment and the external world, such as the skin for intra-body IoBNT applications. In this direction, electronic tattoos, similar to those based on radio frequency identification (RFID) technology, which allow users to authenticate devices within close range, could incorporate a bio-cyber interface able to sense bio- chemical information from cells on the epidermis, sweat glands, or nervous terminations, and communicate it wirelessly to nearby external electronic devices.

Further Challenges

We now briefly mention some further challenges to be faced for the IoBNT development.

The IoBNT enabling technologies discussed in this article could pose serious **security threats** if handled with malicious intent. A new type of

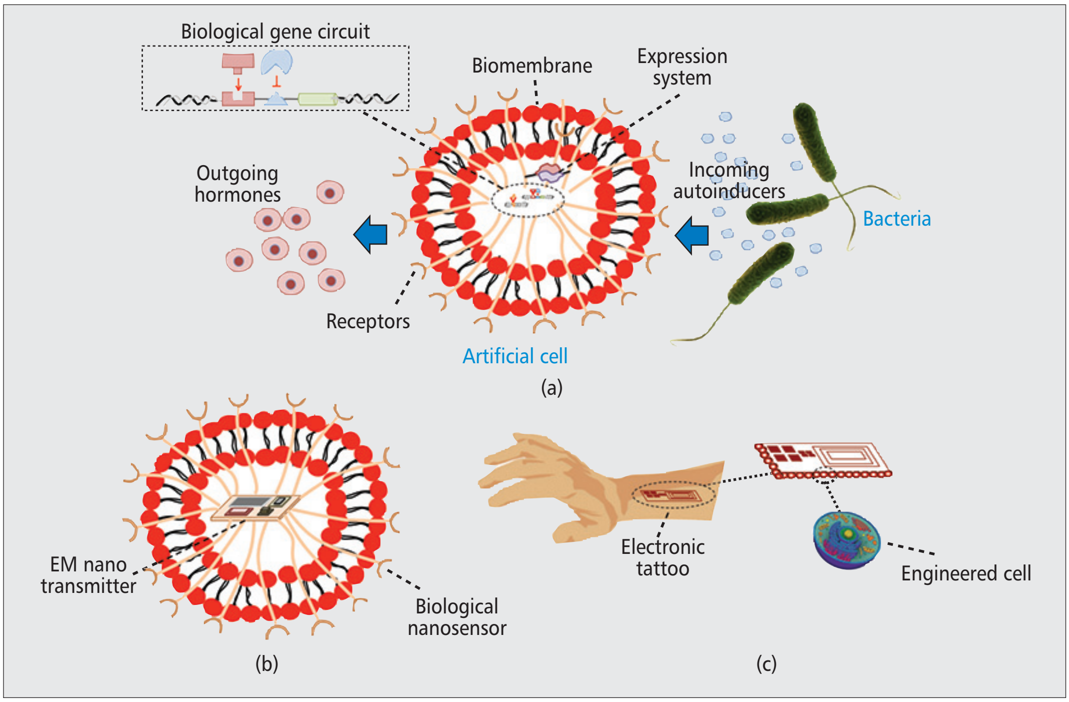


Figure 5. Application of artificial cells for the bio-nanothings networking: a) artificial cells for translating multiple molecule types; b) EM-nano transmitter and nanosensor embedded into an artificial cell for bio-cyber interface; and c) electronic tattoo for bio-cyber interface.

Research within the IoBNT should necessarily address these problems by combining the security assurance methods applied to today' s electrical networks with security solutions developed through evolution by nature, such as the human immune system.

terrorism, which we term as bio-cyber terrorism, could effectively take advantage of the numerous possibilities offered by the IoBNT to control and interact with the biological environment. For example, Bio-NanoThings could be used to access the human body, and either steal personal health-related information, or even create new diseases. Moreover, new types of viruses could be created to hack into already deployed IoBNTs. Research within the IoBNT field should necessarily address these problems by combining the security assurance methods applied to today's computer networks with security solutions developed through evolution by nature, such as the human immune system.

The realization of **localization and tracking** techniques within the IoBNT, in a similar way as realized in wireless sensor networks (WSNs), could enable important applications related, for example, to the monitoring of disease locations in the body or identification of the location and distribution of toxic agents in an environment. One solution could come from the engineering of chemotaxis in Bio-NanoThings, based on the aforementioned capability of bacteria to localize and track sources of particular types of molecules, which could be, for example, biomarkers released by cancerous or infected cells.

In line with the vision of "all-Things connected," an ultimate goal is to **interconnect the paradigms of IoBNT and IoNT to the IoT**. A challenge of incorporating nanoscale devices is the large quantity of information that will emerge, taking the challenges of managing "Big-Data" to a new level. Besides the increase in the quantity of data, new services will need to be designed to semantically map between different types of data that IoBNT and IoNT will feed to the IoT. New service discovery solutions will also be required to search deep into the biological environments and interact with engineered biological entities to actuate or collect information.

Conclusion

While the Internet of Things (IoT) is enabling the pervasive connectivity of real-world physical elements among themselves and to the Internet, the Internet of NanoThings proposes to push the limits of this concept to nanotechnology-enabled nanoscale devices, which can be easily concealed, implanted, and scattered in the environment. In this article we introduced the further concept of Internet of Bio-NanoThings, where synthetic biology and nanotechnology are combined to develop Things based on the control, reuse, modification, and reengineering of biological cells. This article outlined the challenges that will be faced to realize these Things and, more importantly, to enable their communication and networking, with paradigm-shifting techniques for the fields of communication and network engineering. We believe that the IoBNT research field, while still in its infancy, will result in a game-changer technology for the society of tomorrow.

Acknowledgement

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Biographies

I. F. Akyildiz (ian.akyildiz@tut.fi) is Ken Byers Chair Professor in Telecommunications with the School of Electrical and Computer Engineering, Georgia Institute of Technology, Atlanta, the Director of the Broadband Wireless Networking (BWN) Laboratory and the Chair of the Telecommunication Group at Georgia Tech. Since 2013 he has been a FiDiPro Professor (Finland Distinguished Professor Program (FiDiPro) supported by the Academy of Finland) in the Department of Electronics and Communications Engineering at Tampere University of Technology, Finland. He is an IEEE Fellow (1996) and an ACM Fellow (1997). He has received numerous awards from IEEE and ACM. His current research interests are in nanonetworks, TeraHertz band communication networks, 5G cellular systems, and wireless sensor networks.

M. Pierobon (maxp@unl.edu) received the Ph.D. degree in electrical and computer engineering from the Georgia Institute of Technology, Atlanta, GA, in 2013, and his M.S. degree in telecommunication engineering from the Politecnico di Milano, Milan, Italy, in 2005. Currently he is an assistant professor with the Department of Computer Science & Engineering at the University of Nebraska-Lincoln. He is an editor of the *IEEE Transactions on Communications*. He is a member of IEEE, ACM, and ACS. His current research interests are in molecular communication theory for nanonetworks, communication engineering applied to intelligent drug delivery systems, and telecommunication engineering applied to cell-to-cell communications.

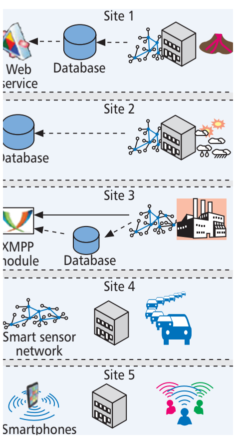
S. Balasubramaniam (sasi.bala@tut.fi) received his bachelor (electrical and electronic engineering) and Ph.D. degrees from the University of Queensland in 1998 and 2005, respectively, and the master's (computer and communication engineering) degree in 1999 from Queensland University of Technology. He is currently a senior research fellow at the Nano Communication Centre, Department of Electronic and Communication Engineering, Tampere University of Technology (TUT), Finland. He was the TPC co-chair for ACM NANOCOM 2014 and IEEE MoNaCom 2011. He is currently an editor for *IEEE Internet of Things* and Elsevier's *Nano Communication Networks*. His current research interests include bio-inspired communication networks and molecular communication.

Y. Koucheryavy (evgeni.kucheryavy@tut.fi) is a full professor and lab director in the Department of Electronics and Communications Engineering at the Tampere University of Technology (TUT), Finland. He received his Ph.D. degree (2004) from the TUT. He is the author of numerous publications in the field of advanced wired and wireless networking and communications. His current research interests include various aspects of heterogeneous wireless communication networks and systems, the Internet of Things and its standardization, and nano communications. He is an associate technical editor of *IEEE Communications Magazine* and an editor of *IEEE Communications Surveys and Tutorials*.

Cloud4Sens: A Cloud-Based Architecture for Sensor Controlling and Monitoring

This article discusses two strategies for managing sensing resources in the cloud and providing them as a service. The first strategy follows the data-centric model, where the cloud offers environmental data to its clients as a service, without any knowledge on how data are measured and processed. The second strategy adopts the device-centric model, which enables the cloud clients to use a virtual sensing infrastructure that they can customize for specific purposes.

Maria Fazio and Antonio Puliafito



Maria Fazio and Antonio Puliafito work at the University of Messina.

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Abstract

This article discusses two strategies for managing sensing resources in the cloud and providing them as a service. The first strategy follows the *data-centric* model, where the cloud offers environmental data to its clients as a service, without any knowledge of how data are measured and processed. The second strategy adopts the *device-centric* model, which enables the cloud clients to use a virtual sensing infrastructure that they can customize for specific purposes. They can access sensed data and customize one or more virtual devices. The article also presents a cloud framework, called Cloud4Sens, that combines both the data-centric and device-centric models, enabling the end-user to choose which type of cloud service he needs. The proposed solution was designed according to the Sensor Web Enablement (SWE) specifications defined by the Open Geospatial Consortium and exploits well known technologies, such as XMPP (Extensible Messaging and Presence Protocol) communications and X509 certificates.

Introduction

This article presents a cloud-based architecture for sensing resources provisioning, called Cloud4Sens. Cloud4Sens exploits cloud technologies to acquire, integrate, and manage heterogeneous sensing and actuation resources made available from different monitoring infrastructures (MIs).

The MI administrative domains (i.e. public/private authorities and companies) share their resources over the cloud to achieve two main goals:

- Improving the quality of the services they already offer.
- Developing new services for cloud end-users.

Indeed, aggregating several resources and cross-relating different types of information bring new opportunities to design innovative applications and services with added value. For example, our ongoing project SIGMA[1] exploits cloud technologies to acquire, integrate, and compute heterogeneous data from several sensor networks for controlling and monitoring both environmental and industrial production systems.

Cloud4Sens offers heterogeneous sensing resources according to two different models:

- The data-centric model [1], to gather sensed data in a specific geographical area and in a given time interval for remote processing.
- The device-centric model [2], to exploit virtual sensors in a geographical area, specifying sensor boards' capabilities and features.

To meet the requirements of both the data-centric and device-centric models, specific abstraction and management techniques have been designed. These techniques provide a uniform description of available resources in the whole system according to the Sensor Web Enablement (SWE) specifications defined by the Open Geospatial Consortium [3]. Thus, Cloud4Sens is able to collect information from different types of sensing devices and sensor networks exploiting different communication technologies (Wi-Fi, ZigBee, UMTS, Ethernet, etc.), and offers such data as services.

The article is organized as follows. We present the data-centric and device-centric models at the basis of the proposed cloud services and provide a reasoned comparison. We discuss related work on available cloud services that provide sensing resources. The Cloud4Sens architecture is described, and we also discuss two use cases specifically developed to test the main features of the data-centric and device-centric oriented services of Cloud4Sens. We summarize our conclusions and provides final remarks.

Type of Services in Cloud4Sens

Cloud4Sens is a cloud-oriented solution to integrate heterogeneous MIs into the cloud according to agreements between the Cloud4Sens provider and MI administrations (e.g. real-time access, time scheduled access, limited access to the physical infrastructure or to the available data, etc.). There are several possible models that lead MI owners to share their data over the cloud:

- The MI provides data as open sensory data through the web. In this case, the Cloud4Sens provider is interested in integrating such type of data in its system and will develop a specific adapter for such type of data provisioning.
- The MI owner is at the same time both resource provider and consumer. In such a scenario, the MI owner exploits Cloud4Sens to extend his physical infrastructure by means of the cloud virtual infrastructure.
- The Cloud4Sens provider and the MI company can make commercial agreements, but the definition of such commercial agreements is out of the scope of this article.

Cloud4Sens offers all the aggregated heterogeneous sensing/actuation and software systems to its clients according to two *as a Service* models. In Cloud4Sens a client could be an end user, a programmer, or a software system. According to the NIST (National Institute of Standards and Technology), which provides the de facto stan-

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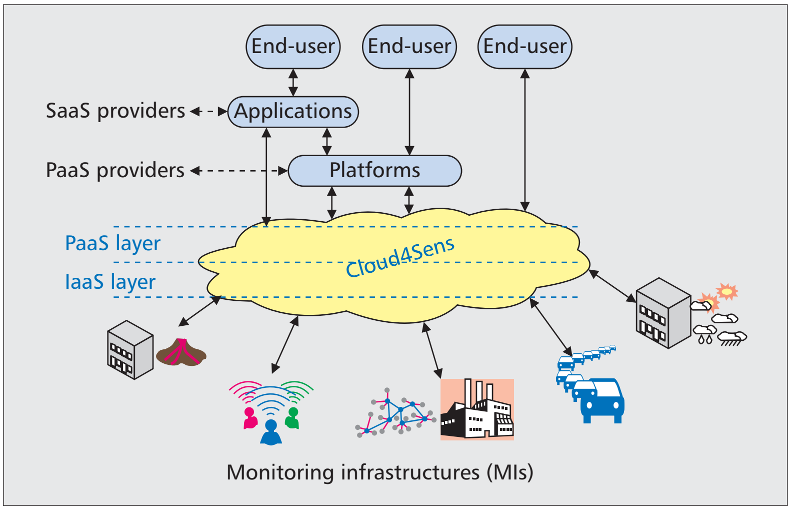

Figure 1. Reference scenario.

dard in the definition of cloud computing, cloud computing services are structured into Software as a Service (SaaS), Platform as a Service (PaaS), and Infrastructure as a Service (IaaS) [4]. Cloud4Sens provides two types of services, at the IaaS and PaaS layers, according to the *data-centric* and *device-centric* models in offering sensing/actuation resources.

The data-centric model gathers physical measurements and environmental information from heterogeneous MI, and organizes them according to a uniform format. To support a data-centric resource provisioning model, Cloud4Sens implements a PaaS, that is a seamless provisioning platform able to abstract and store heterogeneous sensing/actuation data, to provide such data to clients. Thus, clients do not need to have knowledge of the monitoring system's features or technologies, and they access data through high level interfaces regardless of the monitoring system.

The device-centric model offers a sensing/actuation infrastructure to the clients. Such infrastructure aggregates sensing and actuation resources that the applications exploit to deliver services to the end users. Cloud4Sens implements the device-centric model as an IaaS. Cloud principles enable the virtualization of such infrastructure, hiding specific deployment issues from the end user. Sensing and actuation resources are managed to offer a customizable virtual node for environmental monitoring. A virtual device can be equipped with several capabilities, such as sensing, storage, and computation power.

As shown in Fig. 1, Cloud4Sens clients may exploit both the PaaS and the IaaS offered by the Cloud4Sens itself. Clients can be end-users interested in sensing/actuation data and/or virtual sensors. Possible Cloud4Sens clients can be SaaS providers that exploit Cloud4Sens services (or a mashup of services) to offer new applications. Moreover, PaaS providers can offer to their end-users new functionalities at the platform level by using the Cloud4Sens virtual infrastructure.

To better understand the benefits of the proposed cloud solution and their feasibility, we provide a comparison between the services developed according to the data-centric and device-centric models in Table 1, thus to highlight their similarities and differences.

As already mentioned, the data-centric model focuses on data, whereas the device-centric model focuses on the infrastructure (Feature 1 in Table 1. It means that, following the data-centric model, Cloud4Sens provides a PaaS for gathering data from MIs, storing and providing them to clients (Feature 2). Otherwise, Cloud4Sens offers an IaaS to export the device-oriented model, where the sensing/actuation physical resources are virtualized in sensing/actuation components, accessible from clients.

A client requests a data-centric service (that is PaaS) if it needs environmental information and physical measurements, but has no expertise in virtual/physical infrastructure configuration and exploitation. Observations can be the result of multiple or aggregated measurements over several MIs, in order to fulfill client needs.

In contrast, the client requests a device-centric service (that is IaaS) if it needs full control of the sensing/actuation resources. In this case, the client can also tune the device parameters and customize the on-board software (Feature 3).

To achieve the above goals, both the Cloud4Sens services based on the data-centric and device-centric models aggregate heterogeneous and distributed MIs (Feature 4), which can be different both in terms of hardware equipment and software systems. Moreover, the Cloud4Sens services abstract MI information and monitored data, in order to provide a uniform description of available resources, device components, and physical observations (Feature 5).

To provide the device-centric service, specific virtualization techniques decouple physical devices from the virtual devices. Indeed, the virtual sensors on a cloud platform are dynamic in nature, and hence facilitate automatic service provisioning. Clients can supervise virtual sensors using some standard functions and interfaces (Feature 6).

The data-centric model guarantees a good isolation between cloud and MI systems; data gathered from sensors into MIs flow toward the cloud, and actuation directives flow from the cloud toward the MIs. Thus, the service provisioning of the cloud provider does not affect activities inside the MIs. In contrast, the device-centric service extends the cloud functionalities inside the physical monitoring infrastructure, and hence the MIs' behavior can be influenced by the specific requirements of the clients (Feature 7).

Related Work

The interest of the research community in cloud based sensing solutions is proved by the numerous running projects. ClouT (http://cloutproject.eu/) is a joint EU-Japan project that intends to provide reusable infrastructure, services, tools, and applications to allow different city stakeholders such as municipalities, citizens, service developers, and application integrators to develop user-centric applications to improve the quality of life. FI-WARE (http://www.fi-ware.org/) focuses on the design and development of an innovative infrastructure ("core platform") for cost-effective creation and delivery of scalable and secure Future Internet services. The Arrowhead project (http://www.arrowhead.eu/) addresses the technical and applicative challenges associated with collaborative automation. The main challenges of the project are enabling the interoperability and integrability of services provided by networked embedded devices.

The following are examples known in the literature of cloud services for sensing resources provisioning that adopt a data-centric model. Fong *et al.* [1] present a mobile system, where a non-contact ECG measurement method is employed to capture biomedical signals from users. To observe and analyze the ECG signals in real time, a mobile device is used to continuously collect biomedical signals from multiple locations. Health data are synchronized into the cloud service to ensure a seamless healthcare monitoring system, and anytime and anywhere coverage of the network connection is available. The article by Yerva *et al.* [5] investigates the potential of fusing social and sensor data in the cloud. The authors present a cloud framework that blends the heterogeneous social and sensor data for integrated analysis, extracting weather-dependent people's mood information from Twitter and meteorological sensor data streams. A context-oriented data acquisition and integration platform for the Internet of Things (IoT) over a cloud is presented in [6]. The platform collects sensor data from different types of sensor devices and integrates them into semantic contexts, which can be easily shared and reused among different mobile applications. The context information can enhance mobile applications' usability by adapting to conditions that directly affect their operations. The cloud data management service reported in [7] aims at massive sensing data management in the cloud. It includes a framework supporting parallel storage and processing of massive sensor data in cloud manufacturing systems based on Hadoop.

Cloud services for sensing resource management based on the device-centric model usually offer general purpose functionalities. Tomarchio *et al.* [8] present an OSGi (Open Service Gateway initiative) based middleware to abstract sensors from their proprietary interfaces and to offer sensing capabilities to third party applications. Through such a middleware, sensors are no longer low-level devices producing raw measurement data, but can be seen as "services" to be used and composed over the Internet in a simple and standardized way. In [2] a dynamic micro-cloud environment made of embedded devices able to share computational power is proposed. Thus, a dynamic micro-cloud encompassing inter-linked smart objects and smart mobile devices available from a smart environment can be formed on the fly. The Missouri S&T (Science and Technology) sensor cloud [9] connects different networks over a large geographical area, to connect together and be employed simultaneously by multiple users according to an on-demand service model. Virtual sensors compose a multi-user environment on top of resource-constrained physical wireless sensors to support multiple applications. Kyusakov *et al.* [10] propose to integrate wireless sensor and actuator networks (WSAN) with existing telecommunication infrastructures and legacy systems deploying interoperable Simple Object Access Protocol (SOAP)-based web services directly on the wireless nodes and not using gateways. This strategy bridges the gap between sensor nodes and enterprise applications (e.g. factory monitoring, control, and tracking systems) and supports heterogeneity at the lowest level.

#	Feature	Data-centric	Device-centric
1	Type of resources	Data	Infrastructure
2	Type of offered cloud service	PaaS	IaaS
3	Client needs	Environmental information, contextual measurements, and compound observations	Full control of of sensing/actuation resources, and the ability to customize the behavior of devices.
4	Support for heterogeneous distributed infrastructures	Yes	Yes
5	Need of abstraction technologies	Yes	Yes
6	Need of virtualization technologies	No	Yes
7	Decoupling between cloud and MI activities	High	Low

Table 1. Comparison between data- and device-centric models to provide sensing resource management services.

To the best of our knowledge there are no solutions in literature able to offer both the data-centric and device-centric services at the same time. Cloud4Sens is a new cloud framework able to offer different types of services to its clients. This implies high flexibility and adaptability of the cloud to the specific client's needs.

Cloud4Sens Architecture

The Cloud4Sens framework interacts with different administrations deploying several MIs to accomplish specific purposes in terms of types of sensing detection goals (e.g. temperature, humidity, pressure, etc.). Possible examples of heterogeneous MIs, both in terms of purposes and adopted technologies, are shown in Fig. 2, where:

- Site 1 is involved in monitoring the territory, to identify eruption or seismic risk in a volcanic land, to provide support for the preparation of response plans in case of natural disasters. The site provides collected data through a local database (DB) and web services.
- Site 2 delivers weather forecasting, collecting quantitative data related to the current state of the atmosphere in a given area (e.g. evaluating barometric pressure, current weather conditions, and sky condition) that are stored in a local DB.
- Site 3 is a power grid that makes use of sensor networks to manage security, interoperability, and performance required for large-scale smart grid applications. Data on energy produc-

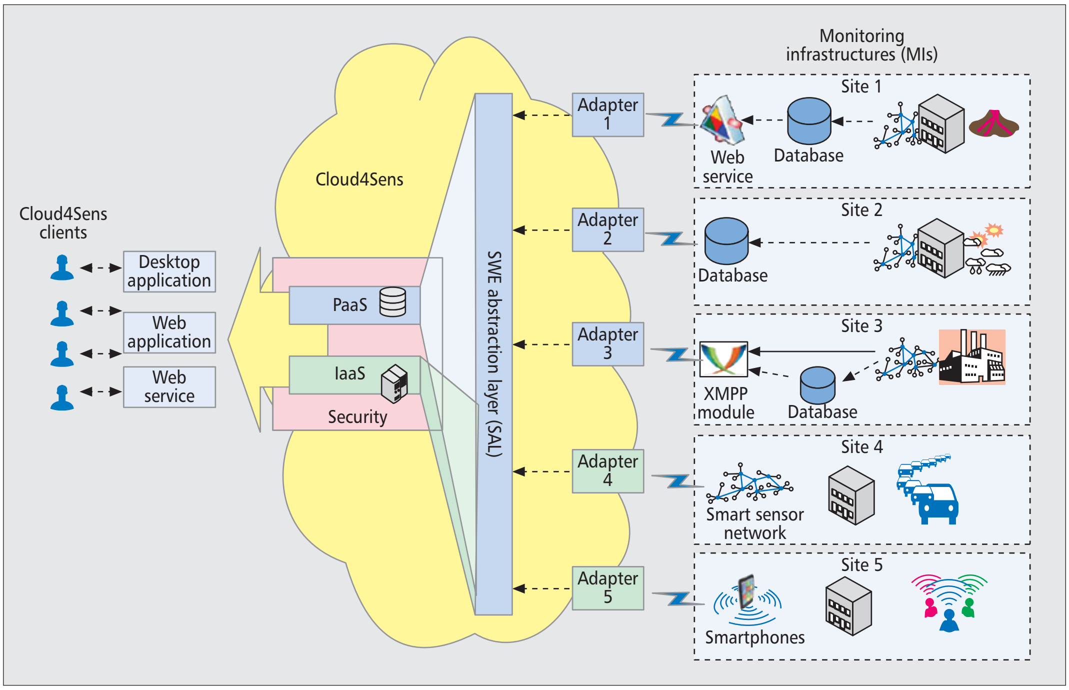

Figure 2. Cloud4Sens architecture.

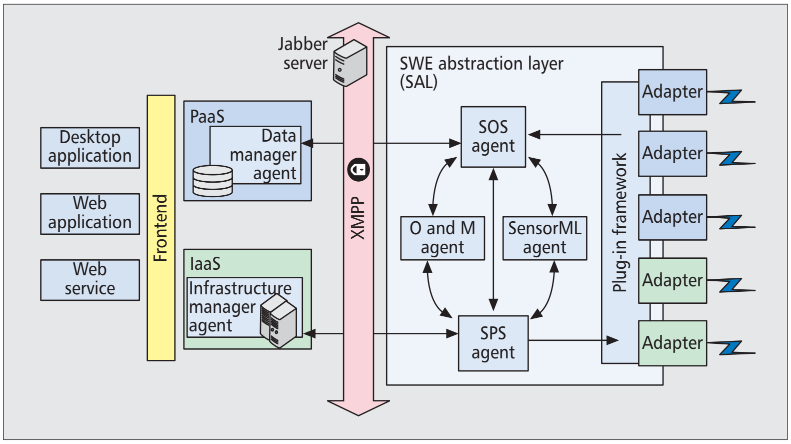

Figure 3. Cloud4Sens main components.

tion and power plant operation are stored into a local DB and moved in the cloud by using the XMPP technology [11].

•Site 4 presents a direct interconnection between Cloud4Sens and a sensor network deployed in a specific geographical area for general purpose applications (e.g. traffic monitoring).

•Site 5 presents similar features as Site 4, but it includes particular types of sensing devices, such as smartphones, PDAs, and all other mobile communication devices equipped with sensing capabilities.

All the MIs access the cloud through the SWE abstraction layer (SAL), responsible for integration and abstraction of the system resources that are necessary to develop both the data-centric and device-centric based services. The *adapter* components of the architecture hide the communication technologies required to interconnect different MIs to the SAL. Then the SAL arranges the information according to a well defined format. The SAL is compliant with the OGC-SWE standard, which includes a set of XML-based languages and Web service interface specifications to facilitate the discovery, exchanging, and processing of sensor observations. These languages provide a means to integrate data from heterogeneous sources in a standard format accessible from cloud users. Thus, the SAL abstracts all the connected systems and observed data by using an XML based semantic according to the SWE specifications.

To offer data-centric based services, the SAL interacts with the PaaS level of the Cloud4Sens architecture, where the data gathered from different MIs are transferred.

The IaaS level of Cloud4Sens offers services based on the device-centric model, exploiting specific virtualization techniques to present available virtual devices to its clients and enable clients to interact with the virtual infrastructure. To support the IaaS, the SAL implements bidirectional communications through adapters to support information flows from MIs to the cloud and vice versa.

Cloud4Sens includes some security components to guarantee data confidentiality, integrity, and non-repudiation. Since they do not depend on the offered service, the security components embrace all the levels of the architecture.

Figure 3 shows the main software agents composing the Cloud4Sens architecture and illustrates how they are interconnected.

To connect heterogeneous MIs to the cloud, a plug-in based framework for systems interconnection has been designed. Implementing a new plug-in implies the development of a new adapter, and hence interconnecting a new MI to the cloud. Each adapter has an interface toward the sensing nodes able to interact with the specific communication technology (web service, DB query, ZigBee, etc.).

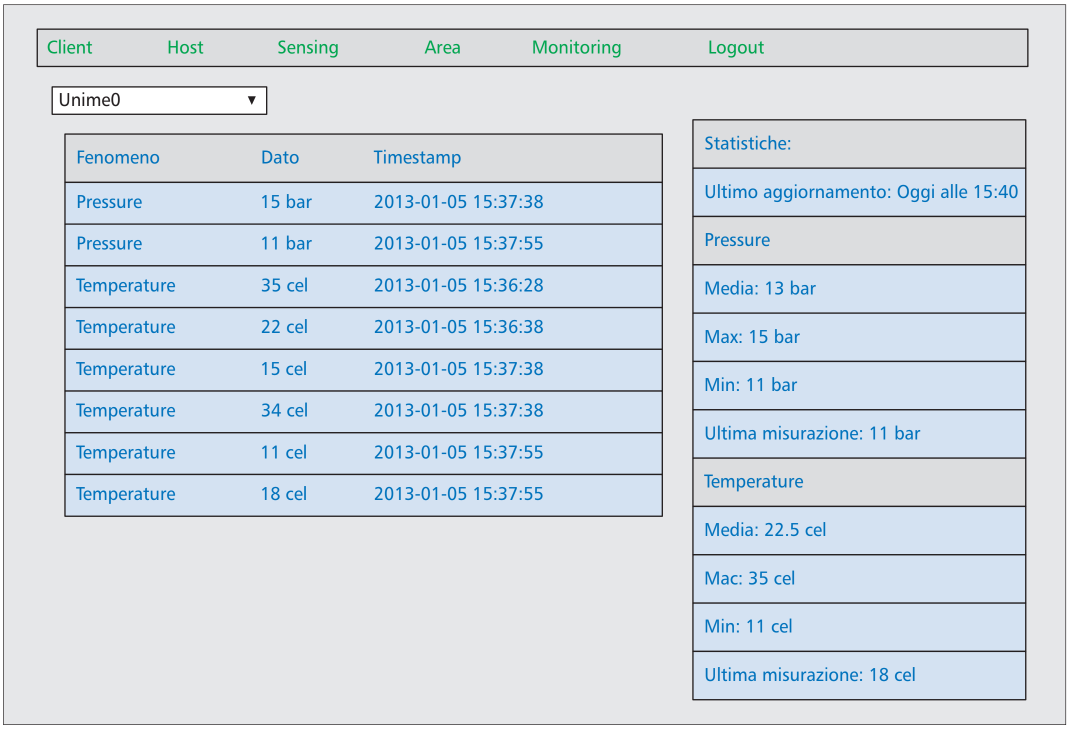

Figure 4. Sensing data.

Designed as a protocol for near-real-time communication, XMPP supports a small message footprint and low latency message exchange. In the Cloud4Sens architecture, a Jabber/XMPP server provides the main XMPP communication functionalities inside the cloud system.

The SAL includes the software agents necessary to implement information abstraction, and it has been designed according to the SWE specifications (Fig. 3):

O&M (Observation and Measurements) Agent: According to the SWE-O&M standard, it holds models and XML schema for encoding observations and measurements from a sensor network.

SensorML Agent: Provides models and XML schemas for describing sensor systems and processes. It manages information to enable the discovery of sensors and location of sensor observations, processing low-level sensor observations and listing taskable properties.

SOS (Sensor Observation Service) Agent: Responsible for requesting, filtering, and retrieving observations and sensor system information. It gathers data entering the system through the adapters, formats observations according to O&M directives, and sensors information according to SensorML directives and pushes them to the cloud layers of the architecture.

SPS (Sensor Planning Service) Agent: Into the SAL, it performs user-driven acquisitions and observations. It receives asset specifications from the cloud layers of the architecture and instructs the MI to execute a task over a sensor. Then, it reserves resources required to perform the planned task for a certain amount of time. Sensing information is collected via the **SOS Agent**, but they are provided on a per-task basis.

The above agents are deployed over a distributed infrastructure into the datacenters of the Cloud4Sens provider. Communications among the distributed nodes of the infrastructure are based on XMPP. XMPP is an open technology for real-time communications and it works according to a decentralized client-server paradigm. Designed as a protocol for near-real-time communication, it supports a small message footprint and low latency message exchange. In the Cloud4Sens architecture, a Jabber/XMPP server provides the main XMPP communication functionalities inside the cloud system.

A great advantage of XMPP is that the XMPP specification supports both the Simple Authentication and Security Layer (SASL) Mechanism and the Transport Layer Security (TLS) technology for the authentication and encryption of communication channels. Moreover, customized features can be built on top of XMPP. To maintain interoperability, common extensions are managed by the XMPP Software Foundation. Specifically, the *XEP 0027* specification [12] allows us to include additional XML tags according to specific *namespaces* that define how a piece of information contained in a tag has to be processed. Thus, we have extended the XMPP security functionalities with new tags, encrypting and signing messages with X509 certificates. For further information refer to [13].

At the PaaS level, Cloud4Sens includes the **Data Manager Agent** that provides cloud sensing services to clients according to the data-centric model. It receives abstracted data from the **SOS Agent** and manages them as follows:

• It stores data in a reserved storage system. Data gathered from MIs can be so large and complex that it becomes difficult to store and process them using traditional database management tools. Thus, the **Data Manager Agent** exploits scalable technologies to guarantee efficient data management in the cloud that are NoSQL technologies such as MongoDB, Hadoop, and Cassandra.

• The **Data Manager Agent** organizes data in order to publish them as a service. Data are clas-

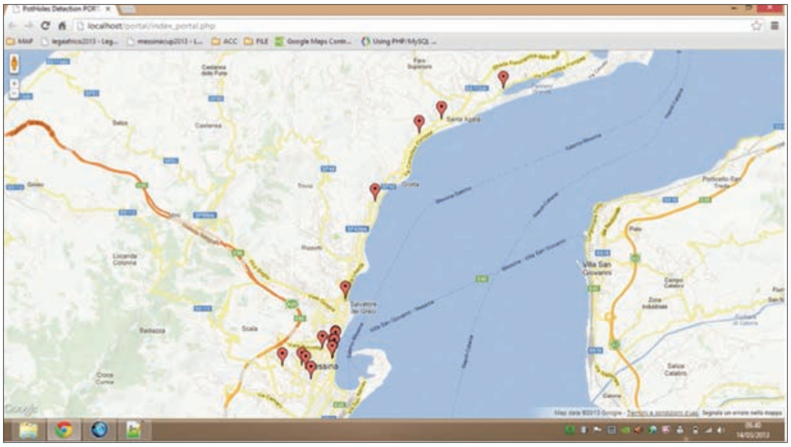

Figure 5. PDM: Web app.

sified into *subscription offerings*, and each subscription offering includes one or more observations.

• The **Data Manager Agent** manages client subscriptions to access specific types of observations or data over a limited geographical area. It guarantees to clients a seamless access to data offered by different MIs, decoupling the activities of sensing devices and clients.

To satisfy device-centric client requests, at the IaaS layer Cloud4Sens implements the **Infrastructure Manager Agent**, in charge of setting up a virtualized sensing infrastructure as a service. To adapt the physical infrastructure to the virtual one customized by the client, the **Infrastructure Manager Agent** submits a set of tasking parameters to the **SPS Agent**, and the SPS tasks the assets on the physical infrastructure.

The main functionalities of the **Infrastructure Manager Agent** are:
- Enrollment, discovery. and monitoring of available resource.
- Orchestration and composition of virtual resources according to client's requirements.

Service Usability: Discussion of Two Use Cases

This section discusses two applications developed by using the Cloud4Sens services. The first application supports risk management in industrial sites [14]. The second application detects potholes and monitors road surface conditions exploiting virtual devices sited in the area of interest [15].

Risk Management

Monitoring industrial risk implies the processing of a set of information useful to characterize the context, such as air pollution, temperature, power consumption, hydraulic charge pressure, air compression, motion, optical positioning, weight, acceleration, chemical composition, gases, liquid flow, and so on. A generic risk management system performs two main tasks:
1) Data collection, to gather data from the environment and/or activities.
2) Event classification, to specify which events may be dangerous and the impact due to their occurrence.

Damaging events include (among the others) on-the-job injury, accidents, explosions, and environmental damage.

We have developed a web site that interacts with the Cloud4Sens PaaS to obtain all the necessary information for risk detection and offers end-users an easy access to the main functionalities of the risk management system. In the testbed, we emulated several sensors to gather observations from an area of interest. The emulated environment gives us the opportunity to easily tune the number of sensors and the sampling rate. Then data are processed by algorithms developed to detect risk in that area. Figure 4 shows how data on temperature and pressure are handled. Each row identifies an observation, specifying its type, the measured value, and the timestamp.

Pothole Detection and Mapping

The pothole detection and mapping (PDM) application leverages commonly available sensing capabilities of mobile devices, carried around by people also during commuting and other driving-related activities, in order to automatically detect and rank road surface conditions. The combined sampling of acceleration values, as measured by on-board motion detection sensors, and of geospatial coordinates, as provided from the GPS subsystem of the mobile device, serves as a first step in generating a qualitative map of travelled roads, highlighting possible conditions of road distress as well as the potential existence of "potholes."

The Android application running on the mobile device performs a continuously ongoing sensing activity with regards to accelerometer-provided measurements. An ad-hoc developed algorithm interacts with the sensors through the Cloud4Sens IaaS to evaluate the fluctuations in the sampled values for acceleration. Intuitively, when stumbling upon a pothole along the path, or driving through a road featuring a distressed surface, these fluctuations may be abrupt. The client of the PDM is able to choose one or more virtual nodes in a specific area over a map, and turn the type of information he needs. Based on these input data, a Web-based application performs data collection and basic filtering, aggregation, and data mining operations. Figure 5 shows a screenshot of the Web UI, leveraging the Maps service to host the Points of Interest placeholders.

Conclusions

This article presented a new cloud architecture, called Cloud4Sens, that is able to offer services based on both the data-centric and device-centric models. We discussed in detail the data-centric and device-centric models for sensing and actuation resource management into a monitoring environment. We also showed their implications in offering resources as a service, according to the cloud computing model.

To identify possible scenarios of interest and explain how administrations involved in monitoring activities can collaborate by using the cloud, we described the strategy adopted to interconnect heterogeneous monitoring environments and the cloud. Then details of the Cloud4Sens architecture were provided to explain how some useful standards and technologies (e.g. OGC-SWE, XMPP, XEP 0027, X509 certificates) have been exploited and extended in our system.

To show the effective benefits of the proposed solution, we briefly presented two applications developed at the University of Messina by using both data-centric and device-centric based services of Cloud4Sens.

Acknowledgements

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Biographies

Maria Fazio (mfazio@unime.it) received the Ph.D. in 2006. She worked as a collaborator on many national and international projects, such as EU FP7 VISION Cloud, EU FP7 RESERVOIR, and EU FP7 CloudWave. Since 2014 she has been a member of the IEEE Internet of Things (IoT) Technical Community. Her current research activities include smart sensing, big data storage, cloud federation, Internet of Things, energy efficiency, and security.

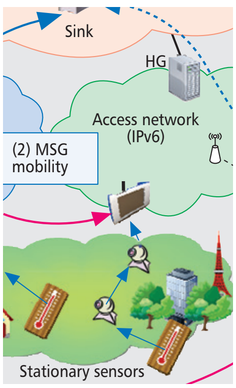
Antonio Puliafito (apuliafito@unime.it) is a full professor of computer engineering at the University of Messina, Italy. His interests include parallel and distributed systems, networking, wireless, and cloud computing. He is acting as an expert in ICT for the European Commission since 1998. He is currently the President of the Centre on Information Technologies at University of Messina. He participated in several European projects such as Reservoir, Vision, CloudWave, and Beacon. He recently constituted DHLabs srl.

The client of the PDM is able to choose one or more virtual nodes in a specific area over a map, and turn the type of information he needs. Based on these input data, a Web-based application performs data collection and basic filtering, aggregation and data mining operations.

Design and Implementation of Dynamic Mobile Sensor Network Platform

Most current sensor networks are application-specific; that is, they are designed and deployed for providing only specified services. The authors present the design and implementation of a dynamic mobile sensor network platform in which users can freely install new applications of their choice.

Ved P. Kafle, Yusuke Fukushima, and Hiroaki Harai



The authors are with the National Institute of Information and Communications Technology.

Abstract

This article presents the design and implementation of a dynamic mobile sensor network platform consisting of stationary sensors, mobile sensors, mobile sensor gateways, and sink servers. The sensor network platform supports heterogeneous protocols in the network layer and performs ID-based communications to deliver sensor data from the sensors to the sinks, as well as to send control and monitoring commands from the sensor owner or administrator to the mobile sensors and mobile sensor gateways. To provide sensor data reliably irrespective of the sensor locations, the mobile sensors and mobile sensor gateways natively support mobility and multihoming and possess network access authentication and data transport security functions. The sensor owners can freely install new applications over already-deployed mobile sensors and mobile sensor gateways. They can configure the mobile sensors to operate in lightweight or full-function modes depending on the sensor application requirements or available networking environments. The technical feasibility of the proposed mobile sensor network platform has been demonstrated by implementing a prototype system.

Introduction

Different types of sensors are becoming available for sensing physical events and transmitting sensor data via wireless communications [1]. Sensor networks are becoming the major component of new network architectures and applications. However, most current sensor networks are application-specific; that is, they are designed and deployed to provide only specified services [2]. Their configurations are mostly static and cannot be adapted easily to different networking environments and applications. Their networking protocols and applications (e.g. ZigBee) are pre-installed as a package, and it is difficult for sensor users to separate, replace, or change them.

In this article we present the design and implementation of a dynamic mobile sensor network platform in which users can freely install new applications of their choice. We apply the ID/locator split protocol stack developed for the Heterogeneity Inclusion and Mobility Adaptation through Locator ID Separation (HIMALIS) [3] architecture to the design of our mobile sensor network platform. In HIMALIS, the communication endpoints or session identification functions do not use IP addresses or locators, whose values depend on the network-layer protocol as well as the topological location, but instead use newly assigned, location-independent, static IDs. We call the communication using these IDs *ID-based communication*. Using IDs to identify the communication endpoints or sessions in the application and transport layers allows ID-based communication to take place between heterogeneous network-layer protocols. HIMALIS has inserted an identity layer between the network and transport layers of the protocol stack to hide the heterogeneity in the network-layer protocols from the transport and application layers. This allows changing of locators present in the headers of data packets as they traverse the gateways connecting the network segments of heterogeneous network-layer protocols. ID-based communication is favorable for mobility and multihoming management because it allows the communicating endpoints to change their locators or to use two or more locators simultaneously without disrupting the ongoing communication sessions or breaking the security context identified by the pair of source and destination IDs.

While the previous publications on HIMALIS described the general architecture [3], mobility management [8], and security scheme [9], this article describes the extension of HIMALIS to the design and implementation of a general-purpose sensor network platform, where the networking protocols and applications are decoupled in such a way that new applications can be easily installed on the mobile sensors. The proposed dynamic mobile sensor network platform natively supports mobility of mobile sensors and mobile sensor gateways, as well as mobility of the sensor network as a whole. It provides multihoming support for mobile sensor gateways, allowing them to use multiple upstream access links for failure resiliency and seamless handovers. The mobile sensors, mobile sensor gateways, and sinks are allowed to use different network layer protocols, e.g. 6LoWPAN [4], IPv4, and IPv6, respectively.

The mobile sensor network platform also includes inbuilt authentication and security mechanisms to securely attach the mobile sensors to the network and transmitting sensor data to sinks, as well as for enabling the network administrator to remotely monitor and configure mobile sensors and mobile sensor gateways. To balance the signaling overhead and energy consumption, the mobile sensors are capable of operating in two different modes: lightweight-signaling mode and full-signaling mode. While running in lightweight-signaling mode, the mobile sensors are not required to participate heavily in the signaling process. Instead, they delegate the mobile sensor gateways to perform security-related and mobility-related signaling, after having completed a simple authentication process. If the mobile sensors have no energy

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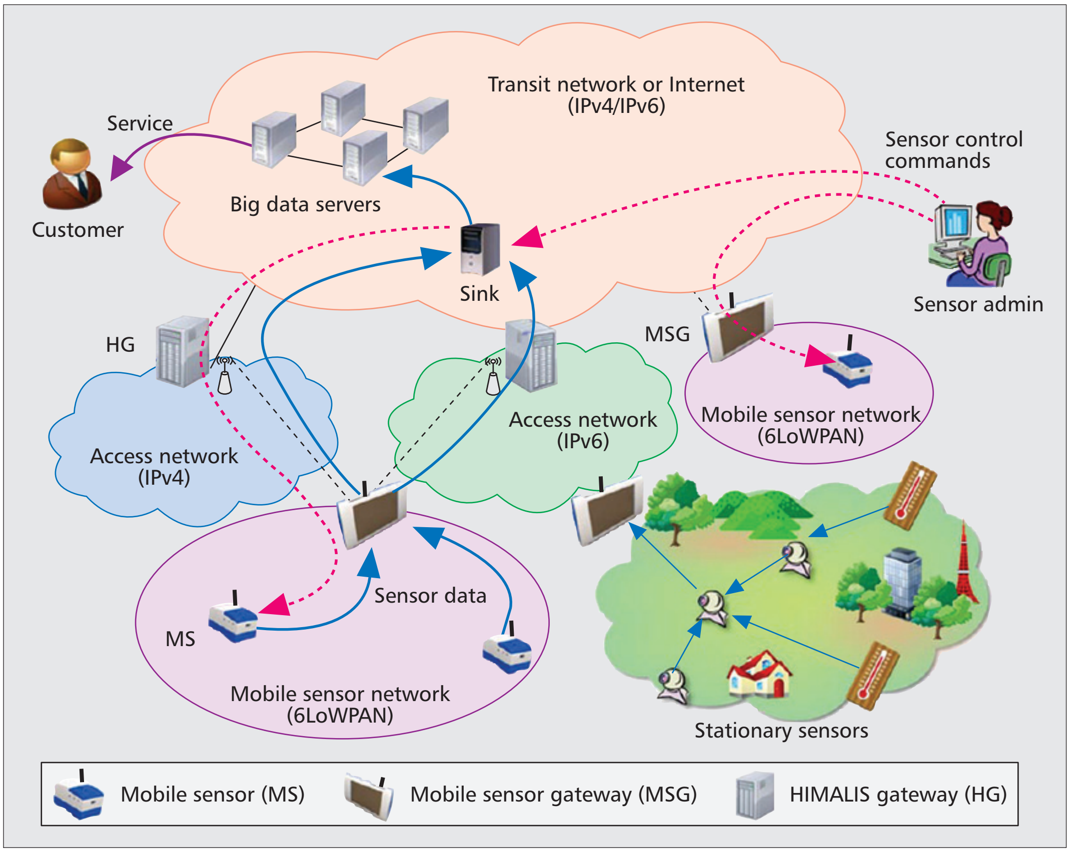

Figure 1. Components of the dynamic mobile sensor network platform.

> The sink can distribute the sensor data to other storage servers, such as Big Data servers, to offer sophisticated sensor application services to customers. The MSs and MSGs are monitored, reconfigured, and controlled by the sensor administrator by issuing control commands.

constraints, they operate in full-signaling mode and execute all security and mobility related tasks by themselves, without depending on the mobile sensor gateways.

The proposed dynamic mobile sensor network platform is based on ID-based communication, which has already been accepted as an appropriate approach for the design of future networks being standardized in the ITU-T Study Group 13 [5]. Since the basic concepts of ID-based communications and ubiquitous sensor networks have been standardized [6, 7], the other component technologies explained in this article can be gradually brought to ITU for standardization. For example, the methods for supporting heterogeneous network protocols, detecting and managing mobility across heterogeneous networks, authentication and network access control, and the interfaces between the sensing and communication units are potential candidates for standardization.

The remainder of this article is organized as follows. The next section describes the components of the dynamic mobile sensor network platform and their functions. The implementation details follow next. Some possible service scenarios and performance results are then illustrated. The final section concludes this article.

Dynamic Mobile Sensor Network

Network Components

Figure 1 shows the components of the dynamic mobile sensor network platform. The mobile sensor network consists of stationary sensors, mobile sensors (MSs), and mobile sensor gateways (MSGs). The mobile sensor network is connected to one or more access networks via the MSGs. The access networks can use IPv4 or IPv6 as the network-layer protocol. Moreover, two adjacent access networks can use different network layer protocols, e.g. one using IPv4 and the other using IPv6. The access networks are connected to the Internet or the transit network through HIMALIS gateways (HGs). The figure shows a sink located on the Internet and which stores sensor data collected from the MSs and MSGs. The sink can distribute the sensor data to other storage servers, such as Big Data servers, to offer sophisticated sensor application services to customers. The MSs and MSGs are monitored, reconfigured, and controlled by the sensor administrator by issuing control commands. These components are briefly described in the following subsections.

Mobile Sensor — The MS has two units: sensing and communication. The sensing unit is composed of sensors to sense the environment, generate sensor data, and provide the sensor data to the communication unit. The communication unit is composed of the communication functions and applications for collecting and transmitting sensor data to the sink through the network. The communication protocol stack is based on a ID/locator split [3] and contains mobility management [8] as well as security functions [9]. Its access link technology can be WiFi (i.e. IEEE 802.11) or IEEE 802.15.4, but the latter has been chosen for our prototype system because of

Network access control functions specify the procedure for the mobile sensor network to attach to the access network. They mainly specify how the MSG and the MS identify themselves and present their credentials to get securely attached to the access network and the sensor network, respectively.

its low power consumption. In conjunction with IEEE 802.15.4, the 6LoWPAN protocol [4] is used in the network layer. The MS possesses capabilities for securely attaching itself to the mobile sensor network, initiating ID-based communications with the sink, and mobility management.

Mobile Sensor Gateway — The MSG possesses all functions of the communication unit of the MS for performing ID-based communication. It also includes additional functions for network-layer protocol conversion, multihoming management, routing, and packet forwarding. It has at least two radio interfaces: one for connecting with the mobile sensor network and another for connecting with the access network. The mobile sensor network link is a low-power radio, i.e. an IEEE 802.15.4 link, and the access network link is a WiFi link. Optionally, it may have additional wireless and wired interfaces for connecting to cellular and wired access networks. The mobile sensor network interface uses the 6LoWPAN protocol in the network layer, whereas the access network interface uses IPv4 or IPv6. The MSG collects sensor data from the stationary and mobile sensors and forwards the sensor data to the sink.

Stationary Sensors — The stationary sensors are tiny devices containing the sensing and communication units. They are not required to establish end-to-end ID based communication with the sink and possess mobility support functions. They sense the environment and provide sensor data to the MSG through a lightweight radio link, e.g. IEEE802.15.4. The MSG collects sensor data from the stationary sensors and sends it to the sink through ID-based communication.

HIMALIS Gateway — The HG is similar to the MSG as it also possesses functions for network-layer protocol conversion, mobility, and multihoming management. It also has at least two interfaces: a radio interface (e.g. WiFi) for setting up a downstream link with the access network, and a wired interface (e.g. Ethernet) for setting up an upstream link to the Internet or the transit network. Optionally, it can have additional upstream links for making the access network multihomed. It supports both IPv4 and IPv6 and performs IPv4-to-IPv6 translation, and vice versa, if needed. It takes part in the network access control of the MS and MSG through an authentication mechanism. Note that some access networks may have no HGs, i.e. they are simply Internet access networks, and the MSG is connected directly to the Internet. The access networks communicate with each other through the transit network or Internet.

Sink — The sink is a server on the Internet or in the access network and possesses an ID/locator split-based protocol stack to perform ID-based communication with the MS and the MSG. It has either IPv4 or IPv6, or both protocols in the network layer. It also possesses sensor applications for gathering sensor data from the MSs and MSGs and distributing them to clients or other storage servers, such as Big Data servers, which create various applications such as Internet of Things (IoT) services. Although only one sink is shown in the figure, the proposed network platform supports any number of sinks for collecting sensor data from the same or different sensors.

Sensor Administrator — The sensor administrator issues commands to monitor, configure, and control the MS and the MSG through ID-based communication. The sensor administrator also provides the MS credentials to one or more sinks so that the sinks can establish secure communication sessions to obtain sensor data from the MS.

Although not shown in Fig. 1, there exist name registry servers on the Internet for storing, updating, and retrieving ID/locator mappings of the MS and MSG. These servers may also exist in the HIMALIS access network to help the MSG in resolving another node's (e.g. the sink's) hostname into an ID and locator, which are required for initiating ID-based communication.

Network Functions

The network functions can be grouped into two categories: control plane and data plane. We briefly describe the various control and data plane functions below.

Control Plane Functions — The control plane functions are used for network access control, mobility and multihoming management, and registering, retrieving and updating of ID/locator mapping in name registries. The data plane functions establish ID-based communication sessions, and transport sensor data from the MS or the MSG to the sink through heterogeneous network domains of various network-layer protocols.

Network access control functions specify the procedure for the mobile sensor network to attach to the access network. They mainly specify how the MSG and the MS identify themselves and present their credentials to get securely attached to the access network and the sensor network, respectively.

Mobile Sensor Gateway's Network Access Functions — The MSG can be attached to an access network that contains an HG or does not contain an HG; that is, the MSG can make direct access to the Internet. The network access signaling sequence performed by the MSG varies in these two cases. To attach to an access network containing the HG (as shown in the left side of Fig. 1), which is referred to here as the *HIMALIS access network*, the MSG executes the signaling sequence specified in [5] to authenticate itself and pass through the network access security check. In this case, the MSG first obtains the access network parameters, such as the ID, locator, and security information of the authentication agent, from the HG in the optional header of Dynamic Host Configuration Protocol (DHCP) and then performs the authentication and local registration processes. The HG supports the MSG for seamless mobility across the heterogeneous HIMALIS access networks. To attach directly to an Internet access network that does not have an HG (as shown in the right side of Fig. 1), the MSG uses a conventional host configuration protocol, such as DHCP. To help the MSG to distinguish the HIMALIS access network from the Internet access network, additional parameters, such as the IDs and locators of the HG, local name reg-

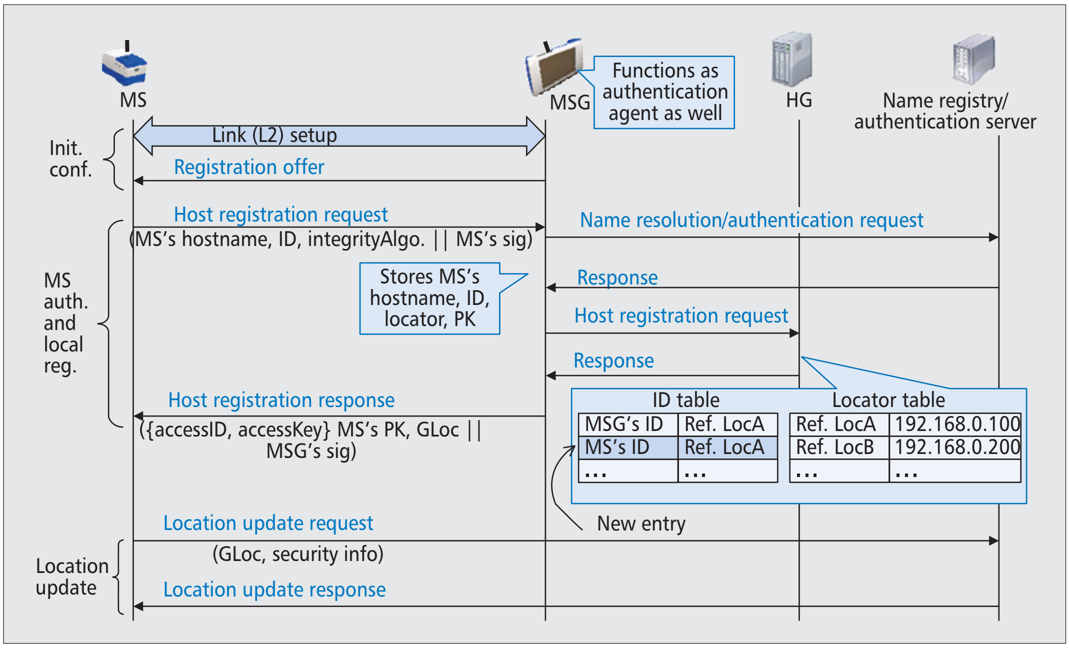

Figure 2. Mobile sensor's network access procedure when the mobile sensor gateway is in proxy mode.

The sequence of signaling messages exchanged by the MS to attach to the sensor network depends on the operational mode of the MSG, which is indicated by the MSG in the registration offer message that it advertises in the mobile sensor network.

istry server and authentication agent, are included in the optional header of the *DHCP offer* message. We have opted to make the MSG capable of accessing both the HIMALIS access network and the Internet access network to ensure flexibility in the deployment of the sensor network. The HIMALIS access network empowers the MSG with the additional features of network access security, seamless mobility, and communication across heterogeneous access networks, whereas the Internet access network simplifies the mobile sensor network deployment as it does not require special nodes in the access network, but may need additional authentication and mobility management mechanisms not specified here.

Depending on the access network type, the MSG can operate in any of the following three modes to help the MS get connected to the mobile sensor network: as an HG, as a relay node, and as a proxy node. In the first mode, the MSG is attached to an Internet access network; in the latter two modes, it is attached to a HIMALIS access network.

Mobile Sensor's Network Access Functions — The sequence of signaling messages exchanged by the MS to attach to the sensor network depends on the operational mode of the MSG, which is indicated by the MSG in the registration offer message that it advertises in the mobile sensor network.

When the MSG is operating as an HG: The MS's network access procedure is similar to that specified for the network access of the host in [9]. The MSG forces the MS to pass through an authentication procedure for controlling the network access, and maintains the MS's ID/locator mapping in its ID table, which is used to help the MS in mobility and ID-based communication across heterogeneous protocols.

When the MSG is operating as a relay node: The MS's network access procedure is similar to that specified for the network access of the host in [9]. In this mode, the HG located in the access network enforces the MS to pass through the authentication procedure. The MSG simply translates the network header from 6LoWPAN to IPv6 used for the signaling packets. Thus, when the MSG is operating as an HG or a relay node, the MS operates in full-signaling mode.

When the MSG is operating as a proxy node: The MS's role in the network access procedure becomes lighter (i.e. the MS can operate in lightweight-signaling mode) as it only needs to initiate the network access procedure, whereas the MSG takes care of the remaining steps of authentication and local registration required for completing the network access by interacting with the HG. Figure 2 shows the sequence of signaling messages exchanged for the network access. After having finished the layer 2 (L2) link setup by using the IEEE 802.15.4 protocol, the MSG sends a *registration offer* message containing the MSG's hostname, ID, local locator (LLoc), i.e. its IP address used within the sensor network, and public key (PK) to the MS. The MS configures and sends a host *registration request* message, containing its hostname, ID, and integrity algorithm used to calculate a signature for protecting the message, to the MSG. The MSG verifies the MS's credentials by retrieving the MS's ID and PK from the name registry or authentication server. Upon verification, the MSG stores these parameters in its ID table, and then sends a host registration request containing the MS's ID and a reference locator to the HG. The reference locator is the locator of the MSG's upstream link, i.e. the locator the MSG has obtained from the access network. The HG stores the MS's ID and reference locator in the ID table and sends a response back to the MSG. The MSG then configures a host *registration response* message containing an access ID and an access key, both encrypted using the MS's PK, and the reference locator. The mes-

Mobility is the major feature of the proposed dynamic mobile sensor network platform. This feature natively supports the mobility of the MS, MSG, and mobile sensor network. The MS can securely change its link from one mobile sensor network to another and continue sending its sensor data through the new MSG as the mobility process completes.

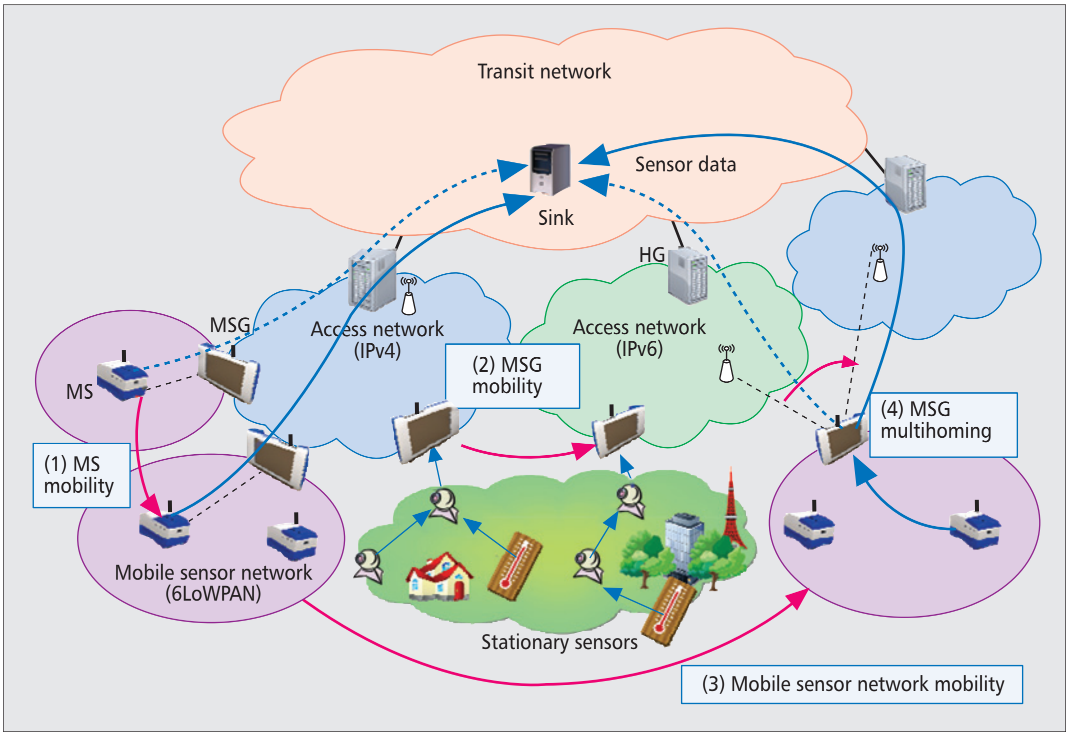

Figure 3. Various mobility scenarios.

sage is signed by the MSG and sent to the MS. When the MS receives this message, the network access procedure concludes successfully. The MS then sends a *location update* request message containing its new reference locator to its name registry. Thus, the name registry stores the up-to-date value of the MS's ID and locator so that the sinks or sensor administrator willing to initiate an ID-based communication session with the MS can retrieve the MS's current ID/locator mapping from the name registry.

In this way, the MS and the MSG are designed to be capable of operating in different modes and are able to select a network-layer protocol and wireless link technology that match the available network environments, device capability (e.g. computation, memory, and battery power), and application requirements (e.g. higher reliability and uninterrupted connectivity).

Mobility Management Functions — Mobility is the major feature of the proposed dynamic mobile sensor network platform. This feature natively supports the mobility of the MS, MSG, and mobile sensor network (as shown in Fig. 3). The MS can securely change its link from one mobile sensor network (i.e. MSG) to another and continue sending its sensor data through the new MSG as the mobility process completes. To move from one sensor network to another, the MS performs the network access procedure by authenticating itself in the new access network and getting a new locator. It then updates its locator stored in the name registry and in the sink by sending location update messages containing the new locator. The sink updates the MS's locator cached in its ID table and sends a response back to the MS. Then the MS resumes the ongoing ID-based communication session with the sink.

The MSG can move alone or together with all sensors located beneath it (i.e. mobile sensor network mobility). In both cases, the MSG performs almost similar handover procedures. Namely, it initiates a network access signaling process to authenticate onto the new access network and obtain its new local locator (i.e. configured with the network prefix of the access network) and global locator (i.e. configured with the network prefix of the HG's upstream interface), and updates the name registry with its new global locator. In the case of mobile sensor network mobility, the MSG informs the MSs located beneath it about the global locator change by broadcasting a new registration offer message. By receiving this message, an MS moving along with the MSG knows that its global locator has changed and it has to update its locator cached in the sink so that it can continue the ongoing ID-based communication session from the new location. In the case where the MSG was moving alone, the stationary and mobile sensors, which were not moving along with the MSG but encountering the registration offer message from the MSG for the first time, start the network access procedure as described earlier to get connected to the MSG. The MSG can achieve seamless mobility when it possesses two upstream wireless interfaces for performing a make-before-break handover in the HIMALIS access networks. The MSG also possesses multihoming functions and can switch the upstream links used for the ongoing ID-based communication when some trouble occurs in the currently used link.

ID/Locator Mapping Registration, Retrieval, and Update Functions — When the MSG and the MS are securely connected to the access network and the sensor network, respectively, through an initial registration

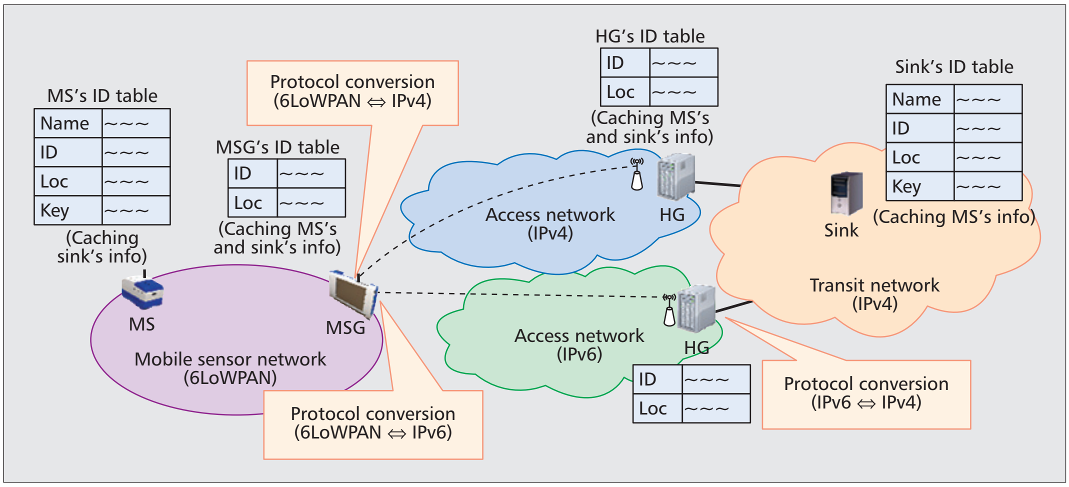

Figure 4. ID-based communication across heterogeneous network-layer protocols.

> To establish an ID-based communication session, the MS, the MSG, or the sink starts a communication initialization procedure by exchanging and verifying each other' s credentials provided by the sensor administrator and negotiating a shared key through a handshake process.

process explained in the previous subsection, they get their hostnames, IDs, and public keys registered in their name registries. Their name registries also provide shared secret keys to them. Whenever they change their locators due to mobility, they send a locator update message secured by the shared keys to their name registries. The name registries verify the security and update their records accordingly. Similarly, the sink and sensor administrator can retrieve the IDs, locators, and security keys of the MS and the MSG from the name registries by sending a name resolution query.

Data Plane Functions — The data plane uses information, such as ID/locator mapping and security keys, provided by the control plane to establish ID-based communication between the MS or the MSG and one or more sinks for transferring sensor data. To establish an ID-based communication session, the MS, the MSG, or the sink starts a communication initialization procedure by exchanging and verifying each other's credentials provided by the sensor administrator and negotiating a shared key through a handshake process. As shown in Fig. 4, the MS and the sink cache each other's hostnames, IDs, locators, and security keys in their ID tables. Similarly, the MSG and the HG also cache the ID/locator mappings of the MS and the sink. These mappings are used to perform network layer header (i.e. locator) translation of the packets carrying sensor data from the MS to the sink or the control commands from the sensor administrator to the MS.

For ID-based communication the application and transport layers use IDs, instead of IP addresses, in functions such as socket identification and checksum computation. Although IPv4 and IPv6 protocols are used in the network layer for routing and forwarding by using IP addresses, i.e. locators, these IP addresses are not used in the application and transport layers. The identity layer inserted between the transport and network layers maps the IDs to locators, and vice versa, by using the ID table. The ID table of the MS and the MSG is updated when they change their locators due to mobility. Because of this provision, the ID-based communication session is prevented from being disconnected even when underlying network protocols or locators of the endpoints have changed due to mobility. Moreover, the MS and the MSG can be located and contacted even if they are connected to different network layer-protocols (e.g. IPv4 and IPv6).

Implementation

We have implemented the components of the proposed dynamic mobile sensor network platform described in the preceding section. Figure 5 illustrates the hardware components of the MS and the MSG, where the small oval shapes represent the software components. The mobile sensor, consisting of the sensor unit and communication unit, currently has four sensors in the sensor unit: light, temperature, pressure, and humidity. Any other sensors can be added as required in the future. The sensor unit is connected to the communication unit through a USB cable. The communication unit is composed of Raspberry Pi (Model B) and XBee S1 RF module. Raspberry Pi includes an ARM 1176JZF-S 700 MHz processor, 512 MB memory, 8 GB SD card storage, and Raspbian OS. The HIMALIS stack and the sensor application are installed on it.

The mobile sensor gateway has been implemented on a Nexus 7 Android tablet with the following specifications: NVIDIA Tegra 3 Quad-core 1.3 GHz processor, 1 GB memory, 16 GB storage, and Android 4.2.2 OS. An XBee RF module and an extra WiFi interface are also attached to it externally. The HIMALIS stack and the sensor gateway application are installed on the tablet.

The sink and the HIMALIS gateway are built from PCs running Linux Ubuntu 12.04 on which the HIMALIS stack is installed. The sink has the sensor application for collecting sensor data from the mobile sensors and distributing them to other storage servers and clients. We have also implemented the name registry functions on PCs by extending the open source DNS implementation, BIND, for storing, updating, and retrieving the ID/locator mappings of the MS and the MSG.

Figure 6 shows the software modules of the

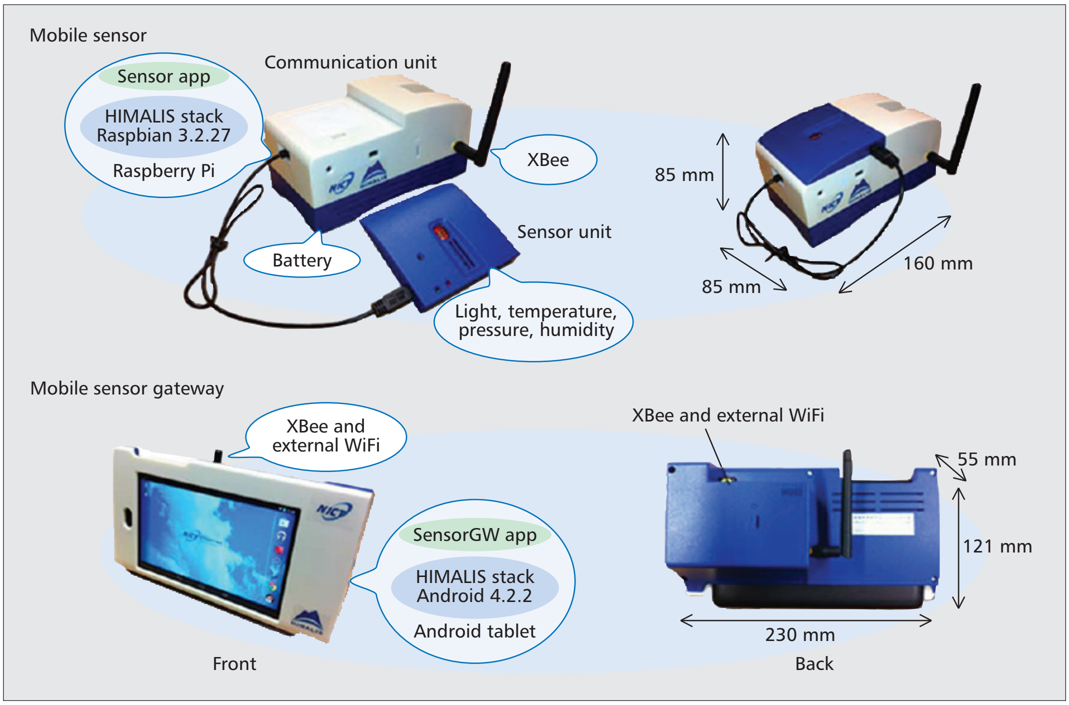

Figure 5. Hardware components.

The control and monitor commands are given through graphical user interfaces and are transmitted to the MS and the MSG securely by the ID-based communication. The sensor administrator can be located in IPv4 or IPv6 networks, i.e. heterogeneous access networks.

MS and the MSG. The sensor application and 6LoWPAN daemon are installed in the user-space and the HIMALIS stack; TUN (a virtual network kernel device), and serial connection modules are installed in the kernel. The sensor application in the MS collects sensor data by activating the sensor unit through the USB connection. It creates sensor data packets and sends them to the HIMALIS stack through UDP sockets. In the HIMALIS stack, the packet passes through the UDP, identity, and IPv6 layers. These layers add a UDP header, an ID header containing the MS's and the sink's IDs, and an IPv6 header containing the MS's and MSG's locators, respectively. The HIMALIS stack gives the packet to the TUN device, which delivers it to the 6LoWPAN daemon in the user-space. The 6LoWPAN daemon converts the 40-octet IPv6 header into a 6LoWPAN header of three octets (if no fragmentation) or eight octets (with fragmentation). The packet may become fragmented if its size does not fit into 102 octets to be carried as the IEEE 802.15.4 MAC Service Data Unit (MSDU). The 6LoWPAN daemon writes the packet in the serial port attached to the XBee RF module. Then the data gets transmitted through the mobile sensor network.

In the MSG side, the processing occurs in the reverse direction. That is, the XBee RF module receives the packet from the sensor network and gives it to the 6LoWPAN daemon through the serial port. The 6LoWPAN daemon assembles the packet fragments, creates a full IPv6 header, and writes the packet to the TUN device. The packet then passes through the HIMALIS stack, which can either give the packet to the sensor application in the MSG or forward the packet to the access network through the WiFi interface. In the former case, the sensor application may add extra sensor data, e.g. location information obtained from the GPS, and then transmits the packet to the access network. In the latter case, the MSG simply acts as a router for forwarding the sensor data packet received from the sensor network to the access network. Note that the HIMALIS stack natively supports the network-layer protocol translation from IPv6 to IPv4, or vice versa. Therefore, the access network can be an IPv4 or IPv6 network.

Applications and Performance

Current Basic Applications And Performance

We have implemented functions to allow the sensor administrator to control and monitor the MS and the MSG remotely and collect sensor data to sink servers. The control and monitor commands are given through graphical user interfaces (GUI) and are transmitted to the MS and the MSG securely by ID-based communication. The sensor administrator can be located in IPv4 or IPv6 networks, i.e. heterogeneous access networks. Figure 7a shows an example of the GUIs used to send control commands to the MS to turn the sensors on/off and change the sensor data sampling and transmission intervals. Here, the MS's hostname is *sensor01#himalis.net*, whose base sampling interval is set to five seconds (i.e. its sensors sense the environment once every five seconds). The temperature, humidity, and illuminance sensors are to be kept on, whereas the pressure sensor is to be kept off. As specified by data transmission interval values, the temperature reading should be transmitted to the sink server via the MSG every five seconds, while the humidity and illuminance readings should be transmitted every 10 and 15 seconds, respectively. Our current implementation can produce and transmit sensor data as fast as one reading every second. The network protocol between the MS and

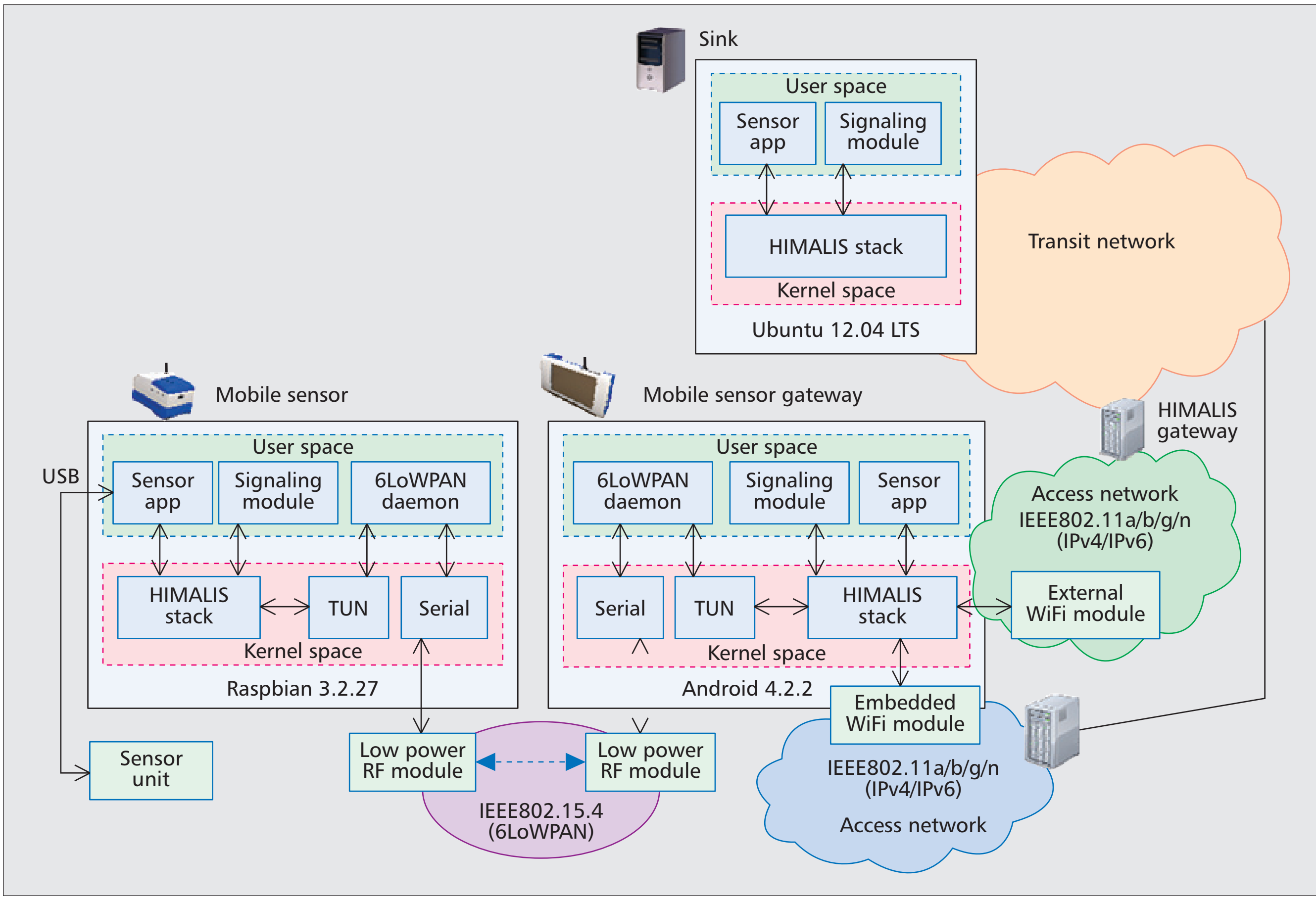

Figure 6. Software modules.

MSG is 6LoWPAN, whereas between the MSG and the HG it is IPv4 or IPv6.

After receiving the control commands from the sensor administrator, the MS configures its sensors to generate data in every base sampling interval and starts sending the sensor data to the sink once in the data transmission interval. Figure 7b shows a sample of sensor data received in the sink server. During this observation, as the sensors were gradually exposed to brighter light and heat generated from a table lamp and to water vapor from a humidifier, the sensor data values changed accordingly. The flexibility in setting the different values for data transmission intervals for different sensors allows us to optimally use the network resources by reducing the transmission rate of sensor data that do not change often or that are less important for a given application.

The MS can transmit sensor data to one or more sinks through the MSG. Similarly, the sink can obtain sensor data from many MSs. The relationship between the MS, the MSG, and the sink is controlled by the sensor administrator by distributing MS credentials (e.g. hostnames and security keys), which are required to securely establish ID-based communication sessions to collect sensor data at the sink from the MS. The ID-based communication session persists even in the event of mobility of the MS or the MSG.

We have tested the following three mobility patterns:

a) The MS moving from one sensor network to another.
b) The MSG and MS moving together from one access network to another.
c) The MSG moving alone and connecting with a new access network as well as with a new MS.

In (a) we found that it took about 10 seconds for the MS to complete the handover process, which included the MS's network access procedure (eight seconds) and location update with the sink (two seconds). The MS could not upload its sensor data during this interval. However, after that it could continue uploading sensor data using the existing ID-based communication session and the associated security context. Similarly, in (b) it took only about 2.6 seconds for the MSG to complete the handover process. Because it has two upstream interfaces, the MSG could get connected to the new access network while still being connected with the old access network and performed a make-before-break type of handover, resulting in no data loss during the handover. Handover pattern (c), which is a parallel combination of handover patterns (a) and (b), took about 10 seconds for the MS to resume uploading sensor data through the new MSG. Note that, irrespective of the handover patterns, there was no sensor data loss in the network once it was received by the MSG and every data was successfully sent to the sink. Even when the

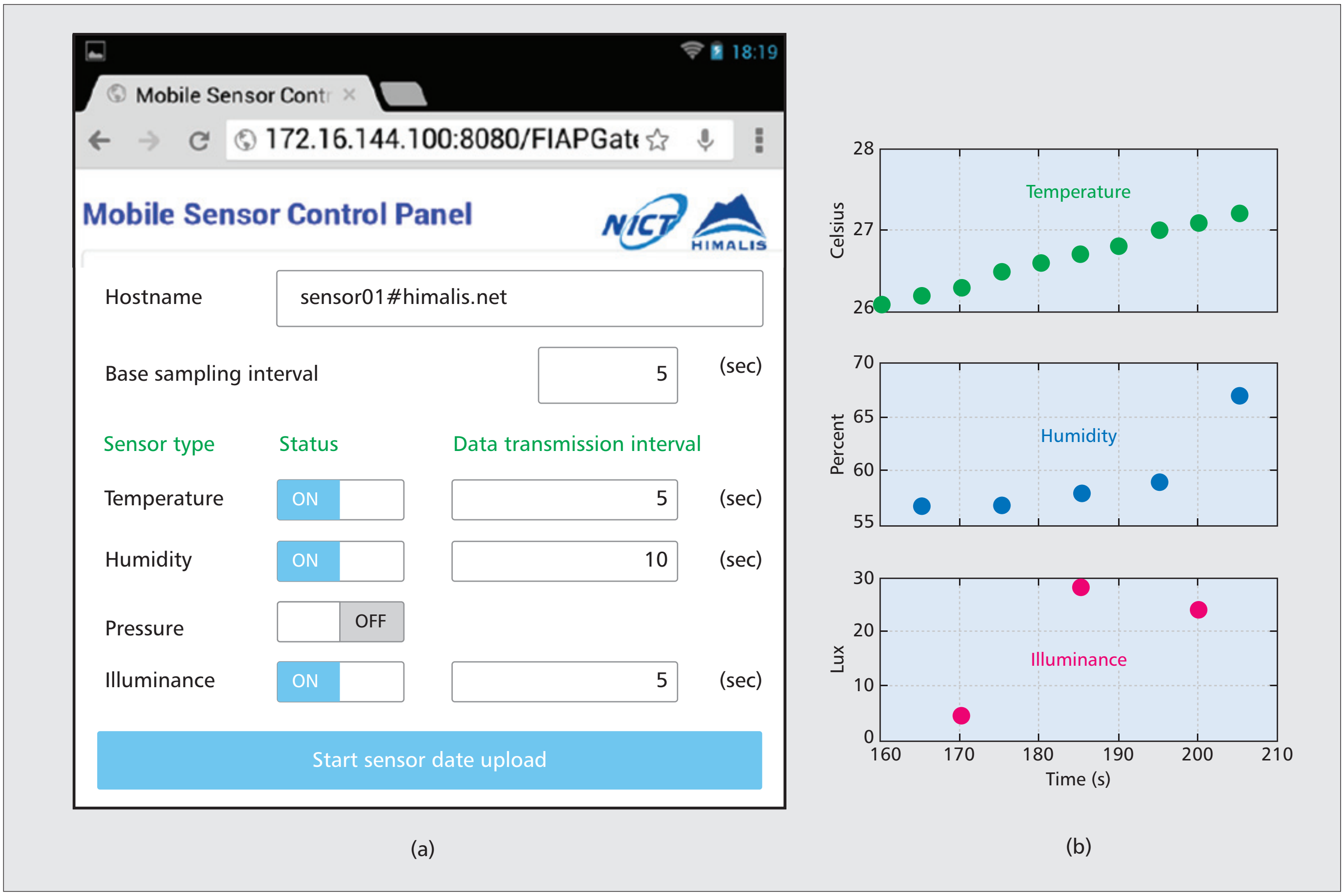

Figure 7. Mobile sensor control panel and sensor data: a) mobile sensor remote control panel; and b) sensor data.

MSG has only one upstream interface, it can cache sensor data for a while, until it completes the handover process, and then forward it to the sink.

Prospective Applications

We envision the following two prospective applications of the proposed dynamic mobile sensor network.

Self-Healthcare —As the proposed platform easily supports the addition of new sensor modules to the mobile sensor nodes, new sensors can be added to read human body parameters, e.g. blood pressure, insulin level, and heart or lung conditions, besides the ambient light, temperature, pressure, and humidity. A patient can carry the MS along with the MSG to continuously send the sensor data, irrespective of her location and mobility, to the sink, where the self-healthcare application uses these data to evaluate the patient's health condition. In an alternative deployment, the patient can carry only the MS and uses the MSGs installed in homes and public places that people visit frequently. In this case the MS and the MSG authenticate each other using the HIMALIS access control mechanism [9] before enabling the MS to send sensor data to the sink. In this way, the patient's condition can be monitored automatically without requiring the patient to be involved in the measurement or data transmission procedure. If the patient wishes to know her health condition, she can use a client device to establish an ID-based communication session with the sink at any time and download information about herself. If the patient is carrying both the MS and the MSG, the MSG can also be used as the client device to contact the online healthcare service.

Automatic Patient Registration at Hospitals — Another application of the mobile sensor network is the automatic monitoring and registration of patients when they visit a hospital. In this scenario, patients carry the MSs, and hospitals install the MSGs at entrances and in waiting rooms. When a patient enters the hospital premises, the MS accesses the sensor network through the MSG and sends her body parameters to the sink server located in the hospital. The healthcare application running on the sink server evaluates the patient's condition based on the past and current sensor data and distributes the results to the client devices carried by nurses and doctors so that they become aware of the patient's arrival as well as her current health condition in advance. Even if the patient moves around the hospital, her location and body parameters can be continuously monitored. Thus, the implementation of this service would be helpful in effectively reducing delays in patient registration and checkup by providing patient information to nurses and doctors automatically as soon as the patient enters the hospital or roams around the hospital.

Conclusion

This article presented the design and implementation of a dynamic mobile sensor network platform in which users can freely install applications of their choice. The proposed platform can overcome the major drawback of current designs, namely, that configurations are mostly static and difficult to adapt to different networking environments and applications. The proposed platform allows various sensor applications to be easily developed and deployed on the same physical sensor network infrastructure. It is mainly useful for sensing scenarios where sensor administrators and mobile sensor gateways as well as sink servers are located in heterogeneous mobile access networks. It can find applications in self healthcare and tracking of mobile objects such as people, animals, and vehicles.

In the proposed sensor network platform, there may be an interoperability issue because IP addresses are not used in the ID-based applications. However, we have easily addressed this issue by using multi-stack protocols and middleware in the sensor gateways and sinks. We have developed middleware as a wrapper program at the C-library level, which intercepts packets sent by the existing Internet applications and forwards them through the ID-based sockets, and vice versa. Some components of the proposed platform, e.g. ID-based communication, are already in the standardization process in the ITU-T Study Group 13, and the other components can be submitted for standardization in the near future.

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Biographies

Ved P. Kafle (kafle@nict.go.jp) is a senior researcher at the National Institute of Information and Communications Technology (NICT), and concurrently holds a visiting associate professor's position at the University of Electro-Communications, Tokyo. His research interests include future Internet, naming and addressing, ID/locator separation, integration of heterogeneous network protocols, integration of resource-constrained sensors into the Internet, distributed mobility management, and privacy, security and trust in communication networks. He received an award from the ITU Association of Japan in 2009, and two Best Paper Awards (second prize) at the ITU Kaleidoscope Academic Conferences in 2009 and 2014. He holds the M.S. degree from Seoul National University and the Ph.D. from the Graduate University of Advanced Studies. He is a senior member of IEEE and a member of IEICE.

Yusuke Fukushima received the Ph.D. degree in information science from Tohoku University, Japan in 2009. He is currently a researcher at the National Institute of Information and Communications Technology (NICT). Before joining NICT, he briefly worked as an assistant professor at the Graduate School of Information Science from Tohoku University, and then as a research associate on the Faculty of Science and Technology of Sophia University, Japan from 2010 to 2012. His research interests include fault-tolerant routing control, parallel and distributed systems, and designing future network architecture. He is a member of ACM and IEICE.

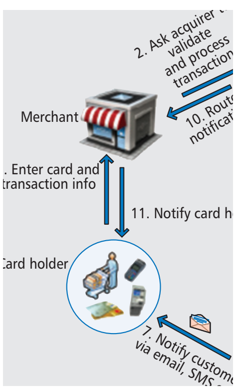
Hiroaki Harai received M.E. and Ph.D. degrees in information and computer sciences from Osaka University, Japan in 1995 and 1998, respectively. He is currently a director at the National Institute of Information and Communications Technology (NICT), Tokyo, Japan, where he is leading the Network Architecture Laboratory for Optical and New-Generation Networks. He is concurrently a visiting associate professor at the Japan Advanced Institute of Science Technology, Ishikawa, Japan. His current research topic is the design and development of new generation network architecture. Dr. Harai was named an Outstanding Young Researcher in the 3rd IEEE ComSoc Asia-Pacific Young Researcher Award, 2007. He received the 2009 Young Researcher Award from the Ministry of Education, Culture, Sports, Science and Technology. He is a member of IEEE and IEICE.

The proposed platform can overcome the major drawback of current designs, namely, that configurations are mostly static and hard to be adapted to different networking environments and applications.

Enhancing Communication Adaptability between Payment Card Processing Networks

Financial organizations communicate with each other using ISO 8583 or its derivatives to complete the request and response cycle of card originated transactions originated from an ATM, POS, or the web. ISO 8583 is a broad standard. Its implementations slightly vary due to the flexibility available within the standard. The authors discuss the problem of adaptability of communication between payment card transaction processing entities due to this flexibility.

Adnan Noor Mian, Abdul Hameed, Muhammad Umar Khayyam, Farooq Ahmed, and Roberto Beraldi



Adnan Noor Mian and Abdul Hameed are with Information Technology University.

Muhammad Umar Khayyam is with TSSG Waterford Institute of Technology.

Farooq Ahmed is with National University of Computer and Emerging Sciences.

Roberto Beraldi is with Sapienza University of Rome.

[1] *3-D Secure* is branded as *Verified by Visa* by VISA and as *MasterSecure* by MasterCard http://www.visaeurope.com/en/cardholders/verified_by_visa, http://www.mastercard.com/us/personal/en/cardholderservices/securecode/index.html

Abstract

Financial organizations communicate with each other using ISO 8583 or its derivatives to complete the request and response cycle of card originated transactions originated from an ATM, POS, or the web. ISO 8583 is a broad standard. Its implementations slightly vary due to the flexibility available within the standard. This paper discusses the problem of adaptability of communication between payment card transaction processing entities due to this flexibility. We first provide an overview of different variations in ISO 8583 implementations and identify the interoperability issue based on our industry experience. We then propose a solution and suggest a way to standardize different implementations so that one organization can communicate with another without or with minimal changes in software. The suggestion is based on the exchange of meta data indicating how the target system is interpreting the header fields during the communication process.Finally, we discuss the benefits of the solution in which the vendors could not only avail the customization flexibility provided by ISO8583 but also ensure that their implementation of the standard is interoperable with others. This reduces the cost of interconnectivity with other partners when a network wants to expands its business.

Introduction

Financial transactions originating from credit, debit, gift, or other typed of cards are considered card-originated transactions. These transactions may include purchase, withdrawal, deposit, refund, reversal, balance inquiry, payments, inter-account transfers, etc. Such transactions are made via ATM, POS (Point of Sale) machines, web, and mobile applications, etc. A customer/cardholder inserts her card in an ATM or swipes it in a POS device and invokes a transaction after providing the necessary information, e.g. transaction amount and PIN (Personal Identification Number). Alternatively, the same information can be provided through a website or mobile-based application. Such user action initiates a card-based transaction that typically travels through a number of systems for its processing and results in either success or failure. Several entities are involved in the processing of the transaction and collectively constitute the payment processing chain. A message being exchanged between different systems carries information about the type of transaction, the card number, the merchant's information, requested amount, PIN, and so on. This is a type of two way communication. System A requests system B and system B responds to the transaction accordingly to system A. The response message contains similar attributes and a response code needs to be returned describing the status of the transaction as either approved, failed, or partially approved.

Figure 1 shows the interaction of entities and the sequence of steps involved in the validation of a payment transaction. The merchant validates the transaction and forwards a request message to the acquirer (step 2). The acquirer merchant's financial institution (MFI) validates the message and hands it over to the bankcard network, which routes it to the issuer processor/customer's financial institution (CFI) (step 5). The issuer, who holds the user account, processes the transaction request and replies with its status to acquirer (step 6). Additionally, the issuer may send a notification, e.g. through SMS or email to the card holder about the transaction done in his/her account (step 7). The acquirer also notifies the merchant about the status of the transaction (step 9), which then notifies the cardholder.

Since different vendors (e.g. VISA, MasterCard, etc.) provide services for such card-based transactions and different entities are involved in the entire process, a need to standardize message formats and communication flow for the exchange of financial data was seen as early as 1987, and a standard for exchanging card originated transactions was proposed by the International Standards Organization (ISO) under the code ISO 8583 [1]. The first version of it was published in 1987, and it was revised in 1993, 1998, and 2003. While the three revisions have made slight changes to the original draft, the basic message structure and communication flow remains the same. Most of the vendors still base their implementation on the 1987 draft [2].The standard defines the message structure and the communication flow among different entities of the transaction processing chain. Consumers may directly interact with ISO 8583-based devices (in the form of an ATM or POS devices), or their interaction may be routed indirectly through other protocols and standards built on top of ISO 8583 for value-added interaction. For instance, the ANSI X.59-2006 standard [3] and its derivative protocols such as CONSEPP [4] and ESCPS [5] are built on top of ISO 8583 to provide a secure electronic payment system in account-based environments (e.g. in web or mobile applications). Similarly, other protocols such as SET [6] and 3-D Secure[1] have been established for the same purpose with the support of major credit card vendors VISA and MasterCard, while other solutions such as IPS [7] have also been proposed to enhance security and usability. Underneath all these application-level solutions, ISO 8583 is at work.

Most vendors (VISA, MasterCard, etc.) differ in their implementation schemes of the standard. Moreover, standards such as ANSI X.59-2006 also require some customizations of ISO 8583. This does not pose as big a problem as long as every merchant supports only a single vendor. However, solutions such as jPOS [8] have made it quite easy for a merchant to support a diverse set of card-based services through a single integrated POS system. The issue with ISO 8583, however, is that it is a very flexible standard, which is simultaneously both good and bad. The standard defines the message structure and the information content that is necessary for the exchange of financial data along with a provision for custom fields from the flexibility perspective. That is the good part. However, it leaves various implementation details to the vendors, which raises interoperability issues. Hence, the standard provides a lot of flexibility, and that is probably why it has been extremely successful in facilitating various electronic payment schemes over the decades. However, this raises problems of interoperability in the modern dynamic environment when new and improved schemes continue to emerge [2].

Currently, if a financial service organization wants to make its own version of ISO 8583 interoperable with another organization, it has to review the target organization's implementation of the standard and make changes to its own implementation accordingly. For example, an acquirer may put a customized header in between the MTI and length indicator field. In this case the only option for the merchant is to make changes to its own implementation of the standard by incorporating the new header. This is not a scalable option, as the process has to be repeated for any organization that has recently entered the business circle.

In [2] a UML-based modeling technique has been discussed to address the interoperability issues in the VISA card processing network. The system provides a UML-based model for specifying the structure, semantic, and lifecycle of the data to prevent misunderstanding and incorrect population of message fields in the payment processing chain. A monitoring system is deployed that compares the field values of the message in every transaction against the specified values of the model and takes corrective actions if required. The problem with this approach is that a monitoring system is required to monitor and compare fields' values of every message of the transaction. Furthermore, the corrective steps required to address an interoperability issue are not addressed after they are identified.

Interoperability issues within card networks occur frequently and are pushing vendors to look for improved solutions [2]. One such effort is being conducted by ISO itself through another standard, ISO 20022 [9], which is aimed at standardizing protocols for all financial information exchanges, including card-based payments. ISO 20022 provides a common development platform

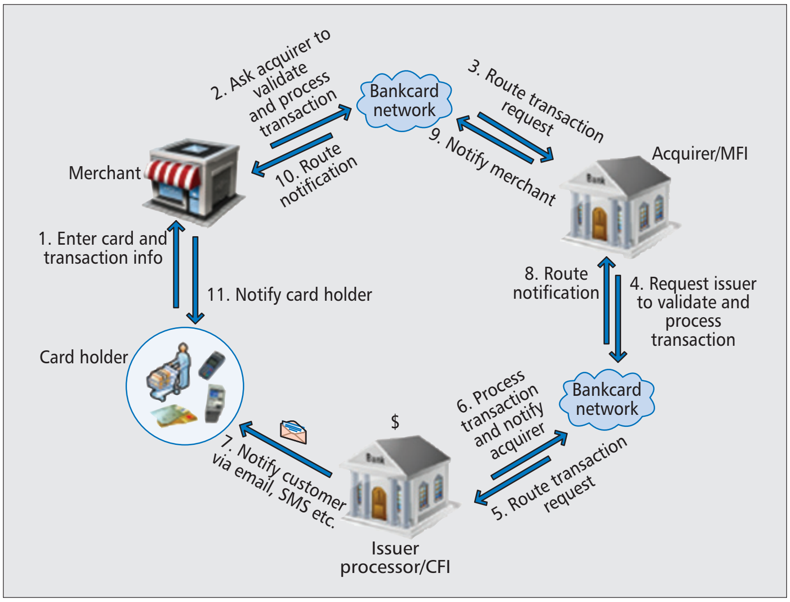


Figure 1. Basic process of a card-originated transaction.

based on UML modeling to define a number of financial business processes. Using this platform, financial institutions can develop a new message exchange protocol for various financial transactions that avoids confusion or misunderstanding of some common concepts. But there are some drawbacks to using this platform. ISO 20022 messages use an XML wrapper and each message carries a fare amount of XML data. Field-level constant information also needs to be shared with every single message and thus needs much more bandwidth. This increased bandwidth requirement of ISO 20022 becomes a bottleneck for millions of real-time transactions. On the other hand, ISO 8583 is simple and more precise, focusing only on the card-originated transactions. Further, ISO 8583 is lightweight and only small-sized bitmaps are exchanged in the messages. Consequently, we see that global adaptation of ISO 20022 is unlikely to occur in the near future. It is thus difficult for a financial institution to start its business based on ISO 20022 because no major payment network is currently accepting ISO 20022-based transactions. Though ISO 20022 may be used as a platform for avoiding misunderstanding of some common concepts in the financial process,it is however a paradigm shift and will take a long time to be globally adapted. The likely scenario is that both ISO 20022 and ISO 8583 will co-exist in order to support card-based payments [10]. The need, therefore, is to propose changes to ISO 8583 to make it more interoperable without compromising the flexible structure of the protocol.

This article is an attempt to identify the interoperability issues and propose changes to ISO 8583 to address these issues.Our proposed solution is based on the inclusion of meta data about the custom fields of the message that help to know how the target system is interpreting and using these fields in the processing chain. By using the current standard and suggesting enhancements to it, ISO 8583 will be become more adaptable, retaining all of its customization

The ISO 8583 standard is a message interchange specification that defines the format for exchanging transactional data in the entire payment processing chain. Transactions can be of various types, e.g. purchase, cash withdrawal, refund, etc.

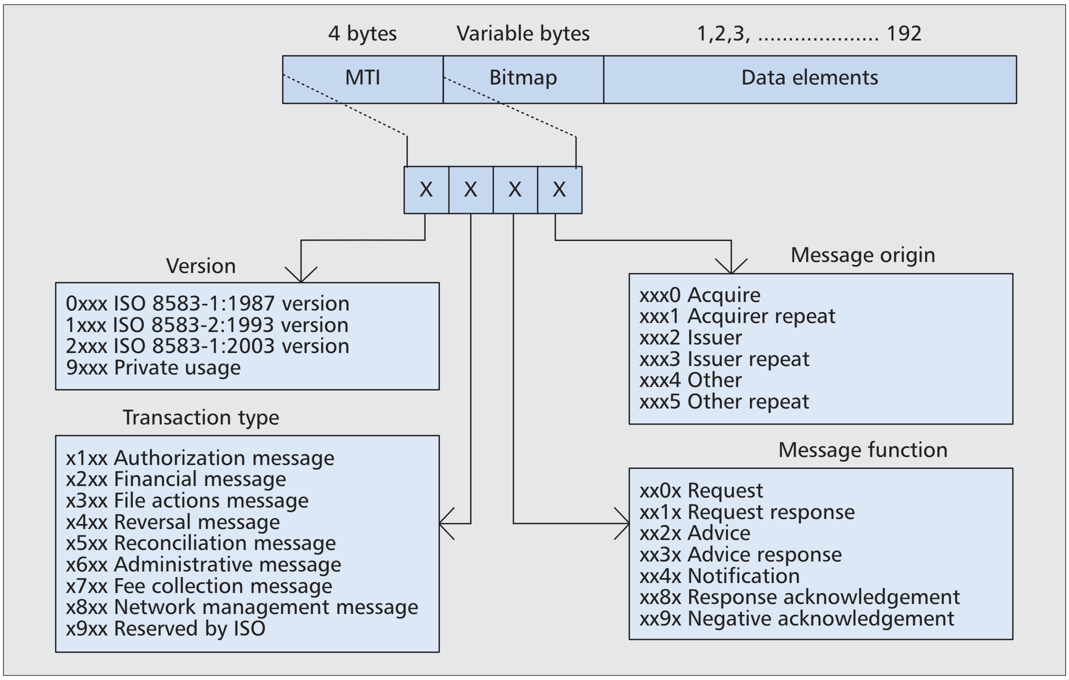


Figure 2. ISO 8583 message structure and MTI field.

flexibility intact. It will reduce the time and resources needed to make changes to existing implementations of the standard that are often needed when an organization wants to expand its network. Moreover, all entities involved in processing card-originated transactions would be on same page, and their software systems would be highly adaptable to welcome any other organization in their communication circle. Further, organizations would be able to focus on their business instead of the plumbing work for connecting with new partners.

The rest of the article is organized as follows. We provide a brief overview of the ISO 8583 message format. Then we present a gap analysis of different ISO 8583 implementations based on our experience in industry. We discuss the proposed solution to address these interoperability issues. We analyze and discusses the benefits and issues relevant to the solution, and we then conclude the article.

ISO 8583 Message Structure

The ISO 8583 standard is a message interchange specification that defines the format for exchanging transactional data in the entire payment processing chain. Transactions can be of various types, e.g. purchase, cash withdrawal, refund, etc. It also carries information regarding the merchant, her type of business, capture method, transaction amount and currency, etc. All of these are supported through the flexible message structure. The basic message structure has three components: MTI, bitmaps, and data elements, as shown in Fig. 2.

The message type identifier (MTI) is a small four digit numeric field that classifies the high-level function of the message. The first digit of the MTI identifies one of the three versions of ISO 8583. The second position indicates the type of transaction, e.g. authorization message, financial message, etc. The third digit of the MTI field indicates the function of the message. The fourth one is the origin of the message. Figure 2 shows various types of MTI messages with respect to the position of each field.

The MTI field is followed by bitmaps that are used to specify which data elements are present in the message considering the nature of transaction. A message should contain at least one bitmap that indicates the presence of data elements 1 to 64. This bitmap is called the primary bitmap. A message may also contain additional bitmaps in the form of a combination of primary and secondary (for data elements 1 to 128), or primary, secondary, and tertiary (for data elements 1 to 192).

Data elements contain the actual information about the transaction (e.g. card number, transaction amount, etc.). Each data element serves a specific purpose and its function is detailed in the standard. There are a total of 192 data elements, but not all are required in order to process a particular transaction. A data element is present only when its specific bit is on in the bitmap. This depends upon the nature of the transaction. Considering that the necessary data elements are made part of the message and their presence is indicated by setting the respective bits in the bitmap, the message length is thus controlled. Figure 3 shows a message with different data elements. The presenceof each data element is indicated by the related bit of the bitmap. Since the last data element present is 62, which is less than 64, only the primary bitmap is required. The absence of other bitmaps is also indicated by the OFF status of the first bit of the bitmap, which is ON only if this bitmap is followed by another one.

Data elements may be either fixed length or variable length. In order to control variable length, the tag length value (TLV) format may be used. It is a series of values of specified length, recognizable by a tag, all written in a single string. For instance, a string such as ad09city42USAag0229gd01M can represent useful information about a person, in this case address, age, and gender. The given string shows that there are three tags, ad, ag, and gd, respectively, representing address, age and gender. The length of the address is nine characters, thus the corresponding value of the address is city42USA. After nine characters, the next tag starts that represents the age whose length is two and hence the corresponding value is 29. Similarly, gender is represented using one character M, meaning male.

The issue with ISO 8583 is the flexibility accorded in the standard with respect to various implementation issues. For instance, the standard does not enforce any specific encoding scheme. A system is free to use any encoding from BCD, EBCDIC, ASCII, HEX, or BITSTRING for different types of its proprietary data fields. Similar flexibility is provided for other representations as well. Systems can use private fields to provide extra information for transaction processing, and different vendors use these in their own way. For instance, MasterCard sends field 48 (reserved for private data) in TLV format. VISA, on the other hand, uses field 48 with different formats depending on the transaction type and consequences. Similarly, some fields of the VISA message format have subfields represented by bitmaps.

Consequently, different vendors have slightly different formats all based upon ISO 8583 and they provide their own customized versions of the standard for development purpose. These customized versions are generally proprietary and not made available to larger audience. These customizations raise interoperability issues.

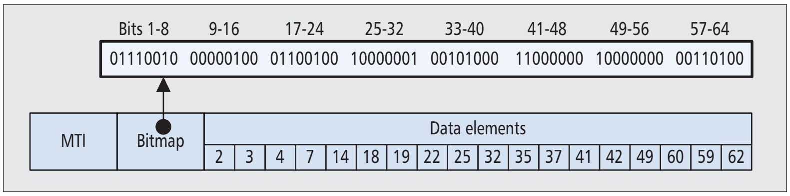


Figure 3. Example bitmap and data elements.

Gap Analysis

Based on our experience implementing ISO 8583 based solutions over the years for various network types including (but not limited to) MasterCard and VISA, we have done a gap analysis to find out where these implementations differ. We classify these implementation differences broadly into two categories: message level and field level.

Message Level Gaps

Message level gaps are the gaps that are related to the whole message and are not concerned with individual fields. Examples of such gaps are:

• According to the standard, the first two bytes of the message indicate the message length. This is followed by the MTI and the rest of the message. However, there is flexibility to place any informed data in between the message length and the MTI. Different vendors are free to add some reserved bytes in the message length and the MTI before transporting the message.

• A header may exist or not before the MTI. If it does, how can its length be specified in a standard way? An example of the header is a reject header indicating that the message received by another system is not in the correct form or information is not correct.

• The encoding of the messages may differ. It can be EBCDIC, ASCII, or any other encoding scheme. It may also vary by fields and subfields.

• The bitmap can be either packed or unpacked. An unpacked bitmap requires a byte each in order to represent the presence or absence of respective data elements. Thus 192 bytes will be required to represent the entire bitmap. A packed bitmap, on the other hand, uses one bit to represent the same information, thus reducing the size of the message.

Field Level Gaps

Field level gaps pertain to differences in representation of the data elements. They generally arise due to length and encoding issues. Examples of such gaps are:

• In case of variable length fields, a length is specified first before its content. Differences arise in the interpretation of length itself as to what it represents: the number of bytes or the number of elements in the field. In the case of bytes as a unit, length 8 means 16 values of some field of encoding hex. If the length is measured in the number of elements in the field, 16 in the length indicates 16 hex values and eight bytes of the original message. Even in a single message the length unit could vary. It can be either in bytes, in data per byte, or both (according to fields).

• A field can be a straight, TLV, or bitmap field. Straight means a simple data field with a constant number of sub-fields. TLV is discussed earlier in this document (section 2). A bitmap field is a field with a bitmap in its beginning, indicating the presence or absence of subfields in it. Multi usage is a field that may vary in its attributes or category in different transactions according to some context or condition.

• Moreover, the character set of a field can be numeric, alpha numeric, or alpha numeric special. Also, its encoding may vary from EBCDIC, ASCII, to others on a field to field basis.

Suggested Solution

As we have already seen, different networks may differ in their implementation of ISO 8583 and hence one vendor may send the information in a slightly different format while routing the information from one network to other. This will make interpretation of the message difficult. Our proposed solution is to keep the vendors flexible in using the standard in their own way as they do currently, but in order to make communication easy, we suggest sharing some meta information that can be made part of the standard. We propose two forms of meta data sharing: handshake level and transaction level.

Handshake Level Meta Data

Handshake level meta data would contain information for variations among systems both at the message level and the field level, only needing to be indicated once because in all upcoming messages that would remain same. Such metainformation may represent message level variations, as follows.

- **Message Beginning/Ending Symbol:** Currently, ISO 8583 does not suggest any symbol to represent the start or end of the message. It just asks for putting the length of the incoming message in the beginning of every message in two bytes. In the case of a noise in the message, the whole queue of messages for the transaction becomes disturbed. The issue could be solved by indicating the message boundaries through start and end symbols.
- **Reserved Bytes:** How many reserved bytes does a vendor send in the beginning? The number of bytes of proprietary data being used between the message length and MTI field should be indicated.
- **Presence of Header:** Is there any header present? If yes, then how much length does it have?
- **Encodings Used:** How is the message encoded? Which option (BCD, EBCD, ASCII, etc.) is used?
- **Bitmaps Information:** Does the message use packed or unpacked bitmaps? The type of bitmap should be indicated in the meta data.

Table 1 provides one possible format for representing these message level and field level meta indicators, but other ways may also be worked out. Similarly, meta information for field level variations, especially for private use fields, may include:

- **Length Unit:** What is the length unit for a specific private field? Is it fixed or variable? If variable, does the specified length represent the number of bytes or the number of elements?
- **Data Format:** Does it use the straight, TLV, bitmap, or multi usage format?
- **Character Set:** Does it use the numeric, alpha numeric, or alpha numeric special character set?
- **Encodings:** Which encoding scheme (BCD, EBCD, ASCII, etc.) does the field use?

The field level indicators are described in Table 1. Figure 4 shows an example of a header that could be used for exchanging the suggested meta indicators for message and field level during the handshake process. Meta indicators are shown by the dotted rectangle boxes within the message. On the receiving end, the header could be parsed to extract each item on the basis of the position and size of the item. The first part of the header is the message level meta data followed by field level meta data separated by a semi-colon (;) symbol. It should be noted that a header may contain meta indicators for multiple propriety fields separated by a semi-colon (;) symbol.

Transaction Level Metadata

Transaction level meta data would contain information for variations that need to be indicated in every message because they can vary from transaction to transaction. Such information includes the following cases:

- **Reject Header:** Is there any reject header present? If yes, then what is its length?
- **MTI:** Where does the MTI lie? Flexibility of putting reserved bytes and headers plus reject headers makes the MTI lost until one parses the headers and reserved bytes first. But if one gets the MTI position or offset, one can proceed to parse the complete data elements from the message accordingly.
- **Field Data Format:** If the multi-usage field

Level	Indicator	Symbols and description
Message	Message beginning/ ending symbol	! marks beginning and ending of message
	Reserved bytes	XX where X represents a digit. Hence, 02 may represent two reserved bytes
	Header	XX where X represents a digit. Hence, 00 may represent absence of a header, where any other representation would give length of header in bytes, e.g. 10 means a 10-byte header
	Encodings	E for EBCDIC A for ASCII B for Binary V for Variable H for Hex 4 for 4-bit BCD
	Bitmaps	P for packed bitmap U for unpacked bitmap
	MTI	Offset of MTI
Field	Field ID	XXX where X represents a digit. For instance, 048 will refer to field number 48 for which the rest of the information is defined by the following indicators
	Length type	F for fixed length V for variable length
	Length unit	E for number of elements B for number of bytes
	Data format	S for Straight M for Multi-usage T for Tag Length Value B for Bitmap
	Field character set	N for Numeric A for Alphanumeric S for Alphanumeric special
	Field encoding	E for EBCDIC A for ASCII B for Binary V for Variable H for Hex 4 for 4-bit BCD

Table 1. Message and field level indicators.

format is used, then what usage is available in the current message.

Figure 5 shows an example header format for indicating transaction level variation. The message level and field level indicators are separated by a semi colon (;) symbol. Multiple field level meta data can be carried in the header. The indicator for a reject header is a two digit number where 00 represents the absence of a reject header; otherwise, it indicates its size. The size of the reject header is followed by offset of the MTI indicating the start of MTI in the message.

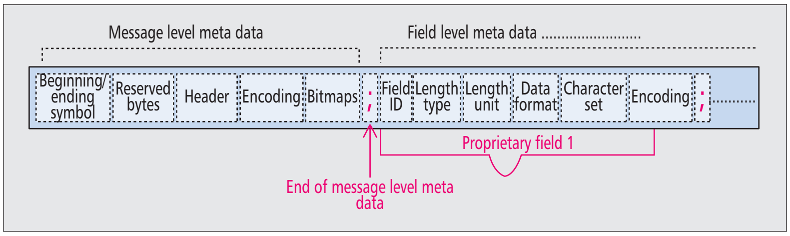


Figure 4. Handshake level meta data header.

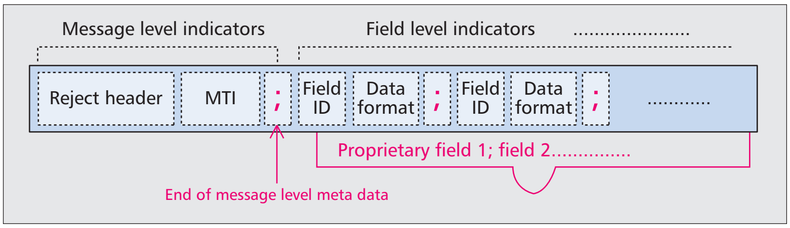


Figure 5. Transaction level meta data header.

Discussion

The solution proposed later has the following characteristics:

- Systems would have to share/receive meta data of private fields and customizations.
- For the first message at the sender side, the system would have to package and send information about handshake level and field level gaps to the receiver.
- For the first message at the receiver side, the system would have to understand the message, parse the information about handshake level and field level gaps, and save the information on a permanent storage.

This raises two issues when compared with existing implementations of ISO 8583: efficiency of the system and effort required to implement proposed changes.

Since extra information will need to be shared as metadata between the two systems for the solution to work, a question on efficiency can be raised. Considering the proposed message structure for metadata and encoding schemes presented in Fig. 4 and Fig. 5, there will be the addition of a few bytes only, and since most of this metadata will be exchanged once at the handshake level, there will be negligible impact on efficient transmission of transaction data.

Another important issue is the implementation effort required. This is surely a problem as many implementations of ISO 8583 are currently in use and will require changes. However, the changes itself are not very complex and can be easily implemented with little effort. This will be far more cost effective when compared with alternatives being explored to the interoperability problem in the form of other standards such as ISO 20022 [9], etc. In fact, these are the bankcard networks who publish their specifications of the customized ISO 8583 standard version. The other business entities, such as issuers, merchants, and processors, follow the specifications. So in order to achieve the objectives, first the bankcard networks such as MasterCard, VISA, Discover, American Express etc. have to agree on same page. They can document the final list of gaps and propose the meta data to be shared in the first message to issuers, merchants, or acquirers. Thus merchants, acquirers, and processors would be responsible for making changes in their system to receive that meta data in the message and store it in permanent storage to understand the upcoming messages accordingly.

Another possibility (even more suitable) is that instead of bankcard networks, ISO 8583 can impose the sharing of the meta data addressing all the gaps in the specification of a bankcard network. This will result in a new version of ISO 8583, also supporting previous versions for backward compatibility. Once adopted as part of the standard, the stakeholders would have to study the gaps in inter system implementations and comply with the proposed solution. Although it would require an additional effort to comply with the requirement of understanding or sharing the metadata, such an effort will result in the following benefits:

- It would reduce the cost of studying and incorporating the gaps for every new partner, resulting in fewer resources required in the form of time and effort.
- The system would be more reliable since handling different implementations for all entities involves multiple risk factors regarding implementation maintenance.
- All the entities involved in processing card originated transactions would be on the same page and their software systems would be highly adaptable to welcome any other organization in their communication circle.
- Organizations would be able to focus on their business instead of the plumbing work for connecting with new partners.

Conclusion

ISO 8583 is a widely used standard in financial transaction processing, but when a financial organization wants to expand its business network to some other organization, it requires making changes in its communication protocol with respect to all the customizations of the target system. Some changes in the standard can make it more interoperable so that once implemented, it can be used for any target system using th same ISO 8583 standard without losing the flexibility it provides. After doing gap analysis of different implementations of ISO 8583 based on our experience in industry, we identified certain fields that raise interoperability issues among different implementation of ISO 8583. To enhance the adaptability in the imple-

To enhance the adaptability in the implementations of this standard, we suggest a solution based on the exchange of meta data indicating how the target system is manipulating the custom fields.

mentations of this standard, we suggested a solution based on the exchange of meta data indicating how the target system is manipulating the custom fields. Two types of headers, handshake level and message level, can address the customizations of the standard being practiced. It would enhance the flexibility of communication and reduce significant waste of resources in porting existing implementations considering interoperability requirements.

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Biographies

Adnan Noor Mian (adnan.noor@itu.edu.pk) is currently working as an assistant professor in the Department of Computer Science at Information Technology University, Lahore, Pakistan. He has a Ph.D. in computer science and engineering from the University of Rome, "La Sapienza," Rome, Italy, and holds master degrees both in physics and computer science. His research interests include distributed algorithms, network protocols, pervasive computing, ad hoc and sensor networks, and application of AI techniques in these areas.

Abdul Hameed (abdul. hameed@itu.edu.pk) is a Ph.D. candidate in the Computer Science Department at National University of Computer and Emerging Sciences, Lahore, Pakistan. He has a master degree in computer science from National University of Computer & Emerging Sciences, Islamabad, Pakistan. His research interests include automation in network management, optimization, switching, network planning and diminishing, software defined networking, and the use of artificial intelligence techniques for computer network problems.

Muhammad Umar Khayyam (m.umar.khayyam@gmail.com) received his master degree from National University of Computer & Emerging Sciences in 2010, and an undergraduate degree in 2006 from COMSATS University, Lahore, Pakistan. Since 2006 he has been working in the software development industry in Pakistan and other countries. His area of interest is research and development of robust, scalable, maintainable and reliable real-time and enterprise software systems. He is currently working with the TSSG research group at Waterford Institute of Technology, Ireland.

Farooq Ahmad (farooq.ahmad@nu.edu.pk) received his master degree from National University of Computer and Emerging Sciences, Lahore, Pakistan in 2006. After that he worked as a software developer in the CS industry. Since 2011 he has been working as an assistant professor in the Computer Science Department of the same university, and he is also a Ph.D. candidate. His areas of interest are wireless ad hoc networks, distributed networks and protocols, and robotics.

Roberto Beraldi (roberto.beraldi@dis.uniroma1.it) received his Ph.D. in computer science from University of Calabria, Italy in 1996. Until 2002 he worked at Italian National Institute of Statistics. Since then he has been with the Department of Computer, Control, and Management Engineering at Sapienza University of Rome, Italy. He has published more than 70 papers in journals and main conferences related to wireless networks. His current research interest includes mobile and cloud computing, wireless networks, and distributed systems.

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