February 2009, Vol. 47, No. 2

IEEE Communications MAGAZINE

LTE Part I: Core Network

Advances in Cooperative and Relay Communications

Optical Communications

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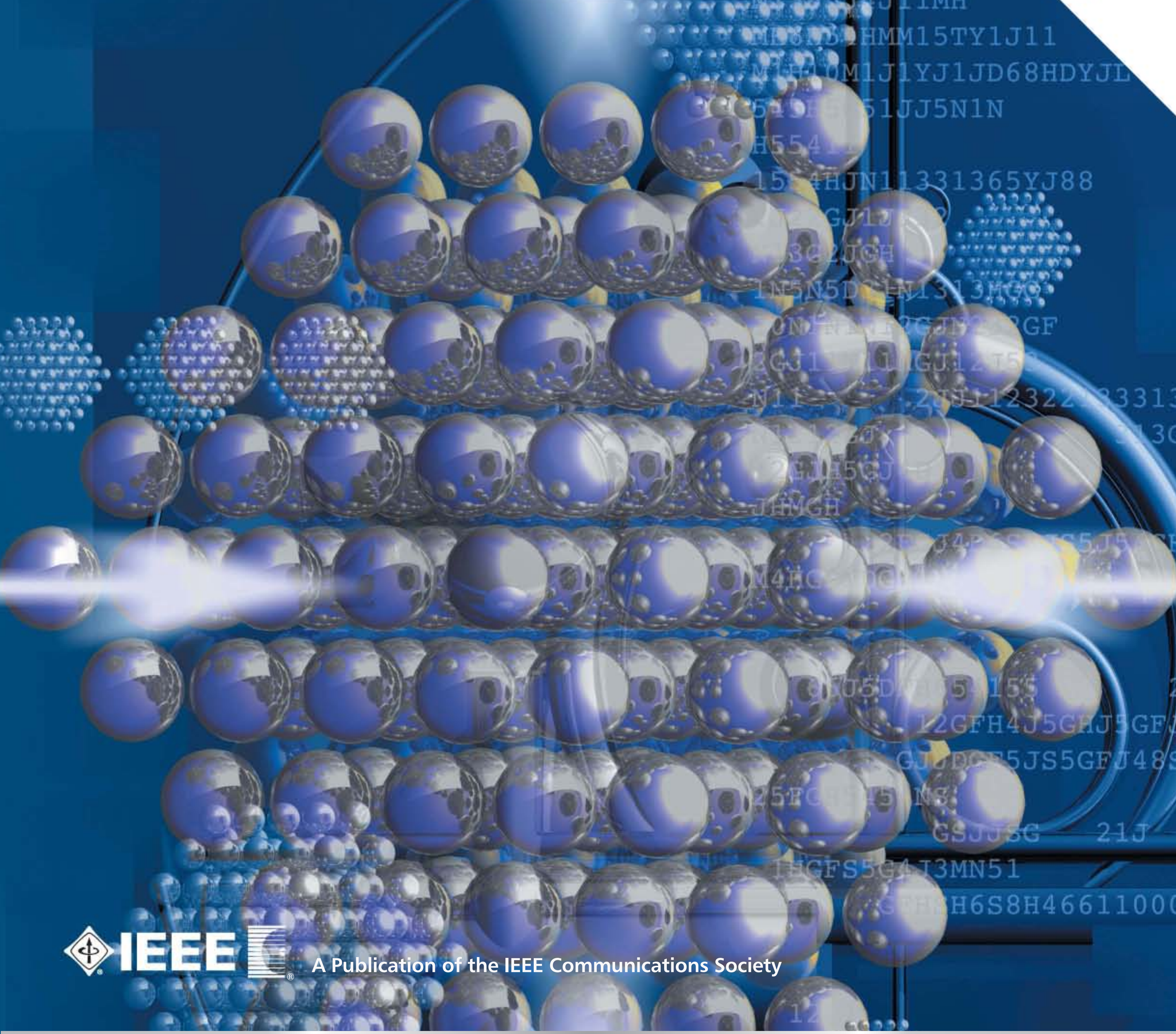
OFC/NFOEC 2009 THE FUTURE OF OPTICAL COMMUNICATIONS IS HERE



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IEEE Communications MAGAZINE

February 2009, Vol. 47, No. 2

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LTE Part I: Core Network

Guest Editors: Kalyani Bogineni, Reiner Ludwig, Preben Mogensen, Vish Nandlall, Vojislav Vucetic, Byung K. Yi, and Zoran Zvonar

Advances in Cooperative and Relay Communications

Guest Editors: Woo Hau Chin, Yi Qian, and Giovanni Giambene

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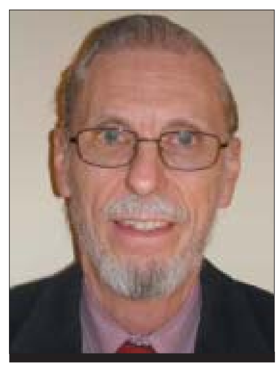
CERTIFICATION: MEETING AN INDUSTRY NEED

DOUG ZUCKERMAN

One of the challenges ComSoc faces is how best to meet the needs of its wide range of constituents from around the world, which includes researchers and practitioners in industry, academia and government, as well as the students and young engineers who are ComSoc's and humanity's future. With support from the IEEE New Initiatives Committee, the Society has been developing and recently launched a trial "certification" program aimed at meeting industry needs for certifying the key skills of engineers who work in the wireless communications area.

The certification program was initially conceived and pursued by Pierre Perra and Celia Desmond. Subsequently, Celia and Rolf Frantz took on responsibility for developing the concept into a viable program. It is my great pleasure to introduce Celia and Rolf, who will tell us more about ComSoc's certification program.

CELIA DESMOND



ROLF FRANTZ

Celia Desmond, President of World Class - Telecommunications, which provides training in telecommunications management, has lectured internationally on programs for success in today's changing environment. As Director - Industry Liaison for Stentor Resource Center Inc., she was the corporate external technical linkage and was instrumental in establishing culture and new processes for service/product development and for project governance, including obtaining employee buy-in. At Bell Canada, Celia provided strategic direction to corporate planners, ran technology/service trials, standardized equipment, and provided technical and project management support to large business clients.

Celia has held many significant IEEE, Communications Society and other leadership positions, including IEEE Director and Secretary, IEEE VP Technical Activities, ComSoc President, IEEE Canada President, IEEE Region 7 Director, IEEE Division III Director, and IEEE Canada Foundation Board Member and Donations Chair. In recognition of her sustained contributions, Celia was awarded the Donald J. McLellan Award for meritorious service to the IEEE Communications Society, the Engineering Institute of Canada John B. Sterling Medal in May 2000, and the IEEE Millennium Medal. She is a Senior Member of IEEE and holds a M.Sc. Engineering, B.Sc. Mathematics & Psychology, Ontario Teaching Certificate and PMP certification. Celia has taught kindergarten, high school, and university at Ryerson School of Business, Stevens Institute of Technology, and the University of Toronto. She is also the author of *Project Management for Telecommunications Managers* (Springer).

Rolf Frantz has over 35 years of experience in the telecommunications industry with Bell Laboratories and Bellcore/Telcordia Technologies. His responsibilities have ranged from product design and development, reliability testing, solving manufacturability problems, and conducting technical analyses on a wide range of telecommunications products, to applications-oriented research projects and project management. He has worked in the areas of telecommunications power, alternate energy sources, field testing and reliability, fiber optic communications, and most recently with wireless communications. Actively involved in standards activities, he has been a member or chair of a half-dozen TIA and IEEE standards committees. Rolf is retired from Telcordia, where his last responsibility was as the technical program manager for the

FIGURE 1: *Practice Analysis Task Force.*

FIGURE 2: *Industry Advisory Board.*

National Fiber Optic Engineers Conference. He currently works as a consultant to the IEEE Communications Society, serving as Industry Relations Manager for the Wireless Communication Engineering Technologies (WCET) certification program. He holds B.S. and M. Eng. degrees from Cornell University and a Ph.D. from Brown University.

BACKGROUND

IEEE Communications Society members live in many countries and work in many different areas of communications. A majority of these members work in industry, and the products and services that ComSoc offers must provide value to these members. ComSoc leadership recognized that more could be done to benefit these current and potential members. There also is little available from the Society today directed specifically at companies — another gap that ComSoc should fill. A certification program was identified as a way to bring value both to the individuals who want to improve their knowledge of key topics in their profession and to the companies that employ these highly-qualified people.

The choice of technical area for focusing the initial certification program was more difficult than the decision to develop certification. Broadband, multimedia, the Internet, and wireless are all hot topics in communications today. Any of these would have been a good choice for a certification program, and more programs will probably surface in the future as people find value in this first program. Wireless was chosen as the initial topic because wireless technologies have been evolving quickly, and growing just as quickly, over the past five to 10 years, and this growth is predicted to continue for years to come. Employers are having difficulty finding enough qualified employees to fill open positions for assessing, designing, developing, and implementing their products and services. Certification will help companies with this and will also assist individuals who want to demonstrate their qualifications to fill these positions.

PRACTICE ANALYSIS

The wireless certification development started with the convening of a Practice Analysis Task Force (PATF) in December 2006 (see Figure 1).

Sixteen professionals from around the world brought a wide range of expertise to the task. In view of the focus on industry and practice, it was important that they all had practical on-the-job experience, enabling them to focus on two key aspects: what wireless practitioners do, and what they need to know to do it. After two days of intense discussions, they developed a draft Delineation — a description of seven technical areas that cover the breadth of the work done by wireless professionals. Each area is described in detail by a series of typical tasks, supplemented by statements of the essential knowledge needed to perform those tasks.

This draft Delineation was discussed at length in focus groups held during industry conferences and IEEE meetings in Europe, Asia, and North and South America. Detailed notes and worksheets generated by the participants in these sessions were provided to the PATF as feedback. In addition, a dozen industry experts each did a detailed review of the Delineation and returned marked-up copies with additional comments for the PATF to consider when it reconvened in May 2007. The revised Delineation resulting from this input from over 100 wireless professionals became the basis for the future development of the WCET certification program.

The Delineation was then validated via an industry-wide survey. Over 1,300 wireless professionals reviewed the seven areas, the specific tasks, and the knowledge statements. Their feedback confirmed that the Delineation accurately described both what wireless practitioners do and what knowledge they need to perform those tasks. Their ratings of the importance of specific tasks and the frequency with which they are performed provided weightings for the seven technical areas, which were used when creating the examination: the more important certain types of tasks are, and/or the more often they are performed, the more questions on the examination are allocated to these topics.

INDUSTRY FEEDBACK AND PARTICIPATION

With this description of practice finalized, the next step was to begin development of a certification examination, along with such supporting materials as a Candidate's Handbook, a guide for organizations wishing to develop training to assist in preparing for the WCET exam, and an overview text covering the wireless field, the Wireless Engineering Body of Knowledge, or WEBOK for short. These efforts needed support in several ways: marketing the program, establishing policies (e.g., for administering the exam and maintaining certification), and building the necessary Information Technology infrastructure to support online application, coordination with IEEE and ComSoc databases, etc. These activities led to the involvement of dozens of wireless professionals — writing exam questions, reviewing and selecting questions to create the exam, drafting the Handbook, authoring and editing material for the WEBOK, developing marketing materials, giving presentations to interested groups, considering the implications of various practices and policies, overseeing the IT efforts, and in many other ways.

In addition, over a dozen executives from leading carriers, manufacturers, regulatory bodies, and other industry segments have joined an Industry Advisory Board (IAB) which was created to provide continuing feedback for the program.

Are there new technologies that should be added as exam topics? Are there technologies that are fading from use and should be removed from the exam? Are WCET credential holders meeting the needs of the industry? Are there policy changes that should be considered to keep the program current, focused, and valuable to the wireless industry? These and other questions are being discussed by and with the IAB members in meetings, via emails, and in conference calls.

RESOURCE MATERIALS

A number of resources have been developed to assist prospective candidates for WCET certification, including a Candidate's Handbook, a Web site, the *IEEE Wireless Communications Professional* bi-monthly e-newsletter, training programs developed by organizations supporting the wireless industry, the practice exam, and the WEBOK.

The free Candidate's Handbook, now in its second issue, contains all of the information needed to apply for the exam, including a list of testing sites worldwide where the exam is offered. It includes a complete copy of the Delineation, some sample references, and even a few sample exam questions.

The WCET Web site (www.ieee-wcet.org) is the source for information about the program, and includes links to the online application and through which candidates can order a Handbook, subscribe to the e-newsletter, or order a copy of the WEBOK.

Subscribing to the *IEEE Wireless Communications Professional* is a great way to keep up with events in the wireless industry. In addition to current information about the WCET program, it includes news about patents and new products, mergers and acquisitions, and standards and regulatory developments.

A number of organizations have approached us about offering training programs to assist candidates in preparing for the WCET exam. While ComSoc does not evaluate, endorse, or recommend any training program, those that we are aware of are listed on the Web site as a service to prospective candidates.

The Practice Exam consists of 75 questions that were developed at the same time and with the same methodology as was used to create the certification exam. It can be purchased online for $75 and can be taken up to four times to enable candidates to assess their preparedness for the certification exam.

The WEBOK is not a study guide for the exam, but it does provide an overview of each of the seven technical areas covered, along with an extensive list of references in which detailed wireless knowledge can be found. It can be ordered through the WCET Web site or directly from John Wiley publishers.

EXAMINATION

All of these efforts culminated in the first WCET certification examination, which was offered during a testing window from 22 September to 10 October 2008. The examination consisted of 150 multiple-choice questions, each of which was carefully reviewed for accuracy and relevance and which was closely tied to a readily-available technical reference in the field. This same care was used later in the Fall of 2008 to collect and review additional questions that were incorporated into the examination for Spring 2009. This ongoing process of creation, review, and revision is essential to keeping the examination fresh and up-to-date with current technologies and practices in the wireless industry.

We were pleased to see that over three-quarters of the candidates who took the Fall 2008 exam passed and have received certificates attesting to their status as IEEE Wireless Communications Professionals, WCP, the formal name of the credential. The Spring 2009 exam will be offered from 16 March through 4 April. The Fall testing window is 12-31 October, and the application period begins on 6 July. We encourage wireless practitioners to visit the WCET Web site, order a Handbook, and apply for the exam.

THE FUTURE

ComSoc has already been receiving inquiries and even requests regarding certification programs to address other areas. If the initial WCET program proves to be of significant value to the industry, it would be quite straightforward to copy the WCET methodology to develop additional programs. In view of this, initial feasibility evaluations have been initiated. One candidate program would be a wireless certification that would require much more in-depth knowledge of a key area (as opposed to WCET's broad and relatively high-level approach). Another option would be a certification addressing a distinctly different technical area such as broadband. Volunteers will be essential to assist with the development of any of these new programs. Anyone who is interested should send a short email to cert@comsoc.org indicating which certification program they believe would be most valuable, why, and how they'd be willing to help.


Life

LOOKING FORWARD, LOOKING BACK

BY ROLF FRANTZ

IEEE Communications Society
Wireless Communication Engineering Technologies

Although this column appears in the February issue, it is being written in early January, a time when it is common to both look back and look ahead.

It is hard to believe it has been almost two years since this column first appeared in March 2007. At the time, the first Focus Groups had just provided feedback on the delineation of technical areas, tasks, and knowledge developed by the Practice Analysis Task Force (PATF) to describe the broad practice of Wireless Communication Engineering Technologies (WCET). More Focus Groups, feedback from Independent Reviewers, industry-wide surveys, and additional meetings of the PATF led to a final delineation, a comprehensive description of what practicing wireless communication professionals do and the knowledge needed to do it. This delineation became the basis for the WCET certification examination, which was administered for the first time this past Fall. Dozens of industry experts volunteered their time and effort to write exam questions, review and revise them, and compile the exam.

Simultaneously, other volunteers were hard at work creating a Candidate's Handbook to provide detailed information such as how to apply for the exam, what to expect when taking it, and how the results will be reported. Still others were writing and editing the Wireless Engineering Body of Knowledge (WEBOK), a survey of wireless technology and practice soon to be published by John Wiley. The WCET Web site was created as a source for the most current information about the program. A bi-monthly e-newsletter, *IEEE Wireless Communications Professional*, was initiated and has attracted thousands of subscribers worldwide. Promotional materials were developed, presentations were given, and industry leaders joined the Industry Advisory Board (IAB). In short, a very busy two years have led to the point where the program is solidly established and poised to become a benchmark for individuals and companies to assess competence in the wireless communications industry.

In this regard, Todd Stockert noted that one reason he took the exam was "to provide an independent assessment of my value to my employer ... (and potentially their customers)." And Mukarram Patrawala noted, "Certification proves to prospective employers (internal and external) that you possess fundamental technical competence and expertise." We congratulate Todd and Mukarram for passing the exam.

The level of activity isn't slackening. The Spring and Fall 2009 exam windows have been scheduled; the 2009 Candidate's Handbook has been issued; the January e-newsletter is being distributed; and plans are underway for future webinars and presentations at industry forums. As this column is being written, volunteers are meeting to review the Spring exam and establish a standard for the performance that will be required to pass. A few days earlier, members of the IAB met and discussed plans to further strengthen the program. The Board identified additional benefits that companies can expect from employing people holding the Wireless Communication Professional (WCP) credential; that value proposition will be part of future presentations and outreach to the industry. New efforts are underway to identify companies and individuals that might particularly benefit from the WCP credential, to encourage them to test during the upcoming exam windows. Even as the first edition of the WEBOK goes to press, initial plans are being made to update the book to keep pace with the rapid changes that mark the wireless communication industry. And an expansion of the IAB is underway, adding new members to broaden both its geographic distribution and the representation of all segments of the industry.

The effort to make the WCP credential the benchmark for skill and abilities in wireless communications has borne fruit, and clearly the effort will continue. You will read about it in future columns (but you will almost always learn about new developments sooner by visiting www.ieee-wcet.org).

OFC/NFOEC 2009: SHOWCASING THE FUTURE OF OPTICAL FIBER COMMUNICATIONS, TODAY

With the optical communications industry advancing at the speed of light and new advances being made in fields such as fiber to the home (FTTH) and photonic networking, it can be overwhelming to stay on the cusp of the latest innovations. That is why the industry will come together to report on achievements and exchange information at the Optical Fiber Communication Conference and Exposition/National Fiber Optic Engineers Conference (OFC/NFOEC) 2009.

OFC/NFOEC 2009 will again take place at the San Diego Convention Center, from March 22–26.

This year's conference includes more than 100 invited talks, panel discussions, workshops, and tutorials on all aspects of the field, and the show floor will host more than 600 exhibitors from every tier of the optical communications market.

Conference organizers are enthusiastic about the trends and advances the industry is making. "Right now, fiber optic communications is in a stage that is both challenging and exciting," said 2009 OFC/NFOEC General Co-Chair Mark Feuer, of AT&T Labs. "It's challenging because our past successes, together with those of other infotech industries, have led people and businesses to demand more data and better services, delivered faster, at a lower cost and with more mobility and ease of use than ever before. It's exciting because the researchers, engineers, and business people that make up OFC/NFOEC have responded with an outpouring of innovation across the spectrum of diverse disciplines that they practice."

With all the advances being made in the industry, how do you know where to start when you arrive in San Diego? "In order to get a feeling for what is going on in the industry, the Market Watch and Service Provider Summit sessions are indispensable, as is, of course, the Plenary Session of the conference," advises Alphion Corporation's Leo Spiekman, another OFC/NFOEC general co-chair. "I would also recommend the tutorials and invited papers as a way to get a more in-depth view for the state of the technology," says Bert Basch of Verizon Network and Technology, also a general co-chair of the conference.

Read on for more in-depth information about these programs as well as the plethora of other offerings and highlights you'll find at OFC/NFOEC 2009.

OFCNFOEC 2009
THE FUTURE OF OPTICAL COMMUNICATIONS IS HERE

PLENARY SESSION

Every year, OFC/NFOEC's Plenary Session proves to be the most-attended event at the conference while serving as a barometer for the industry. With a history of bringing in some of the top names in the field, 2009 will be no different. This year's speakers are a diverse group of industry veterans whose insight into the telecommunications sector will provide attendees with a comprehensive outlook on the future of optical communications.

Phillipe Morin, the president of Nortel's Metro Ethernet Networks, has more than 20 years of experience in the telecommunications industry. In his current position at Nortel, Morin is responsible for P&L, R&D, and product development. Nortel's Metro Ethernet Networks focuses on providing the next generation of solutions for exploding bandwidth demand. Nortel is positioned as one of the top three solutions providers in the world. In his talk, Morin will discuss new directions in optical communications.

Offering a legal perspective on today's digital age will be Stanford University Law Professor Lawrence Lessig. An advocate for innovation commons, a free space where culture, ideas, and expression can flourish, Lessig will offer insight into intellectual property law and its effect on creativity. Lessig is the founder of Stanford's Center for Internet and Society, has been a central figure in the Creative Commons project, and is the author of many acclaimed books on the evolving law of the Internet.

Another pioneer in the field, offering an international perspective on the telecommunications industry, is Shri Kuldeep Goyal. Goyal is currently the chairman and managing director of Bharat Sanchar Nigam Ltd (BSNL) in India. BSNL is India's number one telecommunications company and is the seventh largest telecommunications company in the world. They provide a range of services from wireline through CDMA mobile, GSM mobile, Internet, broadband, carrier service, MPLS-VPN, VSAT, and VoIP services to IN services. Goyal will discuss the advancements and challenges facing the telecommunications industry in India.

The OFC/NFOEC Plenary Session will take place Tuesday, March 24 from 8 to 11 a.m. in Ballroom 20.

TECHNICAL HIGHLIGHTS

From short courses to workshops and cutting edge technical presentations, every industry professional will find something at OFC/NFOEC to help acquaint him or herself with the latest technological advances.

Technical Presentations are a main attraction of the week, and Basch, Feuer, and Spiekman anticipate several hot technical topics to emerge, including FTTH; data commmunications/future Internet; present and future PONs; 100 Gb/s Ethernet and the components to support it; coherent systems; polarization multiplexing, QPSK, OFDM, and other formats once considered "exotic" and now entering the mainstream; quantum dot lasers for operation at elevated temperatures; and novel ultrafast nonlinear devices for optical multiplexing and wavelength conversion at speeds as high as 320 Gb/s.

According to Basch, the increasing use of photonic integration and digital signal processing techniques is in many ways revolutionizing the industry. More than 100 invited talks from the top researchers in the field will be presented in 12 technical categories (listed below), including presenters from Google, Intel NTT, the University of California, and dozens of others. In particular, says Spiekman, "the two technical talks that I will make sure not to miss are the tutorials by Rene-Jean Essiambre and Andrew Ellis, who, from the different points of view of fiber nonlinearities and the use of advanced modulation formats, will illuminate the ultimate limits of transmission capacity that can be expected from a single fiber." Also hot this year are presentations on data commmunications and future Internet, like Intel's "Optical Interconnection Networks for High-Performance Cluster Computing" and a talk on "Energy Footprint of ICT: Forecast and Network Solutions."

Short Courses are an excellent way to brush up on products and technologies in the vanguard of the industry.

(Continued on page 14)

Minimize Your Design – Maximize Your Business

Integrated and Compact 40G Solutions

Two new receiver families are being offered supporting the design of next-generation 40 Gbit/s systems, reducing your integration effort and allowing for highly compact system and subsystem solutions.

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(Continued from page 12)

They offer an in-depth study of subjects such as digital transmission systems, core networks, industry best practices, and optoelectronic devices. Short courses are taught by esteemed industry leaders on a wide range of topics for a variety of educational levels. New courses this year include topics such as "Modeling and Design of Fiber-Optics Communication Systems," "Patent Fundamentals," and "Photonic Integrated Circuits." A full list of Short Courses is available at http://www.ofcnfoec.org/Short_Courses.

Workshops and Panel Discussions provide an opportunity to not only learn about the latest technologies, but also discuss and debate them. Workshops will feature a short presentation from top industry leaders from companies like IBM, Nortel, Corning, Inc., and Bell Labs followed by a panel discussion fielded by audience questions (and are free to all conference registrants!).

Technical Presentation, Workshop, and Panel Discussion Categories:

Category A. Fibers and Optical Propagation Effects
Category B. Fiber and Waveguide-Based Devices: Amplifiers, Lasers, Sensors, and Performance Monitors
Category C. Optical Devices for Switching, Filtering, and Signal Compensation
Category D. Optoelectronic Devices
Category E. Digital Transmission Systems
Category F. Transmission Subsystems and Network Elements
Category G. Optical Processing and Analog Subsystems
Category H. Core Networks
Category I. Access Networks
Category J. Network Experiments and Non-Telecom Applications
NFOEC 1: Optical Networks and Services
NFOEC 2: Network Technologies

Technical presentations take place Sunday, March 22 through Thursday, March 26. Times/locations will be available in advance at http://www.ofcnfoec.org/conference_program

Short courses are offered Sunday, March 22 from 9 a.m. to 7:30 p.m., Monday, March 23 from 9 a.m. to 5:30 p.m. and Tuesday, March 24 from 8:30 a.m. to 12:30 p.m. and 2 to 6 p.m.

Workshops and panel discussions will take place Sunday, March 22 from 4:30 to 7:30 p.m. and on Monday, March 23 from 8 to 11 a.m.

SHOW FLOOR ACTIVITIES

From fiber equipment and components manufacturers to systems and cable vendors, participating companies at OFC/NFOEC will include the industry's largest players like JDSU, Huawei, Bookham, Finisar, Agilent, and Fujitsu. As always, the OFC/NFOEC Exhibit Hall will be at the center of the action. With more than 600 exhibitors expected, not only will the latest market technologies be on display, but attendees will have opportunities to network with peers, develop new business opportunities, and connect with business partners. To get a feel for new product offerings, exhibitors will be giving 30-minute showcase presentations of their latest developments, products, and services throughout the show. The FTTx Center is also full of demos and new information, as well as the expanding Outside Plant/Transmission Systems Pavilion. No matter what your interest, the show floor activities are sure to have something for everyone, including the two most popular floor activities: Market Watch and the Service Provider Summit.

MARKET WATCH

Eager to learn more about the applications and business communities in optical communications? **Market Watch** is the place for open discussions and presentations led by respected leaders on themes like the state of the optical industry, investments, and research developments. This year, presenters will include luminaries from Infinera, Nokia Siemens, Ovum, and Verizon, among others. Market Watch is a free three-day event located on the Exhibit Floor. Topics will range from "More Wavelengths, Higher Bit Rates, More Spectrum ... The Path to Harnessing Maximum Fiber Capacity at the Lowest Cost" to "Optical Switching and Reconfigurable Networks: Balancing Agility, Reliability, and Economy as Networks Evolve," as well as a new session on 100G standards.

SERVICE PROVIDER SUMMIT

This year's **Service Provider Summit** will feature keynote speaker Robert Blumofe, senior vice president of networks and operations at U.S.-based service provider Akamai Technologies. Blumofe currently leads the Akamai team responsible for the global strategy, deployment, operation and security of Akamai's production and corporate infrastructure, which supports all of the company's services. Blumofe's address is titled "Can the Internet Scale for the Coming Explosion of Media and Mission Critical Applications?"

The Service Provider Summit is free of charge for all conference and exhibit-only attendees. It will be held on the Exhibit Floor, so be sure to stop by and engage in the open discussions. This program includes topics and speakers of interest to CTOs, network architects, network designers, and technologists within the service provider and carrier sector.

Market Watch and Service Provider Summit Panel Discussion Highlights:

Market Watch

Panel I: State of the Optical Industry
Speakers from: Corning, Morgan Keegan & Co., Ovum, Qwest, and Verizon

Panel II: More Wavelengths, Higher Bit Rates, More SpectrumELLIPSISThe Path to Harnessing Maximum Fiber Capacity at the Lowest Cost
Speakers from: Fujitsu, Nokia Siemens, Infinera, Infonetics, and Verizon

Panel III: Photonic Integration: Mainstream at Last?
Speakers from: UCSC, Alcatel-Lucent, Luxtera, Heavy Reading, and Infinera

Panel IV: Optical Switching and Reconfigurable Networks: Balancing Agility, Reliability, and Economy as Networks Evolve
Speakers from: Deutsche Telekom, and NTT Network Innovation Labs

Panel V: 100G Standards Update
Speakers from: Ciena and Force10 Networks

Service Provider Summit

Panel I: Core Networks — Keeping Pace
Speakers include: Karen Liu, Ovum (moderator); William Jarr, consultant; Matthew Ma, Tata Communications

Panel II: FTTH — Advancing on Many Fronts
Speakers include: Chris Pfistner, NeoPhotonics (moderator); Andrew Odlyzko, University of Minnesota; Ching-Sheu Wang, ChungHwa Telecom

FTTX CENTER

As deployments by Verizon and AT&T, and around the world have proven, the future of the Internet depends on FTTx. At OFC/NFOEC's FTTx Center, the

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future has become a reality. See all of the latest developments and hardware being developed in the FTTx field and watch this growing technology in action. Experts on FTTx will also be on hand to answer any questions about the technology. Also, pick up a copy of the FTTx Locator Map to help you locate all of the FTTx-related activities going on at OFC/NFOEC.

OUTSIDE PLANT (OSP)/FIBER TECHNOLOGIES SHOWCASE

Ever wonder how an emergency restoration would be performed on a working fiber optics system? Do you have a question about a specific application? Wonder no more as this and many other live controlled demonstrations will be on hand at the OSP/Fiber Technology Showcase. In addition to the demonstrations, OSP instructors under the direction of The Light Brigade will be available to address any pressing questions you might have about FTTx, PMD/CD, or other fiber-related systems.

CAREER CENTER

Also on the exhibit floor is the OFC/NFOEC *Career Center*, which last year assisted 700 job seekers and featured 85 companies. The Career Center will again offer an online database of resumes, job postings, and interview scheduling during and after the conference.

The Exhibit Floor is located in Halls B1–G and is open Tuesday, March 24 and Wednesday, March 25 from 10 a.m. to 5 p.m. and Thursday, March 26 from 10 a.m. to 4 p.m.

Market Watch will take place in the Exhibit Floor Theater on Tuesday, March 24 from 11:15 a.m. to 5 p.m., Wednesday, March 25 from 2 to 4 p.m. and Thursday, March 26 from 10 a.m. to 3 p.m.

The Service Provider Summit will be held in the Exhibit Floor Theater on Wednesday, March 25 from 8:15 a.m. to 5 p.m.

The FTTx Center will be located in Exhibit Hall E on Tuesday, March 24 and Wednesday, March 25 from 10 a.m. to 5 p.m. and Thursday, March 26 from 10 a.m. to 4 p.m.

OSP/Fiber Technology Showcase will take place in Exhibit Hall E on Tuesday, March 24 from 11 a.m. to 12 p.m., Wednesday, March 25 from 11 a.m. to 4:30 p.m. and Thursday, March 26 from 11 a.m. to 4 p.m.

The Career Center will be located in Exhibit Hall B, and will be open Tuesday, March 24 and Wednesday, March 25 from 10 a.m. to 5 p.m. and Thursday, March 26 from 10 a.m. to 4 p.m.

Much more information on conference offerings is available online at http://www.ofcnfoec.org. And if this all seems a little daunting, Feuer says, "Remember that OFC/NFOEC 2009 will have short courses, tutorials, and invited speakers to help attendees put it all in perspective, assessing the virtues, flaws, and commercial significance of emerging technologies." No matter what your motive for attending OFC/NFOEC, you are certain to leave with more insight into the optical communications industry than you ever thought possible.

ADDITIONAL INFORMATION AVAILABLE ONLINE AT OFCNFOEC.ORG

- Conference Schedule-at-a-Glance
- Housing and Registration Information (Advance Registration Deadline: March 5, 2009)
- Short Course Listing
- List of all Invited Presentations and Tutorials
- Exhibiting Companies List

SPECIAL EVENTS

- New last year, the Symposium on the Future Internet and Its Impact on Next Generation Optical Networks will explore the future of the Internet and how it will impact the architecture and technology of next-generation optical networks. The symposium, taking place Tuesday, March 24 from 2 - 6:30 p.m., will feature presentations and predictions from several leaders in the field followed by a question and answer session with the audience.
- New this year is a conference banquet at the San Diego Sea World. Held Monday, March 23 from 7 to 10 p.m., OFC/NFOEC attendees can network with their peers in the relaxed atmosphere of a sit-down dinner, followed by a private Shamu Show. Transportation to and from the park is included in the ticket price, but act quickly as there are a limited quantity of tickets. Tickets are available for purchase at the time of conference registration.

INTRODUCTION TO "THE EARLY HISTORY OF PACKET SWITCHING IN THE UK"

In this issue of the History Column we bring you an article by Prof. Peter Kirstein, one of the original contributors to early packet switching. We are probably all familiar with the history of the Internet, beginning with its genesis in the American-developed ARPAnet of the late 1960s and early 1970s. We may be less familiar with the contributions of British researchers, as well as those in other countries such as France, at about the same period of time, who worked closely with American researchers as well as independently in developing the packet-switching technology so fundamental to the Internet. Prof. Kirstein recounts the early activities by British engineers, led by Donald Davies of the National Physical Laboratory, the British Post Office, those of his own group at University College London, and others as well. He also ties this work into ongoing activities in the United States at the time. In future History Columns we plan to have similar articles by U.S. packet-switching pioneers on their own early activities in the field. This series of articles on the genesis of the Internet should be of great interest to all communication engineers. We commend the article following to your attention.

—Mischa Schwartz

THE EARLY HISTORY OF PACKET SWITCHING IN THE UK

PETER T. KIRSTEIN, UNIVERSITY COLLEGE LONDON

INTRODUCTION

A study of U.K. networking in the 1960s and '70s must start from an understanding of the environment. Clearly anyone, academic or industrial, could do theoretical work. However, the potential for practical work was much more limited. The British Post Office (BPO) had a monopoly on any communication across public rights of way. There were two other sets of players: single organization networks and computer service bureaus; the latter could set up data networks to their computers, and could provide services to remote users. Service bureaus had to be careful; their remit stretched only to data traffic to their own customers. The BPO still had a complete monopoly on facsimile, message switching, and voice traffic. Thus, any technical activity in this field had to take into account that the results would be used only by the BPO as a service provider, or by the computer manufacturers who provided equipment to the service bureaus, the BPO, or data processing centers.

This environment had two other corollaries. The BPO's main telecommunications business was voice, and its thinking was based entirely on circuits and voice calls. It was constrained to think in terms of standards for interoperability with other similar service providers. It was not interested in technical innovations that could not be agreed on universally. By contrast, the computer manufacturers were much more interested in providing all the equipment between their mainframes and the user, including terminals and data communications equipment; their main interest in standardization came when they had to interface to the carrier's equipment, or wished to attach to terminals they were not manufacturing.

In this environment, there was concern with the economics of higher bandwidth. Pulse code modulation (PCM) was providing 2 Mb/s circuits, which would multiplex up to 30 voice channels of the 64 kb/s used for digitized voice traffic. There were already large networks, like those of the U.S. General Electric Information Services (GEISCO) and TYMNET, which used statistical multiplexing to aggregate a number of lower-speed channels used for interactive terminal traffic.

This article is concerned with packet-switched work in the United Kingdom up to the early '80s; therefore it ignores most of the important concurrent U.S. work. It starts with the early work at the British National Physical Laboratory (NPL).. It then considers the contemporary network services situation in the research community, and the first international node of the ARPAnet. The corresponding European activity is considered next. It covers the BPO's response followed by the contemporary work on the Cambridge Ring LAN. Later British activities in the area are considered, and some conclusions are drawn.

THE NPL NETWORK AND RELATED WORK: 1966–1970

From 1965, the NPL, a British government laboratory, investigated, under Donald Davies, the possibilities of putting together a large data network. While there were some published papers from the '60s, the best historical note about the NPL work in this period comes from a paper that Donald sent me two months before he died. It was published posthumously as [1]. Paul Baran had written his first public paper [2] on the principles of packet switching, but Donald arrived at these in parallel. He regarded the work of Len Kleinrock (e.g., [3]) as seminal — but it considered only message switching, not packet switching, at that time, and did not influence his ideas. I return to the question of who should claim precedence at the end of this section. Here I point out only that the '60s was a different era from even a decade later. Only papers like [2], books like [3], or presentations at international working groups would have been known internationally and hence could have influenced the protagonists.

Donald's initial ideas were expressed in unpublished notes that were reprinted as annexes to [1], from which I give some extracts:

Starting from the assumption that on-line data processing will increase in importance, and that users of such services will be spread out over the country, it is easily seen that data transmission by a switched network such as the telephone network is not matched to the new communication needs that will be created. …

The user of an on-line service wishes to be free to push keys sporadically, and at any rate he wishes, without occupying and wasting a communication channel. But he does not expect a reply from the computation service for less than a 'message' of several characters, typically between 10 and 100.

A message communication service in which short messages are temporarily stored in computers situated at the nodes of the network, and forwarded in turn, can give great economies in the use of transmission paths. Further economies are afforded by the use of digital transmission plant, with regenerators in place of linear amplifiers. The result of these two factors is that transmission cost can be extremely low by present day standards.…

Assuming that to carry a message no more than 5 tandem exchanges are ever needed, and therefore that such messages are held 7 times in short buffers and 5 times in output queues and are transmitted 6 times, the total delay time would average about 23t (here he meant 23 times through a tandem exchange — PK). This could be kept down to 100 milliseconds if all the communication channels had a capacity of at least 250 messages per second. With digital transmission, this sort of capacity would easily be provided, and correspond to a few telephone (PCM) channels...”

In the reference Donald went on to analyze the characteristics and costs needed for such exchanges, and the protocols needed to communicate with the nodes. He considered the different components like the packet assembler/disassembler needed to handle terminals (e.g., the development that later became the ARPAnet TIP [4]), and even envisaged services like electronic mail. He outlined the costs and concluded with the prescient comments:

... “Proposal for a pilot service in London and for research and development in the UK. It is important not to find ourselves forced to buy computers and software for these systems only from USA.”

Clearly his ideas needed experimental verification, and over the period 1966–1969, NPL proceeded to build a pilot network for internal services. By 1967, the work was sufficiently advanced that it was possible to give a paper on this subject at a Gatlinburg symposium [5]. This paper had far-reaching consequences. Larry Roberts was then doing the preliminary planning for ARPAnet; the paper showed Larry that there was important work in this area going on outside the United States, and led to the international activity mentioned below. He later stated [6]:

“Donald Davies work ... did show the importance of packet switching for computer communication. This effort had been going on in parallel with the MIT efforts during 1966… …. Although the UK work convinced Roberts to use higher speed lines (50 KB) and to use the word packet, the Rand work had no significant impact on the ARPANET plans and Internet history.”

To what extent Donald’s work was the first in the field and actually influenced the design of ARPAnet is more controversial. Donald said that he invented the concept and that his paper was the first to mention it. Clearly there was a lot of work going on in parallel. Licklider [7] had the vision of a national computer network much earlier — but had no view on its technology. In [8] Len had pointed out the effect of priority and segment size on waiting time even before [9], the first report on such networks in the United States. Len and Larry have pointed out to me that in [8] Len had already analyzed the importance of breaking up messages into smaller parts to reduce queuing delay. Paul had written about many of the design trade-offs for packet-switched networks [9]. Indeed, Larry’s own experiments in connecting the TX-2 in Massachusetts to the Q-32 in California had already shown the need to break messages into fragments to reduce retransmission time. While references such as the annexes of [1, 8, 9] may have been read by others working in the field nationally, they were not known internationally. It is more difficult to establish at this time, however, whether Larry intended to switch the fragments as independent packets in the ARPAnet before he heard of the NPL work; certainly he now claims that this was always his intention. His specification for ARPAnet clearly required such packets. The detailed system design to meet those specifications, mainly due to Bob Kahn, used that concept, and the initial implementation was carried out by a team of BBN engineers during the first eight months of 1969.

By 1968, the experimental NPL network was well enough advanced to be described in the series of papers given at the 1968 International Federation for Information Processing (IFIP) Conference [10–13]. It should be remembered that this was nearly two years before the groundbreaking session on the ARPAnet at the Spring Joint Computer Conference in Atlantic City in 1970.

I do not have space here to outline all the work done at the NPL over that period. It has been described well in [14]. Again I quote from the abstract:

"This paper ... focuses on the construction of the NPL Data Communications Network, which first became operational in 1970. This network served both as a model for a possible U.K. national network and as a practical local area network (LAN) for the NPL site. The report describes the impact of the NPL work on other early networks, such as ARPANET and the British Experimental Packet-Switched Service (EPSS), and on data communications in general."

By 1970, this network was operational as a LAN with 768 kb/s channels — operating at up to 500 packets/s.

NPL was never funded to proceed with a wide area network. Indeed, the BPO felt that this was its prerogative.

EARLY BRITISH RESEARCH AND EDUCATION NETWORKS

The British activity in networks for the research and education communities adopted a different path from those in the United States. The research community had installed its largest computer, an IBM 360/75 replaced in 1969 by a 360/195, in the Rutherford and Appleton Laboratory (RAL). This was to service the whole U.K. research community. To achieve its aim, by 1968 it had installed remote job entry (RJE) terminals, with very limited terminal interaction, in various British universities. Most of these were standard IBM 1130 terminals running a standard IBM system. By the early '70s there was even such a terminal at CERN in Geneva for the British high energy physicists there.

For education, universities had been funded to acquire standard computers; in addition, three regional computer centers had been established at the Universities of London (ULCC), Manchester (MRCC) and Edinburgh (ERCC). These again had RJE connections to allow them to fulfill their regional commitments.

On the whole the above facilities were pure service ones; no network research or development could be done on them. About the only exception was an activity at the University of London Institute of Computer Science (ULICS), in which a DEC machine was connected to the RAL system by a leased line and programmed to provide remote interactive graphics facilities similar to those that could be provided by a local graphics terminal [15]. While this activity was not intrinsically important to the general British network activity, it had a major repercussion, discussed below. ULICS was incorporated into one of the colleges of the university, University College London (UCL), at about the time the equipment discussed later was installed. In the rest of this column the location of this group will be called UCL to avoid confusion.

THE FIRST INTERNATIONAL NODES OF ARPANET

By 1970 the first four nodes of ARPAnet were operational, and nationwide deployment was already under construction in the United States [16]. Shortly after, the Defense Advanced Research Projects Agency (DARPA) started envisaging that more of its research activities might use the network. The first of these was the seismic analysis activity of its Nuclear Monitoring Research Office, which supported two large arrays in the United States and one foreign one [17] in Kjeller, Norway (NORSAR). Other DARPA research activities that might use a similar technology included packet voice, packet radio, and packet satellite.

The original links from the Washington to the NORSAR array went via a satellite circuit to Gonnhilly, United Kingdom, and thence via cable to Kjeller, Norway. In early 1971 Larry proposed to break the circuit and connect in the NPL network to the ARPAnet. Unfortunately for that plan (but not for me), at this time the British government was trying to get the United Kingdom into the European Economic Community (EEC). The EEC governments, particularly France, were in any case suspicious of U.K. links with the United States. It was therefore politically impossible for NPL to be linked directly to a U.S. defense project. Since the opportunity was too good to let slip, Donald suggested that I pursue the offer instead. I took this up enthusiastically; the early history of the British links to the ARPAnet have been detailed elsewhere [18].

DARPA essentially supported only research and development activities; hence, my proposal had to have a strong research component. At the time, all the ARPAnet hosts were local to their communications computers. I proposed three areas of activity:

- Connecting in the RAL IBM 360/195 remotely
- Connecting in the ULCC[PT1] CDC 7600 remotely
- Working with DARPA on a new satellite network project called SATNET

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At the same time, the RAL and ULCC machines discussed earlier were service machines, and I was not permitted to make any changes in those machines. Thus, I had to exactly emulate one of the standard RJE terminals one way and a standard ARPAnet host on the other from my front-end machine. DARPA was going to provide only the communications computers (TIP, [4]) at UCL and Kjeller; they were also going to upgrade the international circuit to the United States to 9.6 kb/s. My problems in getting these activities and the onward link to the NORSAR TIP funded in the United Kingdom are outlined in [18]. The initial support, both financial and political, of the NPL and BPO were vital to the ultimate resolution of the funding and management barriers.

The link to the ULCC CDC computer turned out to be impractical because of the way the machine was being operated. The other two parts of the program were eminently successful. Our linking in of the RAL machine had to be completely transparent, and all its access control was in the IBM host. However, we were concerned from the beginning with security breaches from the United Kingdom, so we devised mechanisms for putting access control into our system — including into the TIP itself. This was vitally important to overcome a reluctance of the BPO to continue permission for the whole project.

Bob Kahn was pursuing two further DARPA programs for new technologies: packet satellite (SATNET [19]) and packet radio (PRN [20]). SATNET required special terminals to sit in the earth stations, which were then operated only by the large carriers. At that time, there were no domestic U.S. satellites; for this and other reasons, the project was carried out internationally. It involved European partners UCL in the United Kingdom, the German Space Research Centre (DFVLR) in Germany, the Norwegian Defense Research Establishment (NDRE) in Norway, and the University of Pisa in Italy; in each case their telecommunications authorities had to host the equipment in their earth stations, and so be partners in the activity. The U.S. equivalent was Comsat, and, in addition to Comsat, the U.S. companies Linkabit and Bolt, Beranek and Newman (BBN) supplied the rest of the earth station equipment.

While all the European partners participated actively in the research project, only UCL went on to make it a component of its service activities. UCL did not participate directly in the PRN project at the time. However, we had one of the packet voice terminals, and its transmission over SATNET was one of the activities in which we did participate. Similarly, we participated in one of the first multinetwork activities when our defense laboratory, RSRE, linked to UCL through SATNET and then ARPANET, communicated via packet radio with a car crossing the Bay Bridge in San Francisco [21]. The great importance of these activities came not only from the technologies themselves, but from the fact that this required the linking of different underlying computer network technologies. It was for this reason that Bob Kahn developed the concept of the gateway, which was fundamental to linking those networks together. It was from this that the IP concept was established, and the TCP/IP protocol of Cerf and Kahn [22] emerged. Because the U.K. networks had to be interfaced at a different level, while we used the U.S. gateways, we also had to further develop our own gateway technology.

Throughout the '70s, the SATNET project was pursued with the other international partners. The work was described in a session of which [19, 23, 24] are three papers. From the beginning, the international dimension had to be considered — as it was in [19].

UCL's activity had another long-lasting activity. Cerf started, in 1978, the International Collaboration Board (ICB). This was to foster unclassified collaboration in command and control between defense departments. The ICB activity continued for 25 years; during its life it included participants from Canada, Denmark, Germany, Italy, NATO, Norway, the United Kingdom, and the United States.

THE EUROPEAN INFORMATICS PROJECT

While NPL was not permitted to take up Larry Roberts' offer to link to ARPAnet, nor could get the funding to work on a Wide Area version of the NPL network, it was encouraged to work with other Europeans. In France the interest in packet switching networks had grown quickly during the early 1970s. In 1973 the first hosts were connected to the CYCLADES network [25], which linked several major computing centers throughout France. The name CYCLADES referred to both the communications subnet and the host computers. The communications subnetwork, called CIGALE, only moved disconnected packets and delivered them in whatever order they arrived without any concept of messages, connections or flow control. Called a "datagram" packet facility, this concept was widely promoted by Louis Pouzin, the designer and organizer of CYCLADES. Since a major part of the organization and control of the network was imbedded in the CYCLADES computers, the sub-network, CIGALE, was not sufficient by itself. The CYCLADES structure provided a good test-bed for trying out various protocols, as was its intent. While, the European Commission and several governments approved the European Informatics Network project (EIN) [26] in 1971, bureaucratic problems delayed its operation until 1976 under project director Derek Barber of NPL. Larry Roberts agreed [27] that it could have been one of the earliest pace-setters in packet networks in the world. However, because of its delay, and because it never had any appreciable usage, its impact was minimal.

The EIN project also had a strong focus on protocol specification — particularly on the transport and network access level. Here it interacted strongly with the TC6.1 working group of IFIP. The NPL group and others in U.K. academia were very prominent in this activity [28].

The EIN activity represented the last experimental activity of NPL on the networks scene. Thereafter, NPL restricted themselves to protocol specification and testing (e.g., [28].

THE BRITISH POST OFFICE ACTIVITY

With the BPO having blocked the NPL activity in public networks, it clearly had to be proactive itself. By 1972, it was considering a separate data network; however, like all the other PTTs, this was still circuit-switched [29]. It was only after the ICCC meeting in Washington, where the large-scale demonstration of ARPAnet was made [30], that their data development department began to see the potential of the technology. From then on, they became both enthusiastic and helpful. They started their own activity in a packet-switched data network, the Experimental Packet Switched Service (EPSS, [31]). This went live in 1975; both the academic and service communities participated in the activity (e.g., [32]). Two of the seven senior

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BPO managers encouraged me to participate in the ARPAnet, funding the link to Norway for its first year. The BPO participated actively in the SATNET activity, mentioned earlier. Indeed, for the latter they made a major precedent of allowing installation of DARPA equipment inside their Goonhilly earth station. They made it a condition that the UCL project use EPSS, and its more standard international sequel IPSS where possible. This led to IPSS and its U.S. counterpart TELENET providing the first public data network service connected to the ARPAnet. Indeed, to aid this activity, the BPO provided my project with a free 48 kb/s IPSS link as soon as that became available; this was used until the early ’80s.

For the next decade, most of the U.K. academic service activities had links to the emerging BPO packet data networks, even if their backbone connections were often via leased lines. The BPO remained very supportive of the U.K. academic service activities. As these developed, the BPO participated strongly in the standards activities that led to the emergence of the colored book protocols discussed later. Even in the early ’80s, when the UNIVERSE project investigated the use of small earth stations connected to LANs [33], the BPO was an active participant in the project.

LAN Activities and the Cambridge Ring

In 1974 Maurice Wilkes, head of the Cambridge University Computer Laboratory, was shown a digital communication ring working at the laboratories of Hasler A.G. in Switzerland, where it was regarded as a contribution to digital telephony. He immediately realized its applicability to computer communication; he immediately started the development of what became known as the Cambridge Ring (CR). The CR was an empty-slot ring, which was believed to be easier to maintain [34]. The data rate was 10 Mb/s, and the original application of the ring was peripheral-sharing. The Cambridge group developed a whole system including interfaces to computers, a terminal multiplexer, and a monitor station. The early versions of the ring were wire-wrapped, and Maurice wanted to go immediately from there to a Cambridge Fast Ring (CFR) [35] based on a chip design, operating at 100 Mb/s. Not wanting to wait for the CFR, UCL copied the Cambridge design of the slower ring, and made a PCB version. A number of universities provided interfaces to other computers. This activity was later advanced by a number of companies. The British Science and Engineering Research Council (SERC) bought several CRs to support an initiative in distributed computing. Unlike the Ethernet being developed at the same time in the United States, Maurice was not interested in pursuing an international standardization activity. In addition, there were development problems with the CFR chip, which delayed its availability. Although the CR and CFR were technically sound, they never had commercial and standardization interests behind them. They were eclipsed by the Ethernet development and were never a real challenger.

Several universities deployed fairly large LANs (for the time) of several rings, dozens of nodes, and hundreds of terminals. Almost all of these used the CR; by the time the CFR became available, the Ethernet had clearly won the day for LANs.

Later Computer Network Activities to the Early ’80s

By 1976, the X.25/X.75 [36] protocols for network access and network interconnection had been standardized. At the same time, the U.K. research councils had decided to network together their main sites with SERCNET [37] and provide access to researchers in the universities. At the same time the U.K. Computer Board had decided to network their main computer centers. This started with the regional computer centers mentioned earlier, but later included all the universities and also subsumed SERCNET into JANET. JANET’s remit included all higher education and research; thus, the United Kingdom avoided the proliferation of agency networks that occurred in the United States, funded by DARPA, the Department of Energy, NASA, and the National Science Foundation — to name a few. In the interests of economy, the main efforts for the next few years were in the definition of standards for such services based on the open systems interconnection (OSI) model. The result was the colored books [38], covering terminal protocols, transport, LANs, file transfer, remote job entry, and mail. The academic community were heavily involved in this work — almost to the exclusion of other activities.

By 1975, UCL was funded by the British SERC, DARPA, and the U.K. Ministry of Defense. Because of the continued support first from DARPA, and hence our strong links with the DARPA program, we participated in the first TCP/IP experiments and the SATNET ones. The DARPA support for UCL started with Larry Roberts and Bob Kahn; later many other luminaries including Vint Cerf and Paul Mockapetris supported us. Our SERC support was restricted to work on the Colored Books. Indeed, in 1978 I was requested to refrain from TCP work — which I refused! Instead, Vint Cerf and I benefited from our complementary experience to write [39]. The OSI model and the relevant high-level protocols were being finalized under the auspices of the International Standards Organization (ISO). However, these always had many options — in the typical way ISO worked. The colored books represented the attempt to specify a subset that would guarantee interoperability of computers.

While there were several experimental implementations of TCP/IP in the ’70s, including ones from BBN and Stanford University, the UCL link to the ARPAnet moved to TCP/IP as their total service activity a year before others in the United States. However, our work was always dual-track between U.K. and U.S. interests; we provided, and continued to develop, an interconnection service to the ARPAnet and later Internet. As the British networks developed, our gateway systems became more complex, following the U.S. developments on one side and the British colored books on the other. This produced many challenges, such as maintaining connectivity between the Domain Name System (DNS) [40] in the Internet and its incompatible equivalent, the Name Registration System (NRS), in the United Kingdom. At the lower levels, the UCL gateway used SATNET technology, IPSS, and leased links. This gave UCL a unique experience of interconnection during the years as is evidenced by [41, 42]. It also allowed the British to develop their own technologies for another decade, until JANET finally converted to Internet protocols — partly because of the universal success of the Ethernet, which required TCP/IP. Moreover, the spanning of the two communities allowed UCL to smooth out some potential problems for the later transition; thus, for example, UCL was largely responsible for the Grey Book mail protocol

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[38] of the colored books mirroring the Internet SMTP protocol [43] over different lower-level transport.

CONCLUSION

This article shows the usual dilemma of research and development in the United Kingdom. On one hand, the early work of Davies and the NPL were important pointers; on the other, lack of government (or commercial) vision and support made it difficult to reap a commercial benefit from the advanced thinking. The history of the Cambridge Ring had a similar pattern in LANs. Next, the perennial tug between ties to Europe and to the United States precluded official participation in the ARPAnet; however, the usual strong personal links allowed close collaboration with the United States to continue in spite of official indifference from British research funders. The more unified research funding in the United Kingdom allowed computer networks to develop in a much more integrated fashion than in the United States with its competing agencies; however, by choosing an insular approach, the research networks went along a rather limited path. Because of high-level concern with maintaining their links to the United States (from both the civil and military sides), the United Kingdom adopted a path that allowed connectivity to continue — and even to recover quickly from the earlier protocol mistakes. Good links at the national level between the British Post Office and the research funders ensured that academia and PTTs worked well together; but the shorter-term commercial interests, compared to those of DARPA, ensured that the objectives of the activity were much more pedestrian.

I have ignored here the U.K. Defense involvements. Indeed, they supported UCL throughout the period consistently; however, their own activity was very limited, so they did not give the same strong impetus to the research that was given in the United States by DARPA.

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BIOGRAPHY

PETER KIRSTEIN received his B.A. in mathematics and mechanical science from Cambridge University, his Ph.D. from Stanford University, and D.Sc. from the University of London. After short spells as a lecturer at Stanford University, an accelerator physicist at CERN (Geneva), and Scientific Representative for the U.S. GE (in Zurich), he returned to the United Kingdom, where he is a professor of computer communications systems. For 35 years he led the Computer Networks research group at University College London, and was the first head of its Department of Computer Science for 15 years. He is a Fellow of the Royal Academy of Engineering, the Institute of Physics, the Institution of Electronic Technologists, a Distinguished Fellow of the British Computer Society, and a foreign member of the American Academy of Arts and Science. For his work in computer networks, he was made a Commander of the British Empire and received the lifetime achievement award from the Royal Academy of Engineering, the Postel Award, and the Senior Medal of the Institution of Electrical Engineers. He has always maintained strong international contacts. He was responsible for one of the two first international nodes of the ARPAnet and had DARPA contracts for 30 years. He has been involved in many European, U.S., and national projects in computer networks, satellite communications, multimedia conferencing, document processing, security, and emergency communications. Currently he is responsible for various NATO and European projects bringing the Internet to the Caucasus and Central Asia, IPv6, multimedia, and emergency communications.


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Book Reviews

Edited by Andrzej Jajszczyk

Broadband Wireless Access and Local Networks: Mobile WiMAX and WiFi

By Byeong Gi Lee and Sunghyun Choi, Artech House, Inc. 2008, ISBN-13: 978-1-59693-293-7, hardcover, 618 pages

Reviewers: Katarzyna Kosek and Marek Natkaniec

This book can be treated as a helpful reference on the most important aspects of WiMAX and WiFi technologies, which currently play an important role in wireless networking.

The main body of the book is organized into 18 chapters. Chapter 1 contains the principles of wireless communications. Chapters 2-10 are grouped into Part I entitled "Mobile WiMAX: Broadband Wireless Access Network". Chapters 11-18 are grouped into Part II - "WiFi: Wireless Local Area Networks". Furthermore, the reader can find five additional sections: a preface, acknowledgements to the contributors and reviewers, a list of acronyms, About the Authors, and an alphabetical index. In order to make the readers familiar with the book we first give a short overview of all chapters and then, we express our general opinion about its contents.

Chapter 1 (contributor: H. Kim) explains the characteristics of the wireless channel and frequency spectrum, shows the history of the standardization process, and gives a brief comparison between WiMAX and WiFi.

PART I

Chapter 2 (contributors: H. Lee, E. Hwang, H. Choi, D. W. Lee, H. Seo) constitutes an introduction to mobile WiMAX. It briefly explains the main ideas of the key component technologies. Additionally, it shows the internal and external network architectures of IEEE 802.16e. Furthermore, it contains an interesting comparison between WiMAX and cellular mobile networks. Chapter 3 (contributors: H-S. Kim, J. H. Chang, W. Kim) describes the initialization process of a WiMAX network. Its key concepts, such as network discovery, network initialization, connection setup, handover, nonconnected state, paging, and mobility are clearly presented here. Additionally, the chapter shortly explains the most important WiMAX maintenance mechanisms: synchronization, periodic ranging, and power control. Chapter 4 (contributors: S. Maeng, M-K. Byun, Y. Yoon, J. Cho) presents the OFDMA physical layer technology of WiMAX. It contains a detailed description of all important aspects of OFDMA PHY signal processing, frame structuring, and subchannelization. Chapter 5 (contributors: J. Song, C. G. Kang) describes two of the three sublayers of the MAC framework of WiMAX, i.e, MAC service-specific convergence, and MAC common part. The details about the security sublayer are given in Chapter 8. In addition, the automatic repeat request mechanism is discussed here. Chapter 6 (contributors: H-S. Kim, C. G. Kang) explains possible types of bandwidth allocation and combines them with the QoS issues standardized for WiMAX. Furthermore, the chapter contains an interesting discussion on CAC, scheduling and policing functions, which remain implementation specific. Chapter 7 (contributors: J. Song, C. G. Kang, H. Seo) contains the most important concepts of mobility support in WiMAX. Therefore, such important issues as cell based network operation, the handover procedure, and power saving techniques are described here.

In Chapter 8 (contributors: P. J. Lee, H. Kwon) the authors comment on the architecture and operation of the WiMAX security system. They have composed a logically coherent whole from the numerous characteristics of this third MAC sublayer. The chapter includes all meaningful security related issues, from cryptography, ciphers, hash functions, encryption, authentication, and key management to practical implementation proposals. Chapter 9 (contributors: I. Hwang, E. Y. Kim, H. Kwon) explains the concept of multiple antenna technology in relation to WiMAX. Firstly, it covers the basics of multiple antenna technology. Furthermore, it helps to understand the differences between the open-loop and closed-loop technologies. Finally, it presents MIMO receiver algorithms with the stress put on the importance of making an appropriate choice among the available models when building a WiMAX system. Chapter 10 (contributors: H. Kim, J. Lee) describes in detail the first mobile WiMAX system - WiBro. The authors focus mostly on its system design, network deployment, and services, as well as the system requirements and configuration issues. This chapter is especially valuable because the knowledge it contains is based on existing systems developed and deployed by Samsung Electronics and Korea Telecom.

PART II

Chapter 11 constitutes the introduction to WiFi networks. It explains their most important concepts, i.e., infrastructure and ad-hoc network architectures, the IEEE 802.11 reference model, and layer interactions. In addition, the authors briefly explain several key technologies employed by the IEEE 802.11 standard: multiple access, multirate, power saving, mobility, confidentiality, power management, and QoS support. Chapter 12 contains the description of the IEEE 802.11 PHY protocols. Firstly, the authors explain the basic PHY operations. Then they describe in detail two protocols: OFDM and HR/DSSS, previously known as IEEE 802.11a and IEEE 802.11b, respectively. Finally, they shortly mention the ER protocol, previously known as IEEE 802.11g. Chapter 13 explains the baseline MAC protocols of IEEE 802.11 defined in 1999. At first, the MAC frame formats and their purposes are given. Then, the mandatory distributed coordination function and the optional point coordination function are presented. The chapter ends with a description of different MAC operations (such as rate adaptation techniques and multirate support) and MAC management functions (such as time synchronization and power management). In Chapter 14 the reader can find the key ideas of QoS provisioning provided by the IEEE 802.11e extension. Firstly, the limitations of the baseline MAC are explained. Secondly, two channel access functions introduced by IEEE 802.11e are described: HCCA and EDCA. Finally, admission control, scheduling, and optional features of IEEE 802.11e are discussed. Chapter 15 contains a description of the security features of WiFi. After showing the severe limitations of the basic encryption and authentication scheme - WEP, the authors introduce 802.1X and data confidentiality protocols: TKIP and CCMP, employed in 2004 by IEEE 802.11i. Chapter 16 describes mobility support. Firstly, the handoff procedures and their features are shortly introduced. Then the concepts of the IEEE 802.11f for Inter-Access Point Protocol practise specifying the inter-AP communication are provided. Finally, the fast scanning and fast roaming mechanisms are explained on the basis of the draft versions of IEEE 802.11k and IEEE 802.11r, respectively. Chapter 17 deals with the spectrum and power management in the 5 GHz band. The

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requirements of the transmit power control and dynamic frequency selection are given for the USA and Europe. Chapter 18 is the last chapter discussing WiFi technology. It briefly presents the most recent WiFi extensions: IEEE 802.11n (higher throughput support), IEEE 802.11s (mesh networking), IEEE 802.11k (radio resource management).

The main advantages of the book are as follows. The book gives a complete and well organized state-of-the-art of the current theoretical and practical knowledge about mobile WiMAX and WiFi. Additionally, the up-to-date contributions of researchers from industry and academia increase the meaningfulness of the book. The guidance for design and implementation is provided by the engineers from leading companies, such as Samsung Electronics and Korea Telecom. Every chapter contains a nicely written introduction to the topics discussed in it. Furthermore, short overviews of all chapters regarding WiMAX and WiFi are gathered in the introduction to Part I and Part II of the book, respectively. In addition, the alphabetical index and the list of acronyms meaningfully facilitate working with this voluminous book.

The book also has several disadvantages. Firstly, the authors do not mention the fact that in 2007 the 802.11 amendments were gathered into a single standard. Secondly, many contributors to the book are not mentioned as co-authors but only in the acknowledgement section. We think that they may remain unnoticed. Therefore, we decided to present their names in the brackets while discussing the contents of each chapter. Finally, the readers who want to learn more about WiFi related issues currently being standardized may be slightly disappointed after reading this book.

However, the mentioned flaws do not affect our overall positive impression about the book. We believe that readers seeking a comprehensive source of information about the current knowledge on mobile WiMAX and WiFi will find it as a very valuable one. Therefore, we think that the book is worthy of recommendation.

INTRODUCTION TO IDENTITY-BASED ENCRYPTION

BY LUTHER MARTIN, ARTECH HOUSE, 2008, 232 PAGES, ISBN-13: 978-1-59693-238-8

REVIEWER: MARCIN DABROWSKI

Identity-based encryption (IBE) is a promising alternative for well-known traditional public-key systems. As a more distributed and lightweight in its nature, it is free of the public-key distribution, certificate validation and revocation problems which, together with its simplicity from a user point of view, makes it an easy security means that the non-IT professionals have been waiting for. In identity-based encryption a sender can calculate a one-time per-message encryption key based on the recipient's public identity information and the system's parameters. On the other hand, the recipient can obtain a per-message decryption private-key based on the received message, the system's parameters and a master secret it shares with the so called Private Key Generator (PKG).

Currently, there are no comprehensive sources which can introduce the reader into the broad topic of identity-based encryption. The book Introduction to Identity-Based Encryption by Luther Martin is indeed the first complete guide which fully explores the subject.

The book starts with the general explanation of the basic IBE concepts. Then, the following chapters give a brief, yet complete, mathematical background that every cryptographer should know. Among others, the author describes the properties of elliptic curves, he explains the terms of divisors and Tate Pairing and describes the main computational problems in contemporary cryptography. The reader is also given necessary information on all cryptographic algorithms related to IBE.

After the introductory part of the book, the reader is given a complete overview of the most significant IBE schemes, beginning with the Cocks IBE scheme, through Boneh-Franklin and Boneh-Boyen IBE schemes, ending with the Sakai-Kasahara IBE scheme.

At the end of the book, the author presents the topic of hierarchical IBE schemes together with master secret sharing aspects. He ends the book with the chapter about efficient calculation of pairings.

Introduction to Identity-Based Encryption is really a comprehensive guide to identity-based encryption and the first one to fully exploit the topic. It is easy to read, however, some basics on cryptography and mathematics are needed. It all makes the book a good choice for readers who are new to IBE and want to learn its concepts as well as for security professionals who need a complete reference point.

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Highlights from IEEE/IFIP Manweek 2008: Fourth International Week on Management of Networks and Services

By George Kormentzas, University of the Aegean, Greece

The 4th IEEE/IFIP International Week on Management of Networks and Services (Manweek 2008) was held 22–26 September 2008 on Samos, a Greek island with ages-long history. Samos is considered to be the birthplace of the goddess Hera on the banks of the river Imvrassos, and it is the place where mathematician Pythagoras, astronomer Aristarchos, and philosopher Epikouros lived. Manweek 2008 extended the key concept of the three previous Manweeks of putting together relevant workshops and conferences by bringing one more workshop on board (NGNM) and integrating related EC projects (AutoI, PEACE, UNITE), cost actions (TMA), and forum meetings (ACF), thus building an even more structured event in the area of network management and services. In this context the actual duration of Manweek 2008 was 10 days, 18-27 September 2008, for the first time in the history of network and services management conferences.

The organizing committee of Manweek 2008 put together a very interesting program including technical sessions and panels of six workshops and conferences in the area of management of networks and services (i.e., DSOM, MMNS, IPOM, MACE, EVGM, NGNM), three keynote speeches by European Commission (EC) Project Officers, and one demonstration of the operation of a virtual distributed testbed (VDT) for B3G experimentation. The first keynote speaker was Francisco Guirao, and his speech was entitled "European Research on Future Networks." He gave the vision of future networks and services within the context of the European research programs. Special emphasis was given to FP7 where the future of the Internet is going to be a central subject for research on overcoming the structural limitations of the current Internet architecture. The second keynote speaker was Bart Van-Caenegem, also from EC. He discussed EU funded network security research in FP7, presenting both the current project portfolio, as well as the research funding challenges related to security, privacy, and trust in a future Internet environment. The last keynote speaker was George Tselentis, again from EC. He delivered a talk related to future Internet research experimentation (FIRE) and discussed how EC is going to stimulate the building of large experimentation platforms, which could be the drivers for the future Internet. The VDT demonstration was related to this keynote speech.

The keynote speeches were complemented with thee panels that gave the opportunity for broader attendee participation. The first panel had the theme "Network Self-Management and Vertical Policy Interactions in E2E Virtualized Networks" and was moderated by Ralf Wolter, CISCO Germany. The panelists formed a good mix of academic and industry representation. The second panel was about scenarios for a FIRE facility, and the moderator was George Tselentis (EC) with four panelists who act as technical coordinators of four important EC-funded projects in the area of future Internet (PII, OnelabII, Vital++, and UNITE). The third panel had the theme "Large Scale Service Deployment: Research Challenges." The moderator was Filip De Turck (Ghent University, Belgium), and the panelists were IEEE Comsoc President Dr. Douglas Zuckerman, and two representatives from DoCoMo and NTT.

Manweek 2008 received a total of 169 submissions. Specifically, DSOM '08 received 45 submissions (14 papers accepted as full, giving an acceptance rate of 31%), MMNS '08 received 46 submissions (15 papers accepted as full and one paper accepted as short, giving an acceptance rate of 33% or 35%), IPOM '08 received 30 submissions (12 papers accepted as full, giving an acceptance rate of 40%), NGNM '08 received 26 submissions (13 papers accepted as full, giving an acceptance rate of 50%), EVGM '08 received 10 submissions (4 papers accepted as full, giving an acceptance rate of 40%), and MACE '08 received 22 submissions (8 papers accepted as full and 4 papers accepted as short, giving an acceptance rate of 36% or 55%). The overall Manweek 2008 acceptance rate for full papers was 39%. The accepted papers formed a technical program of 21 sessions in two tracks with the exception of the second day of the Manweek 2008 conference, when there were three tracks. The proceedings of DSOM '08, MMNS '08, IPOM '08, and MACE '08 were printed by Springer LNCS (vols. 5273–5276), while the NGNM '08 and EVGM '08 proceedings were printed by Multicom (Lecture Notes vol. 9).

Manweek 2008 hosted plenary meetings for the EC-funded projects AutoI (18–19 September), PEACE (18–19 September), and UNITE (26 September), two-day meetings of Traffic Management and Analysis (TMA) COST Action (22–23 September) and Autonomic Communications Forum (ACF, 24–25 September), a three-day TPC meeting for IM '09 (22–24 September), a joint IFIP WG6.6/CNOM meeting chaired by IFIP WG6.6 chair Prof. Aiko Pras (24 September,) and a two-day IFIP TC6 meeting chaired by its President, Prof. Guy Leduc.

About 200 attendees from 28 countries enjoyed the event as well as both the Doryssa Bay Resort (i.e., the venue of the event), which actually resembles a typical Greek village, and the city of Pythagorio with many historical sites, very good restaurants, and a wonderful beach. The famous samiotiko wine facilitated the sharing of research experiences and results and the identification of common opportunities for research collaboration, under either an academic umbrella or ICT FP7.

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WiMAX Developments in the Middle East and Africa

By Saad Z. Asif, Telenor Pakistan

WiMAX is on the lips of every person associated with the ICT industry in the Middle East and North Africa (MENA) and Pakistan. WiMAX refers to the IEEE 802.16 standard-based technology. It enables the delivery of last mile wireless broadband access as an alternative to wireline broadband technologies such as cable and digital subscriber line (DSL) and wireless technologies such as EV-DO and HSPA. In this article we look into the current WiMAX developments that are taking place in this region.

The current surge in demand for WiMAX in these areas is attributed to four factors:

•Telecommunications market liberalization: TML was necessary for countries to gain World Trade Organization (WTO) membership. TML policies are helping the governments to improve their economic standings, in turn increasing the penetration of broadband users.

•Inadequacy of existing infrastructure: The existing broadband (copper) infrastructure is inadequate in many countries. DSL and cable modems are the primary means to access the Internet. Frequent fiber cuts and right of way are also big challenges for operators.

•Competition: The incumbents have to protect their existing base against new entrants, which results in the introduction of new services such as WiMAX.

•3G spectrum cost: The cost of third-generation (3G) spectrum is very high compared to the cost of WiMAX frequencies. For example, operators only paid $1 million to acquire 21 MHz of WiMAX, whereas they would pay a minimum of $291 million for 20 MHz of 3G in Pakistan.

WiMAX, whether based on 802.16d or 802.16e, will mainly be used in fixed and portable modes, not in its mobile form. The reasons are lack of modern public transportation, lack of telecommuting, lack of awareness, and license obligations. For example, in Pakistan the regulator has enforced zero mobility (handovers from one cell to another are not allowed) in the 3.5 GHz spectrum. 3.5 GHz is the primary frequency band used to offer WiMAX services in MENA.

In 2005 Algeria became the first Arab country to have WiMAX service launched via Smart Link Communication (SLC). Table 1 provides a more detailed outlook on MENA's WiMAX industry, which has more than 70,000 subscribers.

Countries	Operators	WiMAX status
Algeria	SLC	ISS 2005
	LaCom	Conducted a 802.16d trial in 2006; filed bankruptcy in Nov 2008
	Icosnet	ISS summer 2008
	Algeria Telecom	Preparing to launch in 2009
Bahrain	Zain Bahrain	ISS 2007
	MENA Telecom	802.16e commercial launch in 2009
Egypt	Two ISPs - EgyNet and TE Data	ISS 2008 in tourist resorts
Iran	Datak TeleCom, Laser TeleCom, Shatel Telecom	ISS 2007/2008
	Paya Comm. Ltd, Iran Mobin, MTN IranCell, MTCE, RDG	Awarded provincial licenses in Nov 2008
Iraq	IRAQTEL	ISS 2007
	Kalimat Telecom	ISS 2008
Jordan	Batelco's UMC	ISS 2007
	Wi-tribe	ISS 2008
Kuwait	Arab Telecom	ISS 2007
Libya	Libya Telecom and Technology	Launch expected in early 2009
	General Post & Telecom Co.	Launch mid to late 2009
Lebanon	CedarCom	ISS 2006
	Comium	ISS 2008
Morocco	Wania	ISS 2007
	Meditel	ISS 2006
Pakistan	Wateen Telecom	Largest network in the country since Dec 2007 covering 22 cities
	Mobilink	ISS late 2008 in Karachi only
	LinkdotNet (ISP)	ISS 2007
	TeleCard (ISP)	Planning to launch in 2009
Qatar	Wi-tribe	Planning to launch in 2009
	Vodafone-Qatar	Received license in Sept. 2008
Saudi Arabia	STC	ISS 2007
	ITC	ISS 2006
	Mobily	ISS Fall 2008
	Batelco	Has the license
UAE	Etisalat	Under customer friendly trial
	du	In trial
Yemen	Nexen Petroleum (energy company)	Service restricted to oil fields only
Note: ISS: In service since		

TABLE 1: *WiMAX availability status.*

The major drawbacks related to innovation in MENA are:

•Nonexistence of R&D and manufacturing houses

•Absence in the standard development organizations

•Lack of intellectual property work

All these factors have shown their marks in the developments of WiMAX in these countries. For example, there are more than 70 plus operators and Internet service providers (ISPs) pursuing WiMAX, but just a handful (7%) are members (just regular members) of the WiMAX Forum. Their membership is like a health club membership that one barely uses twice a year! Their contributions in the developments of IEEE 802.16 d/e are negligible, and they are not members of IEEE-SA. These factors and political instability have also caused brain drain from MENA to technologically savvy nations. Also, not a single of piece of equipment central to WiMAX technology was researched, developed, or manufactured in this part of the world.

We expect that WiMAX will play a role in migration toward a knowledge-based economy

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Ecuadorian Branches and Chapters Organize an International Telecommunications Event: JST 2008

By: Alex Aguirre, Student Activities, IEEE Ecuador Section

In Quito, Ecuador, during 14–16 May 2008, the third edition of the Jornadas de Sistemas de Telecomunicaciones (Telecommunications Systems Journey) 2008 (JST 2008) was held. This event was organized by the Escuela Politecnica Nacional IEEE Student Branch and its Communications Student Chapter. The event had among its sponsors the Proyecto VLIR-ESPOL Componente 8, the Ecuadorian Communications Chapter, CEDIA, and the IEEE-ESPOL Student Branch, among others. Additionally, there was total support from the IEEE Communications Society, which brought to the event two distinguished authorities, Celia Desmond and Dr. Curtis Siller, who participated with two keynote conferences during the event.

The first day of the event there were a series of tutorials that began at 9:00 a.m. These tutorials were taught by companies such as Nokia Siemens, Telefonica, Geocom, CLA Direct, Fortinet, Uniplex, and Telconet. At the end of the day at 19:00 the formal inaugural ceremony took place with the participation of institutional authorities, international speakers, members of Quito's City Council, members of the IEEE-EPN Student Branch, professors and students from several universities around the country and the world. During this important occasion, the event's General Director, Servio Lima, welcomed the participants and gave a summary of previous events. It is important to mention that there were around 150 participants from different Latin American countries. Then EPN's Director, Alfonso Espinosa Ramon, while declaring the event inaugurated, said that as universities become integrated through joint efforts and exchange of criteria, with certainty there will be better perspectives for development of the institutions. These journeys not only increment scientific and technical knowledge in the telecommunications arena, but also strengthen the personal relationships among the participants, the professors and students of the different universities.

Continuing, Quito's City Council representative declared Honorable Visitors among the international speakers: Dr. Curtis Siller, Celia Desmond, Dr. Jaudelice Calvancante de Oliveira, and Fernando Blácido, giving them diplomas and replicas of the Virgen de Quito statue.

Additionally, the Quito City Council's delegate gave the Escuela Politecnica Nacional IEEE Student Branch special recognition for its valuable work for more than 30 years.

Thursday, May 15, began very early with the registration process, where participants were able to attend the first keynote speech by Dr. Curtis Siller, "Timed-Based Resource Reservation for End-to-End Quality of Services in Packet Networks." There were simultaneous translation services during the three days of the event.

The second keynote talk was offered by telecommunications expert Celia Desmond, who presented "Project Management for Telecommunications Projects." This is the third time Ms. Desmond has visited Ecuador to participate in an international event.

During the afternoon, technical papers received during the Call for Papers phase in previous months were presented. Approximately 100 papers had been submitted, from which the best 30 were selected. These selected papers were subjected to very close scrutiny based on rigorous qualification criteria, managed by Pablo Hidalgo, who coordinated the JST 2008 Technical Committee.

At the end of the day the organizers took the participants to visit the Centro Historico de Quito (Quito's historic district), which is UNICEF's Humanity Cultural Patrimony. The participants were able to know more about the history of the main churches and buildings of this very beautiful and much visited district.

The Distinguish Lecturers: Dr. Jaudelice Cavalcante, Dr. Curtis Siller, and Celia Desmond with Alex Aguirre, Relations & Media Committee President

The event's Organizing Committee during the visit to the Centro Historico de Quito. From left to right: Stephany, Belen, William, Jorge, Karina, Telmo (Local Coordinator), Alejandra, Valeria, Coky, Hoover, Santiago, and Daniel.

The last day of the event, Friday May 16, began with the highly anticipated keynote talk by Dr. Jaudelice Cavalcante de Oliveira, "Dominant Set Based ALLIANCES: A New Approach to Handle Bursty Traffic and Collisions in Sensor Networks."

During the day, Pablo Paredes, Mentor of the IEEE-EPN Student Branch, presented an interesting and short presentation about "Obsessive Compulsive Disorders and Potential Research Engineering Projects to Aid Patients and Health Care Professionals."

Concluding the keynote talks, Fernando Blacido, a distinguished telecommunications professional, presented "Hacia un Mundo Convergente (Toward a Convergent World)." He discussed the latest trends and developments in the Ecuadoran and world markets in new generation networks.

Making the best of the available time, the IEEE-EPN Student Branch volunteers coordinated a meeting with its Women in Engineering (WIE) affinity group and Jaudelice Cavalcante de Oliveira. They were able to share the life and leadership experiences of such a valuable woman, who is a truly inspiring example to many women in their engineering careers.

Many other presentations continued until the end of the day, concluding a very successful journey and fulfilling all the criteria of the organizing committee. The participants all received participation certificates. It is worth mentioning that these certificates were exclusively funded by the IEEE Communications Society. This was a good added value the organizers could offer to the participants thanks to Doug Zuckerman, ComSoc President, and his staff.

The IEEE-EPN student branch wants to express our

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Spanish Government Announces €130 Million Investment to Start a Dark Fiber Model for Its R&D Communication Network

By Juan Pedro Muñoz-Gea and Josemaría Malgosa Sanahuja, Spain

Spanish universities and R&D centers have their own academic and research network called RedIRIS, which is administered by the Red.es public entity. Currently, RedIRIS administers a communication infrastructure composed of 20 broadband nodes scattered around Spain. It consists of a trunk network with a mesh topology (called RedIRIS-10), composed of links with capacity up to 10 Gb/s. Therefore, RedIRIS allows the Spanish research community to use advanced national (and international) communication services. This is the reason today more than 300 academic and research institutions are connected to the RedIRIS network.

From its creation, the RedIRIS infrastructure has been constructed following the capacity-rent model, which consists of contracting a link collection with specific technology, capacity, and operation characteristics to be able to satisfy the demand for a period between two and four years. At the beginning of each period, the commercial operator that offers the most attractive solution to solve the new technical necessities is contracted.

However, with the predictable increase in future high-capacity circuit demand, the cost of the capacity-rent model is unattainable. Therefor, all European countries have studied new commercial models based on the property (or the lack of it in long-term rent) of the physical infrastructure, which is called the dark fiber model. Despite requiring high initial investment, this model also involves important economic savings when a service with a much bigger potential capacity transmission is considered.

In particular, the Spanish government decided to start the RedIRIS NOVA project, whose objective is the design and deployment of a new dark fiber network model for their R&D connectivity services during the 2008–2011 period. The project will connect the RedIRIS regional networks to one another with dark fiber, and all of them with the international academic network (GEANT 2) throughout its neighboring European countries: Portugal and France (FCCN and RENATER networks, respectively). To be precise, this project investment consists of €130 million to acquire an indefeasible right of use (IRU) of the lines offered by the operators for a minimal duration of 10 years, and appropriate optical transmission equipment to implement bearer communication services throughout the lifetime of the project.

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MANWEEK 2008/*continued from page 1*

Actually, these synergies made Manweek 2008 a very successful and inspirational event. Combined with a three-hour guided bus tour around Samos , the event became unforgettable.

The next Manweek will take place in Venice, Italy, back to its usual date, 26–30 October 2009. The admirable effort of Raouf Boutaba (Chair of the Manweek Steering Committee) and the support of IFIP WG6.6/CNOM made it possible to have Manweek 2008 one month ahead of its usual dates, causing a domino effect for IM '09. I would really to thank them from my heart.

EQUADOR BRANCH EVENT/*continued from page 3*

thanks for the total support of the IEEE Communications Society, which through its president Doug Zuckerman and previous President Nim Cheung helped surpass the expectations for this event. In the same way, the organizing committee thanks all the national and international enterprises that through their support helped develop a very rich and valuable set of talks and presentations, adding great value to this type of event. Finally, we congratulate the technical reviewers, who with their experience and knowledge made possible the selection of the best papers.

This text was translated by IEEE Student Branch Mentor Pablo Paredes, who can be contacted at pablo.e.paredes@intel.com.

Alex Aguirre is the Public Relations and Media Committee President for JST 2008. He is also a member of the SAC Ecuador Section and can be contacted at alex_aguirre@ieee.org.

WIMAX DEVELOPMENT/*continued from page 2*

for the rich countries of the region and will mainly be used as a substitute for DSL. The cost of customer premises equipment (CPE), lack of education, and absence of local content will be the major hurdles for penetration of WiMAX in the poorer nations. We hope that in the coming years these countries will start contributing to IEEE and 4G standards, and provide funding and manpower for R&D. Lastly, MENA should take the developments in WiMAX and 4G as an opportunity, avoid being a spectator, and become a valuable contributor to the overall food chain.

RIDING THE WAVE TO GLOBAL CONNECTIVITY

CALL FOR PAPERS AND PROPOSALS

IEEE GLOBECOM 2009 will feature a comprehensive technical program including several Symposia and a number of Tutorials and Workshops. IEEE GLOBECOM 2009 will also include an attractive expo program including keynote speakers, various Business, Technology and Industry fora, and vendor exhibits. Prospective authors are invited to submit original technical papers for presentation at the conference and publication in the Proceedings. Proposals for Tutorials, Workshops, and Fora are also invited. Visit the IEEE GLOBECOM 2009 website: http://www.ieee-globecom.org/2009 for details and submission information.

TECHNICAL SYMPOSIA

IEEE GLOBECOM 2009 will feature the following 11 technical symposia. For further details, please contact individual co-chairs or **Symposia Chair**, **Stefano Bregni**, bregni@elet.polimi.it.

Symposium on Selected Areas in Communications
(Tracks: Cognitive Radio and Networks, Emerging Technologies for Access Systems and Networks, Consumer Networks, Data Storage, Satellite and Space Communications, Other Related Technologies)
Mainak Chatterjee, mainak@eecs.ucf.edu
Masaaki Katayama, katayama@nuee.nagoya-u.ac.jp
Kwang-Cheng Chen, chenkc@cc.ee.ntu.edu.tw
Fatih Erden, fatih.erden@seagate.com
Claudio Sacchi, sacchi@disi.unitn.it
Tarek El-Bawab, telbawab@ieee.org

Ad-Hoc, Sensor and Mesh Networking Symposium
Nizar Bouabdallah, nizar.bouabdallah@inria.fr
Azzedine Boukerche, boukerch@site.uottawa.ca
Chunxiao (Tricia) Chigan, cchigan@mtu.edu
Ashfaq Khokhar, ashfaq@ece.uic.edu

Communication and Information System Security Symposium
A.Benslimane, abderrahim.benslimane@univ-avignon.fr
Stamatios V. Kartalopoulos, kartalopoulos@ou.edu
Qingming Ma, qma@cs.cmu.edu
Guenter Schaefer, schaefer@tu-ilmenau.de

Communication Theory Symposium
Lars Rasmussen, lars.rasmussen@unisa.edu.au
Merouane Debbah, merouane.debbah@supelec.fr
Elza Erkip, elza@poly.edu
Syed Ali Jafar, syed@uci.edu

Communication QoS, Reliability and Modeling Symposium
Fabrizio Granelli, granelli@disi.unitn.it
Hajime Nakamura, nakamura@kddilabs.jp

Communication Software and Services Symposium
Young-Tak Kim, ytkim@yu.ac.kr
Pascal Lorenz, lorenz@ieee.org
Biplab Sikdar, sikdab@rpi.edu
Qian Zhang, qianzh@cs.ust.hk

Next-Generation Networking and Internet Symposium
Nasir Ghani, nghani@ece.unm.edu
Ashwin Gumaste, ashwing@ieee.org
Xiaoming Fu, fu@cs.uni-goettingen.de
Deep Medhi, dmedhi-@umkc.edu

Optical Networks and Systems Symposium
Alberto Bononi, bononi@tlc.unipr.it
Galen Sasaki, galens@hawaii.edu
Naoaki Yamanaka, yamanaka.naoaki@ieee.org
Arunita Jaekel, arunita@uwindsor.ca

Signal Processing for Communications Symposium
Hung Henry Nguyen, hung.h.nguyen@aero.org
Tomohiko Taniguchi, t-taniguchi@jp.fujitsu.com
Hsiao-Chun Wu, wu@ece.lsu.edu

Wireless Communications Symposium
Robert Schober, rschober@ece.ubc.ca
Alberto Zanella, alberto.zanella@ieiit.cnr.it
Cheng Li, licheng@engr.mun.ca
Jingxian Wu, jingxian.wu@sonoma.edu

Wireless Networking Symposium
Sastri Kota, skota@harris.com
Maria Luisa Merani, merani.marialuisa@unimore.it
Tarik Taleb, taleb@aiet.ecei.tohoku.ac.jp
Jiang (Linda) Xie, linda.xie@uncc.edu

EXPO PROGRAM

IEEE GLOBECOM 2009 will feature several prominent keynote speakers, four major business and technology fora, and a large number of vendor exhibits. For vendor exhibits, please contact **Exhibits Chair: Jerry Gibbon**, jtgibbon@hotmail.com; for keynote speakers, please contact **Keynote Chair, Mahmoud Daneshmand**, daneshmand@att.com; and for the fora, please contact appropriate chairs:

Business, Technology, and Industry Fora Chair:
Chi-Ming Chen, chimingchen@att.com

CEO Forum:
Matt Bross, matt.bross@bt.com
Dilip Krishnaswamy, dilip@ieee.org

Designers & Developers Forum:
Jeff Friedhoffer, jafried@ieee.org

ACCESS Executive Forum:
Dave Waring, dwaring@telcordia.com
Gabriel Jakobson, gabejakobson@earthlink.net

EntNet Business Forum:
Daniel Minoli, minoli@comcast.net

TUTORIALS

Proposals are invited for half- or full-day tutorials in communication and networking topics. Proposals should be submitted to **Tutorials Chair, Nelson Fonseca**, nfonseca@ic.unicamp.br. Visit the conference website for detailed proposal guidelines.

WORKSHOPS

Proposals are invited for half- or full-day workshops in communication and networking topics. Proposals should be submitted to **Workshops Chair, Rolf Stadler**, stadler@ee.kth.se. Visit the conference website for detailed proposal guidelines.

IMPORTANT DEADLINES

> **Complete Paper:** 15 March 2009
> **Tutorial Proposal:** 15 March 2009
> **Workshop Proposal:** 15 March 2009
> **Acceptance notification:** 1 July 2009
> **Camera-ready papers:** 14 August 2009

EXECUTIVE COMMITTEE

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CONFERENCE REPORT

IEEE GLOBECOM 2008 EXPLORES THE FUTURE OF NETWORKED COMMUNICATIONS

For over 50 years, IEEE GLOBECOM has become the foremost venue for highlighting the latest advances in voice, data, image and multimedia communications.

Held at the Hilton New Orleans Riverside Hotel from November 30th to December 4th, IEEE GLOBECOM 2008 followed this esteemed tradition by bringing together nearly 2,000 designers, developers, academics, researchers and business professionals to discuss the ongoing convergence of telecommunications technologies that are expected to one day "solve the challenge of time and space."

The event began with a rousing grand opening as General Chair Richard Miller led a local jazz band around the exhibit floor, followed by hundreds of conference members decked out in masks, beards, beads, and other typical Mardi Gras attire. Hours later, attendees were still dancing to the local music while dining on a buffet of quintessential New Orleans cuisine.

On other networking and social fronts, IEEE ComSoc Past President Maurizio Decina was honored with the ComSoc/Exemplary Global Service Award at the Awards Luncheon as the ceremony was transmitted via the Internet for the first time. Also at the luncheon, Andrej Jajszczyk, Professor, AGH University of Science & Technology, received the Joe LoCicero Publication Exemplary Service Award. The award was renamed prior to the conference to honor the lifelong contributions of Joe LoCicero, who served as a leader on numerous IEEE ComSoc committees while working diligently as editor for nearly two decades to advance the quality and value of *IEEE Transactions on Communications*.

During the annual Conference Banquet that was highlighted by a serenade from the New Orleans Children's Choir, nearly a dozen "Best Paper" awards were announced for the authors of the conference's top one percent of accepted papers. Other presentations included the award of $1000 student travel grants at the Author's Breakfast, as well as the gift of numerous prizes including camcorders, DVD players and digital picture frames to attendees browsing the corporate booths and more than 150 poster displays prominently showcased in the conference Exhibit Hall.

Under the theme of "Building A Better World Through Communications," industry experts delivered keynote speeches that boldly described the inevitable creation of wireless ecosystems, merging all communications into one seamless multimedia, interactive experience.

During his keynote address at IEEE GLOBECOM 2008 in New Orleans, John Donovan, chief technology officer at AT&T, noted, "We are all witnesses to the most profound changes in the history of communications. Tomorrow's integrated IP applications will make life a lot easier as service providers like AT&T partner with a range of other companies to combine network connectivity with the right devices and applications. This will deliver not only more connectivity, but the power to get more from — and do more with — that connectivity. More functionality ... delivered more intuitively ... with less complexity."

Dr. Flavio Bonomi, head of research at Cisco Systems, furthered this concept when he described the inevitability of "flat mobility access" and the industry's continued dedication to "learning the language of applications," which will "bridge the gap" toward "cloud computing."

"Soon partnerships will extend across academia, government and industry to create a perfect storm of inter-connectivity," said Bonomi. "Billions of new users and millions of new applications will be available though the Internet over the next decade. In a short time, TV will become a mobile device and the car will become a center for mission-critical activities, business and entertainment. A new platform of computing is emerging there, which will merge infotainment with business to create mobile offices that will also provide interactive entertainment capabilities anytime, anywhere."

Pankaj Asundi, vice president of media and content of Ericsson, Inc., punctuated his presentation by stating that the "consumer multimedia market will be worth $149 billion in 2011." According to Asundi, "Multi-device interaction across the globe will become a reality in the very near future as the online experience continually shifts from mass media to 'me' media and consumption becomes a totally 'on demand' process. We are on the verge of an integrated communications experience that will spin 360 degrees to offer rich, compelling and interactive opportunities provided through the convergence of TV, print, mobile, voice and data."

Other featured IEEE GLOBECOM 2008 speakers included Kaoru Yano, president of the NEC Corporation, who discussed the building of a sustainable information society, and Richard J. Lynch, executive vice president and chief technology officer at Verizon Communications, who shared his views on "The Power of Broadband Innovation" and the increased globalization of communications networks and businesses.

Throughout IEEE GLOBECOM 2008, hundreds of industry experts and researchers exemplified the overall conference theme with demonstrations and presentations exploring the latest innovations in telemedicine, disaster recovery, cognitive radio, seamless mobility, optical switching and wireless multimedia communications. This included the presentation of 153 technical sessions, 27 poster sessions, 18 tutorials and nine technical workshops. In addition, IEEE ComSoc enhanced the educational value of the conference's many tutorials and workshops by distributing the materials on USB flash drives for the first time.

Additional IEEE GLOBECOM 2008 events served to provide premier learning, networking and meeting opportunities for the profession's top researchers, academics and applications specialists. A "Gold Panel" of business and industry leaders discussed the continued growth of opportunities for qualified professionals within the wireless field as well as tutorials on "The History of Communications" and "Who Invented Radio...?" presented by five noted historical researchers.

Another well-attended event was the IEEE International Conference on Enterprise Networking and Services (EntNet) 2008 that hosted discussions outlining "Service-Oriented Architectures" and the "Evolving Technologies in Support of the Oil Industry." In addition, Dr. Nancy Victory, Chair of the FCC Katrina Report offered her expert insights on "Business Continuity, Disaster Preparedness and Disaster Recovery Including Lessons Learned from Katrina," while Dr. Paul Mockapetris, Inventor of the Domain Name System, addressed "DNS Revolutions & Evolutions."

Other key networking and educational events held throughout IEEE GLOBECOM 2008 included:

* 3rd Annual IEEE Communications Industry Forum & Expo, providing attendees with the opportunity to tour the Expo exhibit hall and learn about the latest telecommunications solutions currently under development by leading companies such as Telecordia Technologies, NEC, NIKSUN and OPNET Technologies

* Design & Developers Forum, which offered 18 sessions dedicated to cutting-edge topics like "Wireless Access for Vehicular Environments," "B3G & 4G Mobile Wireless Broadband Technologies" and "Security for Seamless Mobility"

Planning has already begun for IEEE GLOBECOM 2009, which will be held in Honolulu, Hawaii from November 30th to December 4th at the Hilton Hawaiian Village. Under the theme "Riding the Wave to Global Connectivity," researchers, academics, engineers and business professionals are urged to submit presentation proposals to conference planners by March 15, 2009. IEEE GLOBECOM 2009 will detail the latest research, solutions and applications related to leading communications topics such as cognitive radio & networks, communications theory, satellite and space communications, signal processing, optical & consumer networks, data storage, broadband signaling and much more.

Additional information is available at www.ieee-globecom.org/2009 or can be obtained by contacting Heather Ann Sweeney via email at h.sweeney@comsoc.org.

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CONFERENCE CALENDAR

2009

JANUARY

● ***COMSNETS 2009 - 1st Int'l. Conference on Communication Systems and Networks, 5-10 Jan.***
Bangalore, India. Info: http://www.comsnets.org

■ ***IEEE CCNC 2009 - IEEE Consumer Communications & Networking Conference, 10-13 Jan.***
Las Vegas, NV. Info: http://www.ieee-ccnc.org/2009

● ***RWS 2009 - 2009 IEEE Radio and Wireless Symposium, 16-23 Jan.***
San Diego, CA. Info: http://www.radiowireless.org

● ***ICOIN 2009 - Int'l. Conference on Information Networking 2009, 20-23 Jan.***
Chiang Mai, Thailand. Info: http://www.icoin.org

FEBRUARY

● ***WONS 2009 - 6th Annual Conference on Wireless On-Demand Network Systems and Services, 2-4 Feb.***
Snowbird, UT. Info: http://nets.cs.ucla.edu

● ***ISWPC 2009 - Int'l. Symposium on Wireless Pervasive Computing, 11-13 Feb.***
Melbourne, Australia. Info: http://www.iswpc.org/2009

● ***ICACT 2009 - 11th Int'l. Conference on Advanced Communication Technology, 15-18 Feb.***
Gangwon-Do, Korea. Info: http://www.icact.org

MARCH

■ ***OFC/NFOEC 2009 - 2009 Conference on Optical Fiber Communication, 22-26 March***
San Diego, CA. Info: http://www.ofcnfoec.org

■ ***IEEE ISPLC 2009 - IEEE Int'l. Symposium on Power Line Communications and Its Applications, 29 March-1 April***
Dresden, Germany. Info: http://www.comsoc.org/confs/index.html

● ***IEEE Sarnoff 2009 - IEEE SARNOFF Symposium, 30 March-1 April***
Princeton, NJ. Info: http://ewh.ieee.org/r1/princeton-centraljersey/2009_Sarnoff_Symposium/index.html

APRIL

■ ***IEEE WCNC 2009 - IEEE Wireless Communications and Networking Conference, 5-8 April***
Budapest, Hungary. Info: http://www.ieee-wcnc.org/2009

■ ***IEEE INFOCOM 2009 - 28th Annual IEEE Conference on Computer Communications, 19-24 April***
Rio de Janeiro, Brazil. Info: http://www.ieee-infocom.org/2009

● ***WTS 2009 - Wireless Telecommunications Symposium 2009, 22-24 April***
Prague, Czech Republic. Info: http://www.csupomona.edu/wtsi

■ ***IEEE RFID 2009 - 2009 IEEE Int'l. Conference on RFID, 27-28 April***
Orlando, FL. Info: http://www.ieee-rfid.org/2009

● ***WOCN 2009 - 6th Int'l. Conference on Wireless and Optical Communications Networks, 28-30 April***
Cairo, Egypt. Info: http://www.wocn2009.org

MAY

● ***MC-SS 2009 - 7th Int'l. Workshop on Multi-Carrier Systems & Solutions, 5-6 May***
Herrsching, Germany. Info: http://www.mcss2009.org

● ***CNSR 2009 - Communication Networks and Services Research 2009, 11-13 May***
Moncton, NB, Canada. Info: http://www.cnsr.info/events/csnr2009

■ ***IEEE CTW 2009 - IEEE Communication Theory Workshop, 11-14 May***
St. Croix, U.S. Virgin Islands. Info: http://www.ieee-ctw.org/2008/index.html

■ ***IEEE CQR 2009 - 2009 IEEE Int'l. Workshop, Technical Committee on Communications Quality and Reliability, 12-14 May***
Naples, FL. Info: http://www.ieeee-cqr.org/

JUNE

■ ***IM 2009 - IFIP/IEEE Int'l. Symposium on Integrated Network Management, 1-5 June***
Hempstead, NY. Info: http://www.iee-im.org/2009

● ***ConTEL 2009 - 10th Int'l. Conference on Telecommunications, 8-10 June***
Zagreb, Croatia. Info: http://www.contel.hr

● ***IWCLD 2009 - Int'l. Workshop on Cross Layer Design 2009, 11-12 June***
Mallorca, Spain. Info: http://www.iwcld2009.org

■ ***IEEE ICC 2009 - IEEE Int'l. Conference on Communications, 14-18 June***
Dresden, Germany. Info: http://www.comsoc.org/confs/icc/2009/index.html

■ ***SECON 2009 - IEEE Communications Society Conference on Sensor and Ad Hoc Communications and Networks, 22-26 June***
Rome, Italy. Info: http://www.ieee-secon.com/2009/

JULY

■ ***IEEE WiMAX 2009 - 2009 IEEE Mobile WiMAX Symposium, 9-11 July***
Napa, CA. Info: chenkc@cc.ee.ntu.edu.tw

■ ***IWQoS 2009 - Int'l. Workshop on Quality of Service 2009, 13-15 July***
Charleston, NC. Info: http://iwqos09.cse.sc.edu

● ***NDT 2009 - 1st Int'l. Conference on Networked Digital Technologies, 28-31 July***
Ostrava, Czech Republic. Info: http://arg.vsb.cz/NDT2009/

■ Communications Society sponsored or co-sponsored conferences are indicated with a square before the listing; ● Communications Society technically co-sponsored or cooperating conferences are indicated with a circle before the listing. Individuals with information about upcoming conferences, calls for papers, meeting announcements, and meeting reports should send this information to: IEEE Communications Society, 3 Park Avenue, 17th Floor, New York, NY 10016; e-mail: p.curran@comsoc.org; fax: +1-212-705-8999. Items submitted for publication will be included on a space-available basis.

NEW PRODUCTS

ANRITSU COMPANY UPGRADES LTE SOFTWARE AND ANALYSIS TOOLS

Anritsu Company

The improved software packages from Anritsu Company allow Anritsu's MS269xA Signal Analyzer series and MG3700A Vector Signal Generator to conduct highly accurate measurements on LTE Uplink and Downlink signals. With the accuracy required for R&D and the speed needed for manufacturing, Anritsu's test equipment facilitates efficient testing of 3GPP LTE-compliant mobile terminals, base stations, and components.

For the MS269xA, both the LTE Uplink and Downlink packages now include analysis at the resource block level. Analysis resolution can be set to the physical channel, resource block, subcarrier, and symbol. EVM, a key figure of merit for LTE, is available at any resolution. Users can quickly test all varieties of unwanted emissions described in the specification with the MS269xA. By pressing a single button, OBW, ALCR, and Out-of-Band Spurious Emissions measurements can be made.

IQproducer, Anritsu's waveform generation software, has been enhanced to allow users to define test waveforms for the Uplink or Downlink. While standard test waveforms are not yet part of the specification, IQproducer provides users with a rich set of parameters and variations to create standard and unique waveforms for LTE signal analysis. When IQproducer is installed on the MS269xA and the MG3700A, the instruments can up convert and generate any waveform, including Uplink and Downlink data frames, and Uplink Random Access Preamble signals.

About MS269xA

The MS269xA Signal Analyzers support all popular wireless technologies. In addition to LTE, the signal analyzers support GSM, GPRS, EDGE, W-CDMA, HSPA, and mobile WiMAX. Three models covering frequency ranges of 50 Hz to 6 GHz (MS2690A), 50 Hz to 13.5 GHz (MS2691A), and 50 Hz to 26.5 GHz (MS2692A) are available, so users can select the instrument that matches their measurement requirements. With a fundamental band that goes to 6 GHz, these high-performance signal analyzers have excellent level and modulation accuracy.

The basic instrument includes a spectrum analyzer for classic swept mode measurements, signal analyzer for high-speed, wideband FFT analysis, and a digitizing function that allows the capture and replay of any signal received by the analyzer. The basic analysis bandwidth is 31.25 MHz, the widest basic analysis bandwidth on the market. The addition of the broadband hardware option increases the analysis bandwidth to 125 MHz. An optional 6 GHz vector signal generator creates a one-box tester that saves time, money, and space on the bench.

About the MG3700A

With its 160 MHz high-speed arbitrary waveform baseband generator, wide vector modulation bandwidth, and large capacity ARB memory, the MG3700A has the performance to provide signal generation for new and emerging wireless communication systems. Equipped with two waveform memories that simultaneously produce data streams, the MG3700A generates real-world signals with multiple-source interference, AWGN, and other signal impairments. The MG3700A eliminates the need for a second signal generator when evaluating receiver characteristics.

www.us.anritsu.com

CML UNVEILS RF QUADRATURE MODULATOR AT ELECTRONICA'08

CML Microcircuits

CML Microcircuits has unveiled a new I/Q modulator, the CMX993, that is a highly integrated, general purpose, RF quadrature modulator, offering 100MHz to 1GHz operation and excellent wide-band noise performance. Additional features include gain control and uncommitted differential amplifiers. Targeted to meet the challenging requirements of wireless data (digital TV, CATV modulators, ISM transmitters, Wireless LAN, WLL) and two-way radio systems (APCO P25), the product,s I/Q architecture supports a wide range of modulation types. Various integrated, selectable functions are offered, to maintain performance across multiple modulations and bandwidths.

The quadrature modulator provides translation from baseband I/Q signals to a modulated RF signal. The wideband inputs can be driven single-ended or differentially for optimum performance. The device offers two integrated and matched double-balanced mixers, driven from a buffered quadrature split local oscillator. The ŒLO, frequency is divided by either 2 or 4, with the mixers forming an I/Q vector modulator with programmable gain stages, offering 30dB of gain control in 2.5dB steps.

A digital control interface, C-BUS (SPI-compatible interface), allows gain control as well as the power management of individual internal blocks for optimum system performance. The C-BUS interface operates from its own power supply, enabling the CMX993 to be interfaced to baseband devices of differing voltages.

www.cmlmicro.com

ROGERS ROLLS OUT HIGH FREQUENCY MATERIALS AND LEAD-FREE LAMINATES

Rogers Corporation

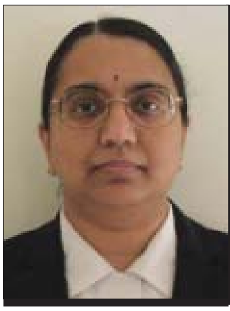
The Advanced Circuit Materials (ACM) Division of Rogers Corporation exhibited its RT/duroid high frequency materials and its RO3000 and RO4000 laminates at the IEEE Radio & Wireless Symposium in San Diego in January.

Featured RT/duroid materials were the RT/duroid 5000 and RT/duroid 6002 as well as the newest addition to the RT/duroid family of high frequency materials: RT/duroid 6202PR for use in designing complex microwave structures such as antennas and multilayer circuits with interlayer connections.

RT/duroid 6202PR offers the same electrical and mechanical properties as RT/duroid 6202, in addition to enabling tight tolerance planar resistors (PR) when clad with resistive copper foils.

www.rogerscorp.com

LTE PART I: CORE NETWORK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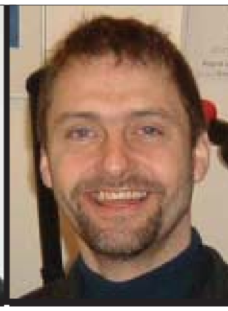
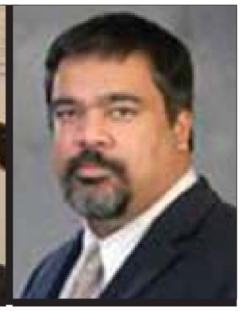
Reiner Ludwig


Preben Mogensen


Vish Nandlall


Vojislav Vucetic


Byung K. Yi


Zoran Zvonar

Current cellular networks based on Third Generation Partnership Project (3GPP) and 3GPP2 technologies provide evolution from circuit-switched technologies, originally developed for voice communications, to packet-switched technologies. Next-generation networks need to deliver IP-based services (voice, video, multimedia, data, etc.) for all kinds of user terminals while moving between fixed (fiber, DSL, cable) and wireless (3GPP-based, 3GPP2-based, IEEE-based) access technologies, and roaming between various operator networks. Users expect the network to originate, terminate, and maintain a session while the user is moving and roaming. Services have to be delivered to users based on serving network functionality (quality of service [QoS], bandwidth, etc.), availability, and user preferences. The network and users must be protected through various authentication, encryption, and other security mechanisms at the access, network, and application layers. Mobility has to be provided through coordinated link, network, and application layer mobility mechanisms that ensure user expectations of service performance are met. Requirements on the radio technology include improved performance as well as reduced system and device complexity. 3GPP Release 8 specifies the architecture to meet the above requirements.

3GPP has finalized the Release 8 specifications of the 3GPP evolved packet system (EPS). The two key work items of 3GPP Release 8 are the service architecture evolution (SAE) and long term evolution (LTE). The standardization work on those two work items, which started in 2005, has led to specifications of the evolved packet core (EPC) and a new radio access network referred to as the evolved universal terrestrial radio access network (E-UTRAN). The completion of the SAE/LTE Release 8 specifications represents a milestone in the development of standards for the mobile broadband industry.

The EPC is a multi-access core network based on the Internet Protocol (IP) that enables operators to deploy and operate one common packet core network for 3GPP radio access (LTE, 3G, and 2G), non-3GPP radio access (HRPD, WLAN, and WiMAX), and fixed access (Ethernet, DSL, cable, and fiber). The EPC is defined around the three important paradigms of mobility, policy management, and security. The EPC provides user terminals with optimized handover schemes between different radio access technologies (e.g., between LTE and HRPD). Standardized roaming interfaces enable operators to offer their subscribers global services connectivity across a range of different access technologies. The network-controlled and class-based QoS concept of the EPC is based on 3GPP's policy and charging control (PCC) framework. This maximizes operator control over all PCC/QoS functions that are distributed across different network elements, including the user terminal.

The LTE radio access is based on orthogonal frequency-division multiplexing (OFDM) and supports different carrier frequency bandwidths (1.4–20 MHz) in both frequency-division duplex (FDD) and time-division duplex

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GUEST EDITORIAL

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(TDD) modes. This provides great flexibility for operators to use existing and future radio spectrum allocations. The LTE radio access is based on shared channel access providing peak data rates of 75 Mb/s in the uplink direction and 300 Mb/s in the downlink direction. Improved coverage and battery lifetime have been key goals in the development of the LTE specifications. Unlike the 2G/3G 3GPP radio access networks, which are connected to the circuit-switched domain of the 3GPP core network, the E-UTRAN is only connected to the EPC. The E-UTRAN protocols and user plane functions have therefore been optimized for the transmission of traffic from IP-based real-time and non-real-time applications/services.

Part I of this Feature Topic will focus on the 3GPP Release 8 EPC, the standard that is the flat SAE architecture. This new architecture is designed to optimize network performance, reduce total cost of ownership, increase cost efficiency, and facilitate the uptake of mass market IP-based services. The system is considered "flat" as there are only two nodes in the SAE architecture user plane: the LTE base station (eNodeB) and the gateway, as shown in Fig. 1. The flat architecture reduces the number of nodes involved in the signaling and media paths. With incorporation of radio network controller (RNC) functionality inside eNodeB, handovers will be negotiated and managed directly between eNodeBs, which will mimic those currently employed in 3G UTRAN networks.

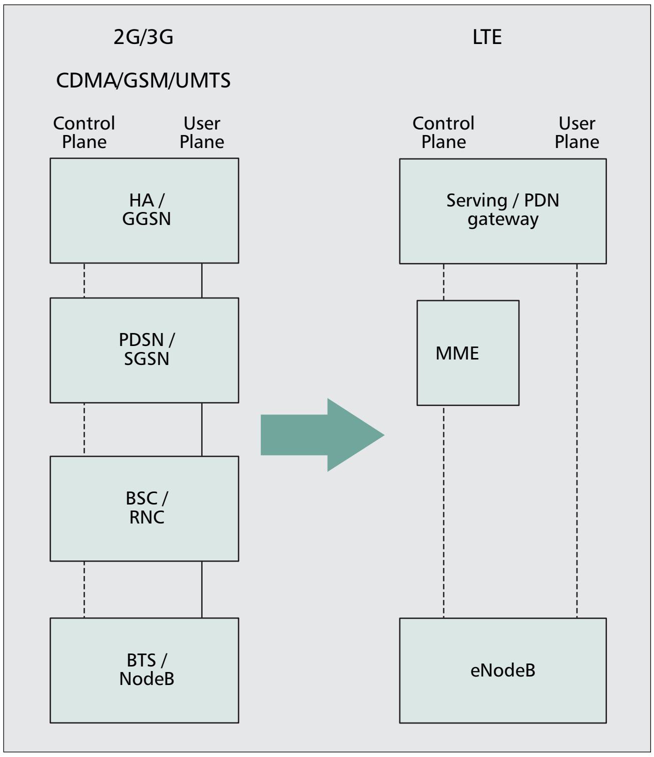

Figure 1. *From hierarchical to a simpler, flatter network.*

A key difference from current networks is that the EPC is defined to support IP packet-switched traffic only. Interfaces are based on IP protocols. This means that all services will be delivered through packet connections, including voice. To this end, voice call continuity between circuit-switched voice systems and packet-switched voice over IP systems has received particular attention in Release 8. It is assumed that voice services will be implemented through the use of an IP multimedia subsystem (IMS). The article "Voice Call Handover Mechanisms in Next-Generation 3GPP Systems," coauthored by Apostolis Salkintzis, Mike Hammer, Itsuma Tanaka, and Curt Wong, provides an overview of the voice call handover techniques and mechanisms that enable handover at any time in the call. They also present scenarios in handover that are also known as single radio voice call continuity (SR-VCC) and circuit-switched fallback (CSFB). The techniques described in this article enable mobility and service continuity between existing and future access networks; that is, interworking from E-UTRAN access to UTRAN/GERAN or 1xRTT access. Access selection is based on combinations of operator policies, user preferences, and access network conditions

Existing 3GPP (GSM and WCDMA/HSPA) and 3GPP2 (CDMA 1xRTT, EVDO) systems are integrated with the EPS through standardized interfaces providing optimized mobility with LTE. This means a signaling interface between the signaling GPRS service node (SGSN) and the evolved core network for 3GPP systems, and a signaling interface between the code-division multiple access (CDMA) RAN and the EPC for 3GPP2 systems. Such integration will support both dual and single radio handover and allow for flexible migration to LTE. It should also be noted that wireless LAN or WiMAX radio access could also be integrated into the EPC. The article "Network-Based Mobility Management in the Evolved 3GPP Core Network," coauthored by Irfan Ali, Alessio Casati, Kuntal Chowdhury, Katsutoshi Nishida, Eric Parsons, Stefan Schmid, and Rahul Vaidya, covers network mobility and functions that enable operators to provide a common set of services and mobility at the IP layer across various access networks.

Release 8 also includes a class-based QoS concept. This provides a simple yet effective solution for operators to offer differentiation between packet services. The policy and charging rules function (PCRF) handles QoS management, and also controls rating and charging. Subscriber management and security is the responsibility of the home subscriber server (HSS). Javier Pastor, Stefan Rommer, and John Stenfelt authored the article "Policy and Charging Control in the Evolved Packet System," and address how to provide access agnostic policy control that can be applied to a variety of access networks, including E-UTRAN, UTRAN, GERAN, eHRPD, and WiMAX. This article covers an overview of policy and control functions and the corresponding implementation in EPS 3GPP Release 8. The article "QoS Control in 3GPP Evolved Packet System," authored by Hannes Ekström, is about QoS concepts that enable network operators and service providers with effective techniques to enable subscriber and services differentiation, and maintain the QoS across the end-to-end systems.

The security mechanisms in wireless systems were essential functional elements of GSM and UMTS. However, most of the focus was placed on the radio path. Through evolution of cellular systems, security aspects have evolved to include several network functionalities as well. In EPS architecture, security is more robust in order to encompass end-to-end security both in the radio as well as the core, and additionally spanning across multiple access networks (inter Radio Access Technology). The article "Network Access Security in Next Generation 3GPP Systems" by C. B. Sankaran is a tutorial covering key attributes that are essential for services offered by the operator related to network access security, including inter-radio access technology (inter-RAT).

Several trial activities in LTE are already underway globally. The article "Multisite Field Trial for LTE and Advanced Concepts," coauthored by Ralf Irmer, Hans-Peter Mayer, Andreas Weber, Volker Braun, Michael Schmidt, Michael Ohm, Andre Zoch, Carsten Jandura, Patrick Marsch, and Gerhard Fettweis, presents LTE performance in a multisite field trial. It includes an overview of LTE standards, and the industry alliances and initiatives driving the requirements and performance (e.g. NGMN and LSTI). Among the requirements defined by NGMN for LTE are increased peak and average data rates, reduced latency, high spectral efficiency, and cell edge throughput. Results from simulation and field tests are provided in this article on key performance indicators such as throughput and latency. Radio interface physical layer features enabling performance, including vhannel coding and physical channel mapping, MIMO diversity, as well as UE physical layer capabilities, are presented in the results.

As summarized above, Part I covers several key aspects of the EPS architecture. Part II of this Feature Topic will focus on RAN technology and standards.

BIOGRAPHIES

KALYANI BOGINENI (Kalyani.Bogineni@VerizonWireless.com) is principal architect at Verizon Communications with extensive experience in architecture and design of telecommunications networks for wireless and wireline technologies as well as various application technologies. She has published extensively in IEEE/ACM peer-reviewed journals and conferences. She has been on the Technical Program Committees for several conferences, and has been a reviewer for various IEEE journals and magazines for over 18 years. She is an active speaker on next-generation converged networks at various conferences and panels. Recently she has been active in the development of 3GPP standards for 4G technologies focused on the development of converged networks for multiple access technologies with IP-based mobility management mechanisms, policy-driven roaming architectures, and converged security architectures. She has B.Tech and M.E. degrees in electrical engineering, an M.S. degree in computer engineering, and a Ph.D. in electrical and computer engineering.

REINER LUDWIG received his Diploma and doctoral degree in computer science from the University of Technology, Aachen, Germany, in 1994 and 2000, respectively. He joined Ericsson in 1994 working within the Research Department on cross-layer aspects of wireless packet-based networks. He has worked within the Internet Engineering Task Force (IETF), where he co-authored standards on operating end-to-end protocols across wireless access networks. More recently, he has been actively involved in the standardization of the policy and QoS framework of the 3GPP EPS, including link layer aspects of the LTE radio access. He currently holds an expert position in the Systems and Technology Department of Ericsson's Business Unit Networks, where he is responsible for policy and QoS control for fixed and mobile access networks.

PREBEN MOGENSEN received his M.Sc.E.E. and Ph.D. degrees in 1988 and 1996, respectively, from Aalborg University (AAU), Denmark. Since 1999 he has been a part time professor in the Department of Electronic Systems, AAU, where he heads the Radio Access Technology (RATE) research section. He also holds a part time position as principal engineer at Nokia Siemens Networks, Aalborg, where he is involved in LTE and LTE-Advanced standardization research. He is author or co-author of more than 170 technical publications within a wide range of areas, including radio wave propagation, advanced antenna technologies, receiver design, frequency assignment, radio resource management, and packet scheduling.

VISH NANDLALL is the chief technical officer for Carrier Networks at Nortel. He is responsible for Nortel's technology vision in 4G and in particular LTE, and has shaped Nortel's product and standards strategy in this field, advocating seamless intertechnology handoff and flat network topologies. He has spent the last 15 years in architecture roles within Nortel, most recently as chief architect for Nortel's CDMA and EVDO wireless access division, contributing to the launch of high-speed data services in North America and Eastern Europe. Prior to his life in wireless, he contributed to Nortel's Metro Optical and DMS product lines, providing key technologies in core computing and private line services. His current research is in cross-layer design for cellular interference control and scheduling in direct relay systems.

VOJISLAV VUCETIC received his Ph.D. degree from Imperial College, London, United Kingdom. In 1988 he joined AT&T Bell Laboratories, where he worked on software design and software architecture for data communications systems, and network designs for data carrier networks internationally. In 1998 he joined Cisco, where he worked as a consulting engineer supporting U.S.-based and international service providers. He also contributed to metro Ethernet and cable-based VoIP development activities. Currently he is a senior manager in the Carrier Standards and Architecture group. He leads a group that is responsible for coordinating industry and standards activities with Cisco carriers' development organizations and service providers. His current focus is on architecture and protocols for access-agnostic IP-based networks to support 3GPP, 3GPP2, WiMaX, and other access technologies.

BYUNG K. YI, senior executive vice president of LG Electronics, has over 32 years of experience in research and development of communication and space systems. He has been working on 3G and 4G wireless communication systems. He served as TSG-C chair of 3GPP2 for two terms, developing cdma2000 air interface specifications, and served as a co-chair of Working Group 5 of 3GPP2 TSG-C, developing 1xEV/DV wireless standards. Under his leadership, TSG-C published three important air interface standards, cdma2000 Rev. D, and High Rate Packet Data (HRPD) Revs. A and B. He is currently heading the LGE North America R&D center, developing mobile terminals for North American carriers. He was in charge of small satellite system engineering for distributed low earth orbiting telecommunication and remote sensing applications at Orbital and CTA as a chief engineer. He taught graduate courses for nine years at George Washington University as an adjunct professor. His current interests are wireless and space communication systems, iterative decoding, and space system engineering. He holds eight U.S. patents and five international patents in the areas of iterative decoding and handoff schemes for cellular-based systems.

ZORAN ZVONAR () is director of systems engineering, MediaTek Wireless, and a MediaTek Fellow. He received a Dipl.Ing. in 1986 and an M.S. degree in 1989 from the Department of Electrical Engineering, University of Belgrade, Serbia, and a Ph.D. degree in electrical engineering from Northeastern University, Boston, Massachusetts, in 1993. From 1994 to 2008 he pursued industrial carrier within Analog Devices. He was a member of the core development team for the baseband platform and RF direct conversion transceiver wireless product families, and has been a recipient of the company's highest technical honor of ADI Fellow. Since January 2008 he has been with MediaTek focused on the design of algorithms and architectures for cellular standards, with applications to integrated chip set solutions and real-time software. He is the Editor of the Radio Communications Series in *IEEE Communications Magazine* and has served as Guest Editor and the member of the editorial boards for a number of professional journals in wireless communications.

The Momentum behind LTE Adoption

By Darren McQueen

Video, flat-rate pricing, and connected devices all contribute to the growing demand for mobile data services.

In the world of telecommunications, people today are more connected and more mobile than ever. We have more devices and more ways to stay in touch with one another. The Internet and wireline worlds are experiencing a rapid convergence of IP video, audio, and data into completely new applications. Users want that same on-demand access and Internet, multimedia experience, and content anywhere from any device.

We are also consuming huge amounts of data while on the go. Some HSPA operators were reporting an increase of 6–14x mobile data usage in 2007, and saw an increase of 30–50x in the first nine months of 2008. That demand for mobile data is quickly pushing third-generation (3G) and 3.5G networks to capacity, motivating operators to pursue 4G solutions like long-term evolution (LTE) today to maintain a competitive edge and add capacity to support mobile broadband take-up.

This unprecedented demand for mobile data is driven by several factors: flat-rate tariffs, a proliferation of connected laptops, devices with large screens and exciting user interfaces, and video. There is a video explosion with more video content embedded on Web pages, and Web 2.0 sites that are relying more heavily on video. Video is already driving much higher data usage on fixed line broadband networks and will soon make its way to mass market mobile broadband, continuing to feed the growth of data consumption on mobile networks.

As an all-IP technology, LTE with its evolved packet core (EPC) can interconnect and hand over between other all-IP-based access technologies like WiFi and digital subscriber line (DSL) to enable media mobility. The fixed/mobile convergence capabilities of LTE offer possibilities for operators to embark on a strategy that transforms them into communications providers that break the "wall" between home connections and the outside world — allowing personalized broadband for users.

For example, using LTE an operator can deploy a host of integrated applications that provide media mobility, allowing content to follow the user from one device and one location to another. That capability to offer "follow me" content can create a new source of revenue for operators and a powerful proposition for consumers who no longer would have to stay home to download big files or upload videos to YouTube.

Mobile devices that have more intuitive user interfaces are also responsible for the increasing demand for mobile data. Web-friendly smart phones are turning mobile customers into prodigious consumers of wireless data services. We are enabling users to do more with a mobile device than just text, email, and voice. With the advent of Web 2.0 sites tuned for mobile device access, the industry is bracing for an unprecedented increase in data traffic associated with devices capable of accessing graphically rich Internet and video-based content. As prices on these powerful multimedia mobiles decrease and choice increases, they will further penetrate the mass market.

The sheer number and types of connected devices, including new consumer electronics like digital cameras, MP3 players, and camcorders, is also fueling mobile data usage. According to ABI Research, shipments of network-enabled consumer devices are expected to go from 92 million in 2007 to 460 million by 2012, further driving network usage and challenging their capacity.

Operators themselves are a key component of the equation. With the advent of flat-rate pricing, consumers are encouraged to make the most of their devices to share multimedia digital content, play games online, and enjoy video while on the go. As operators provide a great mobile media experience, users will continue to rely on their device as part of their lifestyle.

This skyrocketing demand for mobile data presents a challenge for network operators as their existing networks become capacity constrained. The radio access network (RAN) grid, backhaul, radio network controller (RNC), and packet data core (PDC) on these networks were all dimensioned for voice and 3G data usage. So while network traffic is increasing, data pricing is flattening.

This creates a revenue challenge for operators. They need to upgrade their networks to offer a more compelling user experience. What they need is a solution that offers a lower cost per bit, higher capacity, and faster data speeds. For many operators, LTE will be their answer.

Why LTE?

LTE is popularly called a 4G technology. It is an all-IP technology based on orthogonal frequency-division multiplexing (OFDM), which is more spectrally efficient — meaning it can deliver more bits per Hertz.

LTE will be the technology of choice for most existing Third Generation Partnership Project (3GPP) and 3GPP2 mobile operators. It will provide economy of scale and spectrum reuse. LTE also offers smooth integration and handover to and from existing 3GPP and 3GPP2 networks, supporting full mobility and global roaming, and ensuring that operators can deploy LTE in a gradual manner by leveraging their existing legacy networks for service continuity. LTE also brings subscribers a "true" mobile broadband (~5–10 Mb/s/~15 ms latency) that enables a quality video experience and media mobility.

The LTE standard has been defined with as much flexibility as possible so that operators can deploy it in all current existing frequencies as well as new spectrum. Operators can deploy the technology in as little as 1.4 MHz or as much as 20 MHz of spectrum and grow the network as demand for data services grows.

LTE will also appear in a number of different spectrum bands around the world, including the new 2.6 GHz band, which is perfect as a capacity band since operators are able to secure up to 2 × 20 MHz of virgin spectrum. LTE can also be deployed in refarmed GSM bands in 900 MHz and 1800 MHz and digital dividend spectrum (e.g., 700 MHz in the United States), providing superior coverage and global roaming in the rest of the 3GPP market.

With improvements in capacity, speed, and latency, LTE will not only make accessing applications faster, but will enable a wealth of new applications previously available only on a wired Internet connection. The wall between wired and wireless will come down. And moving from one environment to another with your content moving seamlessly will become second nature:

- Continuing to watch the latest TV series recorded on your DVR, automatically transferred to the 4G network as you walk out the door

- Uploading content onto your social networking profile to let your friends know what you are up to
- The PowerPoint file you just saved on your laptop instantaneously becoming available on your Smartphone
- Or even your LTE-enabled digital camera uploading your latest picture onto your home server or social networking site for your family to see

OPERATOR INVESTMENT DECISIONS

So what sort of investment should operators make to meet this growing consumer demand for mobile broadband? Should they continue to invest in the 3.5G technology they have already deployed or make the necessary investments now to move to the inevitable: an all-IP data-optimized technology that offers ultra-fast broadband services and much improved capacity?

The scenario is different for 2G, 2,5G, 3G, 3.5G, and time-division duplex (TDD) operators. Motorola believes LTE will be the technology of choice for most existing 3GPP and 3GPP2 mobile operators because LTE can be deployed in existing and new frequency-division duplex (FDD) spectrum bands. In addition, LTE offers smooth integration with the ability to keep global roaming agreements and hand over calls to existing 3GPP and 3GPP2 networks, thereby offering the coverage benefit of existing 2G and 3G networks.

For most 3GPP2 operators, LTE is their 4G choice. With LTE being imminent, 2G and 2.5G operators in emerging markets that have GSM or EDGE networks may find leapfrogging directly to LTE rather than making an intermediary step to HSPA a better option. With LTE, they will get the lowest cost per bit, and the increased network capacity to support their medium- and long-term business objectives.

TDD operators also can benefit by moving to LTE since LTE is also capable of using TDD spectrum. That gives global operators the ability to standardize on one mobile broadband technology across all their markets and provide roaming between TDD and FDD LTE.

For 3G and HSPA+ operators there are other considerations. While HSPA has effectively spurred mobile data adoption, the next iteration, HSPA+, brings improvements mainly in terms of peak data rate for subscribers nearest to the cell site, but very little in terms of capacity improvement — about 20 percent — over HSPA. This increased peak data rate will benefit only the occasional geographically advantaged subscribers (the ones closest to the antenna) but will not benefit the largest concentration of users in the cell, meaning a much smaller opportunity to drive additional revenue.

Operators with HSPA networks then face the choice of deploying HSPA+ as an interim step to LTE or going straight to LTE to simultaneously address the capacity, lowest cost per bit requirement, and improved subscriber experience. These operators will have to weigh the impact of migrating to HSPA+ on a site-by-site basis. In some cases this is the right natural step for those operators.

In effect, 64-quadrature amplitude modulation (QAM) HSPA+ (as per 3GPP release 7) is possible on some of the latest generation eNodeBs, providing a low-cost upgrade. But going for HSPA+ 2 × 2 multiple-input multiple-output (MIMO) will in most instances require new hardware and ancillaries, making the case for a direct upgrade to LTE even more attractive.

Another consideration is whether, despite an investment in HSPA+, they may still be at risk of running low on capacity in the next few years (based on the predicted data growth explosion and the limited improvement in capacity HSPA+ provides) or/and running the risk of a competitor making the move to LTE early and putting them at a disadvantage.

MARKET ADOPTION OF LTE

More than 20 operators worldwide have already stated a commitment to LTE. Together, they represent more than 1.8 billion of the world's 3.5 billion mobile subscribers. ABI Research forecasts more than 32 million LTE subscribers by 2013, despite the fact that LTE networks will not be commercial before 2010. ABI Research comes to this conclusion because several of the world's largest mobile operators — NTT Docomo, China Mobile, Vodafone, Verizon Wireless, T-Mobile, AT&T, and many others — have announced plans to deploy LTE.

Moreover, the Next Generation Mobile Networks (NGMN) Alliance, a global group that now represents close to 70 percent of mobile operators, has approved LTE as the first technology that meets the requirements set by NGMN. And in the fourth quarter of 2008, Qualcomm announced that it is focusing on LTE rather than pursuing ultra mobile broadband.

In today's challenging economic times, operators have many things to consider before making new investments. But some things are certain. The demand for mobile data is not abating. Users from Millennials to road warriors are consuming more data while on the go, and looking for more ways to have a personalized media experience when, where, and how they want it.

Voice Call Handover Mechanisms in Next-Generation 3GPP Systems

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Mike Hammer, Cisco
Itsuma Tanaka, NTT DOCOMO
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ABSTRACT

The evolved 3GPP system is a hybrid mobile network architecture supporting several radio access technologies and several mobility mechanisms. In this article we briefly review the architecture and key components of this system, with particular emphasis on how it can support voice call mobility in several deployment scenarios. First, we present the so-called single-radio voice call continuity mechanisms that enable mid-call handover of VoIP calls from E-UTRAN access to the legacy UTRAN/GERAN or 1xRTT access. Then we focus on deployment scenarios that do not support voice services on E-UTRAN and present the so-called fallback mechanisms that enable handover from E-UTRAN to UTRAN/GERAN or 1xRTT at the beginning of a voice call. Finally, we address the application-layer voice call handover mechanisms enabled by the IP multimedia subsystem. Our conclusion is that the next generation of 3GPP systems are highly sophisticated mobile communication systems that support extended voice call mobility mechanisms, capable of addressing all commercial deployment needs.

INTRODUCTION

As the wireless industry makes its way to the next generation of mobile communication systems, it is important to engineer solutions that enable seamless integration of the emerging fourth-generation (4G) technologies within the currently deployed 2G/3G infrastructures. This is important because, in most cases, the second- and third-generation (2G/3G) systems provide the solid ground on which the next-generation systems will be built, and will continue providing the main revenue stream for operators for several years along the migration path to 4G. The integration of emerging access and core technologies within the existing 2G/3G networks (e.g., code-division multiple access [CDMA], Global System for Mobile Communications [GSM], General Packet Radio Service [GPRS], and Universal Mobile Telecommunications System [UMTS]) can enable a smooth evolution path, and progressively scale network capacity and service innovation in a economically efficient way. The integrated system is characterized by a heterogeneous architecture (i.e., supporting diverse radio access networks) capable of providing mobile broadband services in strategic geographic areas and ensuring the "best connection" of users at any place, anytime. However, such integration requires us to address a vast range of interoperability and migration issues that arise from the need to support seamless mobility across new and legacy radio access technologies, and migrate the legacy services to new radio accesses. As an example, consider a user that initiates a voice call inside a 4G hotspot. This call is carried out over the 4G access network with voice over IP (VoIP) technologies; but as the user goes out of 4G coverage, the call needs to be sustained and seamlessly handed over to the "umbrella" 2G/3G access network, which typically provides a much wider radio footprint.

It may also happen that voice services are not initially supported over 4G access (e.g., in order to eliminate the cost of deploying VoIP-based services), in which case the user would have to be handed over to the overlay 2G/3G access right after the call request is made. In this case the 2G/3G access is used as a *fallback* access that supports the legacy services when they are not yet available in the 4G access. All these requirements create the need for voice call handover mechanisms from the emerging 4G access technologies to the legacy 2G/3G access technologies.

In this article we focus on such voice call handover mechanisms and address in particular the voice call handover mechanisms in the next-generation Third Generation Partnership Program (3GPP) networks, which typically take place between the evolved 3GPP radio access network (evolved UMTS terrestrial radio access network [E-UTRAN] [1]) and the legacy 3GPP/3GPP2 radio access networks, UTRAN/GERAN and CDMA2000 1xRTT. In this context we present and explain the technical details of three different voice call handover mechanisms, which aim at addressing three different deployment scenarios of next-generation

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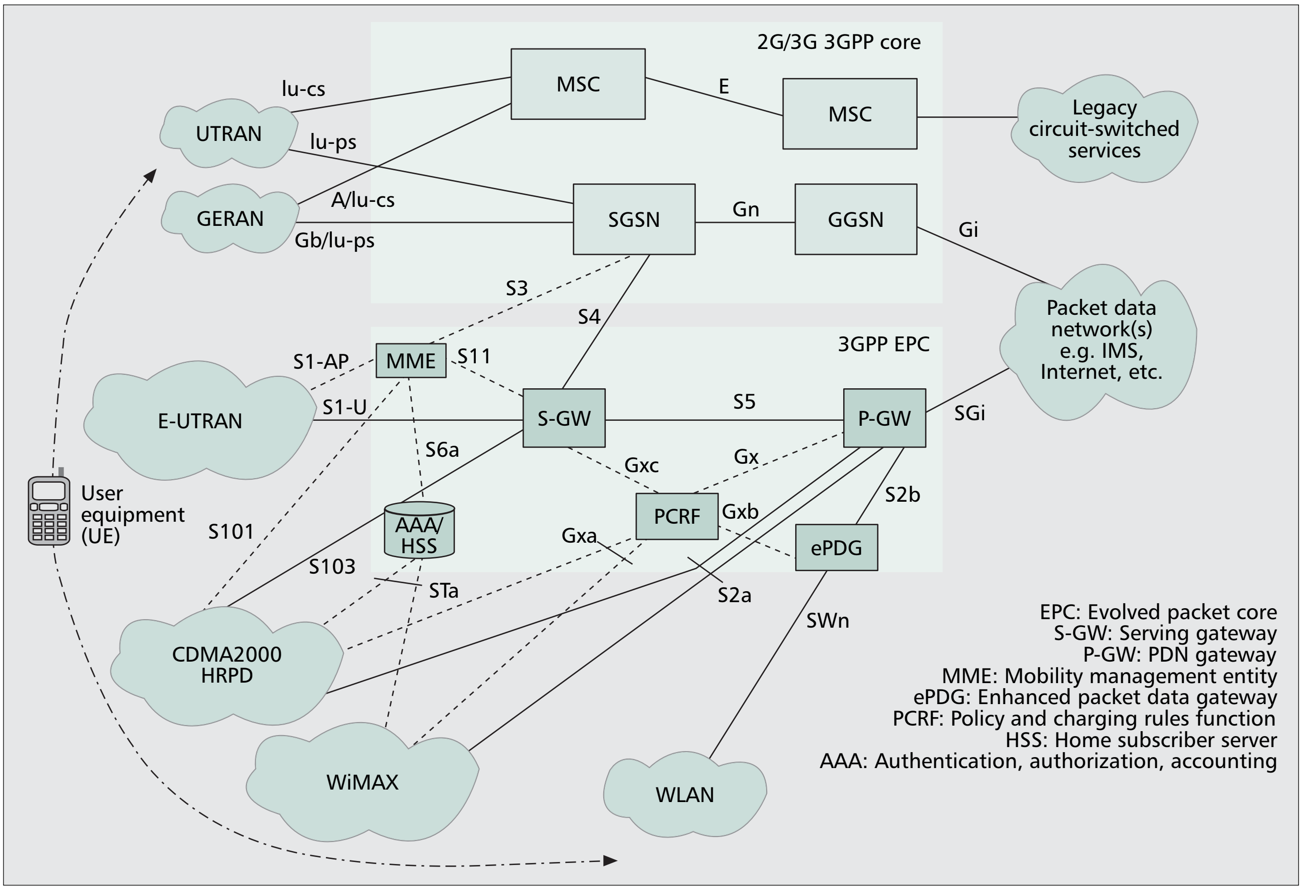


■ **Figure 1.** *Simplified architecture of the evolved 3GPP network.*

3GPP systems. First we present handover mechanisms that enable mid-call handover of VoIP calls from E-UTRAN access to the legacy UTRAN/GERAN or 1xRTT access (note that E-UTRAN access is sometimes also referred to as 4G access). These mechanisms are particularly important in deployment scenarios where voice services are supported on E-UTRAN (as explained later, E-UTRAN supports only IP-based services) but, due to limited E-UTRAN coverage, handover to UTRAN/GERAN or 1xRTT is necessary to maintain seamless voice services over wider geographical areas. Second, we focus on deployment scenarios that do not support voice services on E-UTRAN. This mostly targets initial deployments of evolved 3GPP systems, in which voice services are not yet migrated to E-UTRAN (i.e., not supported yet over IP). For those scenarios, we present the key aspects of a voice call handover mechanism that enables handover from E-UTRAN to UTRAN/GERAN or 1xRTT at the *beginning* of a voice call. The handover is triggered by the arrival of an originating or terminating voice call, which can only be served on UTRAN/GERAN or 1xRTT access and in the traditional circuit-switched (CS) domain [2]. We then address voice call handover mechanisms enabled by the IP multimedia subsystem (IMS). Such mechanisms mostly target deployment scenarios in which voice services are supported on E-UTRAN and non-3GPP-defined radio accesses, such as WLAN and WiMAX, and there is also a need to support voice continuity across these radio accesses. It is also assumed that in such deployments there are no transport layer mechanisms (e.g., based on mobile IP and its derivatives) that can meet the voice continuity requirements. We also provide some background material in the next section, where we present the key aspects of the next-generation 3GPP systems, and introduce the fundamental network elements and interfaces. Finally, we wrap up our discussion by providing a number of concluding remarks.

An Overview of Next-Generation 3GPP Systems

To provide some background material for our further discussion, we present here a short overview of the evolved (or next-generation) 3GPP systems. Readers interested in more details on the evolved 3GPP systems are referred to the 3GPP specifications [1–5] and other articles in this Feature Topic.

As part of Release 8 of the 3GPP specifications, 3GPP has been studying and specifying an evolved packet system (EPS) under the System Architecture Evolution (SAE) work item [2]. The EPS is composed of a new radio access network, called E-UTRAN [1], and a new all-IP core network, called evolved packet core (EPC) [3, 4]. The EPC can be considered an evolution of the legacy GPRS architecture with additional features to improve performance, supporting broadband E-UTRAN access, PMIPv6 mobility, *In order to provide seamless continuity of voice services in wide geographical areas, it is important for the next generation of 3GPP systems to seamlessly handover voice calls between E-UTRAN and UTRAN/GERAN coverage areas.*

and integration with non-3GPP radio technologies such as wireless LAN (WLAN), CDMA2000, and WiMAX. As shown in Fig. 1, the evolved 3GPP network is virtually composed of an evolved version of the legacy 2G/3G network (with the well-known UTRAN/GERAN radio accesses) and the EPC, which supports E-UTRAN access and integration with a range of non-3GPP accesses. Note that for simplicity all network interfaces are not shown in Fig. 1. The interfaces relevant to the voice call handover mechanisms are presented and discussed in subsequent sections.

As shown in Fig. 1, a number of diverse access networks, such as CDMA2000, WLAN, WiMAX, GERAN, UTRAN, and E-UTRAN, are connected to a common core network (the EPC) based on IP technology through different interfaces. All 3GPP-specific access technologies are connected through the serving gateway (S-GW), while all non-3GPP specific access technologies are typically connected through the packet data network gateway (P-GW) or evolved packet data gateway (ePDG), which provides extra security functionality for untrusted access technologies (e.g., legacy WLANs with no strong built-in security features). The S-GW acts as a mobility anchor for mobility within 3GPP-specific access technologies, and also relays traffic between the legacy serving GPRS support node (SGSN) accesses and the P-GW. For E-UTRAN, the S-GW is directly connected to it through the S1 interface, while the SGSN is the intermediate node when GERAN/UTRAN is used. It is important to mention that a mobility management entity (MME) is also incorporated in the architecture for handling control functions such as authentication, security, and mobility in idle mode.

For access to EPC through WLAN, CDMA2000 HRPD, or WiMAX [3], different data paths are used. HRPD and WiMAX are considered trusted non-3GPP accesses and are directly connected to a P-GW through the S2a interface, which is used particularly for *trusted* non-3GPP accesses. On the other hand, a WLAN considered as untrusted access (e.g., because it may not deploy any strong security measures) connects to the ePDG and then to a P-GW through the S2b interface. In this case the ePDG serves as a virtual private network (VPN) gateway and provides extra security mechanisms for EPC access. All data paths from the access networks are combined at the P-GW, which incorporates functionality such as packet filtering, QoS policing, interception, charging, and IP address allocation, and routes traffic over SGi to an external packet data network (e.g., for Internet access) or the operator's internal IP network for accessing packet services provided by the operator. Apart from the network entities handling data traffic, EPC also contains network control entities for keeping user subscription information (home subscriber server [HSS]), determining the identity and privileges of a user and tracking his/her activities (access, authorization, and accounting [AAA] server), and enforcing charging and QoS policies through a policy and charging rules function (PCRF).

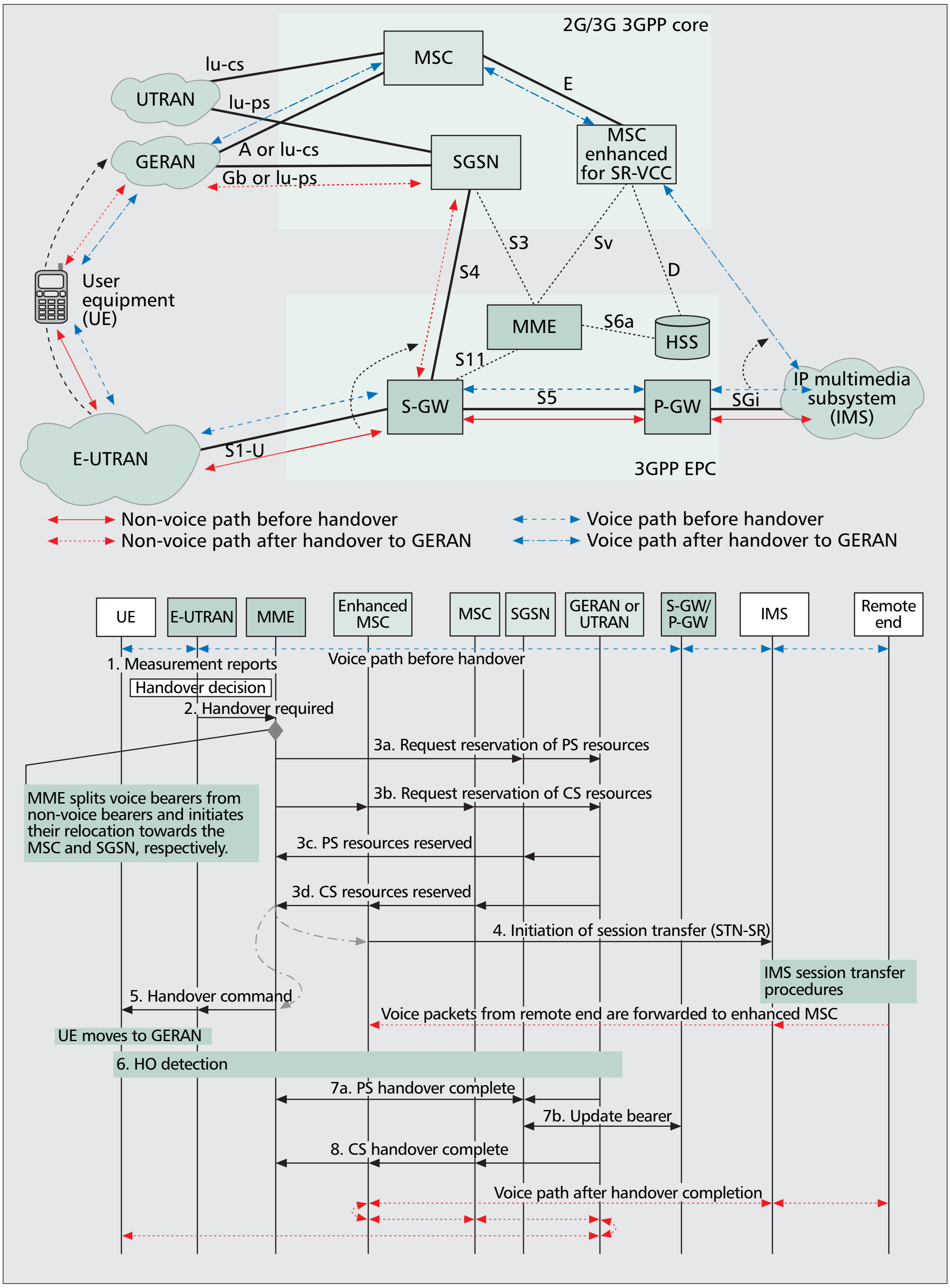
Single-Radio Voice Call Continuity

Especially in the first days of E-UTRAN deployment, coverage will be limited and available only in scattered strategic locations, where wireless broadband services are most needed. Therefore, in order to provide seamless continuity of voice services in wide geographical areas, it is important for the next generation of 3GPP systems to seamlessly hand over voice calls between E-UTRAN [1] and UTRAN/GERAN coverage areas. Although this is not a new requirement (e.g., similar voice continuity requirements exist between UTRAN and GERAN, and between CDMA2000 1xRTT and HRPD), in this case there are new challenges we have to face. First, voice calls on E-UTRAN can only be carried out via IP-based technologies and therefore are provided as IMS-based services. On the contrary, voice calls on UTRAN and GERAN are typically carried out with conventional CS technologies and therefore are provided as CS domain services. This creates the need to transfer and continue voice calls between two different service domains, CS and IMS. Second, the voice calls need to be transferred across service domains and radio access technologies with so-called *single-radio* mobile terminals, that is, mobile terminals that cannot support simultaneous signaling on both E-UTRAN and UTRAN/GERAN radio accesses. This is a key requirement because E-UTRAN and UTRAN/GERAN radio channels may be deployed on the same or neighboring frequency bands; having mobile terminals simultaneously active on both radio channels presents severe technical challenges for the design of the physical layer. This requirement for single-radio mobile terminals makes the use of existing *dual-radio* voice call transfer mechanisms (e.g., mechanisms specified in 3GPP TS 23.206 [6]) infeasible for voice call continuity between E-UTRAN and UTRAN/GERAN access.

Below we present and explain the single-radio solutions recently standardized by 3GPP for enabling voice call continuity between E-UTRAN and UTRAN/GERAN, and between E-UTRAN and CDMA2000 1xRTT (the stage 2 specification can be found in [7]). We primarily focus on voice call handover in the direction from E-UTRAN to UTRAN/GERAN and from E-UTRAN to CDMA2000 1xRTT, since this is considered much more important (due to the limited E-UTRAN coverage) than the other direction of handover. For this reason, only this direction of handover is also supported by the current standards [7].

Voice Call Transfer from E-UTRAN to UTRAN/GERAN

Figure 2a shows the architecture that enables single-radio voice call continuity (SR-VCC) between the E-UTRAN and conventional UTRAN and GERAN radio networks [7]. This architecture is based on the IMS and requires all voice calls initiated on E-UTRAN to be anchored in IMS (a later section explains how this is done). The architecture also requires at



■ **Figure 2.** *a) Architecture for single-radio VCC between E-UTRAN and UTRAN/GERAN; b) message flow diagram for voice and non-voice single-radio handover from E-UTRAN to UTRAN or GERAN (with dual transfer mode capability).*

least one mobile switching center (MSC) server in the traditional CS domain to be enhanced with interworking functionality and a new interface, called *Sv*. Such an enhanced MSC server is referred to as "MSC server enhanced for SR-VCC"; when combined with a standard media gateway (MGW), it is referred to as "MSC enhanced for SR-VCC." In addition, the MME in the EPC (discussed previously) requires additional functionality to support the *Sv* interface and the associated SR-VCC procedures, which are further explained below.

Although the solution for SR-VCC was originally required to enable the handover of *one* ongoing voice call from E-UTRAN to UTRAN or GERAN, during the standardization process it was enhanced to further facilitate the handover of potential non-voice sessions that might be active in parallel with the voice call. To better explain this, we show in Fig. 2a user equipment

Although the solution for SR-VCC was originally required to enable the handover of one ongoing voice call from E-UTRAN to UTRAN or GERAN, during the standardization process it was enhanced to further facilitate the handover of potential non-voice sessions that might be active in parallel with the voice call.

During the execution of the IMS session transfer procedure, the Remote End is updated with new Session Description Protocol (SDP) contact details (e.g., with a SIP re-INVITE message) and subsequent downstream voice packets are forwarded to the MSC enhanced for SR-VCC.

(UE) that has a voice session and a non-voice session (say, a video streaming session) active in E-UTRAN. We assume that both sessions are enabled by IMS, so the data flows of these sessions (indicated with the double-head arrows) enter the IMS cloud. When the UE goes out of E-UTRAN coverage, the SR-VCC procedure is triggered to relocate all data flows to (say) a GERAN radio cell. In the example shown in Fig. 2a, this cell is not controlled by the MSC enhanced for SR-VCC, so another MSC is involved (however, this is not always the case). The voice session is relocated from the P-GW to the MSC enhanced for SR-VCC and the non-voice session is relocated from the S1-U interface to the S4 interface and then to GERAN through the SGSN. In this example the GERAN radio access supports simultaneous access to voice and non-voice (GPRS) services. To meet the seamless voice transfer requirements (i.e., make the voice transfer imperceptible to the user), this handover process should create voice interruption of no more than a few hundreds of milliseconds. For the non-voice session(s), however, seamless transfer cannot be guaranteed given the limited data bandwidth offered by UTRAN/GERAN compared to the broadband data capabilities of E-UTRAN. Therefore, although non-voice session(s) can be transferred and continue in the UTRAN/GERAN packet-switched domain, there is no guarantee that the transfer would be seamless and the quality of service sustained after the transfer.

The SR-VCC solution is further explained below with the aid of the message flow diagram illustrated in Fig. 2b. In this example diagram the UE is initially in E-UTRAN and has active voice and non-voice sessions in parallel. The voice session is anchored in IMS and terminates at the remote end, which is a traditional public switched telephony network (PSTN) phone; thus, IMS/PSTN interworking functionality is invoked. At the top of the diagram, the path of the voice flow is illustrated, which goes through the appropriate (IMS/PSTN) interworking functions in the IMS subsystem and is then routed to the UE via P-GW, S-GW, and E-UTRAN (note that for convenience the S-GW and P-GW are shown in the same box). A similar flow path could be traversed by the non-voice session, although this is not shown in Fig. 2b for simplicity. At the end of the flow diagram the voice and non-voice sessions are transferred, and continue in a GERAN or UTRAN cell.

During its active voice session the UE takes inter-radio access technology (inter-RAT) measurements; that is, it measures the signal strength and quality of the neighbor UTRAN and/or GERAN channels. The list of available inter-RAT channels is regularly transmitted in the system information of the serving E-UTRAN cell. The UE transmits inter-RAT measurement reports as well as measurement reports for the currently selected E-UTRAN cell (step 1 in Fig. 2b), which are typically input to the handover decision algorithms. Apart from the measurement reports, the serving E-UTRAN cell knows:

- If the UE is SR-VCC-capable (this information comes from the MME during attachment or handover)
- If the neighbor UTRAN/GERAN cells can support VoIP
- If the UE has an active voice call

Based on this information, the serving E-UTRAN cell decides when an SR-VCC handover is required and selects the target UTRAN or GERAN cell (which in this case can support only voice services in the traditional CS domain). When this decision is taken, the E-UTRAN sends a *Handover Required* message (step 2) to the MME with some information indicating that SR-VCC handover is required.

This information is necessary in the MME in order to decide what type of handover to execute; in this case it will be a handover toward the CS domain and a handover toward the packet-switched (PS) domain. In turn, the MME separates the active voice bearer of the UE from its active non-voice bearer(s) and initiates two handover procedures in parallel: one to hand over the non-voice bearer(s) to the PS domain and another to hand over the voice bearer to the CS domain of the 2G/3G core. These procedures are typically executed in parallel, but for the sake of presentation, Fig. 2b shows them in sequence: first the PS handover and then the CS handover. Step 3a is used for preparing the non-voice resources in the PS domain according to the standard relocation procedures specified in 3GPP TS 23.060 [5]. Also, step 3b is used for preparing the voice resources in the CS domain according to the relocation procedures in 3GPP TS 23.060 [5] and the inter-MSC handover procedures in 3GPP TS 23.008. Note that in the example message flow of Fig. 2b, E-UTRAN decided to transfer the voice and non-voice sessions of the UE to a neighbor GERAN cell, which supports both PS and CS services in parallel (also called dual transfer mode [DTM]). This GERAN cell is also connected to a legacy MSC (i.e., not enhanced for SR-VCC), which does not interface directly with the MME, so the voice handover signaling must pass through an MSC enhanced for SR-VCC and an inter-MSC handover must be carried out between the MSC enhanced for SR-VCC and the legacy MSC.

Message 4 is a key step in the handover procedure and merits further explanation. This message is merely a new voice call request (e.g., an ISUP IAM) toward a special number, called session transfer number for SR-VCC (STN-SR). This call request is routed to a special application server in the IMS subsystem, called service centralization and continuity application server (SCC AS), which then correlates the received STN-SR with an active IMS voice session, in this case with the voice session between UE and the remote end (see 3GPP TS 23.237 [8] and a section in this article for further details). The reception of a new call request for STN-SR is a signal for SCC AS that the associated IMS voice session needs to be redirected to a new endpoint (in this case to the MSC enhanced for SR-VCC); therefore, it triggers the IMS session transfer procedures specified in 3GPP TS 23.237 [8]. Note that for each UE, a dedicated STN-SR number is configured in the HSS, and the MSC enhanced for SR-VCC receives this number from the MME (in step 3b), which receives this number from the HSS during the UE's initial attachment to EPC. During the execution of the IMS session transfer procedure, the remote end is updated with new Session Description Protocol (SDP) contact details (e.g., with a Session Initiation Protocol [SIP] re-INVITE message), and subsequent downstream voice packets are forwarded to the MSC enhanced for SR-VCC (more correctly, to the MGW associated with this MSC).

After both PS and CS resource preparation is completed (steps 3c and 3b, respectively), the MME responds to the Handover Required message received in step 2 with a *Handover Command*, which is then transmitted to UE in step 5. This command includes information describing the CS and PS resources allocated for the UE (e.g., the target GERAN cell identity, carrier frequency, traffic channel specification). After this step, the UE moves to the indicated target GERAN cell and uses conventional GERAN handover completion procedures. In step 6 the UE is detected in the target GERAN cell, and in steps 7 and 8 the PS and CS relocation/handover procedures are completed, respectively. After step 7b the non-voice bearers are switched from the S1-U interface to the S4 interface, and after step 8 the voice bearer is fully re-established with a new access leg in the GERAN CS domain. Note that the voice gap created by the handover procedure starts from the execution of the IMS session transfer procedures to the completion of step 8 and is expected to last a few hundreds of milliseconds.

Voice Call Transfer from E-UTRAN to CDMA2000 1xRTT

Figure 3a shows the architecture that enables SR-VCC between the E-UTRAN and CDMA 1xRTT radio networks. This architecture requires a 1xCS interworking solution (IWS) function in the traditional 3GPP2 CS domain to interwork with EPS via a new interface, called S102. The 1xCS IWS enables UE to communicate with the 1xRTT MSC while it is connected to the EPC via E-UTRAN. From SR-VCC perspective, this mechanism minimizes the voice gap by allowing the UE to establish the target CS access leg while it is still on the E-UTRAN access, prior to the actual handover to 1xRTT access. The MME in the EPC (discussed previously) requires additional functionality to support the *S*102 interface and mainly functions as a signaling relay station between the UE and 1xCS IWS.

This architecture was designed to handle only the voice part of an IMS session [7]. The transfer of a non-voice component is not specified (different from SRVCC to UTRAN/GERAN).

This SR-VCC solution is further explained below with the aid of the simplified message flow diagram illustrated in Fig. 3b. In this example diagram the UE is initially in E-UTRAN and has an active IMS voice session. The voice session terminates at the remote end, which may be a traditional PSTN phone.

Similar to the SR-VCC for 3GPP system, the VoIP/IMS path is initially established over E-UTRAN. When E-UTRAN detects the need for SR-VCC to 1xRTT handover, it sends a signaling message, including the necessary 1x parameters (e.g., 3G1x overhead parameters, RAND value) to the UE to start the 1xRTT SR-VCC procedure (steps 1–2, Fig. 3b). In step 3 the UE starts the 1x SR-VCC procedure with the 1xRTT MSC (based on 3GPP2 X.S0042-0 [9]). This is done by establishing a signaling tunnel via EPC and 1xCS IWS. When the 1xRTT MSC has received a positive acknowledgment from the 1xRTT radio for traffic allocation and from the IMS for successful domain transfer, it returns an IS-41 handoff message to the UE (similar to the handover command) via the established signaling tunnel (step 5). Upon receiving this message, the UE then performs the handover to 1xRTT access and continues to transmit voice via that system. The voice bearer path is no longer carried by the EPC. E-UTRAN requests the MME to release the UE context (step 6). The MME completes this procedure by requesting the S-GW to suspend the EPS bearer, step 7. (I.e., S1-U bearers are released for all EPS bearers and the voice bearer is deactivated. The non-GBR bearers are preserved and marked as suspended in the S-GW). Upon receipt of downlink data the S-GW should not send a downlink data notification message to the MME.

In some initial deployments of evolved 3GPP systems there will be no support of voice services on E-UTRAN. In such deployments however it is still necessary that users on E-UTRAN be able to participate in voice calls. This can be achieved by conducting a so-called fallback to a legacy 2G/3G access network.

Voice Call Handling with Fallback to 2G/3G

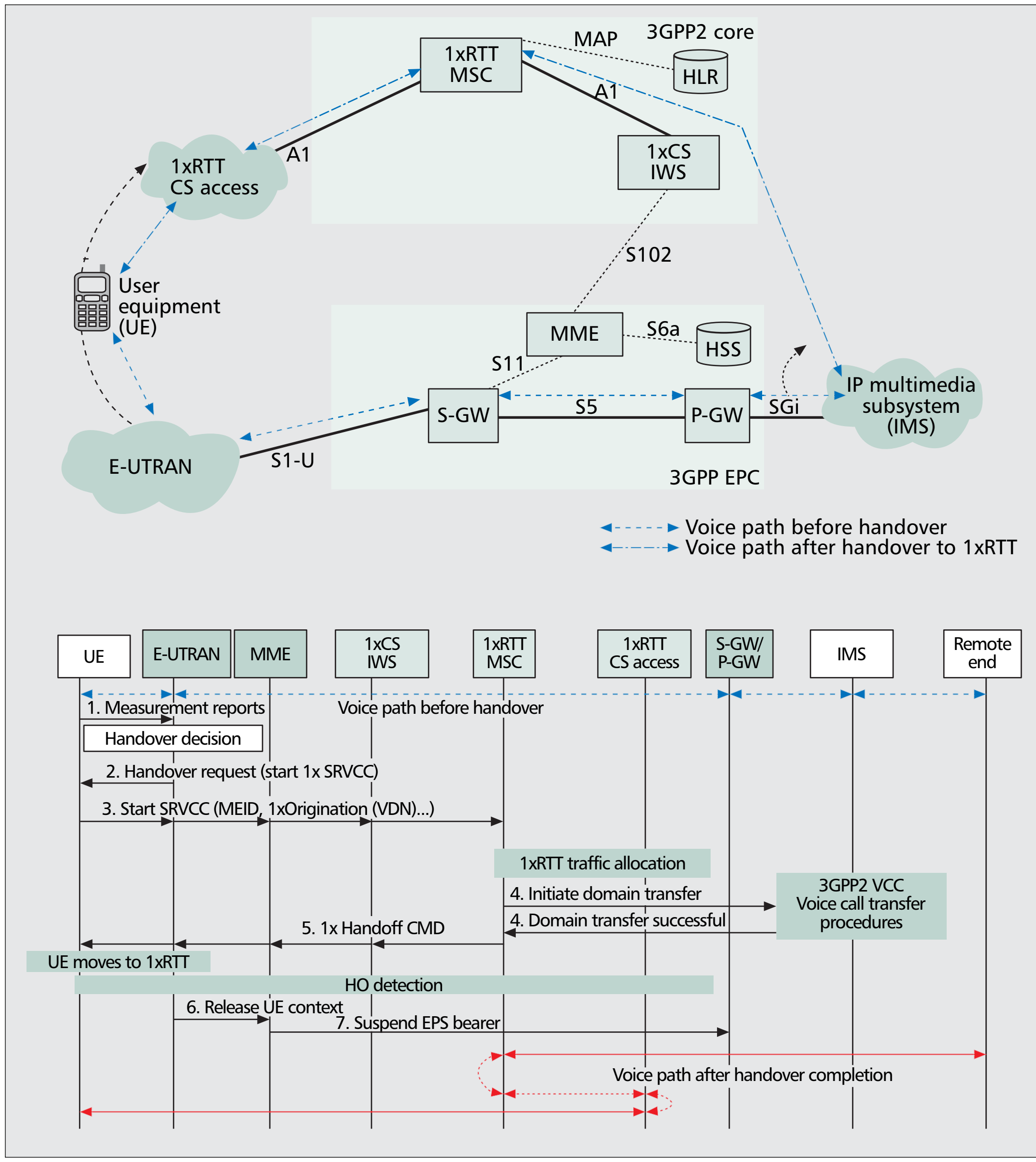
As discussed before, in some initial deployments of evolved 3GPP systems there will be no support of voice services on E-UTRAN. In such deployments, however, it is still necessary that users on E-UTRAN be able to participate in voice calls. This can be achieved by conducting a so-called fallback to a legacy 2G/3G access network that supports traditional voice services (e.g., UTRAN, GERAN, or CDMA2000 1xRTT) at the moment a voice call is requested. This fallback is effectively a *service-based* handover mechanism, a handover triggered by a service request, which in this case is a request for an originating or terminating voice call. After the user falls back to 2G/3G access, standard CS voice call setup procedures are used to establish the voice call in the CS domain. This technique of falling back to the 2G/3G CS domain for voice services is considered an interim solution that can fill the gap between the initial E-UTRAN deployment and the introduction of voice and other IP services over E-UTRAN.

The fallback to the 2G/3G CS domain (or CS fallback for short) is enabled by the following characteristics:

- Mobility management is combined with EPS mobility management.
- Paging request for terminating calls are delivered to the UE via EPS.
- 2G/3G is used for paging responses and further call handling as well as for all originating calls.

Figure 4a shows the architecture that enables CS fallback to the UTRAN/GERAN CS domain. Note that although this figure concentrates on fallback to UTRAN/GERAN, a similar architecture can also be used for fallback to CDMA2000 1xRTT access. Details for both architectures as

The architecture connects the Mobile Switching Center and the Mobility Management Entity via the new SGs interface. This interface is based on the Gs interface which connects MSC and Serving GPRS Support Node and was introduced to reuse the concept of combined mobility management between the CS and PS domains.



■ **Figure 3.** *a) Architecture for single-radio VCC between E-UTRAN and 3GPP2 1x RTT CS; b) simplified message flow diagram for voice session handover from E-UTRAN to 3GPP2 1xRTT CS access.*

well as fallback procedures can be found in 3GPP TS 23.272 [10].

The architecture connects the mobile switching center (MSC) and MME via the new SGs interface. This interface is based on the Gs interface, which connects the MSC and SGSN, and was introduced in order to reuse the concept of combined mobility management between the CS and PS domains [5]. To combine the mobility management between E-UTRAN and 2G/3G, there is need for a mapping mechanism that maps between E-UTRAN tracking areas and 2G/3G location areas. When the UE moves to E-UTRAN, its 2G/3G location area must also be updated so that incoming calls can be correctly delivered to the UE. The mapping of tracking/location areas is performed by the MME, and the combined mobility management procedures (e.g., combined attached [5]) are reused as much as possible to minimize the impact on the MSC.

Figure 4b illustrates an example message flow diagram showing how CS fallback is conducted in order to facilitate a mobile originating call. In this example diagram the UE is initially in E-UTRAN and has active IP packet connection. The UE has attached to the MME in EPC and to an MSC in the 2G/3G CS domain by conducting a combined EPS/IMS Attach procedure [5]. It is assumed also that the target 2G/3G access supports simultaneous PS and CS services. The following procedure takes place when UE and the network are not configured to use IMS. If UE is configured to use IMS voice or IMS SMS services and is registered to IMS, it shall initiate calls via IMS, even if it is attached to the 2G/3G CS domain.

When UE decides to make a mobile originating call, the UE sends a Service Request message with a CS fallback indication to the MME (step 1) so that the MME can initiate the required handover procedures. If CS fallback is allowed for the UE, the MME sends a message

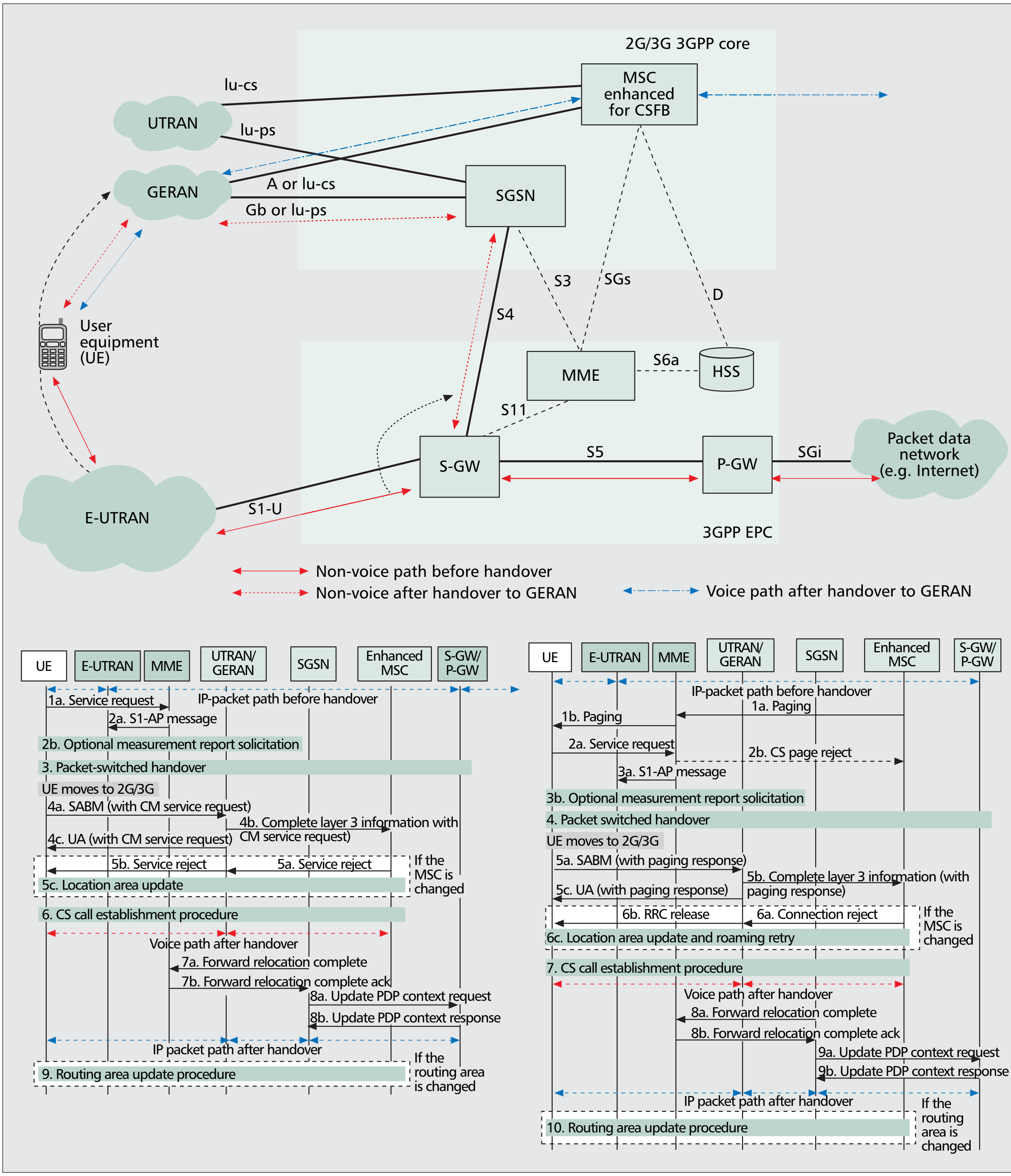


Figure 4. *a) Simplified architecture of 3GPP CS Fallback; b) mobile originating call for CS fallback, UE in active mode; c) mobile terminated call for CS fallback, UE in active mode.*

(step 2) that instructs the eNB to transfer the UE to a neighboring UTRAN/GERAN cell. Before the eNB does so, it may request some radio measurement reports from the UE in order to discover the best target UTRAN/GERAN cell (step 2b). In turn, a standard PS handover procedure is initiated (step 3) in which PS resources are first reserved in the target UTRAN/GERAN cell and then a handover command is sent to the UE to instruct it to move to the target UTRAN/GERAN cell. When fallback to UTRAN/GERAN is successful, the UE sends a standard CM Service Request message to the MSC to request call establishment

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IMS was designed to enable intelligent, IP-based services, and features to be located in the evolved 3GPP network. One key aspect focused on here is the ability to maintain services even when the user is moving across different access networks and terminal types.

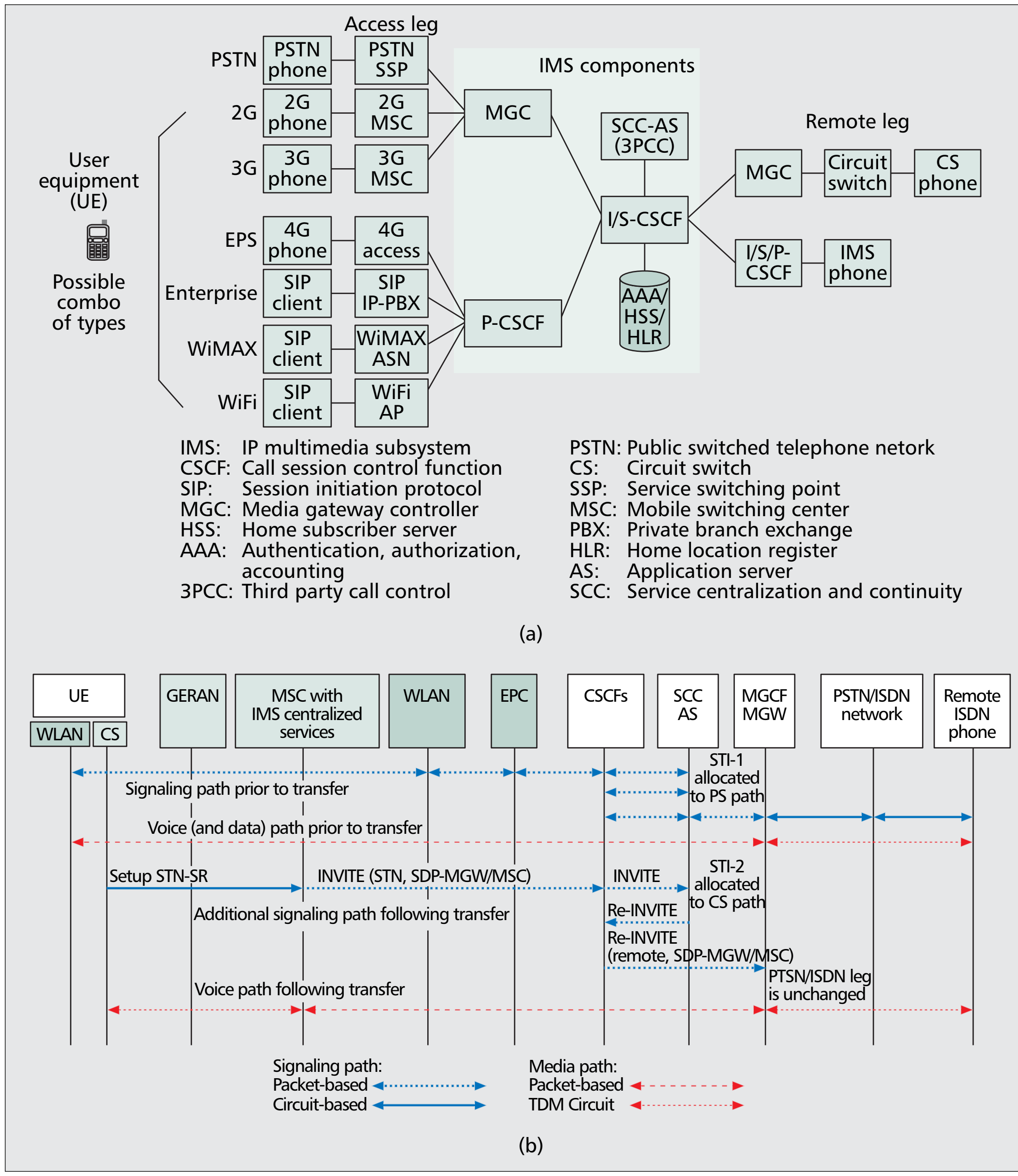

Figure 5. *a) IMS service continuity architecture based around the SCC-AS; and b) simplified message flow diagram for voice call transfer with IMS service continuity mechanisms.*

(steps 4a–4c). In some cases (see [5] for details), the CM Service Request may be sent to an MSC other than the MSC with which the UE was registered in E-UTRAN. In this case, the new MSC rejects the originating call request and a location update procedure is performed (steps 5a–5c) followed by another CM Service Request and the normal voice call establishment procedures (step 6).

In parallel with voice call establishment, the PS handover (initiated in step 3) between SGSN and S-GW/PGW is completed (steps 7a–7b), and the active PS bearers are switched to UTRAN/GERAN access after the update PSP context procedure is finished (Steps 8a, 8b). Finally, a typical routing area update can be performed when the UE detects that it is now in a different routing area (step 9).

Figure 4c shows how CS fallback works for mobile terminating calls. In this example diagram, the UE is initially in E-UTRAN and has an active data connection, and the target access supports simultaneous support of PS and CS services.

When there is an incoming call request, the request is first routed to the MSC/VLR in which the UE is registered through combined CS fallback attach or combined tracking area update procedures. When the MSC/VLR receives the incoming call request, it sends a paging message to the MME over the SGs interface (step 1a). The MME then forwards this paging message to the UE over E-UTRAN access (step 1b). This paging message is an NAS message that can securely carry a caller's ID. By informing the user of the caller's ID prior to fallback, the user can decide whether to accept CS fallback to legacy access for voice calls. This caller ID notification over a paging message is introduced especially for CS fallback to avoid unwanted fallback, especially when a user is engaged in fast data services over E-UTRAN.

Similar to mobile originated calls, PS handover and CS fallback are triggered by sending a Service Request message to the MME (Step 2a). If the user rejects incoming call request before fallback, the MME returns a reject message to the MSC (Step 2b). When UE moves to 2G/3G, UE sends a paging response to the MSC (steps 5a–5c). The paging response can be rejected by the MSC if the UE is sending it to the wrong MSC, and location area update is triggered. Location area update also triggers the originating MSC to redirect the call to the correct MSC by means of an existing roaming retry procedure (steps 6a–6c). When a paging response is successfully returned to the correct MSC, the CS call establishment procedure takes place (step 7). PS handover is also completed in a similar way, as explained in the previous section.

IMS-Based Voice Call Continuity

As mentioned before, voice call continuity mechanisms in the evolved 3GPP system are also provided by the IMS [11]. The IMS is a service control subsystem in the evolved 3GPP architecture (Fig. 1) and is based on SIP defined by the Internet Engineering Task Force (IETF) in RFC 3261 and many related RFCs. IMS was designed to enable intelligent IP-based services and features to be located in the evolved 3GPP network. One key aspect focused on here is the ability to maintain services even when the user is moving across different access networks and terminal types.

The IMS specifications (e.g., 3GPP TS 23.228 [11]) define the functionality of IMS components, including various types of call session control functions (CSCFs) and application servers, which take advantage of SIP procedures to register users with IMS, and originate, terminate, transfer, and release multimedia sessions. Because voice is still the key service for service providers, the first service continuity solution with IMS was designed to enable voice call continuity (VCC) between a conventional access network supporting CS voice (e.g., GSM, UMTS, CDMA2000 1xRTT) and a VoIP access network (e.g., a WLAN). This was defined in 3GPP TS 23.206 [6] as part of 3GPP specifications Release 7. The second solution, service centralization and continuity (SCC), defined in 3GPP TS 23.237 [8] and 3GPP TS 23.292 [12], expands the VCC solution, and provides enhanced call transfer functionality and service centralization in the IMS. There is a common architectural theme, as shown in Fig. 5a.

Figure 5b shows an example of a mobile user's voice call being transferred from WLAN access to GERAN CS domain access using IMS service continuity procedures. One of the benefits of using the IMS to transfer voice calls is that the same service continuity procedures can be used no matter what the source and target accesses are. Hence, although Fig. 5b shows WLAN and GERAN as the source and target access technologies, respectively, it is equally applicable to any other source and target access technologies. In some cases (e.g., when the underlying transport network cannot support voice transfer between two specific access technologies) the IMS service continuity procedures are the only means to enable such voice transfer.

Figure 5b considers an example where a mobile device (UE) has a multimedia session with a fixed device (remote ISDN phone). At the top of the figure, the path followed by signaling (SIP) messages and the path followed by user traffic (voice and data) are shown. Note that the signaling path needs to go through the SCC application sserver (SCC AS) [8], which is the element that effectively enables session transfer by using third-party call control (3PCC) procedures. In particular, the SCC AS splits the session between the UE and the remote ISDN phone into two parts (or legs): one part between the UE and SCC AS (called the access leg), and another part between the SCC AS and remote ISDN phone (called the remote leg). This split is performed when the multimedia session is initiated. The SCC AS terminates the original session request (say from the UE) and initiates a new session toward the remote ISDN phone. By doing so, the SCC AS appears as a *third party* between the UE and remote ISDN phone that controls the call between them (hence the term *third-party call control*).

As opposed to other handover mechanisms (e.g., the SR-VCC discussed previously), in all IMS service continuity mechanisms it is the UE that makes handover decisions based on preconfigured operator policies, user preferences, and the availability of neighbor access technologies. One operator policy could, for example, indicate that GERAN CS has higher preference than WLAN for voice sessions, so when the UE uses WLAN and discovers an available GERAN access in its vicinity, it will try to transfer its ongoing voice and other sessions to GERAN access.

When the UE makes the handover decision, it sends a standard CS Setup message to a special destination number, called a session transfer number (STN), which causes the Setup messages to be translated to a corresponding SIP INVITE message, and then routed to the SCC AS. Note that the Setup message and the translated INVITE message going to the SCC-AS create a third signaling leg (UE-CS to SCC) in addition to the existing access leg (UE-WLAN to SCC) and remote leg (SCC to remote ISDN phone). After receiving the INVITE message from the MSC enhanced with IMS centralized services [12], the SCC-AS generates the re-INVITE to the MGCF, which updates the remote leg at the MGW to point to the MSC (with IMS centralized services) rather than the UE's WLAN IP address. Therefore, the voice media from the remote leg MGW is redirected to the MSC. The path of the voice media component after the session transfer is shown at the bottom of Fig. 5b. Note that in this example, if there are other media components (in addition to voice) active between the UE and the remove ISDN phone,

The SCC AS terminates the original session request (say from UE) and initiates a new session towards the Remote ISDN phone. By doing so, the SCC AS appears as a third-party between UE and Remote ISDN phone, which controls the call between them (hence, the term third-party call control).

Since E-UTRAN coverage will initially be limited compared to the coverage of the legacy radio technologies, many different handover mechanisms have been standardized by 3GPP in order to enable ubiquitous voice call services in many deployment scenarios.

they will either be released after voice is transferred to GERAN CS domain or continue over the WLAN. Of course, the latter case requires dual-radio capabilities in the UE (i.e., capability to operate simultaneously on the WLAN and GERAN radio access.)

CONCLUSIONS

The next generation of 3GPP systems support a new wireless broadband radio access technology called E-UTRAN, and a vast range of mobility mechanisms that facilitate many deployment scenarios and support a flexible evolutionary path toward 4G mobile systems. Since E-UTRAN coverage will initially be limited compared to the coverage of legacy radio technologies (UTRAN, GERAN, CDMA2000, etc.), many different handover mechanisms have been standardized by 3GPP in order to enable ubiquitous voice call services in many deployment scenarios and support service continuity across a wide range of radio environments. In this article we present and explain these handover mechanisms, and identify the key motivation behind them. It is shown that when voice services are supported over E-UTRAN, the so-called SR-VCC mechanisms [7] can be used to seamlessly hand over voice calls to legacy radio technologies such as UTRAN, GERAN, and 1xRTT. On the other hand, when no voice services are supported over E-UTRAN, CS fallback mechanisms [10] can be used to transfer the user to a legacy radio access that supports conventional voice services over the CS domain. Finally, we explain the application-layer handover mechanisms supported by the IMS, which can facilitate additional voice call continuity scenarios, especially when no vertical (or intersystem) mobility is supported by the network layer [8, 12]. All these handover mechanisms allow the next generation of 3GPP systems to support sophisticated voice call continuity in all commercial deployments.

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BIOGRAPHIES

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Network-Based Mobility Management in the Evolved 3GPP Core Network

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ABSTRACT

A key aspect of the 3GPP system architecture evolution is the specification of an evolved packet core that supports multiple access networks. The EPC enables operators to deploy and operate one common packet core network for 3GPP radio accesses (E-UTRAN, UTRAN, and GERAN), as well as other wireless and wireline access networks (e.g., eHRPD, WLAN, WIMAX, and DSL/Cable), providing the operator with a common set of services and capabilities across the networks. A key requirement of the EPC is to provide seamless mobility at the IP layer as the user moves within and between accesses. This article provides an overview of the EPC specifications that use a network-based mobility mechanism based on Proxy Mobile IPv6 to enable mobility between access networks. An important facet of providing seamless mobility for a user's sessions across technologies is to ensure that quality of service is maintained as the user moves between accesses. An overview of the "off-path" QoS model to supplement PMIPv6 is also provided.

INTRODUCTION

The desire of Third Generation Partnership Project (3GPP) operators to maintain a competitive wireless network for the years 2010 to 2020 has been the key driver in the standardization effort known as system architecture evolution (SAE). The standardization effort has two primary objectives. One objective is to create a new radio access network, called evolved-universal mobile telecommunications system (UMTS) terrestrial radio access network (E-UTRAN), based on orthogonal frequency division multiplexing (OFDM) radio technology that significantly increases data rates for mobile terminals, lowers end-to-end latency for real-time communications, and reduces set-up times for new connections. The other objective is to create a common packet core network, the evolved packet core (EPC), to support mobile services, not only over the 3GPP defined-radio access technologies, but also over other non-3GPP defined-radio access technologies, such as wireless local area network (WLAN), worldwide interoperability for microwave access (WiMAX), and code division multiple access (CDMA)2000.

An important aim of the EPC is to provide seamless service continuity for multi-mode terminals as these terminals move from one radio access technology to another. These requirements are specified in 3GPP TS 22.278 [1]. Two different mobility approaches were specified for the EPC to achieve mobility between 3GPP and non-3GPP access systems, namely the network-based mobility protocol Proxy Mobile IPv6 (PMIPv6) [2] and client-based mobility protocols Dual-Stack Mobile IPv6 (DSMIPv6) [3] and Mobile IPv4 [4]. This article provides an overview of how PMIPv6 is used to achieve seamless handovers across different access systems connected to the EPC. Apart from providing session continuity, the EPC also is required to provide quality of service (QoS) support as user equipment (UE)[1] moves between the different access technologies. The interaction of PMIPv6 with QoS support also is covered in this article.

The remainder of this document is organized as follows. The next section describes the basic idea of network-based mobility management. We then discuss how the EPC uses network-based mobility management and the architectural aspects, interfaces, and challenges (e.g., QoS management). The following section provides the handover flows for both optimized and non-optimized inter-access system handovers. The final section concludes the article and identifies future work.

[1] In the context of 3GPP, the user equipment (UE) describes the device that is used to communicate with the network (e.g., a mobile terminal or a data card in a laptop computer).

[2] No interface from the PCRF to ePDG in untrusted non-3GPP IP access has been defined in the Release 8 version of the EPC.

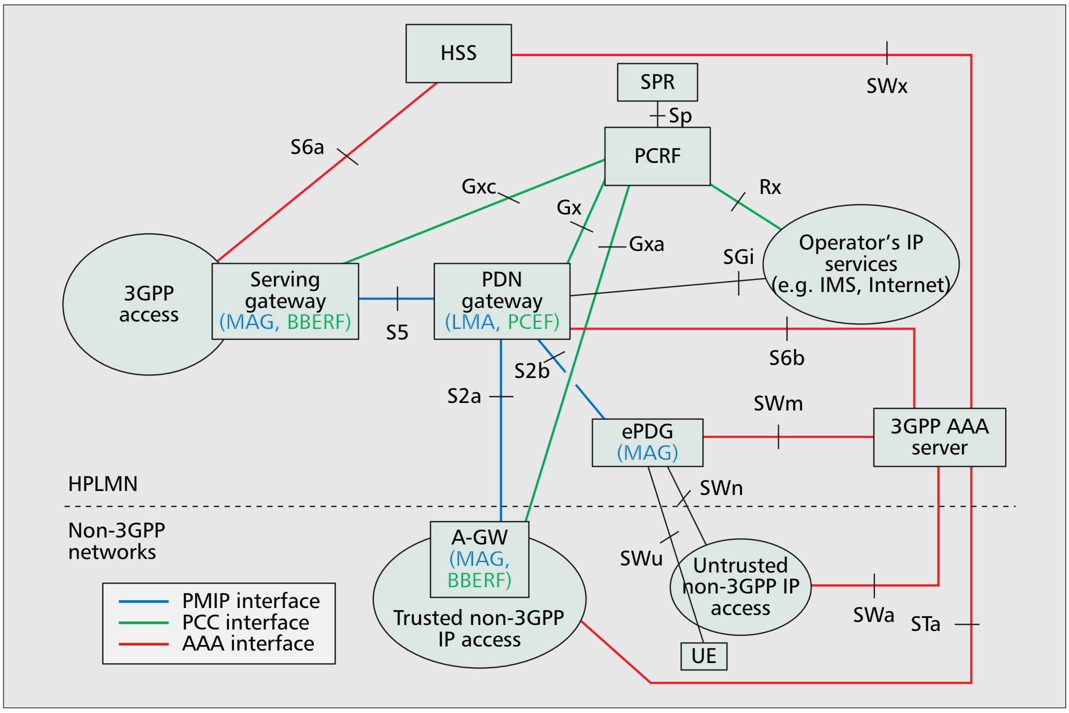

Figure 1. *The PMIPv6-based mobility architecture of the evolved packet system.*

IP-based mobility management enables the UE to preserve IP address(es), even when the UE changes its point of attachment to the network. There are two basic approaches to providing IP-based mobility management: network-based mobility management and client-based mobility management.

Network-Based IP Mobility Management

IP-based mobility management enables the UE to preserve IP address(es), referred to as home address(es) in the rest of the article, even when the UE changes its point of attachment to the network. There are two basic approaches to providing IP-based mobility management: network-based mobility management and client-based mobility management.

In the case of network-based mobility management, the network (e.g., access gateway), on detecting that the UE has changed its point of attachment, provides the UE with the same IP address that it had at its previous point of attachment. The network entity providing the IP address to the UE also handles updating the mobility anchor in the network so that the packets arrive at the new point of attachment of the UE. The UE is not aware of the mobility management signaling within the network. In contrast, for client-based mobility management, the UE obtains a new local-IP address (also referred to as care-of-address) when it moves to a new point of attachment. It is then the responsibility of the UE to update its home agent, which maintains a binding between the care-of-address and the home address of the UE.

3GPP has closely investigated the mobile operator requirements from a service aspect point of view [1]. The requirement to provide handover capability within and between access systems with no perceivable service interruption has been identified. This means that the delay introduced by the mobility management procedure must be minimized. Efficient use of wireless resources is another requirement for mobility management because wireless resources could be a bottleneck. Finally, it is generally desirable to minimize UE involvement in mobility management to improve the battery life of the terminal. Because network-based mobility management fulfills these requirements well, PMIPv6 was adopted as the IP mobility protocol for mobility between 3GPP and non-3GPP accesses and as an option for intra-3GPP access mobility.

PMIPv6 [2] introduces two new functional entities:

- The local mobility anchor (LMA), the equivalent of a home agent, which is the topological anchor point for the home network prefix(es) and manages the binding state of the mobile node.
- The mobile access gateway (MAG), which acts as the proxy (foreign) agent for the terminal and handles the mobility signaling (e.g., a proxy binding update) toward the LMA upon terminal movement.

Network-Based Mobility Architecture of the EPC

The focus of this article is on the analysis of network-based mobility management and in particular on the option based on PMIPv6 [2]. The relevant aspects of the EPC architecture are shown in Fig. 1. The functional entities related to PMIP-based mobility management are indicated in blue, whereas the functional entities related to policy and charging control (PCC) are shown in green. Note that although GPRS Tunneling Protocol (GTP)v2 (an evolution of the existing 3GPP GTP protocol) can be used alternatively for mobility management within 3GPP accesses only, PMIPv6 is used for network-based IP mobility management between 3GPP and non-3GPP accesses.

A UE can access the EPC from both 3GPP accesses and non-3GPP accesses. Non-3GPP accesses are classified into trusted and untrusted accesses. An untrusted access is one that requires the operator to deploy an evolved packet data gateway to provide the appropriate security to enable the UE to securely access the EPC.

Figure 1 shows that a UE can access the EPC from both 3GPP accesses and non-3GPP accesses. Non-3GPP accesses are classified into trusted and untrusted accesses. An untrusted access is one that requires the operator to deploy an evolved packet data gateway (ePDG) to provide the appropriate security, that is, authentication of the UE and data encryption (based on IPsec/IKEv2), to enable the UE to securely access the EPC. For trusted non-3GPP accesses, an ePDG is not required.

The PMIPv6 mobility architecture of the evolved packet system (EPS) includes the following entities:

PDN Gateway (PDN GW): The PDN GW provides access to different packet data networks (PDNs) by assigning an IP address to the UE from the address space of the PDN. This address can be an IPv4 address, an IPv6 prefix, or both. The PDN GW is also the mobility anchor point for the address/prefix of the UE and acts as a PMIPv6 LMA.

Serving Gateway (S-GW): The S-GW includes the PMIPv6 MAG functionality for IP mobility management. The S-GW acts also as a layer-2 mobility anchor, as the UE moves within 3GPP accesses (i.e., E-UTRAN, UTRAN, and GERAN).

Access Gateway (A-GW): The A-GW belongs to trusted non-3GPP accesses and includes the PMIPv6 MAG functionality for IP mobility management. It also can manage layer-2 mobility as the UE moves within the trusted non-3GPP access.

Evolved Packet Data Gateway (ePDG): For untrusted non-3GPP accesses, the ePDG secures the access of the UE to the EPC by means of an IP Security (IPSec) tunnel between itself and the UE. In case local mobility occurs within the untrusted non-3GPP access, MOBIKE [5] is used to update the IPSec security association.

In addition to the functionality provided by the PMIPv6 specification, there are several additional requirements that the EPC must fulfill that have impact on PMIPv6. Some of the key requirements and impacts are as follows:

Support of IPv4 UE: The EPC requires support for IPv4 only, IPv6 only, and dual stack IPv4 and IPv6 hosts. IPv4 support in PMIPv6 is defined in the IPv4 extension draft [6] to PMIPv6.

Simultaneous access to multiple PDNs: A PDN is an IP domain that the UE wants to communicate with. Examples of PDNs are the Internet, a corporate network, and an operator's private network. An access point name (APN), is used to identify a PDN. The EPC assigns to the UE an IP address that belongs to the PDN to which it is connected and allows the UE to simultaneously access multiple PDNs. Extensions defined in RFC 5149 [7] to PMIPv6 enable the MAG to include the APN in the proxy binding update (PBU) request, such that the PDN GW can assign an IP address to the UE from the appropriate PDN. Furthermore, multiple bindings for a particular UE, one for each PDN, must be supported by the LMA.

Support for overlapping address spaces of different PDNs: In addition to the UE being able to simultaneously access multiple PDNs, the IPv4 addresses assigned to the UE in different PDNs can potentially overlap, for example, the use of private address spaces. To allow for overlapping IPv4 address spaces, the generic routing encapsulation (GRE) key extensions for tunneling packets between the LMA and MAG PMIPv6 [8] are employed. This PMIPv6 extension enables the network to disambiguate traffic related to different PDNs based on the GRE key — even when the IP addresses allocated to the UE by the PDNs are identical.

Unique UE identification across accesses on EPC PMIPv6 interfaces: Because the UE can access the EPC from different accesses, and each access can use its own UE identity scheme, the problem of uniquely identifying a UE on the different PMIPv6 interfaces S5, S2a, and S2b arises. To resolve this issue, 3GPP has specified that an international mobile subscriber identity (IMSI)-based network-access identifier (NAI), where the IMSI is the identity that currently is used to identify the UE in GSM/UMTS networks, is used on all PMIPv6 interfaces. Hence, non-3GPP accesses must obtain the IMSI of the UE during access authentication (either from the UE or from the home subscriber server/authentication, authorization, and accounting (HSS/AAA) and use the IMSI-based NAI on the PMIPv6 interfaces. This does not require any extensions to PMIPv6 because the specification in conjunction with RFC 4283 [9] allows for an IMSI-based NAI to be used as the UE identity in the PBU/proxy binding acknowledgment (PBA).

Providing a PDN GW address to the target access: The EPC can support multiple PDN GWs serving the same PDN. As a consequence, the MAG function in the target access network must identify to which PDN GW to send the PBU upon handover. TS 23.402 [10] specifies that the PDN GW identity along with the corresponding APN is stored in the HSS/AAA and provided to the MAG in the target access during authentication at handover attach. An extension to the Diameter protocol addresses this issue [11].

In addition to the above requirements, signaling of QoS and charging information must occur in the EPC as the UE moves across different access networks. An overview of the architectural aspects related to PCC and QoS provisioning is provided in the next subsection.

PCC and QoS Provisioning

The objective of the PCC architecture is to enable operators to provide QoS to subscribers for IP-based service data flows and charge for the resources provided based on the user's subscription information and other policy information related to the access, network, and service. To not overload PMIPv6 signaling with QoS and PCC aspects, an "off-path" PCC model was developed and documented in TS 23.203 [12].

The key network entities and interfaces of the PCC architecture are illustrated in Fig. 1 and are as follows:

- Subscription profile repository (SPR): The SPR contains the QoS and charging subscription policies for the users.

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- Policy and charging rules function (PCRF): The PCRF makes policy decisions for a UE upon request and provides charging and QoS rules to the policy and charging enforcement function (PCEF) and QoS rules to the bearer binding and event reporting function (BBERF) for enforcement. The charging rules contain information to identify flows (e.g., five tuple) along with charging rates. The QoS rules contain information to identify flows along with the QoS behavior to be enforced, such as the QoS class indicator, maximum bit rate, and so on.
- Policy and charging enforcement function: The PCEF performs the function of IP flow detection and charging based on the PCC rules provided by the PCRF.
- Bearer binding and event reporting function (BBERF): The BBERF performs the function of applying the QoS rules to service data flows in the access network, binding of the IP flows to access bearers, and reporting of QoS-related events (e.g., change-of-access technology) to the PCRF.

The following example scenario helps explain the PCC architecture in more detail. Assume a user is placing a voice over IP (VoIP) call through the IP multimedia subsystem (IMS) situated in the operator's IP services domain. During the call set up, a SIP server in IMS provides QoS-related information (e.g., application type, required bandwidth) and the required information for identification of the service data flows (e.g., the set of five tuples to identify the RTP packets of a VoIP flow) to the PCRF, over the Rx interface. Rx is a Diameter-based interface used by the IMS to request QoS resources for a given set of IP flows and also to be informed by the PCRF about the status of the resource allocation. Based on the operator's policies stored in the PCRF, the user's subscription information obtained from the SPR through the Sp interface, and the application-related information dynamically signaled across the Rx, the PCRF determines both the charging rate to be applied and the QoS behavior (e.g., guaranteed bit rate or not, delay and drop targets, maximum bit rates) to be provided to the set of IP flows requested by the application function. The PCRF encapsulates this request in a so-called PCC rule and forwards it to the PCEF located in the PDN GW node for charging enforcement.

The QoS information with the associated IP-flow description also must be provided to the access network through the S-GW or A-GW node (for 3GPP and trusted non-3GPP access, respectively). Because the PMIPv6 protocol is used only for mobility management and has no notion of QoS tunnels, the off-path paradigm relies on the signaling of QoS information off-the-bearer-path and from the PCRF directly to the access network. The PCRF has a separate interface, namely Gxc for 3GPP accesses and Gxa for trusted non-3GPP accesses,[2] toward the functional entity responsible for QoS enforcement in the access network, referred to as the BBERF.

Inter-Access System Mobility Flows

This section provides the handover flows to illustrate how the architecture principles are applied. Inter-access system handover flows according to TS 23.402 [10] are classified into two categories: non-optimized handover flows and optimized handover flows. Non-optimized handover flows cover a situation where the source network is not involved in preparing resources in the target network. In the case of optimized handovers, the source network is involved in preparing resources in the target network. Optimized handovers are typically used when the UE is unable to transmit and receive in both the source and target networks simultaneously. This section first covers the high-level flows for PMIPv6-based, non-optimized handovers between access networks, and then we present the corresponding flows for optimized handovers.

For the flows in the following subsections, it is assumed that the network initiates the set up of QoS resources on behalf of the UE. Details of a UE-initiated QoS set up are provided in TS 23.402 [10] and TS 23.203 [12].

Non-Optimized Handovers

Figure 2 provides the high-level flow when a UE attaches to a trusted non-3GPP access that is connected to the PDN GW using PMIPv6 on the S2a interface, initiates a VoIP call through IMS resulting in the set up of QoS for the VoIP media flows, and then hands over to a 3GPP access with PMIPv6 used on the S5 interface. Steps 1 through 9 are related to the UE attaching to the trusted non-3GPP access network. Steps 10 through 15 are related to setting up the VoIP call and the QoS establishment for the media flow. Steps 16 through 24 are related to the UE discovering and handing over to the 3GPP access. Steps 25 through 28 are related to the cleanup of resources in the trusted non-3GPP access.

The attachment of the UE to the EPC through a trusted non-3GPP access is triggered by the UE sending a layer 2 or layer 3 attach trigger (Step 2), for example, an *attach request* message. The UE authenticates with the trusted non-3GPP access network. In case the non-3GPP access supports multiple PDNs, the PDN to which the UE must be connected is either provided by the UE in the attach request message or is obtained from the HSS/AAA. For the set up of default QoS resources, the BBERF in the non-3GPP access registers itself with the PCRF, providing the UE identity and the APN to the PCRF. The MAG function in Step 5 sends a PBU to the LMA to obtain the IP address of the UE. The PBU contains the IMSI-based NAI of the UE, the APN to which the UE must be connected and a GRE key, which the PDN GW should use to tunnel downlink packets for the UE. Based on the APN in the PBU, the PDN GW provides an IP address to the UE in the PBA that is further relayed to the UE (Steps 8 and 9), completing the attachment of the UE to the non-3GPP access. In the PBA, the PDN GW also provides the GRE key that the MAG should

Because the PMIPv6 protocol is used only for mobility management and has no notion of QoS tunnels, the off-path paradigm relies on the signaling of QoS information off-the-bearer-path and from the PCRF directly to the access network.

For dual-radio-capable UEs, non-optimized handovers can provide a seamless handover experience to the end user. A "make-before-break" can be achieved as the UE can still maintain connectivity through the source access while it establishes connectivity over the target access.

use for tunneling the uplink traffic of the UE. In parallel, the PDN-GW registers itself with the PCRF and obtains charging rules for the default connectivity of the UE. The PDN GW also registers its address and the APN to which the UE is connected with the HSS through the AAA server. The HSS stores this information to be provided to the 3GPP access at a later stage when the UE hands over to the 3GPP access.

When the UE wants to set up a VoIP call, SIP signaling occurs between the UE and the SIP server (Step 10). In turn, the SIP server requests the PCRF to set up QoS resources for the UE in the access network by providing the identity of the UE and the relevant information to enable identification of the IP flow and the QoS to be applied to the flow. The PCRF determines the charging rules and the QoS rules to be applied based on the subscription information. The PCC charging rules are provided to the PDN GW (Step 13), and the QoS rules are provided to the BBERF in the trusted non-3GPP access network (Step 14). The BBERF then sets up a bearer with the appropriate QoS for the VoIP media (Step 15). After these steps, the VoIP data traffic is carried over the dedicated VoIP bearer; the other traffic remains on the default bearer.

When the UE decides to hand over to the 3GPP access, the UE initiates the attach procedure to the 3GPP access (Step 17). In the attach request, the UE indicates that the attach type is *handover attach* and is hence requesting to be attached to the same PDN GW and also to be provided with the same IP address that it had when attached through the trusted non-3GPP access. During the authentication procedure (Step 18), the HSS provides the 3GPP access with the IP address of the PDN GW of the UE. The BBERF function in the 3GPP access registers itself with the PCRF and obtains the QoS rules for the default traffic but also for the VoIP traffic (Step 19). It sets up the required bearer resources during the attach procedure itself to avoid disruptions due to lack of resources in the target access.

The MAG function in the serving GW then sends a PBU to the LMA in the PDN GW in which the hand-off indicator value is set to *hand-off* between two interfaces of the UE (Step 21). The LMA updates the PMIPv6 tunnel to point to the 3GPP access and provides the IPv6 home network prefix and/or the IPv4 home address of the UE to the MAG function in the 3GPP access as part of the PBA (Step 23). In parallel, the PDN GW also updates the PCRF that the UE is connected to the EPC through the 3GPP access and obtains the corresponding charging rules.

In the EPC, the PDN GW initiates the resource release procedure in the source access system after the user-plane tunnel has been switched. The PDN GW initiates proxy binding revocation indication (BRI) as defined in [13], which triggers the resource release procedure in the source access (Steps 25–27). The BBERF in the non-3GPP access also deregisters itself with the PCRF (Step 28).

For dual-radio-capable UEs, where the radios of both access technologies can transmit and receive packets simultaneously, non-optimized handovers can provide a seamless handover experience to the end user. A "make-before-break" can be achieved as the UE can still maintain connectivity through the source access while it establishes connectivity over the target access.

OPTIMIZED HANDOVERS

Whereas the non-optimized handover is well suited for dual-radio-capable terminals, for single-radio terminals it would lead to substantial interruption time during inter-technology handovers. As a result, optimized handovers were defined for specific instances of inter-technology mobility, allowing for seamless handovers even for single-radio terminals.

The architecture used for optimized handover between CDMA2000 eHRPD and an E-UTRAN is shown in Fig. 3.

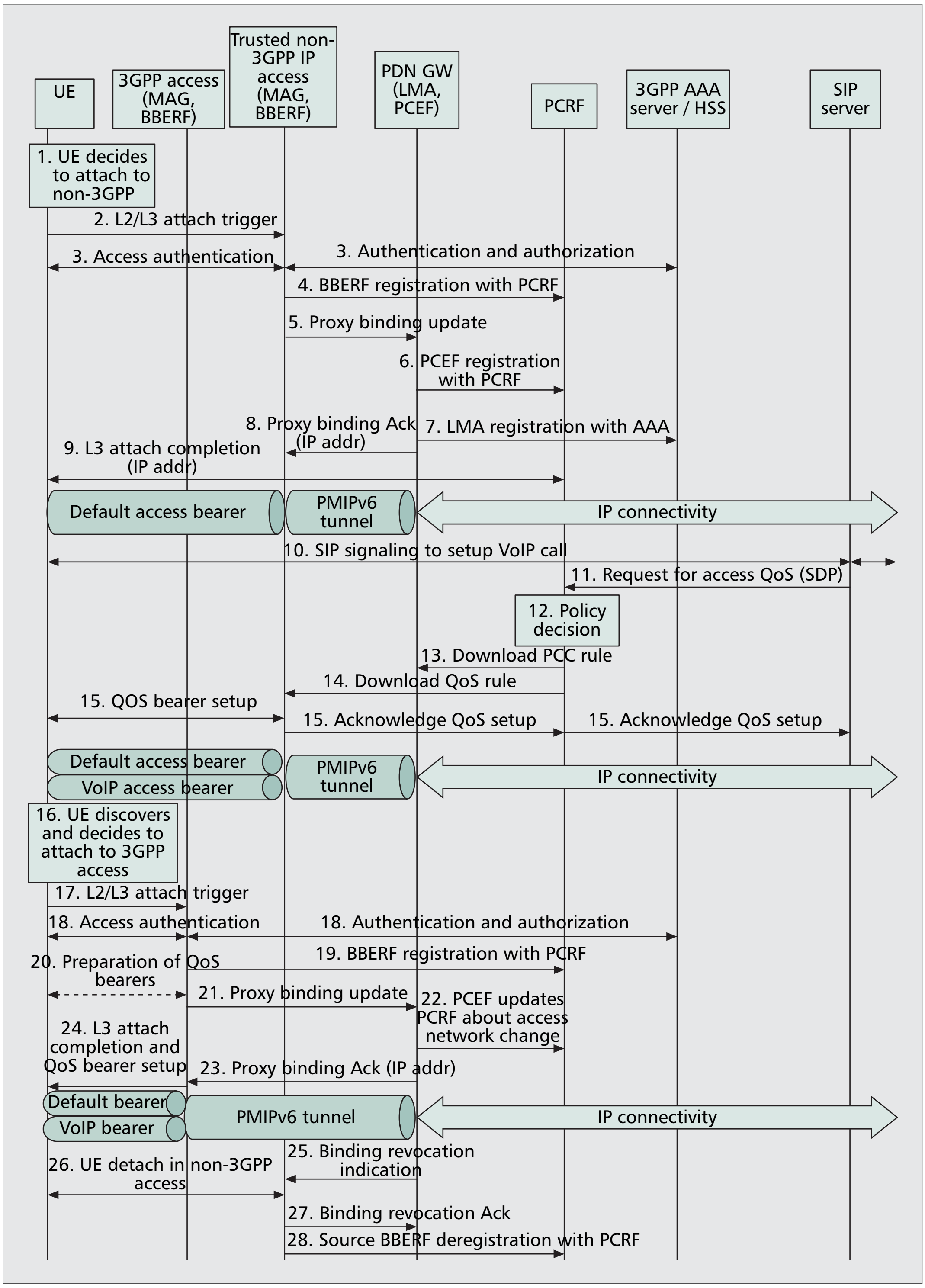
The additional network entities in this architecture that were not discussed yet are:

- Mobility Management Entity (MME): The MME is a control-plane entity in an E-UTRAN access network responsible for managing the mobility of the UE. It also authenticates the UE with the HSS. Details of the MME functions are described in TS 23.401 [14].
- HRPD Access Network (HRPD AN): The HRPD AN is the network entity in the evolved eHRPD access network that is responsible for managing the mobility of the UE.
- HRPD Serving Gateway (HSGW): The HSGW is an entity in the eHRPD access network that performs functions similar to that of the serving GW and MME in 3GPP accesses. The HSGW contains the MAG and the BBERF functions.

Optimized handovers between E-UTRAN and eHRPD rely on the UE managing the context establishment within the target access network while still operating on the source system. This is achieved through a tunnel with the target system allowing the UE to interact with the target system with minimal support from the source system. The S101 interface is used for tunneling the UE traffic to the target access system. The source system still provides network control to trigger interactions between the UE and the target system but otherwise is not involved in the establishment of context in the target system. The S103 interface is used for temporarily forwarding the downlink traffic of a UE from the source to the target system during the handover execution.

To describe the different phases of an optimized handover, the following terms are introduced:

- **Pre-Registration**: In the pre-registration phase, the UE communicates with the target system by tunneling registration signaling through the source system to prepare session context (e.g., authentication, session parameters).
- **Preparation**: The assumption is that the UE has performed pre-registration and now has been instructed by the source system to initiate a handover. In this phase, the target system prepares for the UE to handover and provides the UE with the required



■ **Figure 2.** *Illustration of the scenario when a UE attaches to non-3GPP access, initiates a service, and then hands over to 3GPP access.*

Optimized handovers between E-UTRAN and eHRPD rely on the UE managing the context establishment within the target access network while still operating on the source system. This is achieved through a tunnel with the target system where the UE can interact with the target system with minimal support from the source system.

information to establish radio connectivity over the target access network.
- **Execution**: In the execution phase, the UE uses the information provided by the target system in the preparation phase (delivered through the source system) to switch radio technologies.

Release 8 of the EPC standard only defines optimized handover between eHRPD and E-UTRAN. Future releases may also define optimized handover with other non-3GPP access networks.

Figure 4 provides the flow for optimized handover of the UE from E-UTRAN to eHRPD. It is assumed that the UE already is attached to the E-UTRAN access and has an active VoIP call whose media traffic flows through the VoIP bearer with appropriate QoS in the E-UTRAN access.

In anticipation of possible handover as the UE

At the end of pre-registration, the UE has established its credentials in the target eHRPD access. Also, the BBERF in the target eHRPD access has obtained QoS rules for all active sessions and can prepare resources in anticipation of a handover.

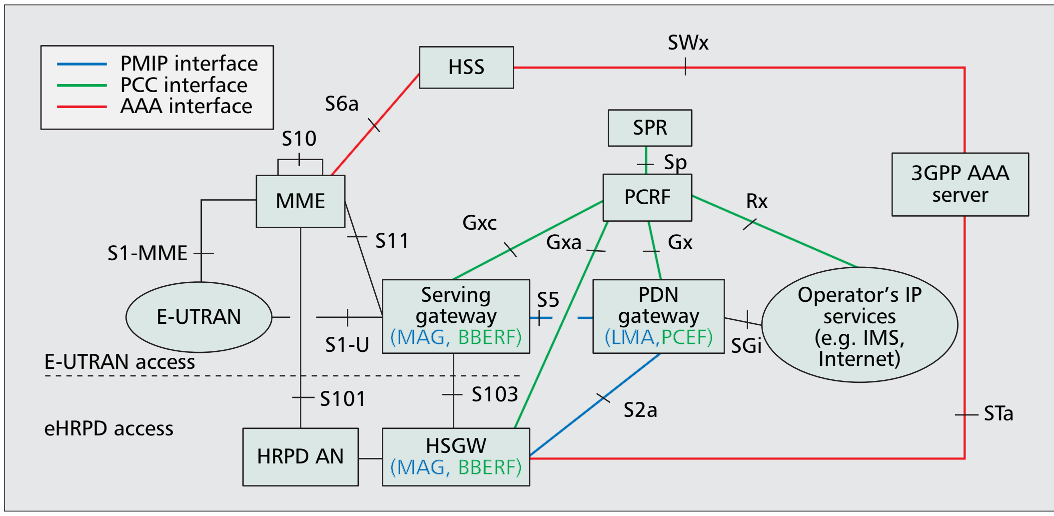


Figure 3. *Architecture for optimized handovers between E-UTRAN and eHRPD access.*

approaches the source technology coverage boundary, the source network provides system information (broadcast or unicast) indicating that the UE should pre-register with the target system. The purpose of pre-registration is to avoid lengthy delays in the handover procedures because pre-registration can take several seconds. The steps for pre-registration are shown in the steps A1 through A7. The pre-registration messages are sent by the UE as direct transfer messages to the MME through the eNodeB. The eNodeB adds the identification of the closest target eHRPD cell to enable the MME to determine the HRPD AN to tunnel the UE messages. At the end of pre-registration, the UE has established its credentials in the target eHRPD access. Also, the BBERF in the target eHRPD access has obtained QoS rules for all active sessions and can prepare resources in anticipation of a handover.

Steps B1 through B4 correspond to the preparation phase of the optimized handover. When the UE approaches the technology coverage area boundary, radio conditions dictate that an inter-technology handover is required. For this, the network configures the UE to report when the source system signal quality drops below a specific threshold, and the target technology is above a specific threshold. After the source system receives this indication, it triggers the UE to initiate handover preparation and execution procedures (Step B2). As part of the preparation phase, the UE requests from the target system (through the source system) to allocate the required radio resources and the establishment of the S103 tunnel for forwarding the UE traffic from E-UTRAN to eHRPD (Step B3–B4).

Steps C1 through C9 correspond to the execution phase of the optimized handover. In this phase, the UE uses the traffic channel allocation information provided during the preparation phase to perform the handover procedures. Although there is no explicit context transfer from the source system to the target system, the entire procedure is controlled by the source system so as to provide the operator greater control. The MAG in the HSGW updates the bearer tunnel by exchanging a PBU/PBA with the LMA in the PDN GW. The PDN GW interacts with the PCRF to obtain charging rules corresponding to eHRPD access for the user traffic. Steps C6 through C9 are for the cleanup of resources in the source network.

The optimized handover procedures in the opposite direction, that is, from eHRPD to E-UTRAN, follow the same basic principles. The main difference is that the handover execution phase takes place immediately after the completion of the pre-registration phase, and no data forwarding is performed through the S103 tunnel.

Summary and Future Work

This article presented the motivation, design, and realization of inter-access system mobility support based on Proxy Mobile IPv6 for the 3GPP EPC, enabling a common packet core architecture to be used for a wide range of access technologies. The document also addresses the issues of QoS provisioning and seamless handover support. Detailed flows illustrating the use of PMIPv6 to achieve non-optimized handovers between 3GPP accesses and other non-3GPP accesses, as well as optimized handovers between E-UTRAN and eHRPD were provided.

Release 8 is the first release of the EPC specification, and additional work is required to enhance and adapt the new system to the ever changing industry requirements. For instance, further study is required to determine how to support the UE to access the EPC through multiple-access networks simultaneously while providing mobility management and controlling the routing of individual IP flows between the different radio interfaces.

Finally, as operational experience for "always best-connected terminals" increases, optimizations, and lessons learned from the field will drive additional enhancements of the EPC.

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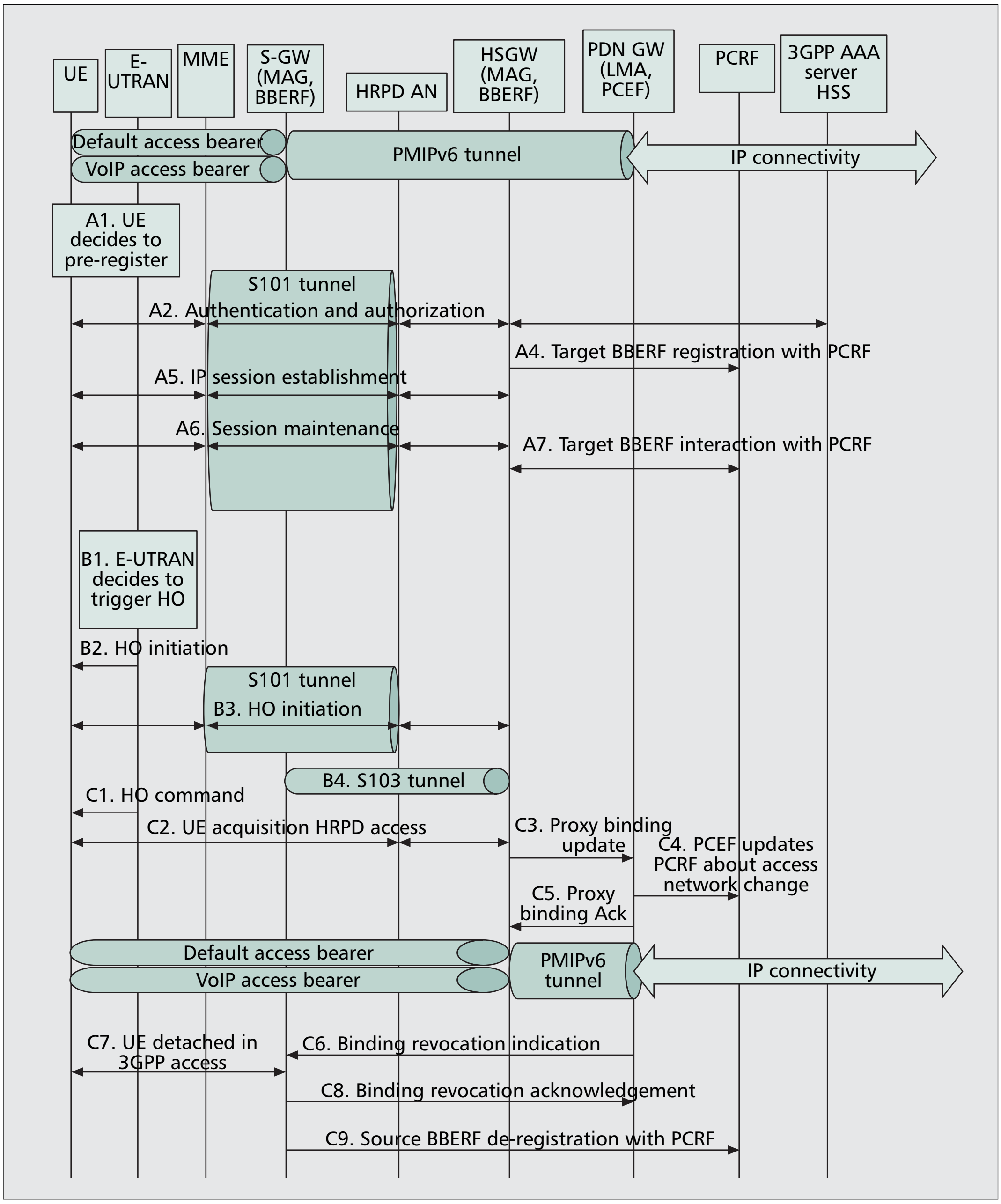

Figure 4. *Flow for optimized handover from E-UTRAN access to eHRPD access.*

Further study is required to determine how to support the UE to access the EPC through multiple-access networks simultaneously while providing mobility management and controlling the routing of individual IP flows between the different radio interfaces.

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BIOGRAPHIES

STEFAN SCHMID (stefan.schmid@neclab.eu) received his M.Sc. in computer science (Dipl.-Inf.) from the University of Ulm, Germany, in 1999. In 2002 he received his Ph.D. in computer science from Lancaster University. After his post-doctoral research, he joined NEC Laboratories Europe in

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Policy and Charging Control in the Evolved Packet System

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ABSTRACT

Policy and charging control provides operators with advanced tools for service-aware QoS and charging control. PCC for the evolved packet system, defined as part of the *3GPP Release 8* specifications, has evolved significantly from previous releases to support multiple-access technologies, roaming, and mobility. Within the PCC framework, a number of protocols have been specified to implement these functions. This article describes key PCC concepts and explains additional amendments to support PCC in the EPS.

INTRODUCTION

Over the years, policy control in the Third Generation Partnership Project (3GPP) has evolved with different features and different architectures. This article provides an overview of the Release 8 policy and charging control (PCC) architecture [1] as it applies to the evolved packet system (EPS) [2, 3]. The target date for the completion of Release 8 was December 2008.

This section contains a brief background on the evolution of PCC in 3GPP. The next section provides an overview of the PCC framework and presents the architecture and enhancements required to support multiple-access technologies in the EPS. Then, we introduce basic PCC concepts, such as how rules are used to make PCC decisions, how these decisions are bound to the access network bearers, how they are renewed, and how they are enforced. The following section provides an overview of PCC roaming and different roaming scenarios. We then describe the protocols defined within the PCC framework. The article concludes with a short summary.

The requirement for resource reservation and access control for individual Internet Protocol (IP) flows arose with the introduction of the IP multimedia subsystem (IMS) in 3GPP Release 5 (completed in 2002). This requirement was addressed by service-based local policy (SBLP) functionality within the IMS domain [4], which provided bearer-level QoS control and service-level access control. The *SBLP* architecture was influenced by the policy control framework defined by the Internet Engineering Task Force (IETF) [5] and efficiently connected the IMS and the general packet radio service (GPRS) domains. SBLP was enhanced further in Release 6 (completed in 2005) by separating it from IMS, thus allowing SBLP to be used by generic services.

Meanwhile, market demand for increasingly sophisticated charging models, for example, those based on application-layer services and content, resulted in the introduction of the flow-based charging (FBC) architecture in Release 6 [6]. *FBC* enables charging on a per-service basis for both offline (charging data record [CDR]-based) and online (e.g., pre-paid) models. FBC provides operators with a centralized mechanism for controlling access to pre-defined services and content, for example, access to a certain Web page or a movie. In addition to pre-defined services, the FBC architecture also provides access control and charging for dynamically defined services, for example, IMS-based services.

SBLP and FBC architectures were very similar. This resulted in mutual interference because both architectures were based on a central node capable of making decisions, and both had the same anchor points: the application function (AF) in the control plane and the gateway GPRS support node (GGSN) in the user plane. This problem was addressed in 3GPP Release 7 (completed in 2007) where SBLP and FBC were merged into a common solution called PCC. PCC is a service-aware control architecture providing operators with a standardized mechanism for QoS and charging control applicable to both IMS and non-IMS based services.

THE PCC ARCHITECTURE

OVERVIEW

Release 8 enhances the basic PCC architecture developed in 3GPP Release 7 by adding new functionality, reference points, and functional entities.

The *evolved packet core* (EPC) defines a single core network for multiple heterogeneous accesses. Because PCC was designed to be access-agnostic in Release 7, it is easily adaptable to EPS. However, amendments are required in Release 8 to accommodate new EPS architectural requirements. These requirements include: support for mobile IP-based protocols in the EPC, roaming, and mobility between heterogeneous accesses.

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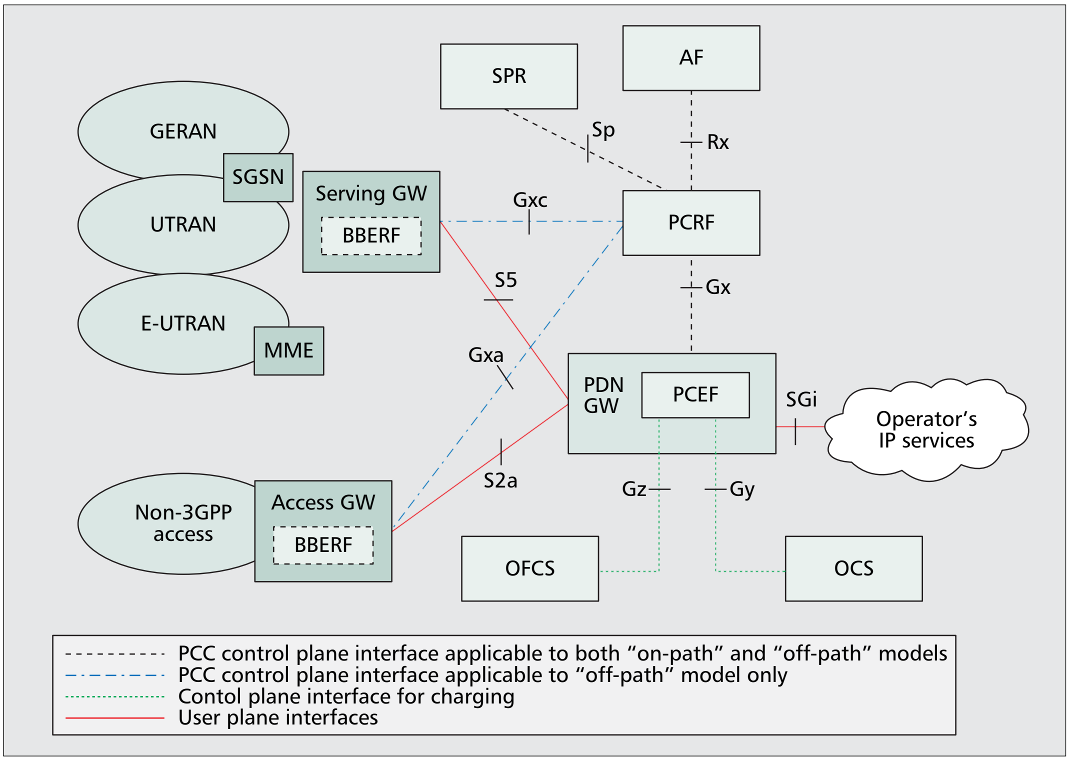


Figure 1. *3GPP Release 8 PCC non-roaming architecture for EPS. Note that only a subset of the EPS reference points and EPS network entities are shown.*

The evolved packet core (EPC) defines a single core network for multiple heterogeneous accesses. Because PCC was designed to be access-agnostic in Release 7, it is easily adaptable to EPS. However, amendments are required in Release 8 to accommodate new EPS architectural requirements.

The *reference network architecture* for PCC in EPS is shown in Fig. 1.

The AF interacts (or intervenes) with applications that require dynamic policy and charging control. It extracts session information and provides this to the policy and charging rules function (PCRF) over the Rx reference point. The AF also can subscribe to certain events that occur at the traffic plane level (i.e., events detected by either the policy and charging enforcement function [PCEF] or bearer-binding and event-reporting function [BBERF]). Those traffic plane events include events such as IP session termination or access technology-type change. When the AF has subscribed to a traffic plane event, the PCRF informs the AF of its occurrence.

The *PCRF* is the policy engine of PCC. It combines the session information received over the Rx reference point and the input received from the Gx and Gxa/Gxc reference points with user-specific policies and data from the subscription profile repository (SPR) to form session-level policy decisions and provides those to the PCEF and the BBERF. The PCRF also forwards events between the BBERF, the PCEF, and the AF.

The *PCEF* enforces policy decisions received from the PCRF and also provides the PCRF with user- and access-specific information over the Gx reference point. It interacts with the online charging system (OCS), which is a credit management system for pre-paid charging and reports usage of resources to the offline charging system (OFCS).

In the PCC architecture for EPS, there are two main architecture alternatives: with and without BBERF in access gateway (e.g., serving gateway [S-GW] or high-rate packet data [HRPD] serving gateway [HSGW]). In this article the two alternatives also are referred to as "off-path" and "on-path" models, respectively. The details regarding these two alternatives are discussed below.

Multi-Access and the Off-Path PCC Model

This section begins with a brief explanation of why two architecture alternatives (with and without BBERF) were defined for Release 8.

EPS allows for different mobility protocols depending on which access technology is used [2]. For the 3GPP family of accesses (global system for mobile communications-enhanced data rates for GSM evolution [GSM-EDGE] radio access network [GERAN], universal mobile telecommunications system [UMTS] terrestrial radio access network [UTRAN], and evolved-[E]-UTRAN), either the GPRS Tunneling Protocol (GTP) or Proxy Mobile IPv6 (PMIPv6) can be used. For connecting other accesses to the EPC, any of PMIPv6, Dual Stack Mobile IPv6 (DSMIPv6) or Mobile IPv4 (MIPv4) can be used. These different protocols have different properties, which results in different requirements for PCC.

The *GTP*, as defined by 3GPP in earlier releases, not only supports functions related to packet routing and mobility, but also the functions that handle QoS, bearer signaling, and so on. When the GTP is used in the EPC between the S-GW and the *packet data network gateway* (PDN GW), the PDN GW can control the QoS

Another EPS requirement on PCC is handover within and between heterogeneous accesses, for example, to support the scenario where the terminal moves between different access GWs (BBERFs). In this case, the PCRF must handle the BBERF relocation and provide the authorized QoS to the new BBERF.

through the bearer procedures toward the serving GW. The term "bearer" refers to a logical IP transmission path between the terminal and the network with specific QoS properties (capacity, delay, packet loss error rate, etc.). The bearer procedures are used to set up, modify, or remove the bearers.

Because bearer procedures are supported by the GTP and the PDN GW, the PCRF can control the QoS through the Gx interface. Such a model is referred to in this article as on-path. It is so named because the QoS/bearer signaling takes place (using the GTP) on the same path as the user plane. The BBERF and Gxa/Gxc have no role here.

Unlike the GTP, *mobile IP protocols* (PMIPv6, DSMIPv6, MIPv4) are defined by IETF for the express purposes of packet routing and IP-level mobility. They are not used to set up, modify, or tear down bearers or to signal QoS parameters. When a mobile IP protocol is used between an access GW (e.g., the S-GW) and the PDN GW, the PDN GW has no knowledge of the bearers. The BBERF was introduced into the architecture to handle this situation. The PCRF provides the authorized QoS to the BBERF over the Gxa and Gxc reference points. This model is referred to as the off-path because QoS signaling takes place (through Gxa/Gxc) on a path different from that of the user plane.

For EPS, the PCEF always is located in the PDN GW. The BBERF location, however, depends on the particular access technology. For example, for the 3GPP family of accesses, the BBERF (if applicable) is located in the serving GW, whereas for eHRPD access, the BBERF is located in the HSGW [7].

The architectural aspects discussed above were added to PCC in Release 8 to support the requirement for multiple accesses. Another EPS requirement on PCC is handover within and between heterogeneous accesses, for example, to support the scenario where the terminal moves between different access GWs (BBERFs). In this case, the PCRF must handle the BBERF relocation and provide the authorized QoS to the new BBERF.

BASIC PCC CONCEPTS

POLICY AND CHARGING CONTROL DECISIONS

Policy control consists of gating control and QoS control. *Gating control* is the capability to block or to allow IP packets belonging to a certain IP flow, based on decisions by the PCRF. The PCRF could, for example, make gating decisions based on session events (start/stop of service) reported by the AF through the Rx reference point. *QoS control* allows the PCRF to provide the PCEF with the authorized QoS for a given IP flow. The authorized QoS can, for example, include the authorized QoS class and the authorized bit rates.

Charging control not only implies service access control by means of online credit management, but it also contains important redirect functionality that is used, for example, for advice of charge and top-up of pre-paid accounts. The OCS may authorize access to individual services or to a group of services by granting credits for authorized IP flows. Usage of resources is accredited on the form of a limited amount of time, traffic volume, or chargeable events. If a user is not authorized to access a certain service, for example, in the case of an empty pre-paid account, then the OCS may deny credit requests and, additionally, instruct the PCEF to redirect the service request to a specified destination (for account top-up). As already mentioned, PCC also incorporates service-based offline charging. However, this functionality does not provide any means for access control in itself. Instead, policy control must be used to restrict access and then service-specific usage can be reported to the OFCS.

The PCRF is the central entity in PCC-making policy and charging-control decisions. The decisions can be based on input from a number of different sources, including:

- Operator configuration in the PCRF that defines the policies applied to given services
- Subscription information/policies for a given user, received from the SPR
- Information about the service received from the AF
- Information from the access network about what access technology is used and so on

An example use case for the on-path model is illustrated in Fig. 2 and further described below:

1. The subscriber initiates a service, for example, an IMS voice call, and performs IMS session signaling through the AF (proxy-call server-control function [P-CSCF] in the case of IMS).
2. Based on the service description information contained in the application signaling, the AF provides the PCRF with the service-related information over the Rx interface. This information typically includes traffic parameters (e.g., IP addresses and port numbers), as well as QoS information (type of service, bit rate requirements).
3. The PCRF can request subscription-related information from the SPR.
4. The PCRF takes the operator-defined service policies, subscription information, and other data into account when building policy decisions. The policy decisions are formulated as PCC rules and typically contain information about the user-plane traffic (packet filters in the form of IP five-tuple). All IP packets matching the packet filters of a PCC rule are designated a service data flow (SDF). PCC rules also can contain the authorized QoS (QoS class and bit rates) and charging information, as well as an indication of whether traffic is allowed (gate open) or not allowed (gate blocked). A subset of the available parameters in the PCC rule is shown in Table 1.
5. The PCC rules are sent by the PCRF to the PCEF for enforcement of the policy decision. The PCEF is located in an edge node where all the user-plane traffic for a given subscriber and the IP connection passes. For EPS, the PCEF is located in the PDN GW.
6. The PDN GW/PCEF installs the PCC rules and performs bearer binding to ensure that the traffic for this service receives appropri-

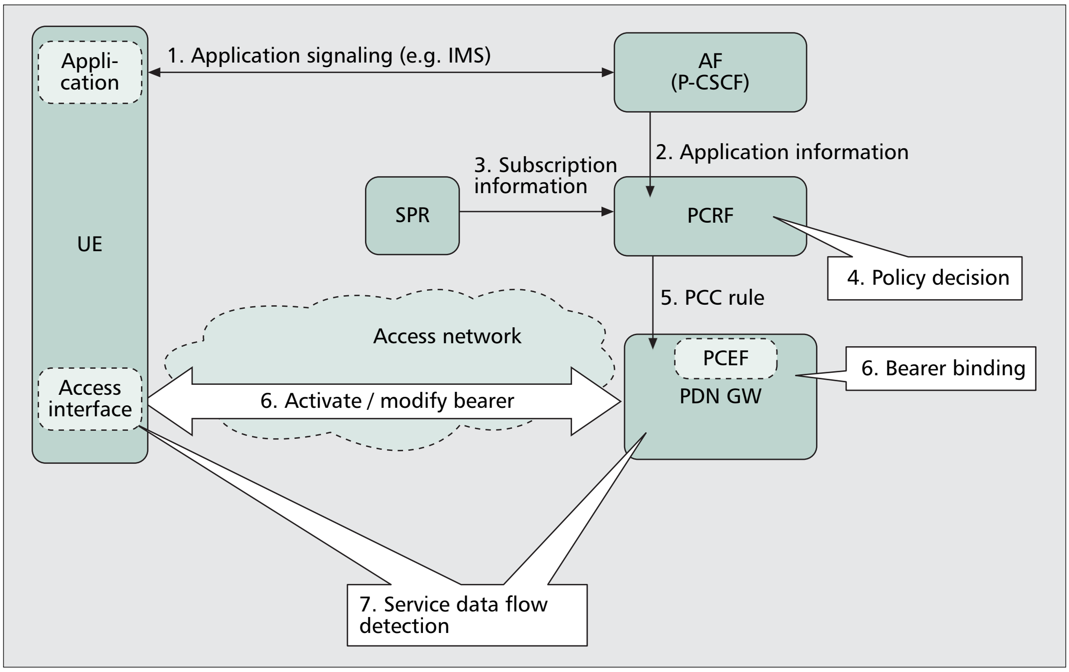


Figure 2. *High-level use case for PCC in the EPS for the on -path model.*

Apart from dynamically provisioned rules (based on IP five-tuple filters) the PCEF may be instructed by the PCRF to take predefined charging rules into account in the evaluation process. The definition of filters for predefined rules is not standardized.

ate QoS. This may result in the establishment of a new bearer or a modification of an existing bearer. More details on bearer binding are presented later in this section.

7 The PCEF performs SDF detection to detect the IP flow for this service. This IP flow is transported over the appropriate bearer. More details on SDF detection can be found later in this section.

For the off-path model, the PCRF also provides the information about the authorized QoS by sending so called QoS rules to the BBERF over the Gxa/Gxc reference points. The QoS rules contain the information required for the BBERF to ensure that bearer binding (see below) can be performed. Thus, the QoS rules contain only a subset of the information of a corresponding PCC rule and do not include any charging-related information.

The policy and charging enforcement in PCEF comprises several different aspects as discussed below.

Bearer Binding in the Access Network

PCC is responsible for the successful interaction with the QoS procedures in the access network.

The PCC rule must be mapped to a corresponding QoS conduit or bearer in the access network to ensure that the packets receive the appropriate QoS treatment. This mapping (i.e., *bearer binding*) function is one of the central components of PCC. The mapping is performed by the bearer-binding function (BBF) that is located in the PCEF (for on-path) or in the BBERF (for off-path). When the PCEF (or BBERF) receives new or modified PCC or QoS rules, the BBF evaluates whether or not it is possible to use the existing bearers. If one of the existing bearers can be used (e.g., if a bearer of the corresponding QoS class already exists) the BBF may initiate bearer modification procedures to adjust the bit rates of that bearer. If it is not possible to use the existing bearers, the BBF must initiate the establishment of a suitable new bearer. In particular, if the PCC rule contains guaranteed bit-rate (GBR) parameters, the BBF also must ensure the availability of a bearer that can accommodate the GBR traffic for that PCC rule. Therefore, the BBF must trigger resource reservation in the access network to ensure that the authorized QoS of the PCC rule can be provided. Further details on the bearer concept can be found in [8].

Service Data-Flow Detection

The PCEF and the BBERF use the packet filters of installed PCC and QoS rules to classify IP packets to authorized SDFs. This process is referred to as *SDF detection*. Incoming packets are matched against the available filters of the installed rules in order of precedence. If a packet matches a filter and the gate of the associated rule is open, then the packet can be forwarded to its destination (note that online charging can prevent this in the PCEF). For the downlink part, the classification of an IP packet to an SDF also determines which bearer should be used to transfer the packet (Fig. 3).

Apart from dynamically provisioned rules (based on IP five-tuple filters) the PCEF may be instructed by the PCRF to take *predefined charging rules* into account in the evaluation process. The definition of filters for predefined rules is not standardized. These filters can take parameters of higher layers into account; in that case they are referred to as deep packet inspection (DPI) filters.

Renewed Policy Decisions During a Session — Event Triggers

During the lifetime of a session, the conditions in the access network may change. For example, the user may move between different access technologies. There also may be situations where the authorized QoS can no longer be maintained over the

Type of element	PCC rule element	Comment
Rule identification	Rule identifier	Used between PCRF and PCEF for referencing PCC rules.
Items related to service data flow detection in PCEF	Service data flow template	List of packet filters for the detection of the service data flow.
	Precedence	Determines the order, in which the service data flow templates are applied at PCEF.
Items related to policy control (i.e., gating and QoS control)	Gate status	Indicates whether a SDF may pass (gate open) or shall be discarded (gate closed).
	QoS class identifier (QCI)	Identifier that represents the packet forwarding behavior of a flow.
	UL and DL maximum bit rates	The maximum bitrates authorized for the service data flow.
	UL and DL guaranteed bit rates	The guaranteed bitrates authorized for the service data flow.
Items related to charging control	Charging key	The charging system uses the charging key to determine the tariff to apply for the service data flow.
	Charging method	Indicates the required charging method for the PCC rule. Values: online, offline, or no charging.
	Measurement method	Indicates whether the SDF data volume, duration, combined volume/duration or event shall be measured.

Table 1. *A subset of the elements that may be included in a PCC rule.*

radio link. In these cases, the PCRF may re-evaluate its policy decisions and provide new or updated PCC rules to the PCEF. The PCRF defines the conditions when the PCEF (and BBERF, if applicable) interact with the PCRF again. It does this by setting event triggers in the PCEF/BBERF. When an event occurs, and the corresponding event trigger is set, the PCEF/BBERF reports the event to the PCRF and allows the PCRF to revisit its previous policy decisions.

When mobile IP-based protocols are used in the EPC, the access-specific bearers terminate in the BBERF instead of the PCEF. This implies that much of the information about the access network (e.g., information regarding available QoS on the radio link, information on access type, etc.) is available only to the BBERF. This being the case, the BBERF detects events and reports them over the Gxa/Gxc reference points. In the off-path model, the Gxa/Gxc and Gx interfaces also are used for more generic information transfer. For example, some of the information provided by the BBERF also is required in the PDN GW/PCEF to enable proper charging in the PCEF. In particular, the PDN GW might be required to know which 3GPP radio technology is used (GERAN, UTRAN, or E-UTRAN), and this information is not necessarily provided through the mobile IP-based protocol. It then must be provided by the BBERF to the PDN GW/PCEF through the PCRF, as illustrated in Fig. 4.

As seen above, most functions of the PCEF are common to both on-path and off-path models. For example, service-level charging, gate control, and QoS enforcement is performed in the PCEF for both models. However, as also seen above, the bearer-related functions and certain event reporting must be performed by the BBERF in the off-path case.

Roaming with PCC

PCC in 3GPP Release 8 supports roaming for both on-path and off-path scenarios. The subscriber can connect through a PDN GW in the home network or in the visited network. However, control of enabled services and authorized resources always are handled by a PCRF in the home network. A new reference point, S9, is defined between PCRFs in the home and visited networks to support certain roaming scenarios. For those roaming scenarios when S9 is used, the PCRF in the visited network can reject but not change policy decisions coming from the home network.

The two main roaming scenarios are home-routed access and visited access (also known as local breakout or LBO) (Fig. 5). In the *home-routed roaming scenario*, an IP connection is established through a PDN GW in the home-public-land mobile network (H-PLMN). Because the PCEF is controlled by the home operator, service-aware charging and control of both dynamically defined (e.g., IMS) and PCEF-predefined services can be used: The PCEF connects (as usual) to the home-PCRF (H-PCRF) through Gx, and online charging is performed (if applicable) through Gy to the OCS.

If the EPC in the visited-PLMN (V-PLMN) is based on mobile IP (i.e., PMIPv6, DSMIPv6, or MIPv4), then the BBERF in the V-PLMN is connected through a visited-PCRF (V-PCRF) to the H-PCRF over the S9 reference point. For this case, the H-PCRF is responsible for control of the gateway in the V-PLMN. Consequently, the H-PCRF provides policy decisions (QoS rules) to the BBERF in the V-PLMN and also forwards information from the BBERF to the PCEF in the H-PLMN.

For certain scenarios, home-routed access may not be desired, for example, if the H-PLMN and the V-PLMN are geographically distant and/or if dynamically provisioned rules are sufficient (i.e., no DPI is required in the PCEF). For those cases, *visited access* can be more suitable. In visited access, a PDN connection is established directly through a PDN GW in the V-PLMN. If a GTP-based EPC is used in the V-PLMN, then the PDN GW in the V-PLMN is connected through a V-PCRF to the H-PCRF through S9. On the other hand, if a mobile IP-based EPC is used in the V-PLMN, then the S9 reference point and also the role of the V-PCRF become more complex for the visited-access case. This is due to the fact that both Gx and Gxa/Gxc procedures are handled using the same S9 session. The V-PCRF must be able to handle splitting and combining of messages to and from S9 on one side and Gx and Gxa/Gxc on the other side. However, certain Gxa/Gxc interactions in the V-PLMN may be hidden from the H-PCRF by the V-PCRF.

For the visited-access case, it is possible to use AFs located in the V-PLMN. In this situation, Rx signaling is proxied through the V-PCRF to the H-PCRF.

In the visited-access case where online charging is used, a proxy OCS can be used in the V-PMLN to connect to the OCS in the home network.

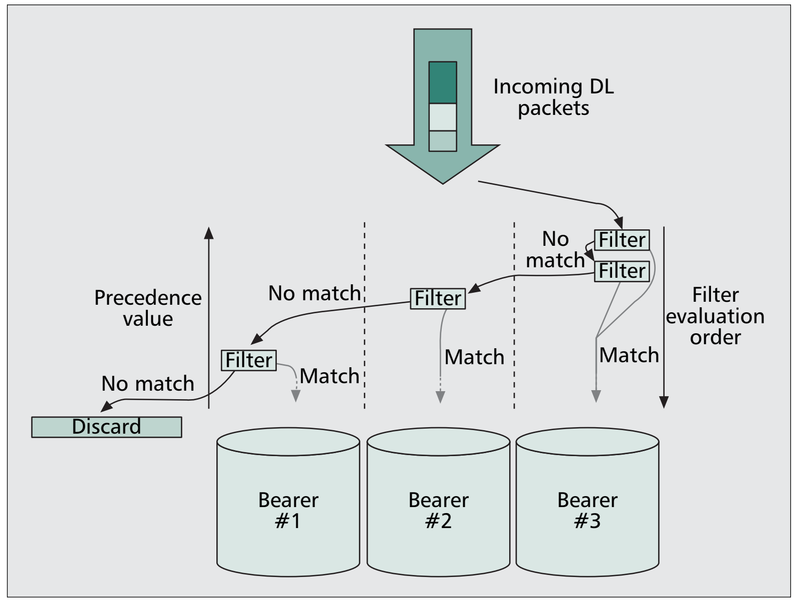


Figure 3. *Example of SDF detection and downlink bearer binding.*

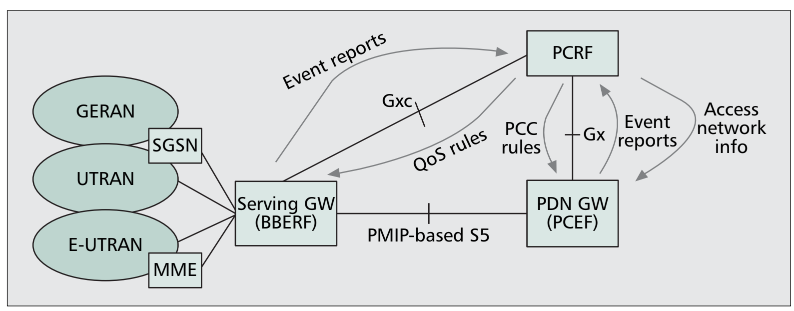


Figure 4. *Information flow for the off-path model. Only the 3GPP family of accesses are shown in the figure for simplicity.*

PROTOCOLS

PCC has defined protocols for the Gx, Gxa, Gxc, Rx, Gy, S9, and Gz reference points in Release 8. The protocol over the Sp reference point is not specified and is left as an implementation option.

All the protocols specified within the PCC framework (except for Gz when the GTP is used) are based on Diameter [9]. The main advantage of using Diameter is the easy reusability, extensibility, and flexibility. The Rx protocol is based on the Diameter network access-server application [10]. The Gx, Gxx, Gy, and S9 protocols are based on the Diameter Credit-Control Application (DCCA) [11]. The fact that these protocols are based on existing applications means that they inherit message types (called commands in Diameter) and mandatory parameters (so-called attribute-value pairs, or AVPs, in Diameter), but they own new application identifiers, add new AVPs, and modify the protocol state machines according to their own procedures.

The Gx protocol (specified in [12]) is used over the Gx reference point, which is located between the PCRF and the PCEF. It was introduced in 3GPP Release 6 and since then has evolved further. Only minor changes were introduced in Release 8.

The Gxx protocol (specified in [12]) is new in Release 8 and is used over the Gxa/Gxc reference points. The Gxa/Gxc reference points are located between the PCRF and the BBERF. The Gxx protocol is based on Gx but reuses only those parameters that are not related to charging. Indeed, instead of using the PCC rules as the main set of parameters, the Gxx protocol uses QoS rules. QoS rules include the same policy-related parameters as the PCC rule, but the charging-related information is left out.

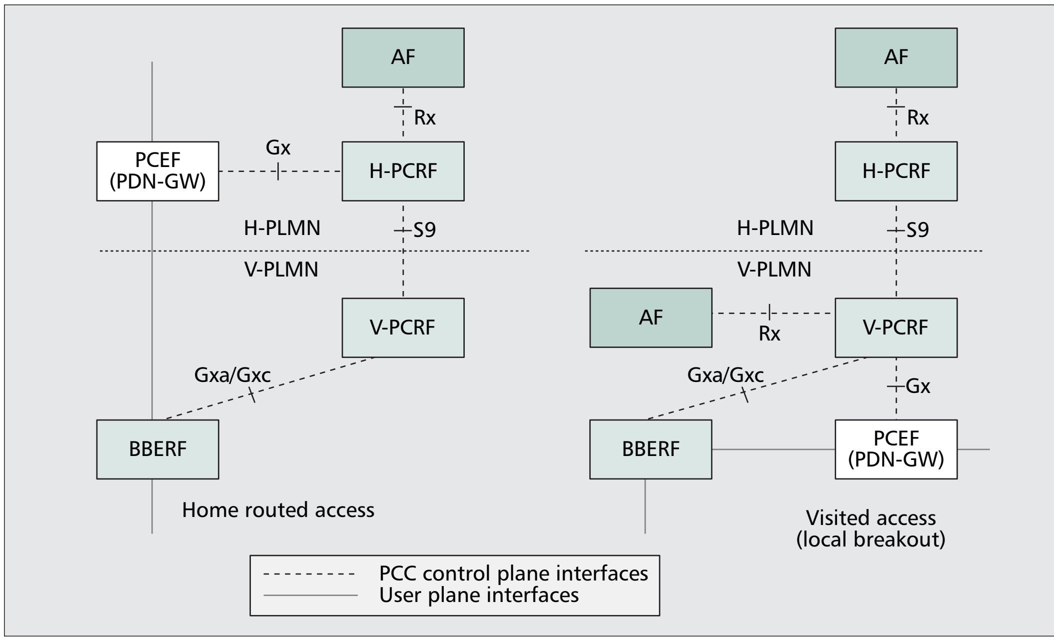
The Rx protocol (specified in [13]) is used over the Rx reference point, which is located between the PCRF and the AF. The main goal for the Rx protocol is transferring the session information from the AF to the PCRF and the bearer events from the PCRF to the AF.

The S9 protocol (specified in [14]) is used over the S9 reference point, which is located between the PCRF in the V-PLMN and the PCRF in the H-PLMN. The S9 was conceived to minimize the number of message exchanges across the roaming interface.

The Gy protocol (specified in [15]) is defined over the Gy reference point, which is located between the OCS and the PCEF. It is a 3GPP-specific variant of the IETF DCCA [11]. As such, it implements the DCCA state machines for session and event-based charging. However, not all functionality of DCCA is used for Gy, and some 3GPP-specific additions (AVPs) also are introduced. The main goal is credit management for online charging.

For offline charging, either the *GTP* or *Diameter base accounting* can be used to report usage of resources over the Gz reference point [16]. In addition file transfer protocol (FTP) can be used to transfer CDRs to the billing domain.

The PCC architecture provides both IMS and non-IMS services with the right framework to successfully control the media plane in all scenarios. With its support for multiple accesses, this is an important step toward a fixed-mobile convergence.



■ **Figure 5.** *PCC architecture for home routed and visited access roaming cases.*

Conclusions

The PCC architecture for EPS is an evolution of the PCC architecture defined in 3GPP Release 7. It is a consolidated architecture for access-agnostic policy control, and as such, it can be applied to a number of accesses such as E-UTRAN, UTRAN, GERAN, eHRPD, and WiMAX.

PCC was updated with the new functional entity, BBERF, and new reference points, Gxa and Gxc, to support new mobility protocols in the access network. This is a step toward service control continuity where a user can move between heterogeneous accesses and have policy and charging control seamlessly applied.

Furthermore, the introduction of a complete roaming model for PCC enables operators to have the same dynamic policy and charging control and provide the same access to services independently of whether a user is making this access through a gateway in their home or visited network.

The PCC architecture provides both IMS and non-IMS services with the right framework to successfully control the media plane in all scenarios. With its support for multiple accesses, this is an important step toward a fixed-mobile convergence.

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Biographies

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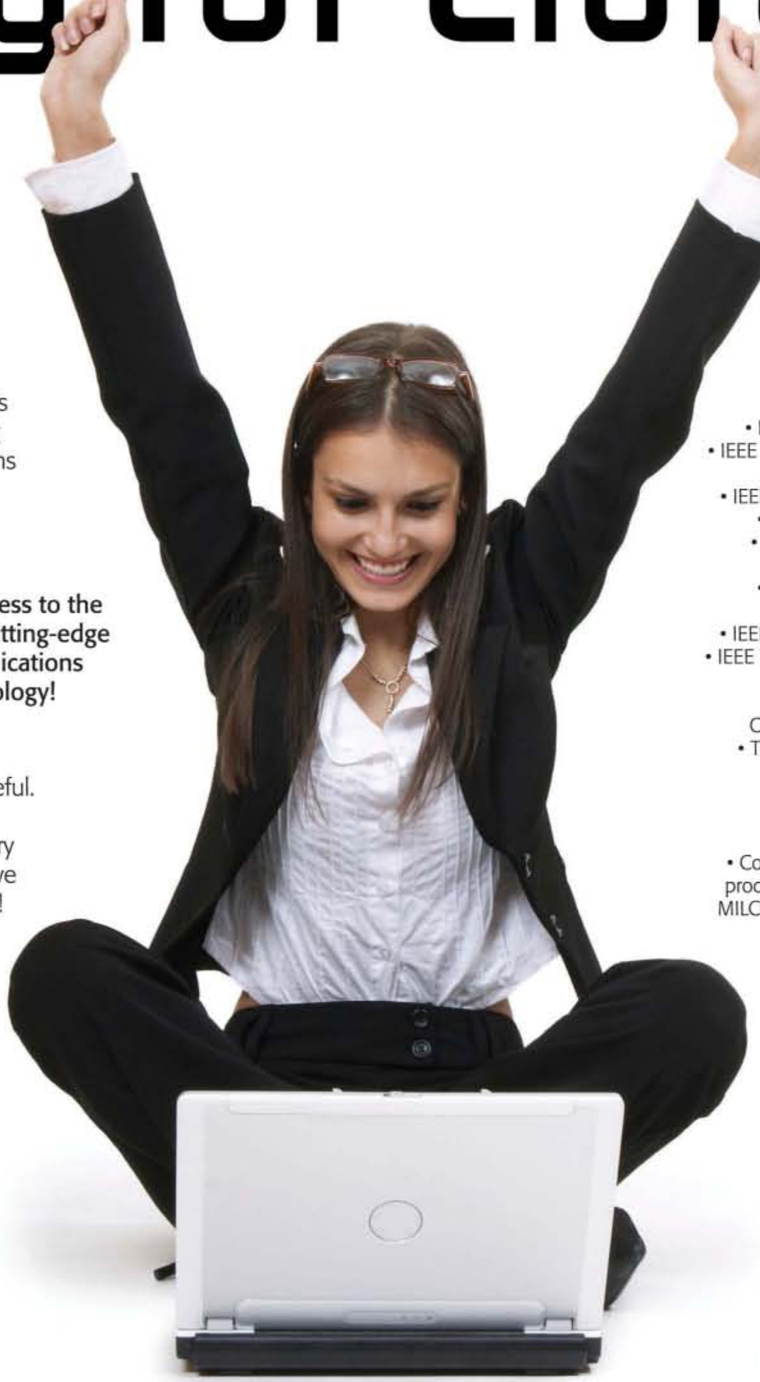

QoS Control in the 3GPP Evolved Packet System

Hannes Ekström, Ericsson

ABSTRACT

In this article we describe the QoS concept of the evolved packet system, which was standardized in 3GPP Release 8. The concept provides access network operators and service operators with a set of tools to enable service and subscriber differentiation. Such tools are becoming increasingly important as operators are moving from a single to a multi-service offering at the same time as both the number of mobile broadband subscribers and the traffic volume per subscriber is rapidly increasing.

The "bearer" is a central element of the EPS QoS concept and is the level of granularity for bearer-level QoS control. The network-initiated QoS control paradigm specified in EPS is a set of signaling procedures for managing bearers and controlling their QoS assigned by the network. The EPS QoS concept is class-based, where each bearer is assigned one and only one QoS class identifier by the network. The QCI is a scalar that is used within the access network as a reference to node-specific parameters that control packet forwarding treatment. This class-based approach, together with the network-initiated QoS control paradigm, gives network operators full control over the QoS provided for its offered services for each of its subscriber groups.

INTRODUCTION AND MOTIVATION

In recent years, cellular operators across the world have seen a rapid growth of mobile broadband subscribers. At the same time, the traffic volume per subscriber is also increasing rapidly; in particular, with the introduction of flat-rate tariffs and more advanced mobile devices. Operators are moving from a single-service offering in the packet-switched domain (Internet access) to a multi-service offering by adding new services that are also provided across the mobile broadband access. Examples of such services are multimedia telephony and mobile-TV. These services have different performance requirements, for example, in terms of required bit rates and packet delays. Solving these performance issues through over-provisioning typically is uneconomical due to the relatively high cost for transmission capacity in cellular access networks (including radio spectrum and backhaul from the base stations).

In addition, operators have started to provide subscriber differentiation, that is, differentiating the treatment received by different subscriber groups for the same service. These subscriber groups can be defined in any way that is suitable to the operator, for example, corporate versus private subscribers, post- versus pre-paid subscribers, and incoming roaming subscribers. Hence, there is a need to standardize simple and effective QoS mechanisms for multi-vendor mobile broadband deployments. Such QoS mechanisms should allow the access operator to enable service and subscriber differentiation and to control the performance experienced by the packet traffic of a certain service and subscriber group as depicted in Fig. 1.

This article presents the network-initiated and class-based concept for QoS control that was standardized for the evolved packet system (EPS). The basis and motivation for this concept was outlined in [1]. This article further describes the QoS mechanisms that are enabled in the EPS by the Third Generation Partnership Project (3GPP) Release 8 specifications.

The article is organized as follows. In the next section, we describe the components of the EPS QoS concept, and we then describee the QoS paradigms standardized for EPS, the network-initiated and terminal-initiated, and further describe the benefits of using the network-initiated paradigm. The following section provides an example of an end-to-end use case of providing service and subscriber differentiation using the EPS QoS concept. The final section concludes the article.

THE EPS QOS CONCEPT

In this section, we describe the details of the EPS bearer, its associated QoS parameters, and the EPS QoS mechanisms that are enabled by the standard.

THE BEARER

An EPS bearer — "bearer" for short — uniquely identifies packet flows that receive a common QoS treatment between the terminal and the gateway. A packet flow is defined by a five-tuple-based[1] packet filter, that is, the packet filters in the terminal (for uplink traffic) and the gateway (for downlink traffic) determine the packet flows associated with an EPS bearer (Fig. 2).

[1] The source and destination IP address, source and destination port number, and protocol ID typically are referred to as the IP five-tuple.

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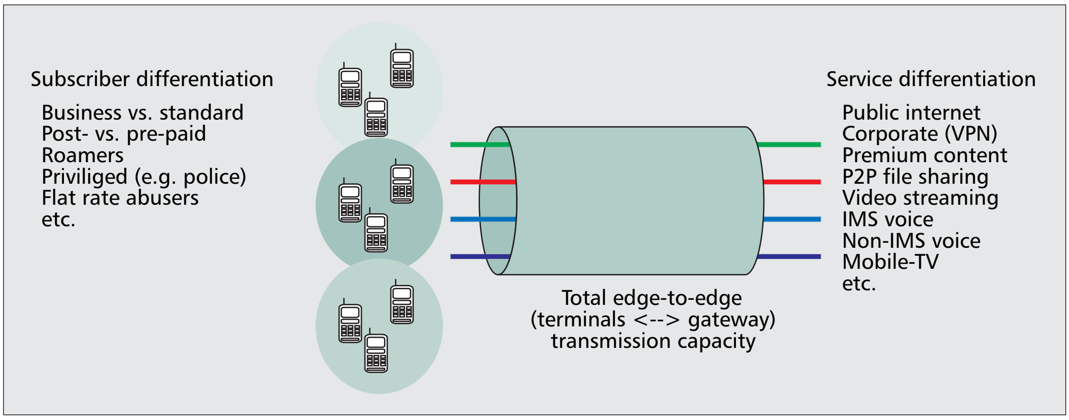
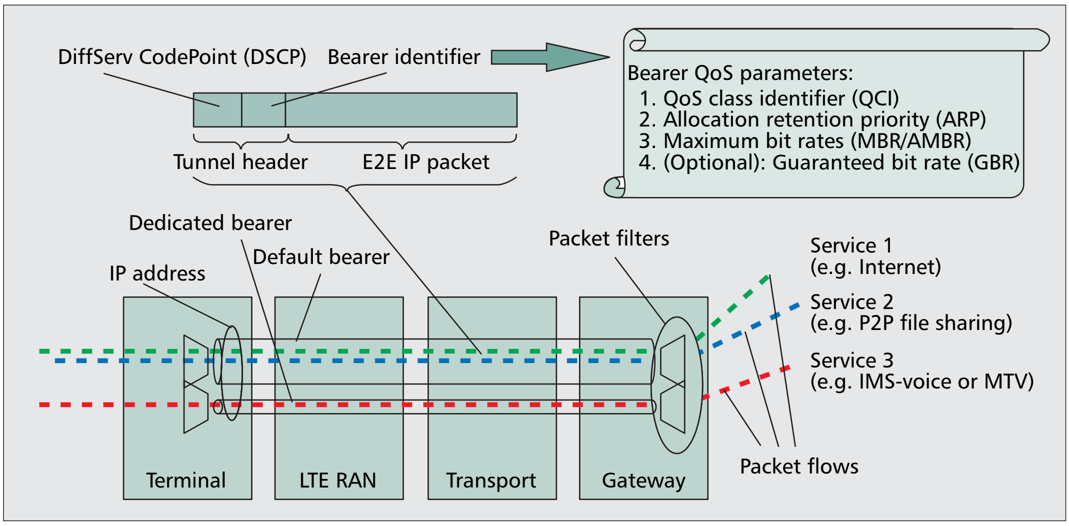


■ **Figure 1.** *Providing service and subscriber differentiation.*



■ **Figure 2.** *The bearer and its associated QoS parameters.*

A bearer is the level of granularity for bearer-level QoS control in the EPS. That is, all packet flows mapped to the same bearer receive the same packet-forwarding treatment (e.g., scheduling policy, queue management policy, rate-shaping policy, link-layer configuration, etc.). Providing different packet-forwarding treatment requires separate bearers.

One bearer exists per combination of QoS class and IP address of the terminal. The terminal can have multiple IP addresses, for example, in case it is connected to multiple access point names (APNs, one IP address per APN). The APN is a reference to the IP network to which the system connects the terminal. That is, the terminal can have two separate bearers associated with the same QoS class to two different APNs.

Each end-to-end IP packet entering the system is provided with a tunnel header on the different system interfaces. This tunnel header contains the bearer identifier so that the network nodes can associate the packet with the correct QoS parameters. In the transport network, the tunnel header further contains a diffserv code point (DSCP) value, as shown in Fig. 2.

The bearer is the basic enabler for traffic separation, that is, it provides differential treatment for traffic with differing QoS requirements. The concept of the bearer and the associated signaling procedures (see later in this section and in the next section) further enable the system to reserve system resources (e.g., processing and transmission capacity) *before* packet flows that are mapped to that bearer are admitted into the system. The latter is performed through an admission control function that operates on a per-bearer level.

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GBR vs. Non-GBR Bearers — Two types of bearers exist: guaranteed bit-rate (GBR) and non-guaranteed bit-rate (non-GBR) bearers. Provided that the traffic carried by a GBR bearer conforms to the value of the GBR QoS parameter associated with the bearer (discussed later in this section), the service(s) utilizing that GBR bearer can assume that congestion-related packet losses (i.e., packet losses caused by overflowing buffers) will not occur. This is realized by admission control functions that may reside in different network nodes (e.g., the long-term evolution [LTE] base station) and are executed at the point in time when a bearer becomes established or modified. A service utilizing a non-GBR bearer on the other hand, must be prepared to experience congestion-related packet loss.

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A GBR bearer typically is established "on demand" because it blocks transmission resources by reserving them in an admission control function. On the other hand, a non-GBR bearer can remain established for long periods of time because it does not block transmission resources.

A GBR bearer typically is established "on demand" because it blocks transmission resources by reserving them in an admission control function. On the other hand, a non-GBR bearer can remain established for long periods of time because it does not block transmission resources.

An operator would choose GBR bearers for services where the preferred user experience is "service blocking over service dropping," that is, block a service request rather than risk degraded performance of an already admitted service request. This is relevant in scenarios where it may not be possible to meet the demand for those services with the dimensioned capacity (e.g., in situations with extreme network load, like New Year's Eve). Whether a service is realized based on GBR or non-GBR bearers is, therefore, an operator policy decision that to a large extent depends on expected traffic load versus dimensioned capacity. Assuming sufficiently dimensioned capacity, any service, both real time and non-real time, can be realized based on non-GBR bearers.

Default vs. Dedicated Bearers — Orthogonal to being classified as GBR or non-GBR, a bearer is either a default or a dedicated bearer. The default bearer is the bearer that is set up when the terminal attaches to the network. One default bearer exists per terminal IP address, and it is kept for as long as the terminal retains that IP address. The default bearer provides the basic connectivity. Because a default bearer can remain established for long periods, the 3GPP specifications mandate that the default bearer is a non-GBR bearer. The QoS level of the default bearer is assigned based on subscription data.

To provide different QoS in the network to two different packet flows for the same IP address of a terminal, one or more dedicated bearers are required. The dedicated bearer can be either a non-GBR or a GBR bearer. The operator can control which packet flows are mapped onto the dedicated bearer, as well as the QoS level of the dedicated bearer through policies that are provisioned into the network policy and charging resource function (PCRF) [2]. In this article, we refer to that node simply as the policy controller. Figure 2 shows a terminal with a default and a dedicated bearer established to the same terminal IP address.

The policy controller defines specific packet flows to be mapped onto a dedicated bearer and typically defines them using an IP five-tuple. The values used in the five-tuple may have been signaled during application-layer signaling, for example, Session Initiation Protocol (SIP) [3] signaling in the case of an IP multimedia subsystem (IMS)-voice session. This is described in more detail later. The default bearer typically is not associated with any specific packet filters. Rather, it typically uses a "match all" packet filter, meaning that any packet that does not match any of the existing dedicated bearer packet filters is mapped onto the default bearer. This has the consequence that if a dedicated bearer is dropped by the system, the packet flows that originally were carried on that bearer are rerouted to the default bearer because in that case, that traffic only matches the "match all" packet filter.

For more details about the EPS bearer, see [4].

QoS Parameters

This section introduces the QoS parameters defined for EPS and explains their purpose and intended use. Additional information about the QoS parameters can be found in [4].

The EPS QoS concept is class-based, where each bearer is assigned one and only one QoS class identifier (QCI) by the network. The QCI is a scalar that is used within the access network as a reference to node-specific parameters that control packet-forwarding treatment (e.g., scheduling weights, admission thresholds, queue management thresholds, link-layer protocol configuration, etc.) and that were preconfigured by the operator owning the node (e.g., the LTE base station).

Each standardized QCI is associated with standardized QCI characteristics. The characteristics describe the packet-forwarding treatment that the bearer traffic receives edge-to-edge between the terminal and the gateway in terms of bearer type (GBR or non-GBR), priority, packet-delay budget, and packet-error-loss rate. See [2] for details. The standardized QCI characteristics are not signaled on an interface. They should be understood as guidelines for the preconfiguration of node-specific parameters for each QCI. The goal of standardizing a QCI with corresponding characteristics is to ensure that applications/services mapped to that QCI receive the same minimum level of QoS in multi-vendor network deployments and in the case of roaming.

Whereas the QCI specifies the user-plane treatment that the packets carried on the associated bearer should receive, the allocation and retention priority (ARP) specifies the control-plane treatment that the bearers receive. More specifically, the ARP enables the EPS system to differentiate the control-plane treatment related to setting up and retaining bearers. That is, the ARP is used to decide whether a bearer establishment or modification request can be accepted or must be rejected due to resource limitations. In addition, the ARP can be used to decide which bearer to release during exceptional resource limitations.

The maximum bit rate (MBR) and GBR are defined only for GBR bearers. These parameters define the MBR, that is, the bit rate that the traffic on the bearer may not exceed, and the GBR, that is, the bit rate that the network guarantees (e.g., through the use of an admission control function) it can sustain for that bearer. In 3GPP Release 8, the MBR must be set equal to the GBR, that is, the guaranteed rate is also the maximum rate that is allowed by the system. Allowing the setting of an MBR greater than a GBR is a candidate for future 3GPP releases. The main scenario that is targeted by such a setting is enhanced support for adaptive video applications where only a minimum video quality is guaranteed by the network.

The main purpose of the aggregate maximum bit rate (AMBR) is to enable operators to limit the total amount of bit rate consumed by a single subscriber. As such, it is not defined per bearer, but rather per group of non-GBR bearers. This parameter gives operators the tools to offer differentiated subscriptions that are widespread among operators employing fixed-line broadband technologies (such as digital subscriber line [DSL], e.g., 10 Mb/s or 100 Mb/s download bit rate).

The 3GPP has agreed on defining two different AMBR parameters:
- APN-AMBR: defined per subscriber and APN and known only to the gateway
- terminal-AMBR: defined per subscriber and know by both the gateway and the radio-access network (RAN)

Both of these AMBR values are defined for an aggregate of non-GBR bearers. Bit rate consumed by GBR bearers is not included in either of the AMBR parameters. It should be noted that each of these AMBR values are defined separately for uplink (UL) and downlink (DL) direction; that is, in total, four AMBR values are defined: UL APN-AMBR, DL APN-AMBR, UL terminal-AMBR, and DL terminal-AMBR.

One example of a scenario where this is useful is if an operator offers two separate services: corporate access for virtual private network (VPN) access into corporate networks and Internet access for general access to the Internet. The operator provides these services using separate APNs. The current AMBR definitions enable operators to differentiate the service level provided for each of these services. For example, a subscriber using both of these services could have a 100-Mb/s downlink limit on the corporate access service (i.e., DL APN-AMBR = 100-Mb/s for that subscriber on that APN) and a 5-Mb/s downlink limit on the Internet access service (i.e., DL APN-AMBR = 5 Mb/s for that subscriber on that APN).

A subscribed terminal-AMBR is associated with each subscription. This subscribed value should be considered to be an upper limit of the total bit rate that can be provided to that subscriber. The actual terminal-AMBR that is enforced by the network nodes is then calculated as the minimum of the subscribed terminal-AMBR and the sum of the APN-AMBR of all active APNs (i.e., APNs where the terminal has set up a default EPS bearer).

For a functional view of where the different AMBRs are enforced, see the second subsection in the following section.

QoS Mechanisms

The mechanisms that are used to provide QoS in the EPS system can be divided into control-plane signaling procedures and user-plane functions, each described in a separate subsection below.

Control-Plane Signaling Procedures — The policy controller in the network determines how each packet flow for each subscriber must be handled in terms of the QoS parameters to be associated with the handling of that packet flow. The policy controller can issue so-called policy and charging control (PCC) rules to the gateway, which in turn are used as a trigger to establish a new bearer or modify an existing bearer to handle a specific packet flow or to modify the handling of a packet flow. The packet flow is described by the UL/DL packet filters. The bearer-level request is forwarded to the LTE RAN and — if admitted by all involved network nodes — to the terminal. A high-level view of the signaling flow is shown in Fig. 3.

The next section provides a brief description and discussion of how QoS control is triggered in the absence of a policy controller (or equivalent node) in the network.

In addition to these dynamic control-plane signaling procedures, the operator must do a semi-static configuration of QoS functions directly in the network nodes through an operation and maintenance (O&M) system. An example of this is the semi-static configuration of node-internal functions (e.g., scheduling functions).

User-Plane Functions — The configuration of the network nodes (both through signaling procedures specified by 3GPP and through an O & M system) enables them to carry out user-plane QoS functions. These functions can be allocated to different nodes and classified into functions that operate per packet flow, per bearer (or group thereof), or per DSCP as illustrated in Fig. 4.

Packet-Flow-Level Functions — 3GPP specifies certain QoS functions that operate on a packet-flow level [2]. Using *packet-flow rate policing*, a gateway (or a physically separate network node) can identify certain packet flows using *(deep) packet inspection* techniques [5] and throttle the bit rate experienced by that particular packet flow, without modifying the bearer-level QoS parameters. This can be a useful QoS function for enabling an operator to limit the throughput experienced by a so-called "flat-rate abuser," that is, a subscriber with a flat-rate pricing plan that engages in extensive uploads or downloads (typically through peer-to-peer applications).

Bearer-Level Functions — The terminal and gateway perform uplink and downlink *packet filtering*, respectively, to map the packet flows onto the intended bearer. These are the underlying functions that provide the network with traffic separation functionality.

The gateway and the LTE RAN can implement functions related to admission control and pre-emption handling (i.e., congestion control) to enable these nodes to limit and control the load put on them. These functions can take the ARP value as an input to differentiate the treatment of different bearers in these functions. For example, the ARP can be used by the pre-emption function to determine which bearers to release from the system in situations when the system is overloaded or when resources must be freed up for other purposes (e.g., an incoming emergency call). In such situations, bearers associated with a low allocation and retention priority are released.

The gateway and the LTE RAN further implement functions related to *rate policing*. The goal of these functions is twofold: to protect the network from becoming overloaded and to ensure that the services are sending data in accordance with the specified maximum bit rates (AMBR and MBR). For the non-GBR bearers,

The main purpose of the aggregate maximum bit rate is to enable operators to limit the total amount of bit rate consumed by a siingle subscriber. As such, it is not defined per bearer, but rather per group of non-GBR bearers. This parameter gives operators the tools to offer differentiated subscriptions that are widespread among operators employing fixed-line broadband technologies like DSL..

To distribute RAN resources between the established bearers, the LTE RAN implements uplink and downlink scheduling functions. The scheduling function is to a large extent responsible for fulfilling the QoS characteristics associated with the different bearers.

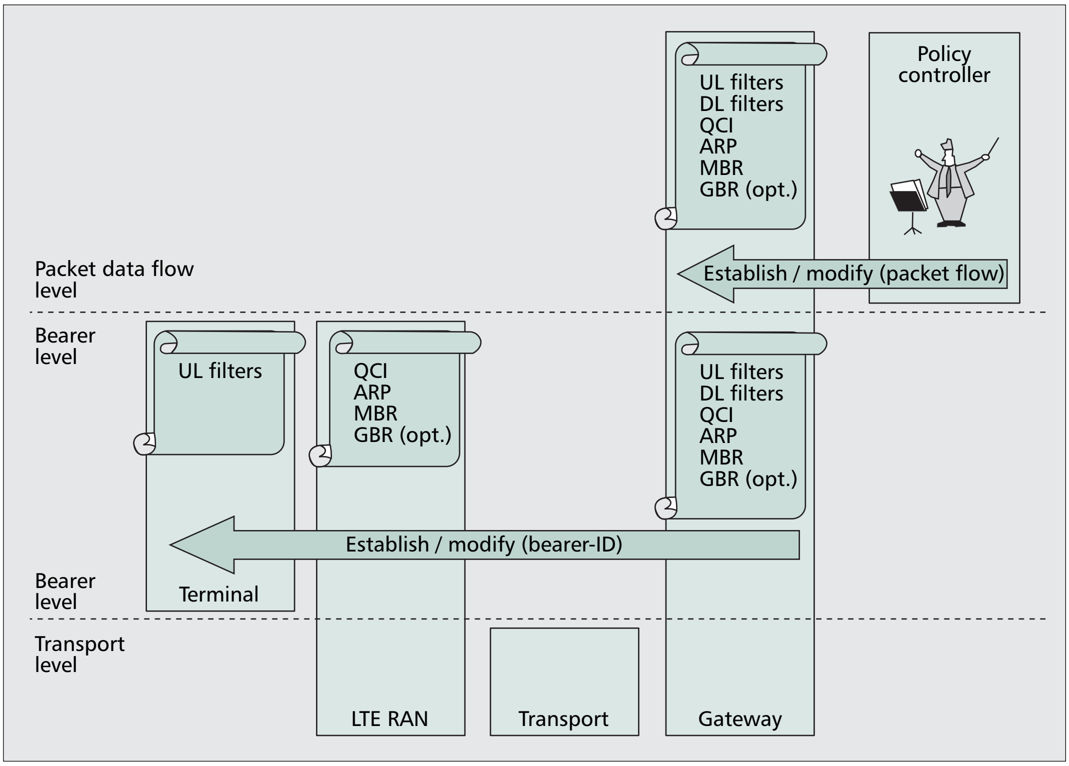

Figure 3. *A high-level view of EPS signaling procedures to control QoS functions.*

the gateway performs rate policing based on the APN-AMBR value(s) for both uplink and downlink traffic, whereas the LTE RAN performs rate policing based on the terminal-AMBR value for both uplink and downlink traffic. For GBR bearers, MBR policing is carried out in the gateway for downlink traffic and in the LTE RAN for uplink traffic.

To distribute RAN resources (radio and processing resources) between the established bearers, the LTE RAN implements *uplink and downlink scheduling* functions. The scheduling function is to a large extent responsible for fulfilling the QoS characteristics associated with the different bearers.

The LTE RAN is responsible for *configuring the L1 and L2 protocols* of the radio connection of the bearer in accordance with the QoS characteristics associated with the bearer. Among others, this includes configuring the error-control protocols (e.g., modulation, coding, and link-layer retransmissions) so that the QoS characteristics packet-delay budget and packet-error loss are fulfilled.

To allow for traffic separation in the transport network, the gateway and the LTE RAN implement a *QCI to DSCP mapping function*. The purpose of this function is to make a translation from bearer-level QoS (QCI) to transport-level QoS (DSCP). Using this function, packets on a bearer associated with a specific QCI are marked with a specific DSCP for forwarding in the transport network. The QCI to DSCP mapping is performed based on operator policies. These are configured into the network nodes through an O & M system. For downlink packets, the gateway performs this mapping while the LTE RAN performs it for uplink packets.

DSCP-Level Functions — Transport network nodes can implement *queue management* schemes and scheduling algorithms for uplink and downlink traffic. In the transport network, the bearer is not visible; and hence, these algorithms determine the traffic forwarding treatment of each individual packet, based on the DSCP value.

NETWORK- AND TERMINAL-INITIATED QOS CONTROL

There are two different paradigms that can be used to establish a dedicated bearer with a specific QoS in EPS. We refer to these as the terminal-initiated and network-initiated QoS control paradigms. The background and motivation for introducing a network-initiated paradigm into the 3GPP specifications was originally described in [1]. This paradigm subsequently was introduced into both the general packet-radio service (GPRS) 3GPP Release 7 specifications [6] (covering 2G/3G accesses), the EPS 3GPP Release 8 specifications (covering system architecture evolution [SAE]/LTE) [4], as well as into the evolved high-rate packet data (eHRPD) system specified in 3GPP2 [7]. The basic principles are shown in Fig. 5.

Using network-initiated QoS control, the network initiates the signal to set up a dedicated bearer with a specific QoS toward the terminal and the RAN. This is triggered by an application function (AF) or a deep-packet inspection (DPI) function [2, 5, 8], and the signal is carried over standardized interfaces (Rx and/or Gx). Using this paradigm, the client application can be left "access QoS unaware," meaning that it is not

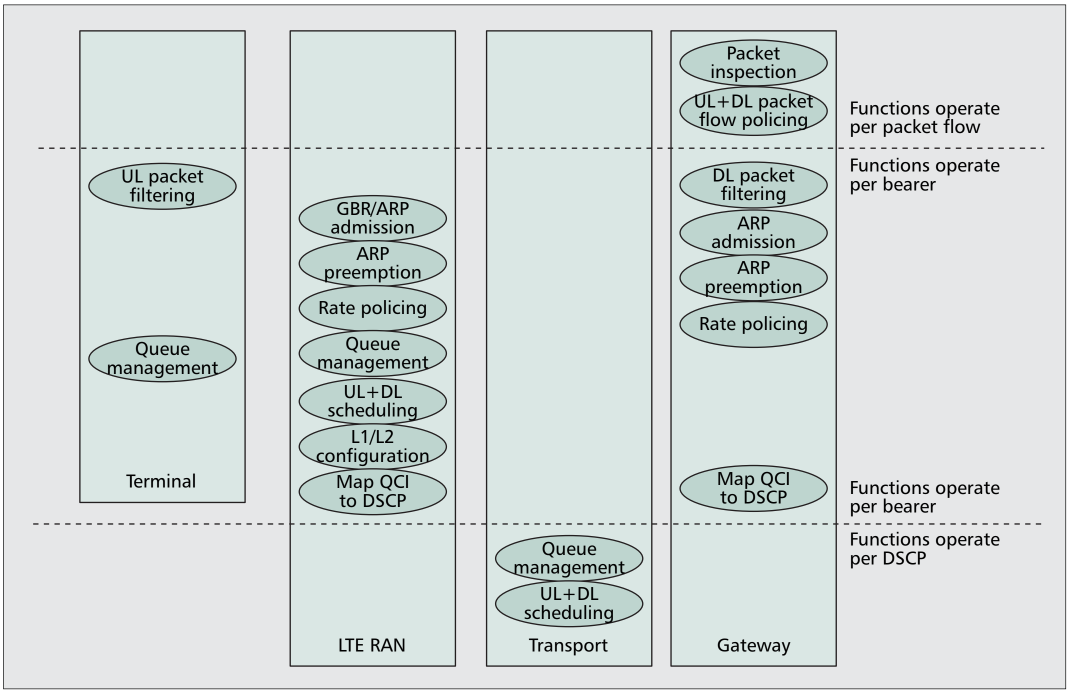

Figure 4. *Overview of user-plane QoS functions in EPS.*

The main motivation for specifying the network-initiated QoS-control paradigm is that services are typically provided by the access network operator. As such, it is natural that the access network and service owner assigns the QoS level per packet flow associated with a particular service.

required to be aware of the specifics of the QoS model of the access network. However, typically, the client application has access-agnostic knowledge of the QoS with which it wants to be provided. For some services, the QoS to be applied to the session can be negotiated with the network by means of application-layer signaling, such as SIP [3] and Real Time Streaming Protocol (RTSP) [9]. It is important to note, however, that there is no access-specific information in this signaling.

Using a terminal-initiated QoS control paradigm, it is the terminal that initiates the signal to set up a dedicated bearer with a specific QoS toward the network (which in turn triggers a command to the RAN). The trigger for this signal is carried over a terminal vendor-specific QoS application programming interface (API). This means that to specify the QoS information for the bearer, the client application must be "access QoS aware," meaning that it must be aware of the specifics of the QoS model of the access network. In this case, there is no policy controller communicating any QoS information to the network.

The main motivation for specifying the network-initiated QoS-control paradigm is that services (e.g., Internet access, mobile-TV, IMS voice) are typically provided by the access network operator, potentially through peering agreements with third-party service operators. As such, it is natural that the access network and service owner assigns the QoS level per packet flow associated with a particular service.

Network-initiated QoS control minimizes the terminal involvement in QoS and policy control. It has the following key advantages when compared to terminal-initiated QoS control:

- It can be used to provide QoS to access-agnostic client applications, such as applications that are downloaded and installed by the subscriber. This is not possible for terminal-initiated QoS control, which requires access-specific client applications that must be programmed toward a QoS API that is specific to the terminal vendor.
- As a direct result of the previous, network-initiated QoS control enables QoS to be provided in the "split-terminal" case where the client applications resides in a node (e.g., a laptop or set top box) that is physically separated from the terminal.
- It enables the deployment of more consistent exception-handling policies. One example of such a policy is the specification of the action to take when the request to initiate a service (or associated bearer) is rejected. There are numerous possible actions to take (e.g., give up, retry N times, or retry with a lower QoS-level). This consistency can be achieved more easily because using the network-initiated QoS control paradigm, the policies are centralized in the network rather than distributed in the numerous terminals from multiple vendors, as is the case using the terminal-initiated QoS control paradigm.

Due to the advantages listed above, we regard the network-initiated QoS control paradigm to be the most useful in cases where the operator controls the service (see [10] for a definition of operator-controlled services). For non-operator-controlled services, there is also the possibility to use the terminal-initiated QoS control paradigm. However, this possibility is not elaborated in this article.

End-to-End Use Case

In this section, we present an end-to-end use case in which a subscriber sets up an IMS voice call where the EPS QoS concept is used to real-

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When the media flow starts, the packet filters in the terminal and gateway map the IMS voice-over-IP (VoIP) packets onto the dedicated bearer, and the IMS VoIP service for this subscriber receives the QoS treatment defined by its service and subscriber differentiation policies.

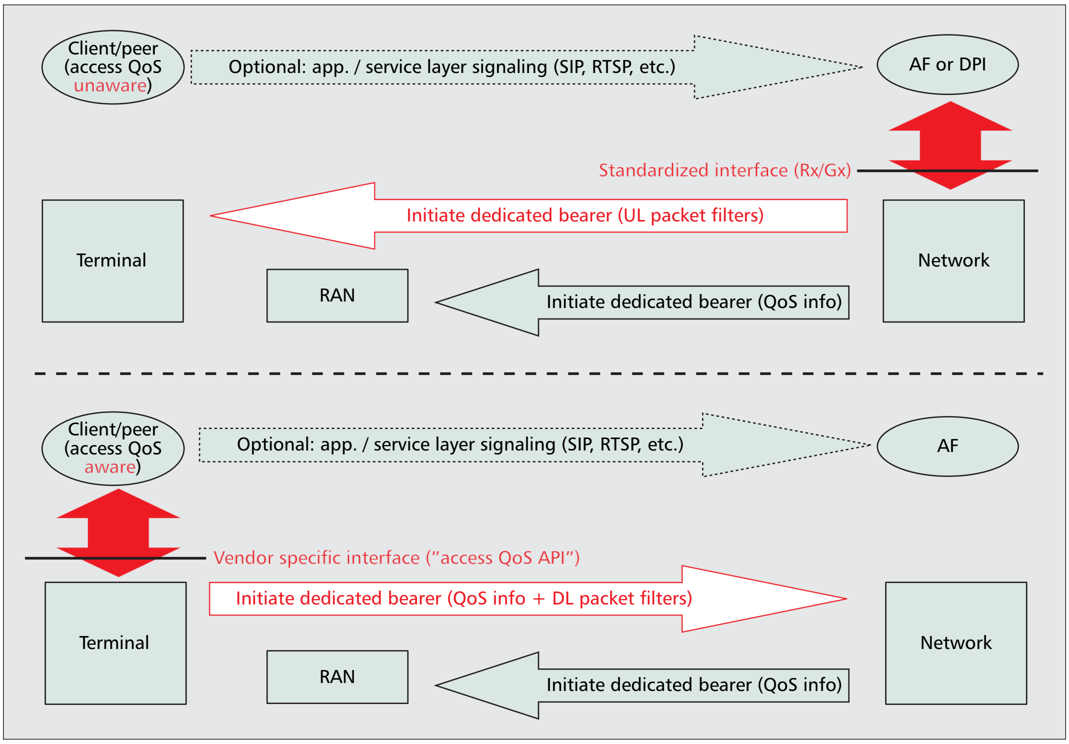

Figure 5. *Illustration of differences in the network-initiated (top) and terminal-initiated (bottom) QoS control paradigms.*

ize the QoS. In particular, the network-initiated QoS control paradigm as described in the previous section is applied. The intent of this use case is to illustrate how an operator can make use of the described mechanisms for providing subscriber and service differentiation.

In addition to the terminal, LTE RAN, transport network, and gateway, the system consists of a policy controller and an application function. The latter is a call-state control function (CSCF) in the IMS architecture [11]. The system and the signaling in the use case are illustrated in Fig. 6.

At the start of the use case, the subscriber is engaging in two services, Internet browsing and peer-to-peer file sharing. These services are both mapped onto the default bearer (shown in grey). The IMS application in the client was preconfigured with the IP address of the CSCF so that signaling messages are directed toward this node.

The subscriber places an IMS voice call, and the media flow is preceded by application-layer signaling using the SIP protocol to set up the call (1). This end-to-end signaling is intercepted by the CSCF in the network, and the messages reveal the IP five-tuple, as well as access-agnostic QoS information (see description in previous section) to the CSCF. One example of this information is codec rates. Based on this information, the CSCF detects a new packet flow and passes this information to the policy controller (2). The policy controller uses the information provided by the CSCF, operator-defined service policies, and subscription data when determining the appropriate QoS treatment that the packet flow should receive. This treatment is signaled to the gateway through the QoS parameters, and the packet flow is described in the defined uplink and downlink packet filters (3). At the reception of this information, the gateway initiates a dedicated bearer-establishment procedure in the control plane. This procedure sets up the dedicated bearer (4) and configures the user-plane QoS functions so that the packets carried on that bearer receive the appropriate QoS treatment. When the media flow starts, the packet filters in the terminal and gateway map the IMS voice-over-IP (VoIP) packets onto the dedicated bearer, and the IMS VoIP service for this subscriber receives the QoS treatment defined by its service and subscriber differentiation policies.

Summary and Conclusions

In this article, we described the QoS concept of the EPS that was standardized in the 3GPP Release 8 specifications. This concept is based on two fundamental principles:

- Network-initiated QoS control
- Class-based mapping of operator services to packet-forwarding treatment in user-plane nodes

The driver for introducing both principles was to simplify and enhance operator control over the provisioning of services and their associated QoS. This is achieved with the evolved QoS concept because it minimizes terminal involvement in QoS and policy control and centralizes the execution of operator policies in the network.

With the network-initiated QoS control paradigm, only the network can make the decision to establish or modify a bearer. This is a shift from the terminal-initiated QoS control paradigm in pre-Release 7, where this decision

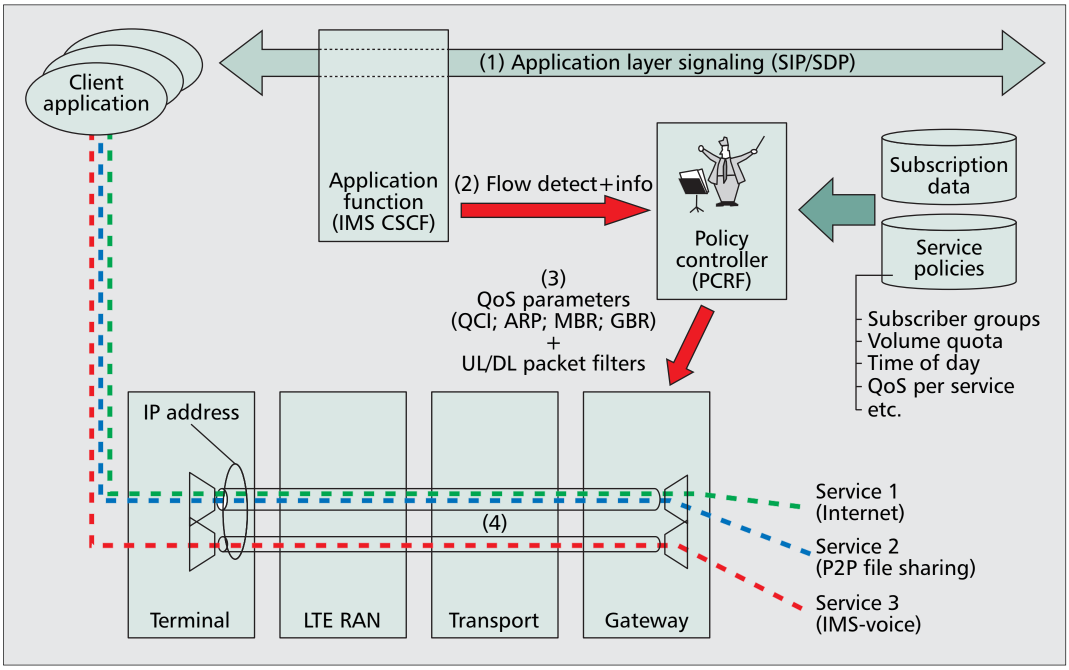

Figure 6. *Illustration of a use case where a subscriber sets up an IMS voice call with the EPS QoS concept.*

The goal of standardizing a QCI with corresponding characteristics is to ensure that applications or services that are mapped to that QCI receive the same minimum level of QoS in multivendor network deployments and in the case of roaming.

can be made only by the terminal. Network-initiated QoS control has a number of advantages: it can be used to provide QoS to access-agnostic client applications (such as those downloaded and installed by the subscriber), it enables QoS to be provided in the split-terminal case, where the client application resides in a node (e.g., a laptop or set top box) that is physically separated from the terminal, and finally, it enables the deployment of more consistent exception-handling policies.

This paradigm assumes that there is network intelligence, for example, application functions or deep-packet inspection functions, that can both identify the service that a subscriber is initiating and trigger QoS control (e.g., setting up a new bearer) when required.

The class-based approach to mapping of operator services to packet-forwarding treatment is a shift from the flow-based approach specified in 3GPP Release 7. With the class-based approach, an operator maps supported applications or services to a small set of QoS classes. Thereby, each packet flow is associated with one and only one QCI. The QCI is a scalar that is used as a reference to node-specific parameters that control packet-forwarding treatment and that are preconfigured by the operator owning the user-plane node. The 3GPP Release 8 specifications include nine standardized QCIs with corresponding standardized characteristics in terms of bearer type (GBR versus non-GBR), priority, packet delay, and packet-error-loss rate. The goal of standardizing a QCI with corresponding characteristics is to ensure that applications or services that are mapped to that QCI receive the same minimum level of QoS in multivendor network deployments and in the case of roaming.

The combination of these two fundamental principles, network-initiated QoS control and class-based mapping of services, provides access-network operators and service operators with a set of tools to enable service and subscriber differentiation. These tools are becoming increasingly important as operators are moving from a single to a multi-service offering at the same time as both the number of mobile broadband subscribers and the traffic volume per subscriber is increasing rapidly.

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Biographies

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Network Access Security in Next-Generation 3GPP Systems: A Tutorial

Sankaran C. B., Motorola

ABSTRACT

The 3GPP Release 8 Long Term Evolution/System Architecture Evolution marks the advancement of mobile cellular technology after UMTS-3G. The evolved packet system (EPS) architecture proposed in Release 8 introduces fundamental changes on top of UMTS in several design areas, including security. This article provides a tutorial overview of the proposed security mechanism in EPS. It first gives the background, a brief overview of the overall EPS architecture. It goes on to list the various requirements to be met for EPS security. A description of the EPS security architecture and detailed security procedures are given subsequently. The innovations that have been introduced in EPS, on top of UMTS, are highlighted all through the article. The article concludes by listing some open security issues at the moment.

INTRODUCTION

The Third Generation Partnership Program (3GPP) Long Term Evolution/System Architecture Evolution (LTE/SAE) system seeks to take mobile technology to the next level through the realization of higher bandwidths, better spectrum efficiency, wider coverage, and full interworking with other access/backend systems. LTE/SAE proposes to do all this using an all-IP architecture [1, 2] with well defined interworking with circuit-switched systems. In order to handle future requirements, the system is defined to work across multiple access networks (both 3GPP-defined and non-3GPP-defined). 3GPP access networks include E-UTRAN, UTRAN, and GERAN [1, 3]. Non-3GPP access networks include both trusted and non-trusted networks (CDMA-2000, WiFi, etc.) [4] In this heterogeneous framework, there is a greater risk of unlawful accessing and tampering with information that travels between the various entities. Hence, security functions assume paramount importance to ensure that the setup works as intended and is future proof [5].

The security mechanism in wireless systems has evolved right from the original analog systems through Global System for Mobile Communications (GSM) and Universal Mobile Telecommunications System (UMTS). In GSM the focus of security was largely on the radio path. In UMTS its scope was enhanced to include several network functionalities too. The ever-increasing focus on IP-based mechanisms meant more threats to security; hence, a more robust security architecture is needed in the evolved packet system (EPS). In EPS the 3G security framework has been enhanced to handle the more diverse nature of the architecture and increase robustness. These enhancements include adding security (both integrity protection and ciphering) on the non-access stratum (NAS) plane, additional layers of abstraction to protect important information like keys, security inter-working between 3GPP and non-3GPP networks, and so on. We look at these in detail in later sections.

EPS ARCHITECTURE

The EPS architecture and protocols are given in Fig. 1. The overall architecture has two distinct components: the access network and the core network. The access network is the evolved universal terrestrial radio access network (E-UTRAN), based on orthogonal frequency-division multiplexing (OFDM) and single-carrier frequency-division multiple access (SC-FDMA) technologies [6]. The core network is called the evolved packet core (EPC); it is different from the UMTS core network. E-UTRAN and EPC together constitute the EPS.

Some of the highlights of the LTE and EPC architectures are listed below. For the UMTS architecture, refer to [7], and for the EPS architecture, refer to [1, 2].

E-UTRAN

The E-UTRAN consists of just one node, the eNode-B, which has the functionality of the Node-B and radio network controller (RNC) in UTRAN. The emphasis is on self-configuration and self-optimization of eNodeBs. The eNode-B talks to the mobility management entity (MME) on the signaling plane and directly to the serving gateway (S-GW) on the data plane The eNode-B hosts the physical (PHY), medium access control (MAC), radio link control (RLC), PDCP, and RRC layers. The access stratum (AS) security mechanism consists of ciphering and integrity protection of RRC signaling messages and ciphering of user plane (UP) packets. The base AS security keys are generated using the NAS authentication and key agreement (AKA) proce-

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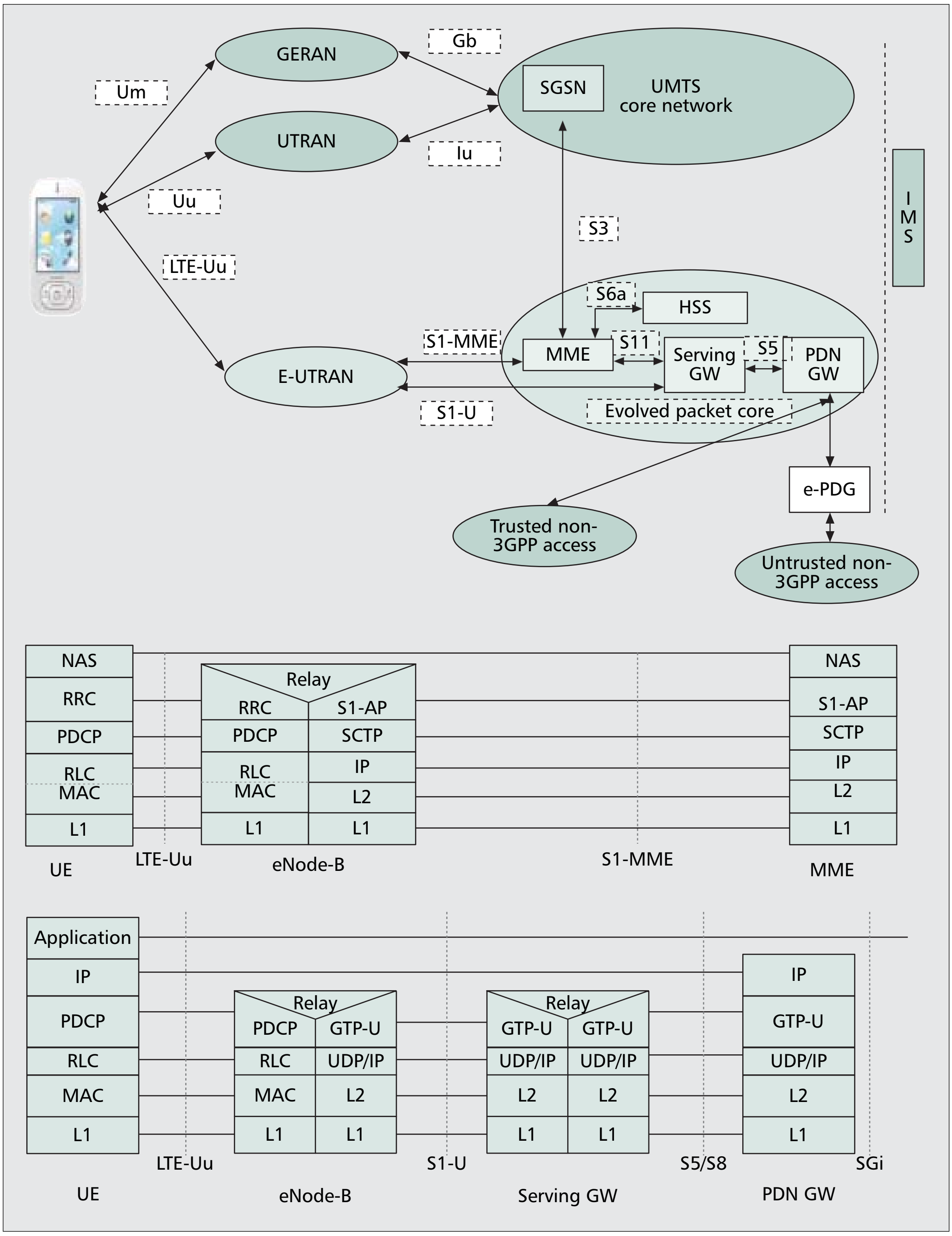

■ **Figure 1.** *EPS architecture, signaling, and user plane protocols.*

In GSM, the focus of security was largely on the radio path. In UMTS, its scope was enhanced to include several network functionalities too. The ever-increasing focus on IP based mechanisms meant more threats to security and hence, a more robust security architecture is needed in EPS.

dure. The configuration and activation of AS-level security is done through the AS security mode command (SMC) procedure. The lifetime of an AS security context is tied to the RRC connection; the keys are generated when the UE moves to connected mode and deleted when the UE goes to idle mode.

EPC

The EPC is an all-IP network (AIPN) and is fully packet-switched (PS). Services like voice, which are traditionally circuit-switched (CS), will be handled using the IP multimedia subsystem (IMS) network. Network complexity and latency are reduced as there are fewer hops in both the signaling and data planes. The EPC is designed to support non-3GPP access networks too. A relevant point is that the EPC supports mobile IP. To improve system robustness security, integrity protection, and ciphering have been added at the NAS level also, on top of the security that exists in the access network. Both integrity protection and ciphering will be applicable to all NAS signaling. This would ensure that even if there is a security breach at one level, the other one can ensure that there is no compromise in overall security. The other factor is that the EPC will be in a more controlled environment than

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Key requirement	Level I	Level II	Level III
Improve overall robustness over UMTS	Add NAS security, keys and identities are better protected, security association alive through idle	Support IPSec Add NAS security	Same as UMTS
User identity confidentiality	Usage of temporary identities	Support IPSec	Secure storage of IMSI
Mutual authentication of user and network	AKA procedure	N/A	N/A
Data confidentiality	Ciphering at the AS (both signaling and data) and NAS levels (signaling only)	Support IPSec NAS ciphering(signaling only)	N/A
Data integrity	Integrity protection at the AS and NAS levels	Support IPSec NAS integrity protection	N/A
Interworking with GERAN/UTRAN	The specs have defined the gracious handling of security in all the inter-RAT mobility scenarios	The specs have defined the gracious handling of security in all the inter-RAT mobility scenarios	N/A

Table 1. *Summary of LTE security functions and procedures.*

the access network; hence, security at this level becomes a must.

The major EPC elements are:

- **Mobility management entity**The MME is equivalent to the GERAN/UTRAN serving general packet radio service support node (SGSN), and hosts the NAS plane in EPS and interfaces with the home subscriber server (HSS, S6a) to enable the transfer of subscription and authentication data for authenticating/authorizing user access. It terminates the NAS level security.
- **Serving gateway and packet data network gateway**: The S-GW handles the IP data from eNode-Bs directly and terminates the interface towards E-UTRAN. The packet data network gateway (PDN-GW) provides the interface to the PDN.

Major Security Threats and Requirements in LTE/SAE

Some of the key security threats in EPS are:

- Illegal access and usage of the user's and mobile equipment's (ME')s identities — to access network services
- Tracking the user based on the user equipment's (UE's) temporary identity, signaling messages, and so on
- Illegal access and usage of the keys used in security procedures to access network services
- Malicious modification of UE parameters (e.g., failure timers, retry timers) to lock out the phone from normal services either permanently or for an extended period of time
- Willful tampering with the system information broadcast by the E-UTRAN
- Eavesdropping and illegal modification of IP packet contents
- Denial of service to the UE
- Attacks on the integrity of data (signaling or user traffic) by replaying

The key requirements could be summarized as:

- **Improved overall security robustness over UMTS** — to take care of the added/new functionality and the use cases thereof, and work in a secure environment
- **User identity confidentiality** — to ensure that any illegal identification and tracking of any user is not possible
- **Mutual authentication of the user and network** — to ensure that both sides are sure they are communicating with the correct entity, authorized to make that transaction
- **Data confidentiality** — to ensure that any eavesdropping of exchanged data is not possible
- **Data integrity** — to ensure that data received by any entity cannot be tampered with
- **Interworking with GERAN/UTRAN** — to ensure that inter-radio access technology (RAT) procedures work as designed without allowing any security weakness of the other access technologies to compromise LTE/SAE security
- **Replay protection** — to ensure that an intruder is not able to replay control messages already transmitted
- **Allowing/requiring dynamic setup** of all respected security association as much as possible

These are generic high-level requirements that are applicable across multiple entities and interfaces in the LTE/SAE architecture. The specific implementation of the same requirement could be different across these different entities. We look at the detailed architecture below, where we see how these requirements are met.

Security Architecture for EPS

There are four levels of security defined in the specifications. These are:

- **Network access security (level I)**: These are security features that protect the radio link

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and provide users with secure access to the EPC and the backend networks. This level has security mechanisms between the USIM, ME, E-UTRAN, and elements in the EPC (both serving and home networks). The integrity protection and ciphering defined in EPC are examples.
- **Network domain security (level II)**: These are security features that protect the wireline networks and enable them to exchange data in a safe manner. This could be, for example, the IPSec used to protect the S1 control plane.
- **User domain security (level III)**: Here the scope is between the USIM and the ME. It would include the mutual authentication of the USIM and the ME before they can access each other, using a secret PIN.
- **Application domain security (level IV)**: The security features that enable applications in the UE and the backend network to exchange information in a secure manner. For example, the IMS architecture provides the framework for this level of security for voice over IP (VoIP).

In Table 1 the different security features and procedures that implement these requirements at each level are summarized. EPS uses the following procedures and mechanisms as effective counter-measures at levels I, II, and III:
- Usage of temporary user identities: Mandating AKA-based mutual authentication during initial attach and as needed before allowing a user to access the network
- Use encryption/ciphering to safeguard the content of user data and signaling messages
- Guarantee signaling messages' integrity using an applicable integrity protection mechanism
- Dynamic key distribution and management in a secure manner using a keying hierarchy based on a preshared master key
- For IP transport (within the network), IPSec with IKE is used for effective protection

Level IV security is out of scope of this article.

Key Management

The various keys play a critical role in the working of the overall security mechanism. Their lifetimes, scope, hierarchy, and properties are clearly defined in the 3GPP Release 8 specifications [8], right from the master key down to the various temporary keys. The E-UTRAN keys are cryptographically separated from the EPC keys used, making it impossible to figure out one from the other. Figure 2 shows the keys' hierarchy and the levels where they are relevant.

The various keys are derived using the Key Derivation Function (KDF) interface defined in [8]. While the inputs are different for the various keys, they are concatenated into a common format S, which is then input to the respective algorithm:
- **KeNB** is a key derived by UE and MME from KASME when the UE goes to connected state or by UE and target eNode-B during eNode-B handover.
- **KNASint** is a key used to protect NAS traffic with a particular integrity algorithm. It is derived by the UE and MME from KASME and an identifier for the integrity algorithm, using the KDF.
- **KNASenc** is a key used to protect NAS traffic with a particular encryption algorithm. It is derived by UE and MME from KASME and an identifier for the encryption algorithm, using the KDF.
- **KUPenc** is a key used to protect UP traffic with a particular encryption algorithm. It is derived by UE and eNode-B from KeNB, and an identifier for the encryption algorithm, using the KDF.
- **KRRCint** is a key to protect RRC traffic with a particular integrity algorithm. It is derived by UE and eNode-B from KeNB and an identifier for the integrity algorithm, using the KDF.
- **KRRCenc** is a key used to protect RRC traffic with a particular encryption algorithm. It is derived by UE and eNB from KeNB and an identifier for the encryption algorithm, using the KDF.

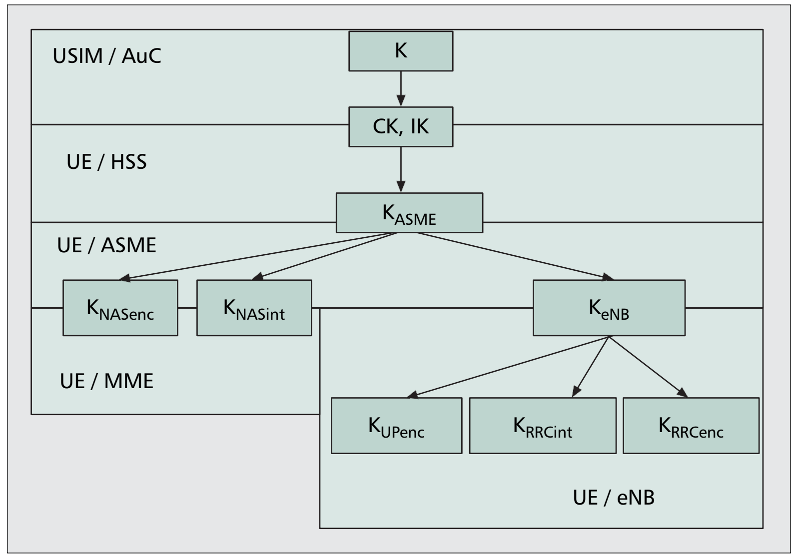

Figure 2. *EPS key hierarchy.*

Security Robustness Improvement over UMTS

The EPC is designed to handle multiple access and backend networks. This means use cases such as eNode-B, where the access network could be unreliable. Also, since all traffic would be wholly IP-based, there is an increased risk of security breach. The major improvement over UMTS is the addition of security functions at the NAS level (between the UE and the MME), on top of the existing ones at the AS level. A separate security sublayer is introduced for doing this and is positioned in between E-MM and RRC in the protocol stack. This sublayer would cipher and integrity protect NAS signaling messages. This would mean that all the NAS signaling (with the exception of a few 3GPP defined messages) would be ciphered and integrity protected twice — once in the NAS security sublayer and once within AS. This adds to the overall

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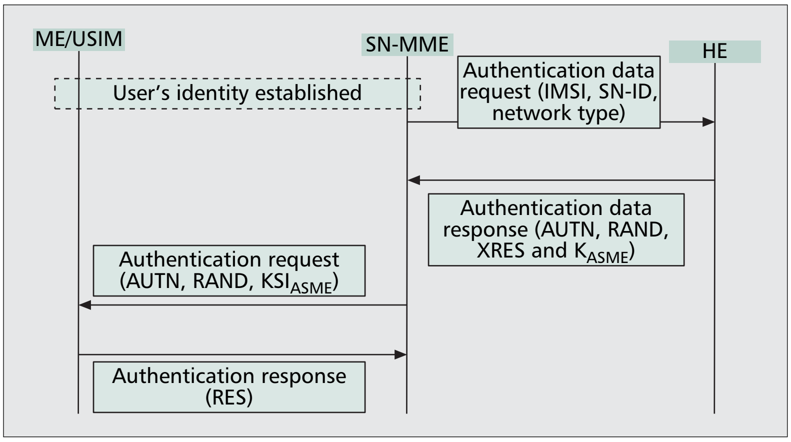


Figure 3. *Message flow for EPS AKA.*

robustness of the architecture; even if one fails, there will be protection from the other.

On the network side, the protection of IP-based internetwork interfaces in EPC and E-UTRAN shall be done using IPSec. This is done for both the signaling and user data planes.

One more improvement is that during the AKA procedure, the ciphering and integrity keys (Ck and Ik) are computed by the HSS in the user's home public land mobile network (HPLMN) when the serving network (SN) queries for the same. While in UMTS these keys are actually communicated back to the SN, it is not so in EPC. Instead, another key, KASME, is computed by the HSS and sent back to the SN. The advantage of KASME is that it is bound to the MS identity and the identity of the SN. Another advantage is that KASME is returned to the SN only after the UE authentication response is validated by the HSS. The NAS security context has a longer lifetime than the AS security context. It can also stay alive when the UE goes to idle.

User Identity Confidentiality

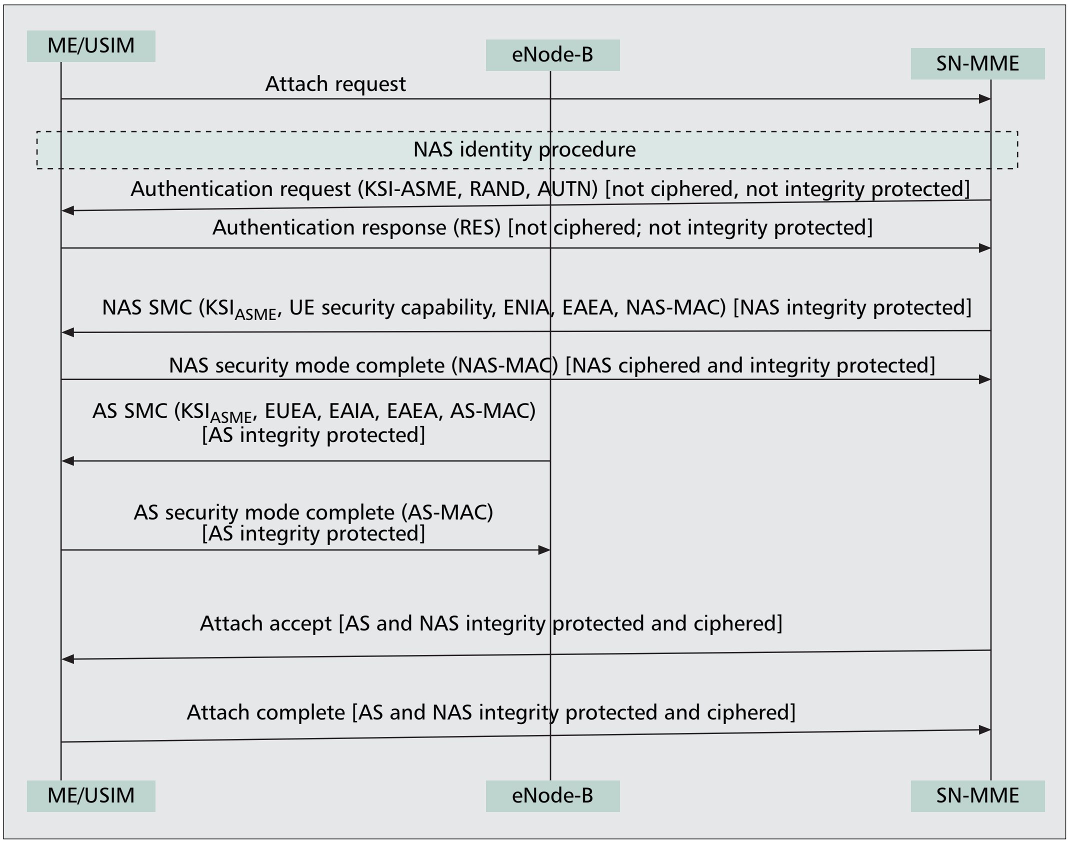
The identity of the user is to be protected to prevent unlawful reading. Threats include tracking and profiling the user's movements, getting information on the network's identity (from the international mobile subscriber identity, IMSI). There are defined countermeasures to prevent these threats. Foremost among these are the usage of temporary identities. Temporary identities are assigned and used wherever possible to avoid unnecessary exchange of permanent identities between entities. Some temporary identities are M-temporary mobile subscriber identity (M-TMSI), which is used to identify the UE within the MME, the S-TMSI, which is constructed from the MME code and the M-TMSI, used for paging the UE, and the globally unique temporary UE identity (GUTI), allocated by the MME with two components, one uniquely identifying the MME that allocated the GUTI and the other uniquely identifying the UE within that MME. GUTI is used to support subscriber identity confidentiality, and, in the shortened S-TMSI form, to enable more efficient radio signaling procedures (e.g., paging and service requests). Apart from the temporary identities, the permanent identities (IMSI and IMEI) shall be stored securely. There is one allowed security breach: if the MME queries for the UE's IMSI in the Identity Request message, the UE should send it, even if security is not configured.

Mutual Authentication of User and Network

The AKA procedure ensures that the serving network (SN) authenticates the user's identity (in the USIM) and the UE validates the signature of the network provided in the authentication token (AUTN). Apart from the authentication itself, this procedure is used to by the HN and UE to generate the Ck and Ik from the same material by the same functions; the KASME is also computed as part of this procedure. The KASME key is subsequently used to derive different session keys for ciphering and integrity protection for AS and NAS.

During the registration, when the UE's identity is established with the MME, it sends a request to the home environment (HE) querying the authentication vector for a specific SN-ID and IMSI. The HE responds with an authentication vector (the use of multiple vectors is part of UMTS and is discouraged for LTE/SAE because there is no need for it with the current keys hierarchy). Each vector has AUTN, RAND (a random value), XRES (which is calculated by the HE using a predefined authentication algorithm using AUTN, RAND, and a master key K unique to each IMSI), and KASME, which is computed from Ck and Ik (these two keys remain in the HSS and are never sent to the MME). There is an additional level of abstraction, where KASME, Ck, and Ik are stored in a key set and identified by a key set identifier (KSIASME). The KSIASME is sent by the MME to the UE in the Authentication Request message along with the AUTN and RAND. The USIM computes the KASME, Ck, Ik, and RES, stores KASME along with the received KSI-ASME, and sends back the calculated RES in the Authentication Response message. The MME compares the RES with the XRES it got earlier and completes the procedure if they are found to be the same. This enables the MME to start ciphering and integrity protection at the next establishment of an NAS signaling connection without executing a new authentication or SMC procedure. It is to be noted that the UE derives the KASME using the SN-ID as a parameter; hence, successful use of the keys derived from KASME implicitly authenticates the network's identity.

Some of the concepts used during the authentication procedure have been described here. The authentication vector has to be *fresh* (i.e., not used before). This is ensured by the sequence numbers exchanged in the messages that serve as an input to the ciphering and integrity algorithms. Also, the algorithms used in the HE and USIM to compute the authentication vectors are largely *one-way* mathematical functions, where an output is gotten given a set of inputs, using a defined algorithm. However, to get back the



■ **Figure 4.** *Establishment of NAS and AS security contexts during initial attach.*

While the Ck and Ik are assigned in the AKA procedure, the ciphering and integrity algorithms and other inputs to these algorithms are communicated in the Security Mode Command (SMC) procedure. There are 2 SMC procedures defined, one at the NAS level and the other one at the AS level.

inputs using the output is extremely complex. This helps in countering security threats. All the different keys used in the AKA and other procedures are interlinked with a defined hierarchy with the single source being the master key K, which is unique to a user and is stored in a secure manner in both the USIM and the HN. Please refer to [1, 8] for more details.

Data Confidentiality: Ciphering and Integrity Protection

Once the UE and network have completed their mutual authentication, they can start communicating in a secure manner, using ciphering and integrity protection. For a detailed description, please refer to [1, 8].

Ciphering is an encryption methodology consisting of adding a random-looking mask bit by bit to the plaintext data to encrypt it. The ciphered message is unintelligible to any third party as the inputs to the algorithm are confidential and protected. On the receiving side the same mask, when added again, retrieves the original plaintext data. This ensures the confidentiality of the data communicated over the radio link. Ciphering is applied on signaling messages (in both NAS and AS) and user plane data (at AS).

Integrity protection ensures that the data received at an entity is what was sent by the sender. In other words, it ensures that the data has not been tampered with midway. This is to ensure protection at the individual message level. Integrity protection is applied to all signaling messages at both the NAS and AS levels. Integrity protection is not applied to user plane data because it would become too much of an overhead at the packet level, impacting the data rates. The basic methodology is computing an integrity tag, which is appended to the message being sent; the same integrity tag is generated on the receiving side too; the message is accepted only when the tags match. Any change in the input parameters (inputs include the original signaling message as well) to the algorithm affect the output in an unpredictable manner; hence, it protects the message from tampering.

While the Ck and Ik are assigned in the AKA procedure, the ciphering and integrity algorithms and other inputs to these algorithms are communicated in the security mode command (SMC) procedure. There are two SMC procedures defined, one at the NAS level and the other one as the AS level.

NAS Security Mechanism

The NAS SMC procedure is triggered by the MME immediately after the AKA — the SMC message contains the replayed security capabilities of the UE, the selected NAS algorithms, and the KSIASME for identifying KASME. This message is integrity protected with an NAS integrity key based on KASME indicated by the KSIASME. The NAS security mode complete message from UE to MME is integrity protected and ciphered with the algorithms indicated by the MME NAS uplink. Downlink ciphering at the MME starts after sending the

3GPP-Rel 8 defines the handshaking between the UE and the different network elements to handle security during mobility within EPS as well as between EPS and UTRAN/GERAN/ non-3GPP networks.

NAS security mode command message. NAS uplink and downlink ciphering at the UE starts after receiving the NAS security mode command message.

Once the AKA and SMC procedures are completed, the NAS security context is said to be created and will be applicable during the time the UE is registered. Depending on what triggers a subsequent detach procedure, the context could be maintained or deleted. While the NAS security context exists, all NAS messages shall be integrity protected and ciphered The inputs for the integrity and ciphering algorithms would be the KNASint/KNASenc, NAS COUNT, BEARER identity, and DIRECTION bit.

AS Security Mechanism

The AS SMC procedure is triggered by the eNode-B by sending the AS SMC to the UE; the UE replies with the AS security mode complete message. The SMC contains the selected AS algorithms (for ciphering and integrity protection) and the KSIASME. It will be integrity protected using the KRRCint tied to the KSIASME that comes in the SMC.The AS security mode complete message shall also be integrity protected with the selected RRC algorithm indicated in the AS security mode command message and RRC integrity key based on the equivalent KASME.

RRC and user plane ciphering at the eNode-B shall start after receiving the AS security mode complete message. RRC and UP ciphering at the UE shall start after sending the AS security mode complete message. The input parameters for the ciphering and integrity protection algorithm would be the KRRCenc/KRRCint/KUPenc, the PDCP count, the bearer ID, and a DIRECTION bit.

Refer to Fig. 4 for the case of establishing the AS and NAS security contexts during initial attach.

EPS Security and Mobility

3GPP-Rel 8 defines the handshaking between the UE and the different network elements to handle security during mobility within the EPS as well as between the EPS and UTRAN/GERAN/non-3GPP networks [5, 8]. The major challenge in security during mobility is how the security algorithms, KDFs, and keys are handled. A UE could move either with or without a security context active. Here, we look at the mobility of the UE with the security context active. Some common points are:

- The UE includes its security capabilities, listing the algorithms it can support in E-UTRAN/UTRAN/GERAN, Attach Request (in all the RATs), TAU Request (in E-UTRAN), and RAU Request (in UTRAN/GERAN) in Rel-8.
- The MME and eNode-Bs are configured by the operator with a priority list of algorithms and KDF to use.
- When the AS security context is established in the eNode-B, the MME shall send the UE's security capabilities to the eNode-B, containing the algorithms supported by the UE.

Intra E-UTRAN Mobility in Connected Mode — At the time of a handover (HO), the source eNode-B forwards the UE's security capabilities to the destination eNode-B. The destination eNode-B selects an algorithm to use (based on the priority list), and lets the UE and MME know about it. If the MME also changes during the HO, the source MME shares the UE security capability with the target MME, and the target MME selects a set of algorithms and KDF (based on the priority list) and assigns them to the UE in the TAU Accept message(if the new values are different from the old ones).

Intra E-UTRAN Mobility in Idle Mode — The UE and network use the RAU/TAU signaling to post the mobility to synchronize on the algorithms to use. If there is data to send at the time of the UE movement, the AS keys need to be recomputed. The KeNode-B is computed using the KASME and NAS count; and from the new KeNode-B, the KRRCenc, KRRCint, and KUPenc are derived. An AKA could be either run or not as part of the TAU; key recomputation is different in each case.

Inter-RAT Mobility in Connected Mode — At the time of the HO, the UE's target RAT security capabilities are shared between the MME and the SGSN, and these are further transferred to the eNode-B or RNC. The selected algorithms (in the target RAT) are then conveyed to the UE in the HO command. The keys are recomputed on both the UE and network side at the time of the HO.

Inter-RAT Mobility in Idle Mode — There are two cases here: the UE has a cached security context in the target RAT, or it does not. When the cached context exists, either the old keys are accepted between the UE and the network or a new AKA run is done. In the other case (called the mapped security context), the UE sends KSI or KSIASME or the source RAT in the TAU/RAU request; there is additional signaling between the network elements to set up the security context.

Conclusion

This article has sought to give an introduction to the ideas and concepts that have been used in EPS security. While most of the threats and requirements have been identified, much more remains to be done given the wholly heterogeneous nature of the EPS and the immense possibilities it throws up in applications, network configuration, and so on. Some specific issues being addressed now are:

- Whether or not a single set of high-level security requirements for all types of eNode-B (i.e., femto, pico, and macro) is enough is being discussed. There is a view that the different deployment environments should dictate the requirements.
- Per user activation of UP ciphering is under discussion.
- Negotiation of KDF, used to derive the KASME, and the handling during mobility

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(e.g., handover between two eNode-Bs with different KDFs) is under study.
- Key handling during handover is another major area of study.
- It is not yet fully defined which messages should be security protected and which need not be (under certain scenarios). The scenarios and exceptions are being discussed and worked on now.

We conclude that security is one area in 3GPP that needs focus in the coming years to ensure that there is no compromise in security while realizing the potential of the next generation of mobile systems.

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Biography

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Multisite Field Trial for LTE and Advanced Concepts

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ABSTRACT

The 3GPP LTE standard is stable now in its first release (Release 8), and the question is how good its performance is in real-world scenarios. LTE is also a good base for further innovations, but it must be proven that they offer performance advantages for the price of their complexity. This article evaluates the performance of LTE Release 8 as a baseline and advanced concepts currently in discussion such as cooperative MIMO based on system-level simulations, and measurements in the laboratory and a multisite field testbed within the EASY-C project.

INTRODUCTION

The mobile Internet has finally arrived with the worldwide deployment of high-speed packet access (HSPA) networks and broad availability of third-generation (3G) terminals, mobile broadband USB sticks, and, increasingly, notebooks with integrated HSPA modules. With flat-rate data tariffs, the usage of mobile Internet has skyrocketed in 2008. Third-generation technology was developed more than a decade ago, and the uptake after launch was below expectations in many cases. There are various reasons for that, including initial lack of handset availability and initial technology performance below predictions.

The Next Generation Mobile Networks (NGMN) Alliance has set out requirements for future mobile networks [1], and the Third Generation Partnership Program (3GPP) is addressing them with the development of long-term evolution (LTE). Among the requirements for LTE are increased average and peak data rates, reduced latency, spectrum flexibility addressing bandwidths of up to 20 MHz, and, last but not least, reduced cost of ownership. The targets in NGMN and LTE are set challenges to ensure a significant performance step from HSPA to a new technology generation.

The performance of LTE meets the essential NGMN requirements, but not the preferred requirements in important key performance indicators (KPIs) like spectral efficiency and cell-edge throughput. Therefore, development of LTE technology is continuing beyond Release 8 to address operator requirements as well as those of the International Telecommunications Union (ITU) for future technologies in the newly identified spectrum. 3GPP has initiated the "LTE-Advanced" study item and defined requirements in [2].

The research project Enablers for Ambient Services and Systems — Part C Wide Area Coverage (EASY-C) is developing technologies for future wireless systems such as LTE-Advanced. The special feature of EASY-C is that research ideas are tested in research field testbeds at the system level. In EASY-C, 16 partners work together across the value chain, including academic institutions, mobile operators, network infrastructure, antenna, and test equipment providers, terminal chipset vendors and semiconductor companies, and network planning specialists.

OVERVIEW OF LTE RELEASE 8

The radio interface of 3GPP LTE/SAE Release 8 uses orthogonal frequency-division multiple access (OFDMA) with cyclic prefix in the downlink and single-carrier frequency-division multiple access (SC-FDMA) with cyclic prefix in the uplink. The physical layer of LTE is defined in a bandwidth agnostic way and supports various system bandwidths up to 20 MHz. Radio resources are subdivided into physical resource blocks (PRBs) consisting of 12 subcarriers with 15 kHz spacing and a time duration of 1 ms. PRBs are dynamically allocated to users in order to realize multi-user diversity gain in both time and frequency domains, leveraging adaptive modulation and coding (AMC) with hybrid automatic repeat request (HARQ).

To meet the performance requirements [3], LTE Release 8 relies on multi-antenna-based multiple-input multiple-output (MIMO) transmission and reception techniques, with 2×2 MIMO as the baseline for downlink and 1×2

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MIMO for uplink. However, higher order antenna configurations are supported. In the downlink closed-loop MIMO with code-book-based linear precoding can be applied, which allows for spatial multiplexing with dual code-word transmission on up to four transmission layers with fast rank adaptation. Additionally, an Alamouti-type transmit-diversity technique called space-frequency block coding (SFBC) is supported. In the uplink multi-user ("virtual") MIMO is used for capacity enhancement, in which pairs of spatially near-orthogonal users may transmit concurrently on the same physical resource blocks.

Evaluation of Next-Generation Networks

As mentioned previously, the performance of 3G network equipment and terminals was not verified to the full extent when 3G was launched in early deployments. This is one lesson learned; therefore, NGMN [1] requested performance evaluation and field trials in parallel to standards development. NGMN and 3GPP have initiated the LTE/SAE Trial Initiative (LSTI), which conducts trial activities and facilitates interoperability tests of LTE equipment.

Provided the metrics are meaningful and the methodology reflects realistic networks, simulations are a good way to compare different concepts and predict absolute values of network performance. The NGMN performance evaluation methodology [4] is well established and allows comparison of different standards. However, there are still a lot of effects that are hard to foresee or model realistically in simulations; therefore, field trials are essential to assess the performance. Also, field trials are a good proof of concept for innovative system-level concepts with lots of interdependencies such as advanced concepts addressing interference. Field trials also allow the calibration of simulations and allow research and development to be focused on tackling the right issues.

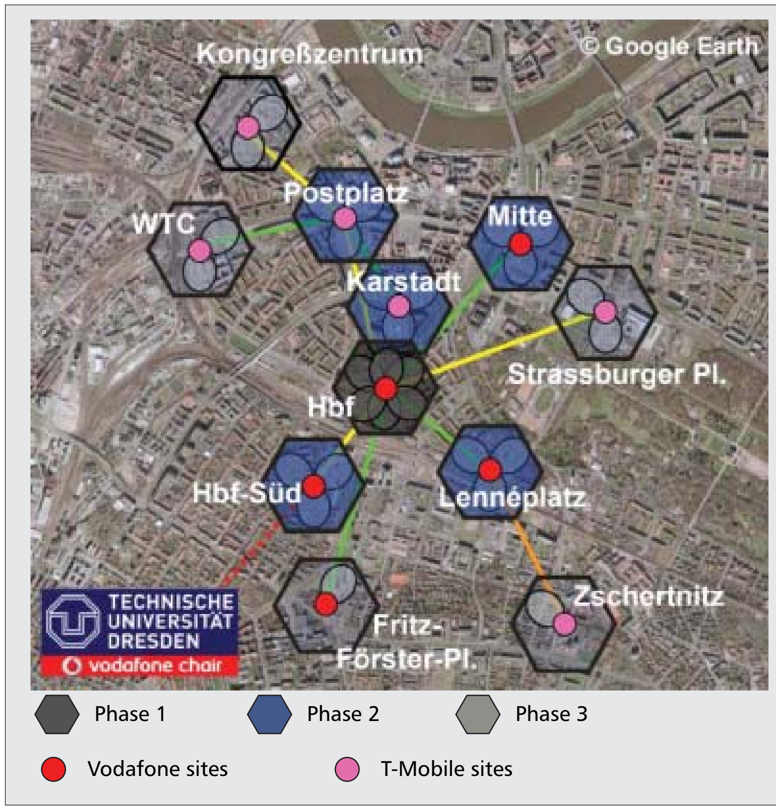
Cellular networks cannot be characterized well by single links. Interference, resource allocation, and propagation environment all impact system performance. To capture all effects, a sufficiently high number of interferers must be present, and multiple sectors and sites have to be involved.

Simulations and field trials focus mainly on technical KPIs such as throughput and latency. However, it has to be kept in mind that the ultimate criterion for the mobile Internet is *user experience*. This is hard to define, depends on particular applications, and changes over time — and is beyond the scope of this article.

EASY-C: A Field Testbed in Dresden

Field Testbed Area and Measurement Scenario

Two testbeds have been built and operated within the above mentioned research project EASY-C. In this article we concentrate on a physical layer focused testbed in downtown Dresden,



■ Figure 1. *Field test area in Dresden.*

Germany, using existing 2G/3G network sites of operators Vodafone and T-Mobile. Both operators are also involved in the trials. An additional testbed focused on applications enabled through LTE and advanced concepts is being set up in Berlin.

The chosen testbed location in downtown Dresden covers various propagation conditions, which are of special interest for evaluation of fourth-generation (4G) systems with MIMO links and interference conditions typical in *frequency reuse one* networks like LTE, and for the development of advanced algorithms such as cooperative MIMO:

- A representative area of a medium-sized European city
- Hills in the south causing signal reflections
- A river through the city causing superrefractions and tropospheric refraction
- Urban areas with multistory buildings, leading to shadowing effects
- An average intersite distance of 500 m

The testbed is being built in three phases. In the first phase, one site with three cells started operating in April 2008. This central site is located near Dresden's main railway station, as shown in Fig. 1. The antenna height is 55 m. The second phase will cover a tier of sites around this central site and consist of six sites with a total of 18 cells. In the final stage the testbed will comprise 10 sites with a total of 25 cells. Additional interferers will surround outer cells in order to emulate the interference intensity and distribution of a network with three tiers of sites. Such a rather extensive setup is necessary to capture all effects of a real-world deployment.

At these locations new base station antennas, feeders, and microwave link equipment are installed.

With this three-tier network, realistic scenarios can be set up to investigate LTE and advanced algorithms beyond LTE Release 8.

Furthermore, the baseline trial setup consists of a testbed platform of base stations and mobile equipment provided by the project partner Signalion. Other project partner's equipment (i.e., base stations, mobile, and chip prototypes) will be inserted into the testbed for various test cases.

This infrastructure enables well defined and reproducible interference scenarios in both uplink and downlink.

■ Figure 2. *Test base station with antennas and microwave link at central site.*

■ Figure 3. *Signal coverage of the trial area based on drive tests.*

Field Testbed Equipment

Figure 2 shows a test base station with a baseband unit, radio frequency (RF) hardware (including duplex filters and power amplifiers), antenna columns, and microwave backhaul units. LTE does not require GPS controlled reference clocks for synchronization, but they are included in this trial to investigate advanced multicell algorithms. The sensitivity of these algorithms to synchronization errors is one major research topic. The backhaul between the sites is accomplished by low-latency microwave links. These links operate in the 5 GHz frequency band and have a maximum throughput of 300 Mb/s.

Field Testbed Measurements

Within the testbed a number of tests are planned. For characterization of the radio environment, channel-sounding campaigns and coverage measurements are conducted. The objective of these measurements is, on one hand, the calibration of raytracing tools and the development of prediction algorithms for multicell MIMO operation in LTE-Advanced. On the other hand, cell edges can be identified: geographical locations where signals from several cells impinge with similar signal strengths.

Figure 3 shows the coverage map of the trial area based on drive tests.

Lab Results

Laboratory tests with pre-standard equipment and fading emulators have been carried out to assess the data rates and latencies that can be expected with LTE Release 8. The results were partly used by Alcatel-Lucent and Signalion to leverage the proof-of-concept work of LSTI.

Examples of the earlier laboratory test results are presented here to highlight inherent LTE capabilities such as AMC or frequency-selective scheduling. Further laboratory tests will be performed to prepare and complement the planned field trial activities, with particular emphasis on MIMO features.

Figure 4 depicts the physical layer cell throughput measured from downlink single-input single-output (SISO) as a function of average signal-to-noise ratio (SNR) with the following configurations:

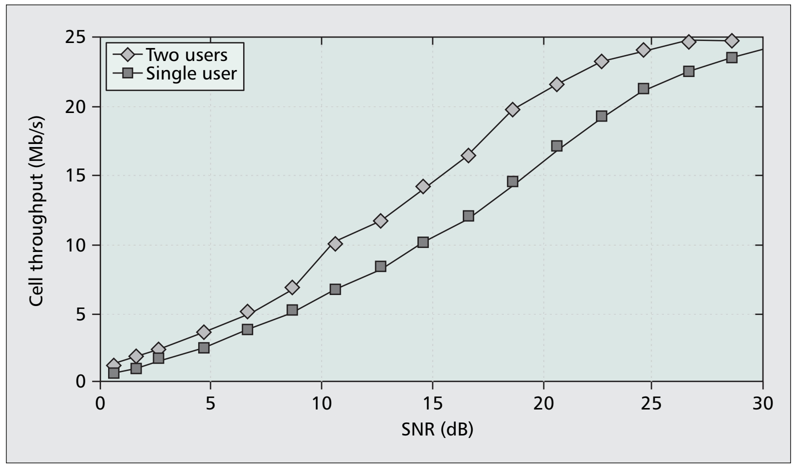
- 10 MHz system bandwidth
- AMC, hybrid ARQ, and multi-user scheduling in downlink (DL) under control of the eNodeB
- Acknowledgment (ACK)/negative ACK (NACK) and channel quality reporting in the uplink (UL)
- Single cell with a single user (blue curve) or two users (red curve) in the cell
- Full queue in the eNodeB for each user
- Pedestrian A 3 km/h fading channels in downlink and static channels in uplink

Figure 4 illustrates the capability of the AMC to finely adjust the user data rate to the channel quality. This is achieved in the downlink by reporting channel quality indicators (CQIs) back to the eNodeB from the user equipment (UE). In this example the update rate is 1 kHz/subband. The peak data rate observed in Fig. 4 can be scaled to the often quoted 88.7 Mb/s by assuming 20 MHz system bandwidth, code rate 1.0, and pilot/signaling overheads < 15 percent as achievable with LTE Release 8.

Figure 4 further illustrates the gain in cell throughput obtained by applying a time- and frequency-selective multi-user scheduling algorithm. This gain can be quantified by relating the multi-user cell throughput to the throughput of a single user. For slowly moving user terminals, this gain can be substantial, particularly in the low and moderate SNR regions. It is enabled by the particular definition of the CQI, which allows the full system bandwidth to be divided into subbands and apply the CQI reporting at the subband level.

In another type of measurement, the two-way air interface latency between UE and eNodeB was demonstrated to meet the 3GPP requirement of below 10 ms in an unloaded cell with a prescheduled UL channel [3]. Measured latencies are summarized in Table 1 for different scenarios, each using a PING application with 64 bytes payload size triggered from a PC connected to the UE.

While laboratory tests are a valuable means of system characterization, they have limitations due to complexity and cost of laboratory equipment, particularly when multiple sites or antennas are involved, and often also are not representative of real-world conditions. There remains, therefore, a strong motivation to carry out field trials, in particular to gain insight into the performance of a multicellular network.



■ Figure 4. *Downlink cell throughput vs. SNR with one or two SISO users in 10 MHz system bandwidth.*

Scenario	Latency
Single user Unloaded cell UL channel prescheduled No channel impairments	9.9 ms
Single user Unloaded cell UL channel set-up after scheduling request No channel impairments	19.4 ms
Single user Unloaded cell UL channel prescheduled Channel impairments in DL	17.9 ms
Two users Cell fully loaded in DL by second user UL channel prescheduled No channel impairments	9.8 ms

■ **Table 1.** *Measured average air interface latencies.*

System-Level Simulation Results

The scope of EASY-C is to prepare and support the standardization of LTE-Advanced and prove the enhanced concepts by field trials. The field trials are accompanied by system simulations in order to evaluate and optimize candidate algorithms for, say, collaborative or network MIMO before they are implemented in the trial system. On the other hand, the accuracy of simulation results will be investigated by comparing these results with measurements from the field trial system.

The system simulators are compliant wtih 3GPP and NGMN performance verification frameworks [4, 5]. The interference is modeled and simulated in detail. In order to avoid boundary effects and, hence, an overestimation of system performance, wrap around is applied. Both interfering and data channels are modeled by a spatial channel model. Furthermore, the simulations shall be realistic in terms of channel estimation loss and delays. In order to obtain the full capacity of the simulated radio access network, full buffer services are assumed. The simulators are able to simulate different receive and transmit antenna configurations with different antenna spacings. For quick randomization of measurements, the event-driven simulation is subdivided into drops in which new mobile positions are randomly chosen. CQI feedback and precoding matrix identifier (PMI) feedback are modeled with realistic granularity and with all relevant delays based on measured pilot SINR. The receivers are explicitly modeled and, for block error rate (BLER) calculation, the so-called mutual information effective SINR mapping (MIESM) link to the system interface is applied.

Baseline: LTE Release 8

Table 2 shows exemplary results for the DL performance of the 2 × 2 closed-loop baseline system for two different intersite distances of 500 and 1732 m, respectively. The performance is presented as sector spectral efficiency and cell border throughput, which is defined as the 5th percentile of the UE throughput. The bandwidth applied is 10 MHz. The operation point has been set to 30 percent BLER for the first transmission. HARQ is adaptive and asynchronous; that is, retransmissions are adapted to the instantaneous channel quality and can be postponed if, for example, the subframe foreseen for the retransmission is already occupied by other retransmissions. The scheduler is proportionally fair and frequency selective. Twenty-seven different modulation and coding schemes (MCSs) have been used for link adaptation and cover

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Antenna configuration	Intersite distance (m)	Spectral efficiency (b/s/Hz)	Cell border throughput (kb/s)
2×2	500	1.46	345
2×2	1732	1.37	255

Table 2. *LTE Release 8 downlink baseline performance.*

Antenna configuration	Intersite distance (m)	Spectral efficiency (b/s/Hz)	Cell border throughput (kb/s)
1×2	500	0.97	295
1×2	1732	0.85	57

Table 3. *LTE Release 8 uplink baseline performance.*

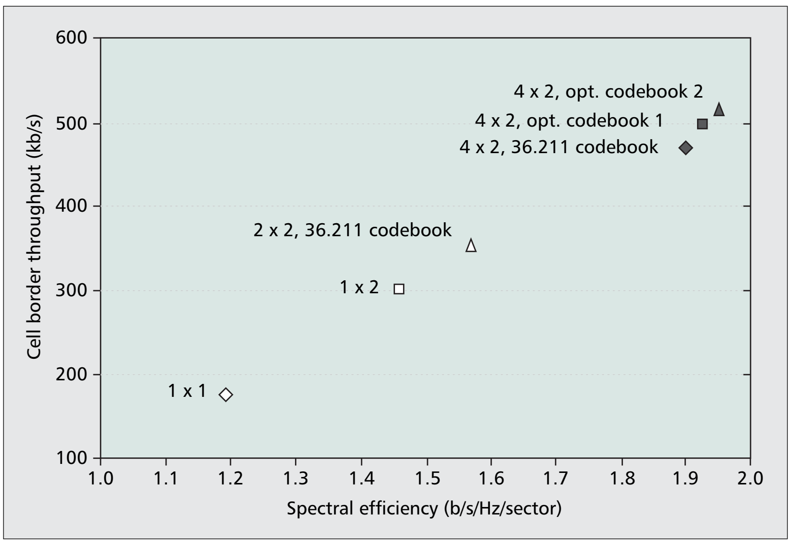


Figure 5. *Cell border throughput vs. system spectral efficiency for LTE DL with different antenna systems and precoding matrices.*

channel qualities from –6 up to 20 dB SINR. Please note that a rather pessimistic channel estimation loss model has been assumed, which causes the decrease of cell border throughput in case of larger intersite distances.

Table 3 shows exemplary results for the corresponding uplink performance for single antenna transmission and receive diversity. Path loss compensation has been applied in order to keep the per mobile average received signal power spectral density at the eNodeB constant. The maximum UE transmit power is 24 dBm. The frequency-selective proportionally fair scheduler considers this maximal transmission power so that the transmission reaches the required power spectral density. An exception to this rule is allowed if the required power for only one assigned resource block exceeds the maximal transmission power. Due to the applied SC-FDMA, the scheduler assigns only adjacent resource blocks. Obviously, due to the limited transmit power of the mobiles, the cell border throughput decreases significantly for the 1732 m ISD case. Different techniques such as interference coordination, or cooperative or network MIMO may enhance cell border throughput and spectral efficiency, and will be investigated in the EASY-C project.

Optimized Codebooks for 4 ×2 SU-MIMO in the Downlink

In this section we show exemplary results of a study on enhancements of LTE beyond Release 8.

Figure 5 shows LTE downlink results for different antenna configurations. The results for 2 ×2 and 4 ×2 are shown for precoding matrices conformant with 3GPP standard TS 36.211 [6]. Additionally, in the same diagram results for optimized codebooks are shown. These codebooks are optimized for linear arrays of X-polarized antennas with an antenna spacing of half of the wavelength of the carrier frequency. This approach saves up to 50 percent of feedback signaling load in the uplink, and at the same time improves cell border throughput and spectral efficiency in the downlink.

Evolution of LTE Beyond the Initial Release 8: LTE-Advanced

With the standardization of LTE Release 8 nearing completion, 3GPP has already created a new study item in order to explore candidate technologies for further technology evolution called LTE-Advanced, which are targeted to meet operator requirements and ITU-R's *IMT-Advanced* requirements. While maintaining backward compatibility with LTE Release 8, these ambitious performance targets include, among others [2]:

- Average spectrum efficiencies of up to 3.7 b/s/Hz/cell in the DL (4 ×4) and 2.0 b/s/Hz/cell in the UL (2 ×4)
- Cell edge spectrum efficiencies of 0.12 b/s/Hz in the DL (4 ×4) and 0.07 b/s/Hz in the UL (2 ×4)
- Peak data rates of up to 1 Gb/s in the DL and 500 Mb/s in the UL
- Peak spectrum efficiencies of 30 bit/s/Hz in the DL and 15 bit/s/Hz in the UL using antenna con-figurations of up to 8×8 in the DL and 4×4 in the UL
- Low cost of infrastructure deployments and terminals and power efficiency in the network and terminals

Within the EASY-C project, the following concepts are investigated among others that appear to be promising to address the above-mentioned targets:

- Advanced single-site MIMO
- Multisector coordination/cooperation
- Multisite coordination/cooperation

The schemes are illustrated in Fig. 6.

Using a high number of transmit and receive antennas in both the DL and UL addresses the demanding requirements for peak and average performances. Single-user MIMO with a large number of transmit and receive antennas is the enabler of high peak data rates. DL multi-user MIMO with optimized fixed beams or user-specific beams is the key to high spectrum efficiencies.

User-specific beams are especially suited for low mobility where accurate channel information at the transmitter must be available in order to create beams with high interference suppression. Fixed beams are suitable for moderate to high mobility, because a user's preferred beam is directly related to its position in the cell and only changes on a rather slow timescale. For compact X-polarized antenna configurations, fixed beamforming can further be elegantly combined with diversity for link enhancement and/or spatial multiplexing.

Multisector and multisite cooperation additionally boosts spectral efficiency and especially cell edge performance. For instance, interference coordination can be used to mitigate the impact of multisector interference, and joint signal processing concepts — often referred to as network MIMO or coordinated multipoint transmission/reception — actually allow interference to be exploited, and yield additional array and diversity gain. From a theoretical point of view, vast performance gains have been predicted for these schemes for both UL [7] and DL [8]. However, major research is still required for various practical aspects connected to network MIMO, such as:

- Synchronization of jointly processed terminals in time and frequency, and detection under remaining synchronization offsets
- Multisector channel estimation, feedback of channel information to the base stations, the impact of imperfect channel estimation on network MIMO, and robust signal processing algorithms
- Performance of network MIMO and concrete signal processing algorithms under a limited backhaul infrastructure between cooperating base stations [9]
- Cooperative scheduling for network MIMO

In EASY-C all the above mentioned aspects are researched, and the first laboratory and field test results for LTE-Advanced technologies are expected in 2009.

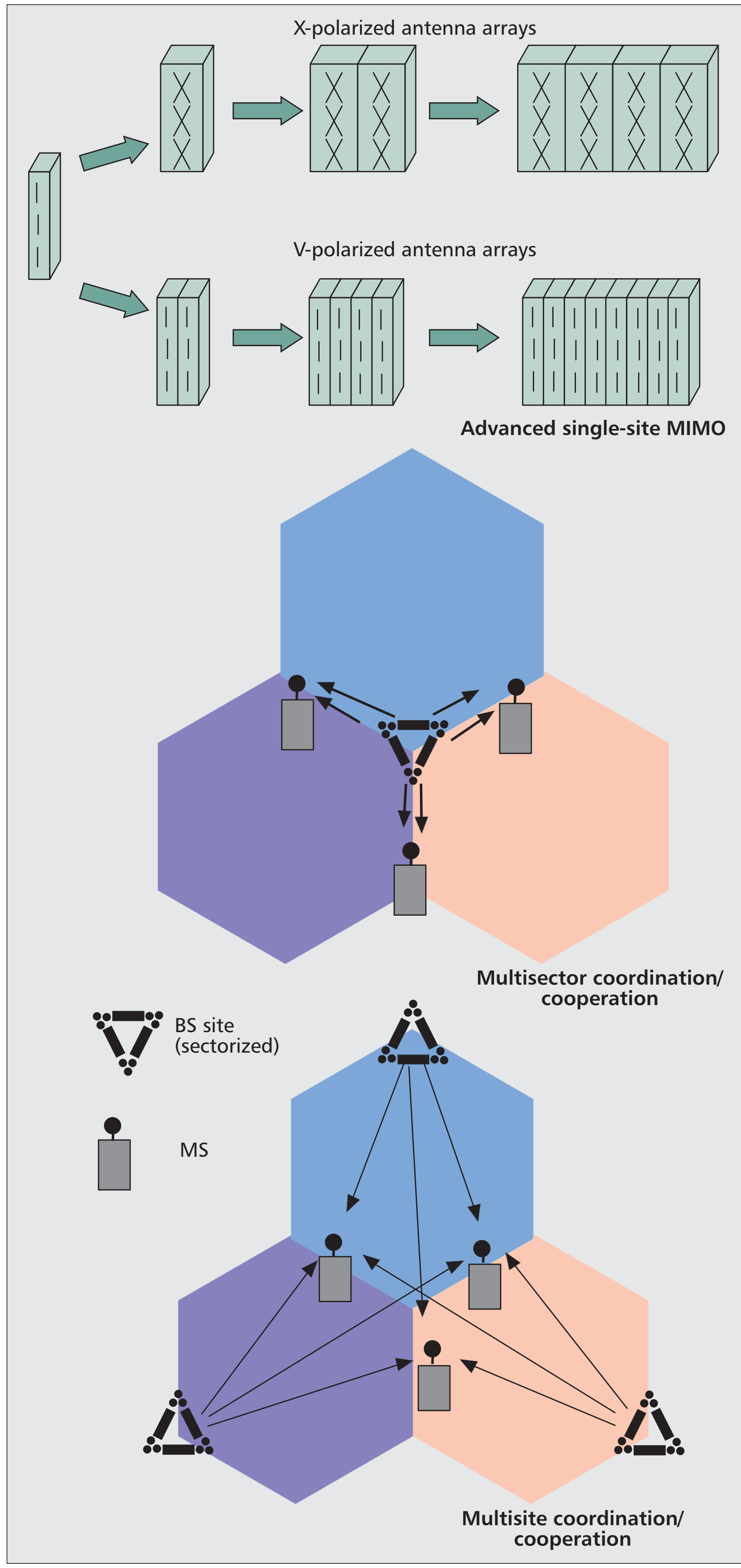
In order to optimally exploit all the various features, generalized multisite multi-user schedulers taking advantage of single-user, multi-user, and multisite technologies must be designed.

Conclusions and Outlook

This article shows performance results of the LTE standard based on system-level simulations and laboratory tests for user throughput, spectral efficiency, and latency. However, it is essential for realistic assessment of LTE and the development of further improvements to the standard to conduct trials with multiple sectors that reflect real-world interference conditions. Such a trial environment has been established within the EASY-C project. Advanced algorithms such as advanced single-site MIMO, and multisector and multisite cooperation are promising from a theoretical point of view, and the established testbed will be used to develop such concepts in detail and evaluate their real-word performance.

Acknowledgment

The authors acknowledge the excellent cooperation of all project partners within the EASY-C project and the support by the German Federal Ministry of Education and Research (BMBF).



■ Figure 6. *Candidate technologies for LTE advanced.*

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BIOGRAPHIES

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GUEST EDITORIAL

Woon Hau Chin *Yi Qian* *Giovanni Giambene*

ADVANCES IN COOPERATIVE AND RELAY COMMUNICATIONS

Traditionally, relays have been used to extend the range of wireless communication systems. However, in recent years, many exciting applications of relay communications have emerged. One such emerging application is to assist in the communication between the source and destination terminals via some cooperation protocol. By controlling medium access between source and relay terminals, coupled with the appropriate modulation or coding in such cooperative schemes, it has been found that the diversity of the communication system can be improved. In multi-user systems, different users can also act as cooperative partners or relays to share resources and assist each other in information transmission, thereby creating a cooperative network. One other emerging application is the exchange of information between multiple users through relay(s). In such cases, by exploiting the knowledge of one's own transmitted signal, the throughput of these systems can be drastically increased.

For cooperative and relay communications, the medium access control (MAC) layer also has many unique features. The MAC in this case is concerned with more than one-hop communication, is distributed and cooperative, and works for multipoint-to-multipoint communication. The MAC also needs to have knowledge about network topology and account for node mobility. Accordingly, new MAC layer designs must be devised to include new functionalities as well as MAC layer routing. With the large benefits to be reaped from employing cooperative and relay techniques, several standardization groups, such as IEEE 802.16 and IEEE 802.11, have started standardization processes to include such technologies in their prevailing standards.

With interest from both the research and industrial communities gaining momentum, there is an urgent need to better understand as well as keep track of cutting edge research in cooperative and relay communications. We have planned this feature topic to help address that need, as well as to help researchers looking to jump on the bandwagon. Therefore, we focus on recent advances, but also include survey articles on cooperative and relay communications.

The response to our Call for Papers on this feature topic of *IEEE Communications Magazine* was overwhelming, with over 50 articles submitted. All the papers were reviewed by experts in the relevant area, with at least three independent reviews for each paper and a rigorous two-round review process. Due to the lack of space, we can only accommodate six excellent articles covering various aspects of cooperative and relay communications involving physical layer (PHY), MAC, network layer, and cross-layer modeling and design.

The first article, "Distributed Transmit Beamforming: Challenges and Recent Progress" by R. Mudumbai *et al.*, reviews promising recent results in architectures, algorithms, and working prototypes on distributed transmit beamforming, and the challenges that must be surmounted. Directions are also discussed for future research needed to translate the potential of distributed beamforming to practice.

In the second article, "Cooperative Relay to Improve Diversity in Cognitive Radio Networks," Q. Zhang *et al.* give a brief overview about the interplay of cooperation and cognitive radio technologies, propose a relay-assisted D-OFDM for data transmission as the fundamental component for the whole system, and also present a new MAC protocol in a Universal Software Radio Peripheral (USRP)-based testbed.

The third article, "Link Layer Diversity and Above in Multihop Wireless Networks" by Y. P. Chen *et al.*, summarizes the causes of channel diversity in wireless communications and how it is perceived at different layers of multihop wireless networks. They concentrate on link layer diversity, and discuss the challenges and possible diversity schemes at the network layer.

H. Shan *et al.* analyze the issues and challenges in designing an efficient MAC scheme for multihop wireless ad hoc networks in the fourth article, "Distributed Cooperative MAC for Multi-hop Wireless Networks." They propose a cross-layer cooperative MAC protocol that is backward compatible with 802.11 networks, and can adapt to the channel condition and payload length.

In the fifth article in this topic, "Cooperative Network Implementation Using Open Source Platforms," T. Korakis *et al.* describe two programmable cooperative communication testbeds built at the Polytechnic Institute of NYU to demonstrate that cooperative techniques indeed work in practice.

Last but not least, J. J. Garcia-Luna-Aceves *et al.* present in the sixth and final article, "Context Aware Protocol Engines for Ad Hoc Networks," an example of context-aware packet switching in MANETs known as CAPE. CAPE is based on nodes storing the entire context within packets to be switched, and each data packet consists of only its payload and a pointer to bind it to the stored context.

In closing, we would like to thank all the authors for their excellent contributions. We also thank the reviewers for their dedicated time in reviewing the papers, and providing valuable comments and suggestions for refining the quality of the articles. We appreciate the advice and support of former and current Editors-in-Chief of *IEEE Communications Magazine* Drs. Thomas Chen and Nim K. Cheung, and Sue Lange and Joseph Milizzo for their help in the publication process.

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Distributed Transmit Beamforming: Challenges and Recent Progress

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ABSTRACT

Distributed transmit beamforming is a form of cooperative communication in which two or more information sources simultaneously transmit a common message and control the phase of their transmissions so that the signals constructively combine at an intended destination. Depending on the design objectives and constraints, the power gains of distributed beamforming can be translated into dramatic increases in range, rate, or energy efficiency. Distributed beamforming may also provide benefits in terms of security and interference reduction since less transmit power is scattered in unintended directions. Key challenges in realizing these benefits, however, include coordinating the sources for information sharing and timing synchronization and, most crucially, distributed carrier synchronization so that the transmissions combine constructively at the destination. This article reviews promising recent results in architectures, algorithms, and working prototypes which indicate that these challenges can be surmounted. Directions for future research needed to translate the potential of distributed beamforming into practice are also discussed.

INTRODUCTION

In wireless communication systems, transmit beamforming refers to a technique in which an information source transmits a radio frequency signal over two or more antennas and aligns the phases of the transmissions across the antennas such that, after propagation, the signals combine constructively at the destination. Fixing the power radiated by a given antenna element, ideal transmit beamforming with N antennas results in an N^2-fold gain in received power. Compared to single-antenna transmission, transmit beamforming can therefore yield increased range (an N-fold increase for free space propagation), increased rate (an N^2-fold increase in a power-limited regime), or increased power efficiency (an N-fold decrease in the net transmitted power for a fixed desired received power). In addition, since more power is directed in the desired direction, less is scattered in undesired directions, resulting in reduced interference and increased security.

Given the many advantages of transmit beamforming, it is natural to ask whether it can be emulated in distributed fashion using a network of cooperating single-antenna sources. In order to operate as a "distributed transmit beamformer," the sources must agree on a common message, transmit it at the "same time," synchronize their carrier frequencies, and control their carrier phases so that their signals combine constructively at the destination. Hence, practical realization of this concept requires the development of implementable distributed techniques for information sharing, timing synchronization, and carrier synchronization. While these constitute a daunting set of challenges, recent results from several different research groups provide promising approaches for addressing them. The goal of this article is to take stock of the current state of the art, and to suggest directions for future research in the design and implementation of wireless networks that exploit distributed beamforming.

In addition to the N^2-fold power gain from distributed beamforming, there is also a potential advantage in terms of wireless propagation. Consider the Friis formula for free-space propagation,

$$P_R = P_T G_T G_R \frac{\lambda^2}{16\pi^2 R^2},$$

where P_T and P_R are the transmit and receive powers, respectively, G_T and G_R are the directivity gains of the transmit and receive antennas, respectively, R is the range between the antennas, and λ is the carrier wavelength. For fixed antenna gains, the propagation loss

$$\frac{P_T}{P_R}$$

is smaller at longer wavelengths. However, antenna gains take the form

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$$G = \frac{4\pi A}{\lambda^2},$$

where A is the effective area. Thus, in order to maintain a given directivity as wavelength increases, one must also scale the effective area of each antenna by λ^2, which can make longer wavelengths unattractive. With distributed transmit beamforming, it is possible to have the best of both worlds: low propagation loss by operating at a long wavelength and high directivity by exploiting the natural spatial distribution of the cooperating nodes to emulate a large antenna array. While this argument is presented for free space propagation, longer wavelengths provide even more of an advantage in cluttered environments, since the radio waves are better able to diffract around obstacles.

As an example application, consider the scenario shown in Fig. 1, where a terrestrially deployed network of low-power single-antenna sensor nodes collects measurements and transmits these measurements to an overflying unmanned aerial vehicle (UAV) using a carrier frequency of 3 GHz and a bandwidth of 10 MHz. For a sensor transmit power of –10 dBm, the received power at an altitude of 3000 m (typical for intermediate range UAVs) is –110 dBm, assuming a sensor transmit antenna gain of G_T = 2 dBi and receive antenna gain G_R = 10 dBi for the aircraft. For a receiver noise figure of 6 dB, the noise power is –97 dBm (thermal noise at 300 Kelvin has a power spectral density of –173 dBm/Hz).Thus, the signal-to-noise ratio (SNR) for a single sensor transmission is –13 dB, making communication with reasonable spectral efficiency infeasible. On the other hand, the SNR increases to +13 dB if 20 sensor nodes form a distributed transmit beamformer. This could enable, for example, upload of image/video data or summaries of sensor data gathered over days or even months. Other interesting applications include reach-back using low-power soldier radios in battlefield communication, and collaboration between subscriber terminals for uplink transmission to a base station receiver, especially in rural or disaster recovery settings where longer range might be required.

As the preceding examples indicate, distributed transmit beamforming has the potential to enable fundamentally new functionalities in wireless communication and sensor networks. In the remainder of this article we discuss some of the technical issues that must be addressed in order to realize this potential. We review recent progress on the crucial distributed carrier synchronization problem in the next section and later describe two working prototypes that suggest this problem is solvable. We then discuss the characteristics of beam patterns realizable using distributed transmit beamforming with randomly placed sources. These results lay the foundation for "physical layer" feasibility of distributed transmit beamforming. In the following section we discuss the cross-layer design considerations for information sharing and coordination among sources. We end in the final section with a discussion of directions for future research.

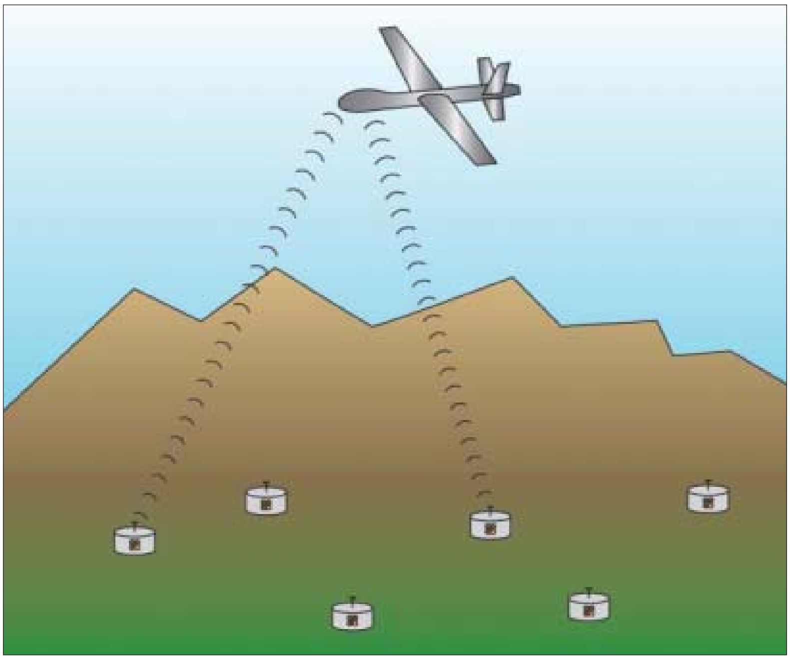


Figure 1. *Sensor network transmitting measurements to an overflying aircraft.*

Distributed Carrier Synchronization

A key distinguishing feature of distributed transmit beamforming with respect to conventional beamforming is that each source node in a distributed beamformer has an independent local oscillator (LO). These LOs are typically generated by multiplying the frequency of a crystal oscillator up to a fixed nominal frequency. Carrier frequencies generated in this manner, however, typically exhibit variations on the order of 10–100 parts per million (ppm) with respect to the nominal. If uncorrected, these frequency variations among sources are catastrophic for transmit beamforming since the phases of the signals may drift out of alignment over the duration of the transmission and may even result in destructive combining at the destination. The first goal, therefore, is to synchronize the carrier frequencies for the different sources to minimize or eliminate frequency offset.

One approach to frequency synchronization is to employ a *master-slave* architecture [1, 2], where "slave" source nodes use phase-locked loops (PLLs) to lock to a reference carrier signal broadcast by a "master" source node. Alternatively, the destination node could broadcast a reference carrier to facilitate frequency synchronization among the source nodes [3–5]. A source node that estimates its frequency offset to be Δf can multiply its complex baseband transmitted signal by $e^{-j2\pi\Delta f t}$, where the operation can be implemented in a digital signal processor (DSP) prior to digital-to-analog conversion and carrier multiplication. Depending on the stability of the sources' oscillators, the process of frequency synchronization may need to be repeated, and should be inherent to any networking protocol built around distributed beamforming.

Once frequency synchronization is achieved, the phase of the transmissions from the different

Regardless of the synchronization approach, it is known that beamforming gains are quite robust to moderate errors in phase alignment. For example, 90 percent of an ideal two-antenna beamforming power gain is attained even with phase offsets of the order of 30°.

sources must be synchronized to arrive with "reasonable" alignment at the destination. To understand why carrier phase synchronization is critical for distributed beamforming, consider first transmit beamforming using an N-element centralized array. To send a complex baseband message signal $s(t)$, the signal transmitted from antenna i is $w_i s(t)$, and the received signal is $\Sigma_i w_i h_i s(t)$, where h_i is the complex channel gain from antenna i to the receiver. The received SNR is therefore proportional to $|\Sigma_i w_i h_i|^2$. Given a constraint on the total transmitted power $\Sigma_i |w_i|^2$, it can be shown that the SNR is maximized by choosing $w_i \propto h_i^*$, i.e., $|w_i| \propto |h_i|$ and $\angle w_i = -\angle h_i$. Another option, appropriate for a peak power constraint per antenna element, is to use a fixed amplitude $|w_i| = w_{\max}$ and $\angle w_i = -\angle h_i$. When the channel gains are approximately equal in magnitude, both methods have similar performance: the received signal $|\Sigma_{i=1}^{N} w_i h_i| \propto N$, so that the received SNR scales as N^2. In either case, the transmitter requires channel state information (CSI) regarding the $\{h_i\}$, with the phase $\angle h_i$ being the critical information required to obtain beamforming gains. Techniques for obtaining CSI at the transmitter fall into two broad categories: implicit feedback (e.g., using reciprocity in a time division duplexed [TDD] system), and explicit feedback, where the CSI is quantized and sent over a separate feedback channel. A detailed review of different beamforming techniques is given in [6].

In distributed beamforming scenarios the sources are assumed to be unsynchronized a priori. This lack of synchronization leads to ambiguous phase estimates at each source. To see this, consider first implicit channel feedback using reciprocity. Ignoring modulation and noise for simplicity, source node i receives the passband signal Re $(h_i e^{j2\pi f_c t})$. When this is down converted using the local oscillator (LO) at node i, using the quadrature carriers $\cos(2\pi f_c t + \theta_i)$ and $\sin(2\pi f_c t + \theta_i)$, the complex baseband channel estimate at node i will be $\hat{h}_i = h_i e^{-j\theta_i}$. Without carrier synchronization across nodes, the local oscillator phases $\{\theta_i\}$ may be modeled as independent and uniformly distributed over $(-\pi, \pi]$, which implies that the phase of the channel estimate contains no information about the actual channel phase. In other words, the channel phase cannot be disambiguated from the relative LO phase at node i with this approach. Now, suppose instead that the receiver measures the channel gains from each node, and feeds them back explicitly. When node i employs these explicit channel estimates, however, it must upconvert the baseband message using its LO, which means that it is effectively using the beamforming weight $w_i = h_i^* e^{j\theta_i}$. Again, the phase of the beamforming coefficient is essentially random without prior carrier synchronization across the nodes.

The preceding observations show that even under ideal timing synchronization across nodes, distributed beamforming is impossible without distributed carrier synchronization. Nonidealities in timing synchronization can also affect distributed beamforming, but the effects are easier to handle. Timing synchronization is required to ensure that all of the cooperating nodes transmit the same symbol at a given time; timing errors between the nodes lead to misalignment between the symbols transmitted by each node, causing intersymbol interference (ISI) at the receiver. For relatively low data rates (say around 100 kb/s), the required level of timing synchronization can be obtained using well-known algorithms such as RBS [7]. These algorithms are capable of achieving accuracy on the order of 1 μs with low complexity. For higher data rates, customized timing synchronization techniques might be needed to achieve the desired level of accuracy. Even with accurate timing synchronization, however, ISI can arise due to dispersive channels from each node to the receiver. A natural approach to handling this is multicarrier modulation: in this case, distributed beamforming would be performed separately for each subcarrier. For single-carrier modulation, we may wish to use transmit precoding to ensure that the same symbol sent by different transmitters appears at approximately the same time at the receiver. Thus, while timing synchronization does pose a challenge, it is not as fundamental a bottleneck as carrier synchronization; hence, we focus on the latter in this article.

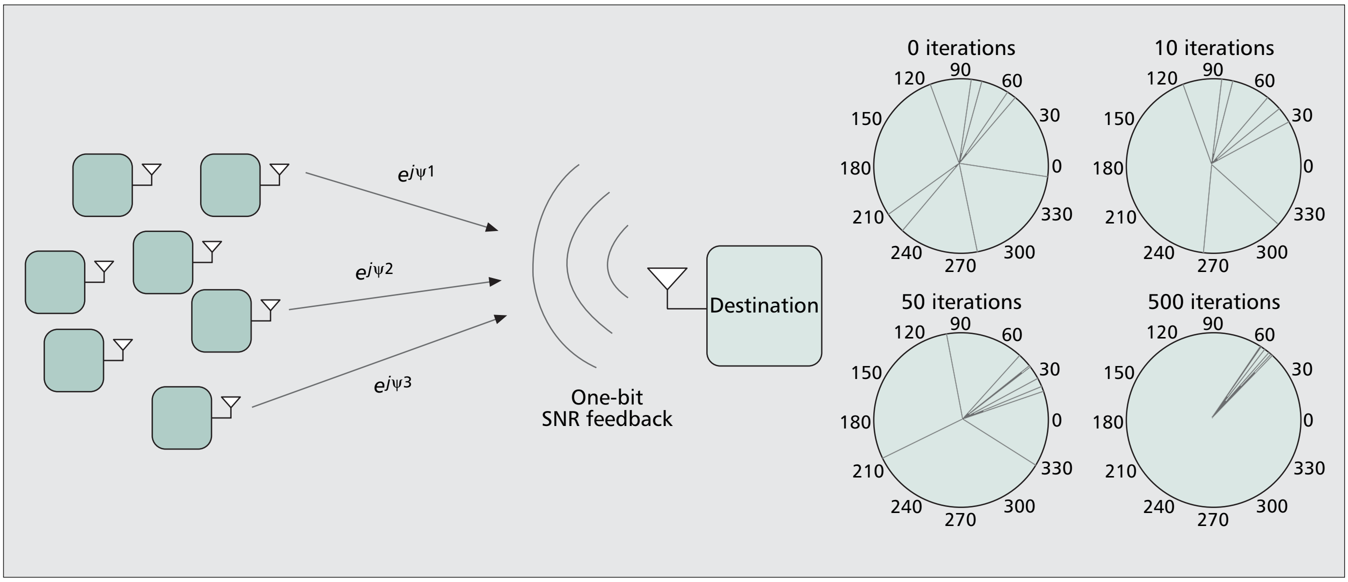
There are two basic approaches to phase synchronization distinguished by the interaction between the sources and the destination:

- **Closed-loop phase synchronization**: In closed-loop systems, the destination directly controls the phase alignment among the sources by measuring a function of the received phases of the source transmissions and then transmitting digital feedback signals to the sources to allow each source to compensate for its overall phase offset (LO and channel). Interaction among the sources can be minimal in closed-loop systems since the destination coordinates the synchronization process.
- **Open-loop phase synchronization**: In open-loop systems, the sources interact among themselves with only minimal signaling from the destination. Rather than providing feedback to be used for adapting the source phases, the destination may simply broadcast an unmodulated sinusoidal beacon to the sources. The sources use this beacon, as well as the signals from other source-source interactions, to achieve appropriate phase compensation for beamforming to the destination. The emphasis of open-loop systems is on using local interactions between the sources to minimize interaction with the distant destination.

Regardless of the synchronization approach, it is known that beamforming gains are quite robust to moderate errors in phase alignment. For example, 90 percent of an ideal two-antenna beamforming power gain is attained even with phase offsets on the order of 30° [1, 2].

Full-Feedback Closed-Loop Synchronization

The first carrier synchronization scheme suitable for distributed beamforming is described in [3]. Carrier frequency synchronization is



■ **Figure 2.** *One-bit feedback closed-loop carrier synchronization system.*

achieved using a master-slave approach, with the intended destination acting as the master node. The unknown phase offset between the destination and the nth source node is corrected via a closed-loop protocol realized in the following steps:

1 The destination broadcasts a common master beacon to all source nodes.
2 Each source node "bounces" the master beacon back to the destination on a different frequency than the master beacon. The source nodes use distinct codes in a direct-sequence code-division multiple access (DS-CDMA) scheme in order to allow the destination to distinguish the received signals.
3 Upon reception of the bounced beacons, the destination estimates the received phase of each source relative to the originally transmitted master beacon. The destination quantizes these estimates, and then transmits the estimates via DS-CDMA to the source nodes in a "phase compensation message." The phase compensation message may also contain clock correction information to facilitate symbol timing synchronization.
4 Each source receives the phase compensation message, extracts its own phase compensation estimate, and then adjusts its carrier phase accordingly.

Assuming that the phase offsets have not changed significantly between the synchronization and beamforming intervals, the bandpass transmissions from each source will combine coherently when the sources transmit to the destination with compensated carrier phases. The effect of energy allocation between synchronization and information transmission on the error probability of digital signals transmitted by a distributed beamformer was also studied in [3]. The results showed that an optimal energy trade-off exists and that allocating too much or too little energy to carrier synchronization is inefficient.

One-Bit Feedback Closed-Loop Synchronization

The rate of feedback necessary to establish and maintain reasonable phase alignment among the sources in the full-feedback closed-loop carrier synchronization system described in [3] may be prohibitive in some scenarios. Recently, a closed-loop carrier synchronization system was proposed using only one bit of feedback for all source nodes [8, 9]. The basic idea behind the one-bit feedback closed-loop synchronization system shown in Fig. 2 is as follows:

1 Each source node adjusts its carrier phase randomly.
2 The source nodes transmit to the destination simultaneously as a distributed beamformer.
3 The destination estimates the SNR of the received signal.
4 The destination broadcasts one bit of feedback to the sources indicating whether its SNR is better or worse than before the sources adjusted their phases. If it is better, all source nodes keep their latest phase adjustments; otherwise, all sources undo their latest phase adjustments.

These four steps form one iteration of the system. Since each source retains only those random phase adjustments that lead to performance improvement, the algorithm may be viewed as a randomized ascent procedure; hence, the number of iterations to achieve a desired degree of phase convergence is a random variable. The average number of iterations required to achieve phase convergence was shown to scale roughly linearly with N, where N is the number of source nodes [8]. Numerical and analytical results in [8] also showed that 75 percent of the ideal beamforming amplitude is achieved in roughly $5N$ iterations on average. Under mild conditions on the distribution of the source nodes' random phase adjustments, the one-bit feedback system was also shown to converge to full phase coherence with probability one [9]. Figure 2 shows the

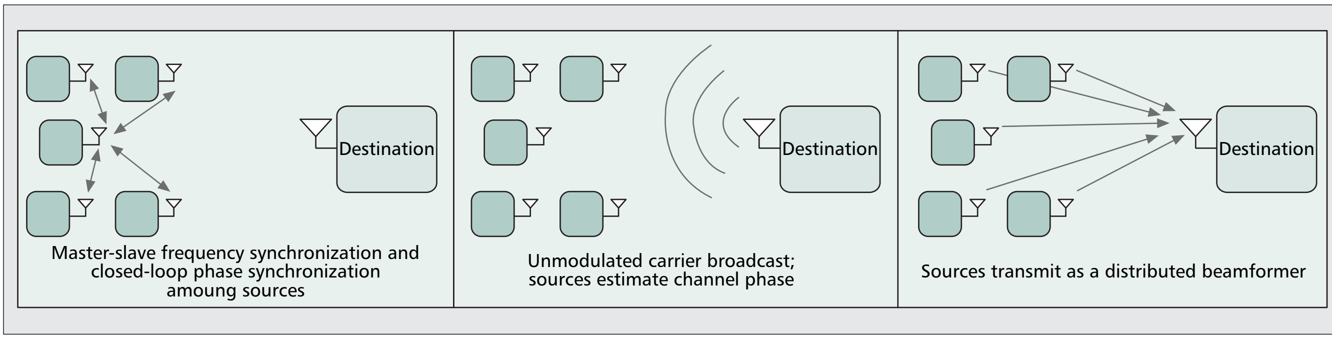


Figure 3. *Master-slave open-loop carrier synchronization system.*

evolution of the received phases from each source node in one instance of the algorithm with N = 10 nodes. In this case, after 500 iterations, all the received phases are between 45° and 60° (i.e., a spread of 15°). This is sufficient to achieve approximately 99 percent of the beamforming gains. Note that the precise value to which the received phases converge is irrelevant to the beamforming process; it only matters that the *differences* between the phases converge to zero to achieve coherent combining.

Like the full-feedback closed-loop synchronization system, the one-bit feedback closed-loop system corrects the overall phase offset for each source caused by both the LO and the channel. As such, the iterations can be continued indefinitely to track both channel time variations and oscillator drift. Moreover, while the system described in [8, 9] assumes that the sources are already synchronized in frequency (e.g., by using the master-slave approach), this approach can be extended to also explicitly include carrier frequency synchronization [10].

The simplicity and scalability of the one-bit feedback synchronization system make it an attractive candidate for practical implementation where closed-loop feedback from the destination is possible. Two experimental prototypes based on the one-bit feedback closed-loop approach are discussed later.

Master-Slave Open-Loop Synchronization

In some applications, closed-loop feedback from the destination to the sources is undesirable due to the relatively high cost of communication over this link and the increased complexity incurred at the destination. Open-loop carrier synchronization systems minimize interaction between the source nodes and the destination by increasing the level of inter-source interactions. One open-loop approach inspired by master-slave frequency synchronization was described in [2] and is illustrated in Fig. 3.

In open-loop master-slave synchronization, one source node is designated as the master and the remaining source nodes are slaves. For frequency synchronization, the master source node broadcasts a sinusoidal signal to the slave nodes, and each slave node estimates and corrects its frequency offset. Phase synchronization among the source nodes is then achieved through a closed-loop method similar to [3] except that a TDD protocol is used between the master source node and the slave source nodes. The primary difference in this case is that the feedback is from the master source node to the slave nodes and does not involve the destination.

Up to this point the synchronization process has been coordinated among the source nodes themselves without requiring any interaction with the destination node. In order for the sensors to beamform toward the destination, each source must estimate its channel response to the destination. This is achieved by having the destination broadcast a beacon (e.g., a sinusoidal signal at the carrier frequency) to the source nodes. Since the sources have already been synchronized, each source node can independently estimate its own complex channel gain to the destination using its frequency and phase-synchronized LO. The source nodes can then transmit as a distributed beamformer to the destination by applying the complex conjugate of these gains, typically at baseband, to their transmitted signals.

Round-Trip Open-Loop Synchronization

A different open-loop carrier synchronization system that eliminates the need for digital signaling during synchronization was proposed in [4, 5, 11]. The scheme is based on the equivalence of round-trip propagation delays through a multihop chain of source nodes and thus is called the *round-trip* carrier synchronization scheme. Like the open-loop master-slave synchronization system, the round-trip system requires minimal interaction between the source nodes and the destination.

A two-source round-trip system model is shown in Fig. 4 (round-trip carrier synchronization of more than two source nodes is discussed in [5]). The basic idea behind the round-trip synchronization system is that an unmodulated beacon "bounced" around the clockwise circuit shown in Fig. 4 will incur the same total phase shift as an unmodulated beacon "bounced" around the counterclockwise circuit shown in Fig. 4 when channels are reciprocal. The equivalence of the accumulated phase shifts for both round-trip circuits is the key feature of the round-trip carrier synchronization technique. The beacons are bounced around the circuit by having each source transmit periodic extensions of received beacons. Beamforming is achieved since the destination is essentially receiving the sum of two beacons, modulated by the common message, after they have propagated through circuits with identical phase shifts.

The actual implementation of a round-trip distributed beamformer is complicated, however, by the constraint that wireless transceivers may not transmit and receive on the same frequency at the same time. One approach is to use continuously transmitted beacons with distinct frequencies (also distinct from the carrier frequency). This approach, called the frequency-synthesis round-trip carrier synchronization system, was considered in [4] where each source employed a pair of frequency-synthesis PLLs in order to generate appropriate frequency-scaled periodic extensions of the beacons it received. An audio-frequency prototype of the frequency synthesis round-trip carrier synchronization system is discussed later. While the continuously transmitted beacons allowed for high rates of source and/or destination mobility, the use of distinct frequencies for the beacons and carriers resulted in non-reciprocal phase shifts and degraded performance in general multipath channels.

To ensure channel reciprocity in general multipath channels, a single-frequency time slotted round-trip carrier synchronization technique was proposed for a two-source distributed beamformer in [11] and extended to $N > 2$ sources in [5]. A total of $2N - 1$ synchronization time slots are needed to synchronize the sources prior to beamforming. The protocol is also repeated in order to avoid unacceptable phase drift, resulting from frequency estimation errors as well as phase noise and/or mobility, between the sources during beamforming. Long duration synchronization time slots tend to result in low estimation error but increased drift due to phase noise and/or mobility. Short duration time slots reduce the effects of phase noise and mobility, but lead to increased drift from low-quality frequency and phase estimates. Guidelines for achieving an efficient trade-off with low synchronization overhead are discussed in [5].

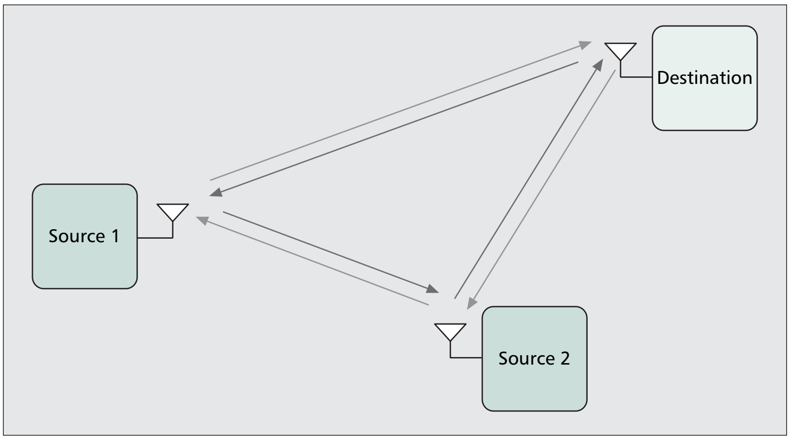


Figure 4. *Round-trip open-loop carrier synchronization system, two source nodes.*

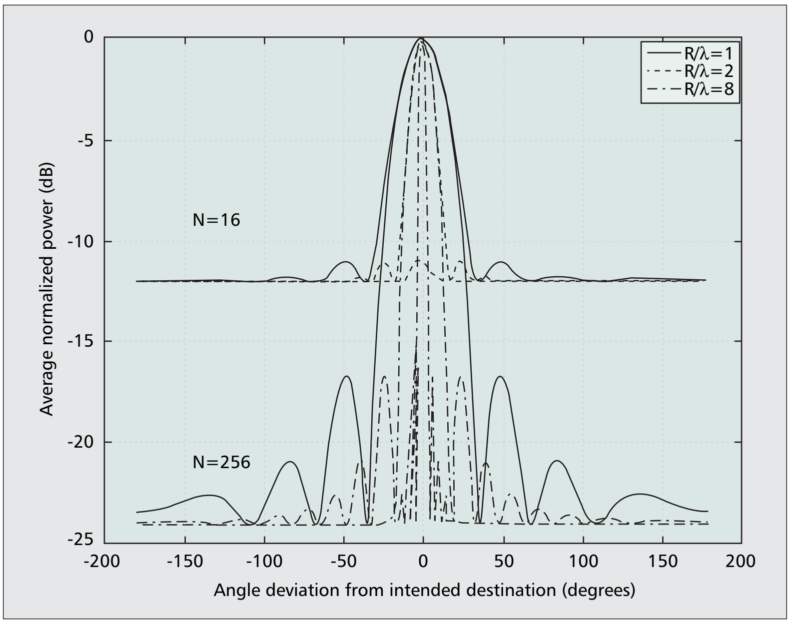


Figure 5. *Average beam pattern of a distributed beamformer with randomly placed nodes.*

Beampatterns for Randomly Placed Sources

The carrier synchronization techniques described previously are necessary to ensure that the directional gain of the distributed beamformer is close to that of an ideal conventional beamformer. Given a particular antenna geometry and the sources' carrier phases, it is also possible to use standard techniques to compute the beamwidth and sidelobe characteristics of a distributed beamformer. These characteristics may be of interest in applications where, for example, security or interference is important. Since the "antenna geometry" of a distributed beamformer may be random, however, a statistical characterization of the beampattern is necessary. This section summarizes recent work in this area.

The probability distribution of the far-field beam pattern of a distributed beamformer with node locations uniformly distributed on a two-dimensional disk of radius R was analyzed in [12]. The average far-field beampattern for a N-source distributed beamformer was shown to be

$$P_{\mathrm{av}}(\phi)=\frac{1}{N}+\left(1-\frac{1}{N}\right)\left|\frac{J_1(4\pi\tilde{R}\sin(\phi/2))}{2\pi\tilde{R}\sin(\phi/2)}\right|^2,$$

where $\tilde{R} = R/\lambda$ is the radius of the disk normalized by the wavelength of the transmission, ϕ is the angle with respect to the intended destination, and $J_1(x)$ is the first-order Bessel function of the first kind. The average far-field beam pattern is shown in Fig. 5 for two different values of N and three different values of $\tilde{R}$.

For sufficiently large N and $\tilde{R} \gg 1$, the average 3 dB beamwidth of the main lobe was shown to be inversely proportional to $\tilde{R}$ by numerically solving Eq. 2 for the case $P_{\mathrm{av}}(f) = 1/2$. The dependence of the average 3 dB beamwidth on $\tilde{R}$ is also evident in Fig. 5. These results suggest that very narrow beamwidths can be achieved in typical sensor network applications. For example, a sensor network with $N = 10$ randomly placed source nodes on a disk of radius 25 m

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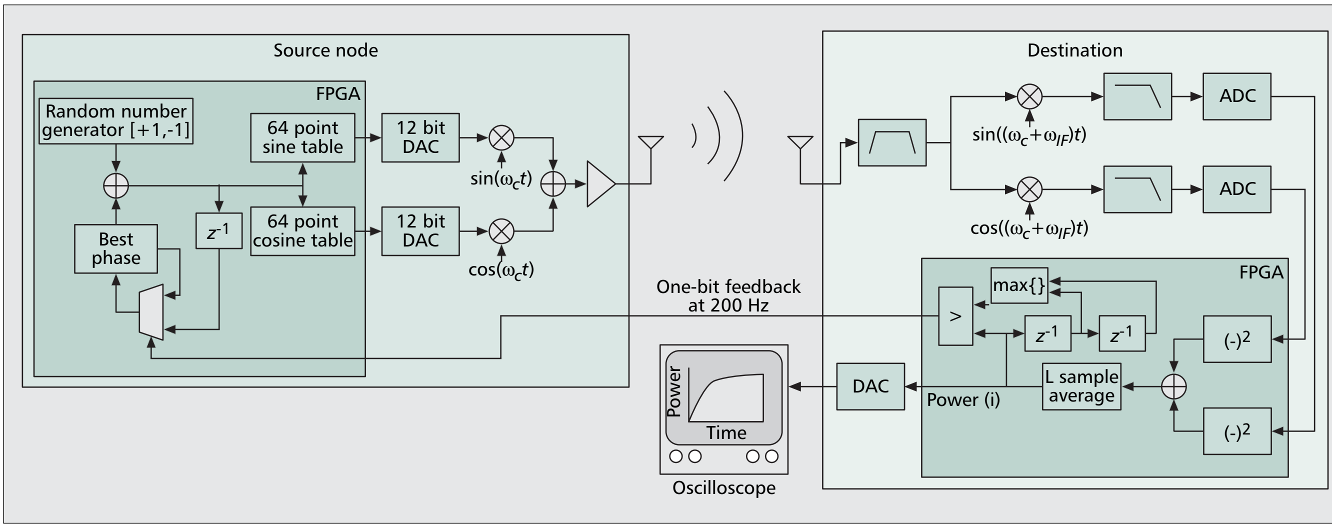


■ **Figure 6.** *Block diagram of the one-bit feedback closed-loop carrier synchronization prototype described in [15].*

will, on average, achieve a 3 dB beam width of less than half a degree if the sources transmit with 900 MHz carriers. The average sidelobe power of a distributed beamformer with N randomly placed source nodes was also shown to be on the order of $1/N$, plus some margin for sidelobe peaks near the main beam in [12]. The dependence of the sidelobe power on N is also evident in Fig. 5.

Information Sharing among Beamforming Nodes

In conventional transmit beamforming, a common message is transmitted across all antennas in the array. The transmitted signal at each antenna element is simply a complex weighted version of the common message with weights selected to achieve a desired beam pattern. In distributed beamforming systems, nodes must share information prior to beamforming. When the links between cooperating nodes in a distributed beamformer are short with respect to the link to the destination, it is reasonable to assume that energy required for information sharing prior to beamforming is negligible with respect to the energy required to transmit to the intended destination. The time required for information sharing may not be negligible in some cases, however, and depends to some extent on the architecture of the network.

The problem of information sharing in distributed beamforming systems has primarily been studied for the case of heterogeneous networks with K master source nodes, each with distinct information to convey to a distinct destination [13]. These master source nodes share a pool of N "non-master" source nodes (or relays) that can transmit as a distributed beamformer. A straightforward approach in this scenario is to use time sharing: one master source node broadcasts its message to the relay pool. The relays then transmit this message, including any noise in the received signals, as a beamformer to the destination. This broadcast beamforming cycle is performed one master source node at a time; hence, the throughput per master source node for this scheme is inversely proportional to the number of master source nodes.

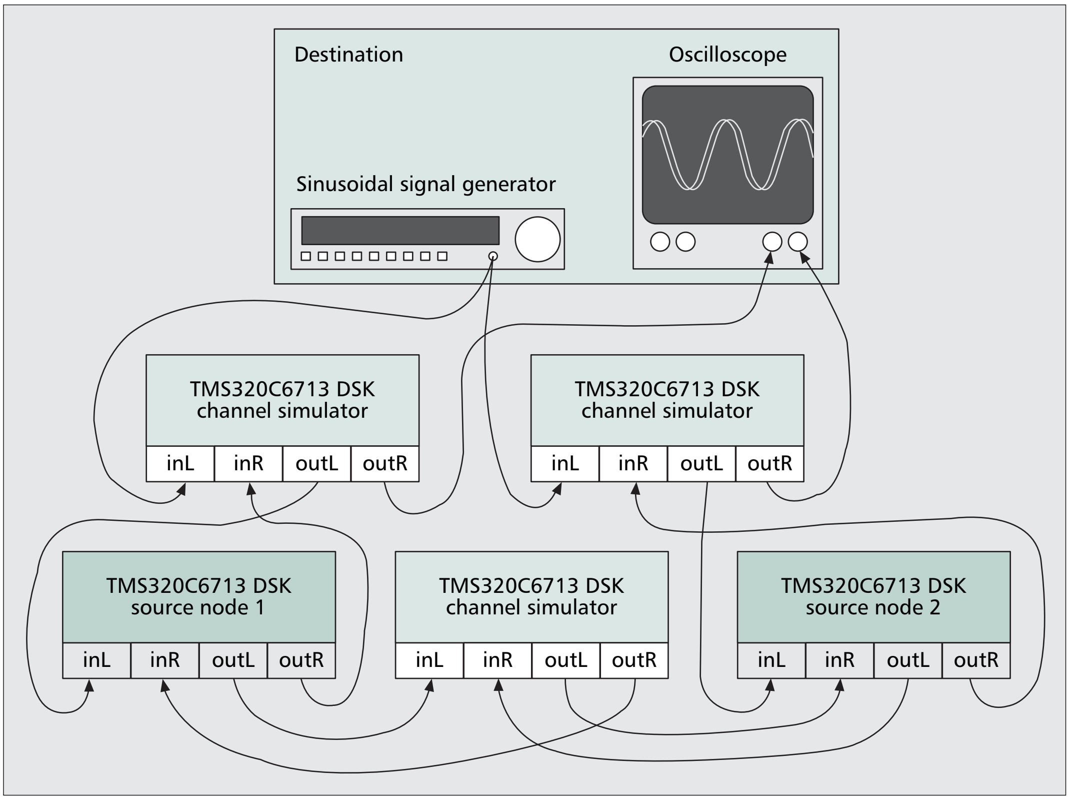
A higher-throughput space-division information sharing strategy is proposed in [14] in which all K master source nodes simultaneously broadcast their independent information to the pool of non-master source nodes. The N non-master source nodes then simultaneously beamform to the K destination nodes so that the message of the kth master source node combines coherently at the kth destination. While the throughput of this space-division approach is clearly better than that of time sharing, the simultaneous broadcast of messages by the master source nodes and simultaneous beamforming by the non-master source nodes may result in interference in the signals received at each destination.

Proof-of-Concept Prototypes

As theoretical research on distributed transmit beamforming has advanced, experimental prototypes have recently been constructed to confirm theoretical predictions and to better understand the inherent nonidealities in practical realizations. This section describes two such prototypes and summarizes the results of the laboratory experiments.

One-Bit Feedback Closed-Loop Synchronization Prototype

In 2006 a prototype of the one-bit feedback closed-loop synchronization system described earlier was built at the University of California at Berkeley in collaboration with the University of California at Santa Barbara [15]. A block diagram of a single source node and the destination node is shown in Fig. 6. Carrier frequency synchronization was achieved by distributing a common clock to the source nodes, which was multiplied up in frequency by each source node separately using a PLL. The one-bit feedback was conveyed from the destination to the source nodes via separate wired links. Using an FPGA-based power estimator, the destination fed back



■ **Figure 7.** *Block diagram of the two-source round-trip open-loop frequency-synthesis carrier synchronization prototype described in [16].*

The next step is to investigate and demonstrate distributed beamforming in a networked context, with a detailed design that spans information sharing, timing synchronization, carrier frequency synchronization, and carrier phase alignment.

a value of 1 to the source nodes when the current received power was greater than the averaged power estimates from each of the last L iterations ($L = 4$ for the results reported in [15]).

In a bench-top experiment performed with three source nodes, the measured received power was better than 90 percent of ideal. Convergence took approximately 60 iterations, which for a 200 Hz feedback rate corresponds to a convergence time of approximately 300 ms. The experiment was performed with unmodulated carriers as well as binary phase shift keying (BPSK) modulated carriers; as expected, data modulation did not affect convergence time or beamforming gain.

The one-bit algorithm has also been extended to provide distributed frequency and phase synchronization; this was demonstrated in 2007–2008 for a millimeter-wave sensor network testbed at the University of California at Santa Barbara [10].

Two-Source Frequency-Synthesis Round-Trip Synchronization Prototype

In 2005–2006 a two-source distributed transmit beamformer using the round-trip open-loop carrier synchronization technique described in [4] was built and tested at Worcester Polytechnic Institute, Massachusetts [16]. The source nodes were realized by using Texas Instruments TMS320C6713 digital signal processing starter kits (DSKs) [17]. All synchronization functionality was realized in software running in real time on the 225 MHz floating point digital signal processor. By using audio carrier frequencies and exploiting the built-in AIC23 stereo codec, a real-time proof of concept was built without custom hardware development.

A block diagram of the round-trip open-loop carrier synchronization prototype is shown in Fig. 7. A total of five TMS320C6713 DSKs were used to realize the system. Three of the DSKs were programmed to work as single-path channel simulators to facilitate repeatable simulation of time-invariant or time-varying channels. The remaining two DSKs were programmed to work as source nodes. Each source node simultaneously ran two PLLs, one for each analog channel.

Several distributed transmit beamforming experiments with unmodulated carriers and time-invariant and time-varying channels are reported in [16]. For time-invariant channels, convergence typically occurred in less than 5000 carrier cycles (at a frequency of 5.4 kHz), with a received power almost 99 percent that of an ideal beamformer. For single-path time-varying channels, sources moving at constant velocity suffer no performance loss, as long as the PLL filters are of at least second order.

Discussion and Conclusions

The results reviewed in this article indicate that distributed transmit beamforming is on the cusp of feasibility. The prototypes reported in the literature thus far have focused on demonstrating that the critical task of aligning carrier phases at the intended destination is feasible. The next step is to investigate and demonstrate distribut-

Protocols must be designed both for local coordination among the sources and for communication between the sources and the destination. The gains from distributed beamforming must, of course, be traded off against the overhead required to implement it.

ed beamforming in a networked context, with a detailed design that spans information sharing, timing synchronization, carrier frequency synchronization, and carrier phase alignment. Protocols must be designed for both local coordination among the sources and communication between the sources and the destination. The gains from distributed beamforming must, of course, be traded off against the overhead required to implement it.

The N^2-fold power gain provided by distributed transmit beamforming with N collaborating sources can be exploited in different ways, depending on the needs of the application. If each source is constrained in transmit power, collaboration can be used to increase the range beyond what is attainable by a single source, which can be exploited for extending network access in rural settings, for example. If the link budget is sufficient for a single source to communicate with the destination, collaboration can be used to significantly increase the rate of communication, assuming that the system operates in a power-limited rather than bandwidth-limited regime. This could dramatically increase the upload rate from a network of sensors or soldier radios. On the other hand, if a single source can already communicate with the intended destination at the desired rate and range, distributed beamforming can be employed to reduce the transmit power per source by a factor of N^2 and reduce the energy radiated in undesired directions, which can be exploited for energy efficiency in sensor networks or low-probability-of-intercept communication in military applications. While each application may require a different cross-layer protocol and physical layer design, we hope that this article has conveyed the fundamental issues that must be addressed by such a design.

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Cooperative Relay to Improve Diversity in Cognitive Radio Networks

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Abstract

Recent studies demonstrated that dynamic spectrum access can improve spectrum utilization significantly by allowing secondary unlicensed users to dynamically share the spectrum that is not used by the primary licensed users. Cognitive radio was proposed to promote the spectrum utilization by opportunistically exploiting the existence of spectrum "holes." Meanwhile, cooperative relay technology is regarded widely as a key technology for increasing transmission diversity gain in various types of wireless networks, including cognitive radio networks. In this article, we first give a brief overview of the envisioned applications of: cooperative relay technology to CRNs, cooperative transmission of primary traffic by secondary users, cooperative transmission between secondary nodes to improve spatial diversity, and cooperative relay between secondary nodes to improve spectrum diversity. As the latter is a new direction, in this article we focus on this scenario and investigate a simple wireless network, where a spectrum-rich node is selected as the relay node to improve the performance between the source and the destination. With the introduction of cooperative relay, many unique problems should be considered, especially the issue for relay selection and spectrum allocation. To demonstrate the feasibility and performance of cooperative relay for cognitive radio, a new MAC protocol was proposed and implemented in a universal software radio peripheral-based testbed. Experimental results show that the throughput of the whole system is greatly increased by exploiting the benefit of cooperative relay.

Introduction

The radio spectrum is a limited and valuable resource that is tightly managed by governments. Recent reports showed a significantly unbalanced usage of spectrum; with a small portion of spectrum (e.g., cellular band, unlicensed band) increasingly crowded, most of the rest of the allocated spectrum is underutilized. Spectrum utilization can be improved significantly by introducing primary-licensed users and secondary-unlicensed users and allowing secondary users to access *spectrum holes* unoccupied by primary users. Cognitive radio [1] was proposed as the means for secondary users to promote the efficient utilization of the spectrum by exploiting the existence of spectrum holes.

The main challenges to the efficient development of cognitive radio networks (CRNs) include *primary user detection and transmission opportunity exploitation*. Here we focus only on the latter issue, which means that after a spectral hole is identified, secondary users must exploit the transmission opportunity so as to maximize their own performance while not interfering with the primary users. Secondary users might be competing for the resource or cooperating to improve efficiency and fairness of resource sharing [2].

The recent study illustrated that large benefits can be gained from cooperation among different terminals. In the following, we first provide a brief overview of the application of the cooperative technology to CRN and then focus on a specific example that shows the advantages of cooperation relay to improve spectrum diversity.

Cooperative Transmission to Improve Spatial Diversity

Cooperative transmission — where the original idea comes from the basic relay model that consists of three terminals: a source S, a relay R, and a destination D — is well known as a powerful technology that combats signal fading due to multipath propagation in a wireless medium. By enabling a set of cooperating relays to forward received information, this regime exploits spatial diversity through cooperation among distributed antennas belonging to multiple terminals in wireless networks [3].

In the context of CRN, cooperative transmission can give rise to the following two different but basic scenarios.

1 Cooperative transmission between secondary users: In this scenario, a secondary user acts as a relay from the transmission of another (source) secondary node. General considerations that are valid for cooperative transmission can be applied here; the only difference is that secondary nodes continuously must sense the spectrum for possible transmissions by the primary users.

2 Cooperative transmission between primary and secondary users: In this case, secondary

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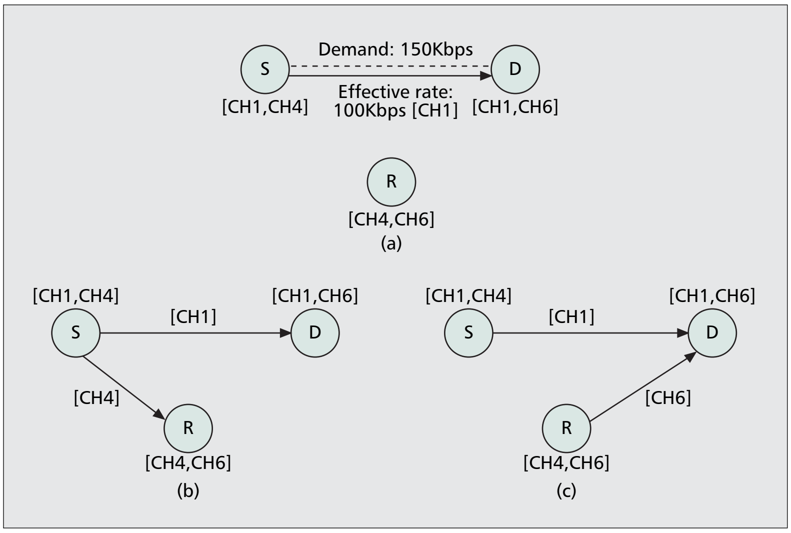


Figure 1. *Motivated example for cooperative relay to improve spectrum diversity: a) network setup and performance without cooperative relay; b) time slot 1 with cooperative relay scheme; c) time slot 2 with cooperative relay scheme.*

users can relay the traffic of a primary transmitter toward the intended destination. The rationale behind such a decision is that helping primary users finish their transmissions as quickly as possible will, in turn, lead to more transmission opportunities for secondary users [4].

We can see that cooperation transmission between secondary users aims to increase the secondary throughput for a given spectral hole, whereas cooperation transmission between primary users and secondary users aims to increase the probability of transmission opportunities. In summary, both of the above cooperation transmissions try to improve spatial diversity for the same spectrum frequency band.

Cooperative Relay to Improve Spectrum Diversity

The resource unbalance in CRNs is much more severe than in traditional wireless networks. The spectrum availability of secondary users is heterogeneous due to the location difference among different users, the dynamic traffic of primary users, and the opportunistic nature of the spectrum access of secondary users [5]. Moreover, the traffic demands of secondary users also can be quite different. Then, a natural yet important question is how to handle the unbalanced spectrum usage within the secondary network to fulfill the heterogeneous traffic demand from secondary users that has not drawn much attention before (Fig. 1).

Our observation is that some secondary users may not be required to use their entire available spectrum because of the low traffic demand. Utilizing these nodes as helpers, to relay the other secondary users' traffic with their otherwise wasted spectrum, can significantly improve system performance. In particular, suppose a transmission from *S* (with available channel *CH1*, *CH4*) to *D* (with available channel *CH1*, *CH6*) has 150 kb/s demand, but their common channel *CH1* can support only 100 kb/s. Meanwhile, a neighbor *R* of both *S* and *D* has abundant channels: channel *CH4* common with S and another channel *CH6* common with *D*. We can involve *R* as a helper in this transmission: while *S* and *D* still communicate on their original link over *CH1*, S sends additional data on *CH4*, and D receives on *CH3*, with R switching between *CH4* and *CH6* to relay data from *S* to *D*. In this way, the data rate between *S* and *D* is increased, and spectrum resources are efficiently utilized.

Starting from such a simple, yet interesting, observation, in this article we propose to use cooperative relay for CRNs with single-radio end users to more effectively utilize spectrum resources. However, the realization of this idea has several challenges:

- Traditionally, one radio can transmit or receive only at one channel at a specific time. The new relay-involved transmission raises a question from the start: how can such three-node, multiple-channel (probably discontinuous) transmission be possible? We propose and implement a new relay-assisted discontiguous orthogonal frequency division multiplexing (D-OFDM) scheme with which the receiver can receive one flow of data from the source directly; at the same time, the relay node can decode another flow of data transmitted on another channel and shift it to a third channel to forward it to the receiver.
- The cooperative transmission scheme brings in new issues of resource allocation. For a network of secondary users, we must address how to select the proper node as relay node and also how to allocate the proper spectrum for secondary users. These problems are coupled together. In the following section, we formalize the joint-relay-selection and spectrum-allocation problem and propose a heuristic algorithm to address it.

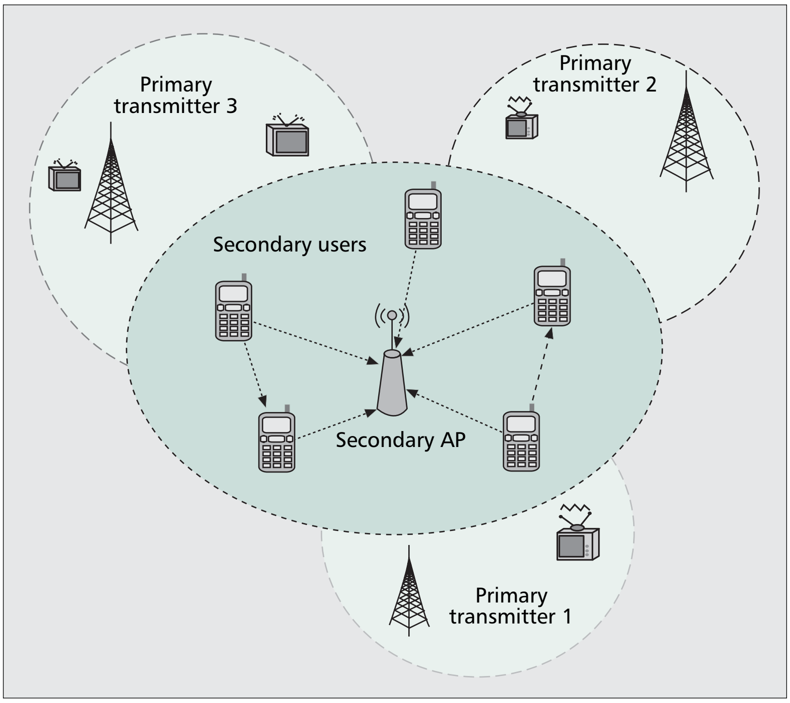
This work is the first one to explore the cooperative relay in the context of CRNs to improve spectrum diversity smartly by allowing spectrum-abundant nodes to help the spectrum-short ones.

System Architecture

In this section, we present an overview of the system architecture of a CRN with cooperative relay. In this article, we design the secondary system in an infrastructure mode as shown in Fig. 2: secondary end users, equipped with a single cognitive radio, connected to a local secondary access point (AP) to enjoy last mile connections. Primary users with different spectrums (channel 1 to 3, in this example) previously are deployed in the same region but are observed with quite low spectrum utilization. Secondary users can access these spectrum holes opportunistically, meaning they use the spectrum of the primary users only when it is not currently used by primary users. The secondary AP uses an OFDMA modulation scheme similar to the IEEE 802.22 standard [5]. Note that the spectrum availability of secondary users can be heterogeneous due to the location difference among different spectrum users, the dynamic traffic of primary users, and the opportunistic access nature of secondary spectrum users.

As shown in the previous motivated example, we can improve the throughput of secondary users by leveraging cooperative relays. However, to make such an idea practical, we must address several challenges to:

- *Signal processing*: The three-node cooperation requires a signal for a single packet to be transmitted from a single radio on potentially discontinuous channels to avoid harmful interference to primary users. This can be realized by the existing approach of D-OFDM technology [6]. However, the signal partition, shifting, and combination required by the introduction of a relay node bring special challenges. Therefore, in the next section, we develop a novel signal processing scheme, named relay-assisted, discontiguous OFDM, to address those challenges.
- *Resource allocation*: Here the resource includes relay and channel resources — adjacent secondary transmission (pairs) content for relays and secondary node (pairs and relays) content for channels. Thus, the allocation of both relay and channel resource is coupled together and must be addressed jointly. Moreover, the resource allocation must take the transmission demand of each end user into consideration. This problem has not been observed in previous papers about CRNs yet.
- *MAC-layer coordination*: The unique operation of secondary nodes also brings challenges in the medium access (MAC) layer. The MAC protocol must coordinate the signal forwarding and packet transmission. In addition, the primary users' signal detection is of fundamental importance for correct operation of secondary networks. We design our system as a synchronized system to ease these operations. The synchronization is easy to implement because the AP can make the coordination.



■ **Figure 2.** *System architecture of secondary CRNs with cooperative relays.*

Relay-Assisted, Discontiguous OFDM

In this section, we address key challenges during the realization of the three-node cooperation technique and present our solutions to overcome these challenges.

First, the sender must be able to transmit multiple packets on multiple channels at the same time using single-radio equipment. For this, we can simply adopt D-OFDM as the physical-layer technique, where signals on multiple channels can be transmitted simultaneously on single-radio equipment. Second, both relay and receiver should be able to alleviate the interference from other simultaneous transmitting channels to achieve a higher signal-to-noise ratio (SNR) on the specific channel. Third, relay and receiver should be able to decode the packet correctly using only some of all subcarriers that correspond to their working channel. We address these two challenges in the following two subsections, respectively.

Radio-Frequency Configuration

For the relay node or receiver, it is important to correctly decode the transmitted signal from multiple concurrently transmitted signals on multiple channels. To achieve this, it should filter out the signals on the working channel while it suppresses the noise and interference on other channels; this can be achieved by proper radio-frequency (RF) configuration.

Assume that the frequency band of channel 1 is from *fu1* to *fv1*, and the RF of the transmitter is set to be *f0* at the receiver end. If RF is set to be the same frequency as the transmitter, when we use a filter to keep the signals on [*fu1*, *fv1*], signals in the symmetric frequency band [2*f0*-*fv1*, 2*f0*-*fu1*] also are kept. However, there can be severe interference in this frequency band. Therefore, to avoid the interference in the symmetric channel, the receiver set the receiving frequency to be *f1*, which is on the right side of the whole bandwidth. Then, a bandpass filter is used to filter the signals in band [*fu1*, *fv1*]. Finally, the resulting baseband signal is multiplied by a sine signal of frequency *f1*-*f0* to be moved back into subcarriers [*u1*, *v1*].

Demodulation Using Part of the Subcarriers

Another key challenge for cooperative relay is that both the receiver and the relay should be able to decode the packet from a fraction of all subcarriers that correspond to the working channel. After the signals are filtered out, there are many other important functions that must be performed, including time synchronization, frequency alignment, and channel estimation. All these are performed based on the preambles added before the packet. Different from traditional OFDM, we must add preambles individually on each group of the subcarriers.

Each MAC frame includes three parts: control-information exchange, downlink transmission, and uplink transmission. The length of the frame is fixed and is properly selected so that it can provide the required protection for a primary user by periodical spectrum sensing and can prevent high transmission delay.

Based on the preambles added before each channel, we use the delay-and-correlate algorithm to detect the beginning of the packet and use the cross-correlation-based algorithm to conduct symbol synchronization. Frequency error estimation is performed after the fast Fourier transform (FFT) module using the frequency domain algorithm to align the carrier frequency offset. Frequency domain channel estimation is conducted to overcome the influence of the wireless channel. Given the assumption that the channel is quasi-stationary and does not change during the time of transmitting a packet, we do channel estimation once for each packet. The implementation in the real testbed, introduced later, demonstrates that the proposed relay-assisted D-OFDM is feasible and can achieve significant throughput gain.

Joint Relay Selection and Channel Allocation

In this section, we focus on a network perspective: how secondary nodes in a CRN with cooperative relays coordinate with each other to allocate the relay and spectrum resource.

The relay selection issue emerges when there are multiple resource-short nodes and resource-abundant nodes. We must make a decision of how to match resource-short nodes to their helpers. In addition, the channel allocation issue inherited in multi-channel networks still exists for both direct and relay links. What further complicates the problem is that these two issues are coupled. This is because relay selection has an impact on the channel allocation and vice-versa.

Suppose time is partitioned into time frames with length Δt. In any frame, there can be two types of transmissions going on in the network. One is the traditional transmission between the AP and a node vi. The other one is the advanced transmission among three nodes (AP, vi, and its relay vj). We should make a decision in each frame on how to arrange the active transmissions into these two types. The detailed formulation for such a resource allocation problem can be found in a technical report [7], which is an NP-complete problem.

To tackle that problem, we propose a heuristic solution based on the observation that *if one user's demand is not fulfilled, it will not act as a relay to help others because this definitely decreases the total throughput of the whole system* (formal proof can be found in [7]). Based on this observation, we first partition nodes into two parts according to their traffic and spectrum availability. Then, we greedily select the best pair of destination and relay from the two parts to increase the system throughput.

MAC Design for Relay-Assisted CRNs

The previous sections present the feasibility of leveraging a helper to improve the throughput of a CRN. Then, we design a MAC protocol to coordinate physical (PHY)-layer operations among multiple nodes. Considering the existence of APs, global synchronization is rather easy to implement. The time is divided into time frames. At the start of each frame, secondary nodes with a single radio switch their radios to the common control channel to exchange control messages and negotiate the resource allocation, including both relays and channels. Then, in the remaining time of a frame, they switch to their assigned channels to conduct data transmission.

Frame Structure

Each MAC frame includes three parts: control-information exchange, downlink transmission, and uplink transmission. The length of the frame is fixed and is properly selected so that it can provide the required protection for a primary user by periodical spectrum sensing and can prevent high transmission delay.

Primary User Detection

With frame synchronization, primary-user detection can be controlled easily and provided with high accuracy compared to cases without synchronization. The detection process is put at the beginning of the information collection period. During the process, each secondary node is silent and senses the spectrum. Different sensing methods (energy detection, feature detection) can be applied.

Coordination

At the beginning of every frame, the AP sends a frame-control header on a common control channel. Nodes receive this header and synchronize with the AP. After that, a short period is used to collect information about the data demand and spectrum availability from each end user. Random access is used here. When the information is collected, a centralized algorithm at the AP is executed to allocate resources for all the secondary end users. Then, the allocation decision is broadcast on the control channel.

Data Transmission

After receiving the resource allocation message, each node adheres to the allocation decision made by the AP for downlink and uplink transmission. During the downlink time of one frame, if one node is assigned to communicate directly with the AP, it switches to the assigned channel to receive the transmission of the AP. For the relay node, it uses half of the downlink time to receive data from the AP and the other half of the time to forward data to the destination. For the node assisted by a relay, it spends the whole downlink time to receive data from the AP and uses half of the time to receive data from a relay on another channel.

The uplink case is similar to the downlink one. The duration of downlink and uplink can be determined by the demand adaptively.

Experimental Evaluation

In this section, we set up a CRN testbed and implement our proposed relay-assisted D-OFDM, resource allocation algorithm, and MAC protocol. The purpose of this section is to demonstrate:

- An infrastructure-based CRN testbed set up. The core technologies to support cooperative relay were implemented through this testbed.
- Under various numbers of users and various traffic demands, our proposed cooperative relay scheme can consistently achieve good performance. The effectiveness of relay selection and channel allocation was verified through the experiments.

Architecture for the CRN Testbed

We design a cognitive radio system with the architecture shown in Fig. 3. Each node is composed of an open-source, reconfigurable-RF front end connected to a general-purpose computer in which the relay-assisted D-OFDM, resource allocation algorithm, and MAC functions are implemented in software. Each node serves upper-layer requests and interacts with the radio spectrum, with the input of spectrum usage rules from the external radio-spectrum management entity.

Reconfigurable Physical Layer — The whole system is based on a reconfigurable hardware platform, and the physical layer is based on the platform of the universal software radio peripheral (USRP) and GNU radio. The USRP [4] recently became a popular experimental platform for wireless communication research projects. It implements front-end functionality and A/D and D/A conversion. RFX2400 daughter boards are used in our work, which operate in the 802.11 frequency range (i.e., 2.4 GHz) and have a transmit power of 50 mW (17 dBm). GNU radio [8] is an open-source toolkit for building software radios. It is designed to run on a PC; combined with minimal hardware, it enables the construction of simple software radios. GNU radio with version 7246 is adopted in our work, in which relay-assisted D-ODFM is implemented. With such a physical layer, we freely can adapt multiple dimensions of data transmission (power, frequency, and modulation).

Spectrum Information — This part is responsible for collecting and managing the spectrum information from the internal physical layer and for external radio spectrum management. The GNU radio of the internal physical layer can provide interfaces to deliver spectrum status, including both spectrum availability and channel condition, measured by the USRP.

MAC Layer and Network Layer — The MAC layer component is a self-developed part of the architecture. Depending on the detailed system configuration, the MAC protocol can be implemented in a standalone module, as a part of the GNU radio, or as a part of a routing layer. We leverage the work of the Click project as the routing part, which is built from fine-grained components, called elements, which perform packet processing.

Cross-Layer Management — Our cross-layer management component interfaces with other layers to collect network information and also to receive spectrum policy information from the radio-spectrum-management component. With such information, the cross-layer management component adjusts operation parameters to optimize network performance according to certain communication objectives.

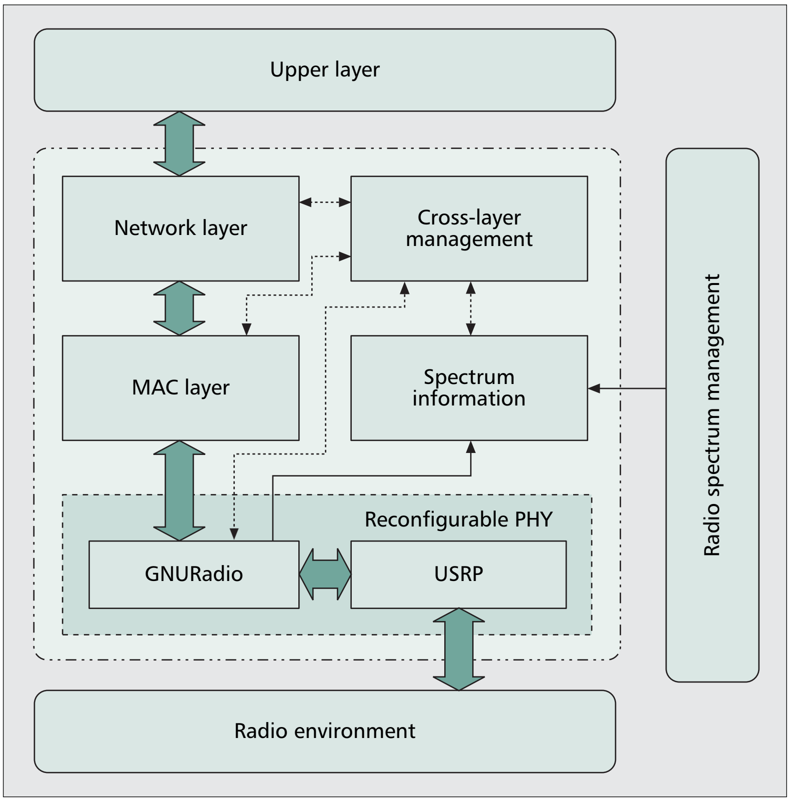


Figure 3. *CRN testbed architecture.*

Performance Evaluation

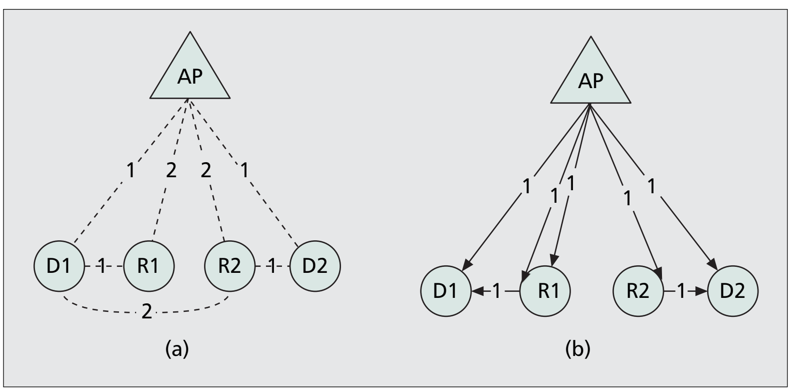
The CRN consists of a secondary AP and several secondary users (Fig. 4). We evaluate a relay scheme in different network configurations. Due to space limitations, only the topology with four nodes as shown in Fig. 5a is presented. Here, the numbers on the dashed lines indicate the number of commonly available channels between two connected nodes. The traffic demands are 100 kb/s, 100 kb/s, 30 kb/s, and 0 kb/s, for node *D1*, *D2*, *R1*, and *R2*, respectively. The link between AP and *D1* via relay *R2* can support a data rate equal to one entire channel. However, if R2 is used by *D1*, it has too much bandwidth compared to the demand of *D1*, whereas *D2* has no relay to fill its spectrum requirement. With our algorithm, an optimal transmission decision is made as shown in Fig. 5b: *D1* uses *R1* as relay, and *D2* uses *R2* as relay.

Figure 6 shows the cumulative distribution function (CDF) of throughput gains for this topology. Our scheme provides a 35 percent (in theory, 40 percent) increase in throughput, which comes from *D1* and *D2*, with 44 percent and 43 percent, respectively.

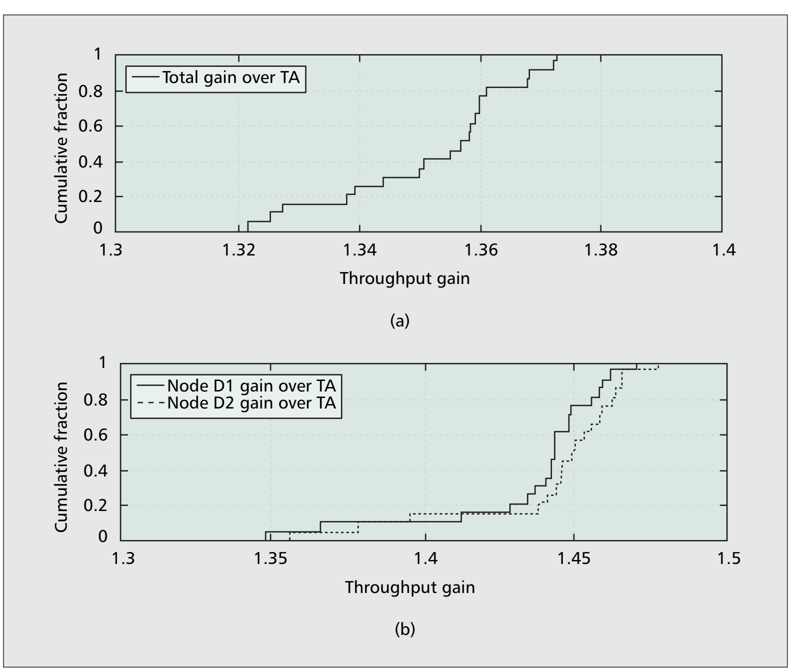
Further Discussion

This article demonstrates the key concept of using cooperative relay for CRN. As this is a new research direction, additional research topics can be explored from this starting point.

■ **Figure 4.** *A photo of our real testbed.*



■ **Figure 5.** *Experiment for four end-users: a) topology with channels; b) optimal transmission.*



■ **Figure 6.** *CDF of throughputs for topology with 4 end-users: a) total throughput gain; b) throughput gain of nodes* D1 *and* D2*.*

TRANSMISSION CONSTRAINTS

This work imposes several assumptions on the secondary users' transmission that can be relaxed in future research. For example, we limit the secondary users' transmission to be exclusive for each channel, meaning that within each time slot, a channel can be used by only one active transmission pair. System performance can be further improved if we allow simultaneous non-interfering transmissions among several distant destinations and relay pairs.

COMMON CHANNEL

So far, we assume that there is always a common channel for the control message exchange between the AP and the end users, for example, industrial, scientific, and medical (ISM) bands. When this is not the case, we must change the control channel dynamically according to spectrum availability. Another way is to use an OFDMA scheme such that each node uses its own channel common with the AP for control message exchange.

MULTIPLE RADIOS

In our article, we assume all nodes are equipped with a single cognitive radio. This is reasonable considering the cost and size of end users. However, if this is not a concern, better spectrum utilization can be achieved. For example, if each node has two cognitive radios, a relay node can act as a pipeline between the AP and its relayed destination, with one of its radios communicating with the AP and the other one communicating with the destination on different channels simultaneously. Therefore, channels in the relay node are fully used, in contrast to half the effective time of the current design.

AD HOC COGNITIVE RADIO NETWORK

Only infrastructure mode is discussed in this article. The cooperative relay concept also can be extended easily to ad hoc networks. In that case, multiple-source-destination pairs can exploit more spectrum channels with the help of relay nodes. However, coordinating ad hoc nodes is much more difficult than using the infrastructure mode. Synchronization, the multichannel hidden terminal problem, and distributed-spectrum sensing are several important issues that must be discussed.

CONCLUSIONS

Cooperative transmission appears to be a promising approach for improving the throughput of secondary nodes by increasing the spatial diversity and spectrum diversity.

In this article, we gave a brief overview about some of the aspects of the interplay of cooperation and cognitive radio technologies. Facing the challenges brought by the heterogeneity in spectrum availability and also the traffic demand for secondary users, we explored the use of a cooperative relay node to assist the transmission of CRNs and improve spectrum efficiency. We observe that spectrum resources can be better matched to traffic demand with the help of a relay node. As the first work about exploiting spectrum resources using relay nodes for CRNs, we focus on the design and implementation in the infrastructure mode. To achieve this, we address several issues. First, we propose relay-assisted D-OFDM for data transmission as the fundamental component for the whole system. Second, we identify a new resource-allocation problem of joint-relay selection and channel allocation that is NP-hard. Finally, we design a practical MAC protocol to coordinate the relay and spectrum allocation. To demonstrate the feasibility and performance of cooperative relay for CRNs, we implement a testbed consisting of the USRP and GNU radio. Experimental results confirm a significant gain compared to traditional transmission.

Leveraging a cooperative relay to improve spectrum diversity is a totally new research direction for CRNS; some interesting future research topics also are discussed.

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Biographies

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Link-Layer-and-Above Diversity in Multihop Wireless Networks

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Jian Zhang and Ivan Marsic, Rutgers University

ABSTRACT

The instability of wireless channels was a haunting issue in communications until recent exploration in utilizing variation. The same transmission might present significantly, and usually independently, different reception quality when broadcast to receivers at different locations. In addition, the same stationary receiver might experience drastic fluctuation over time as well. The combination of link-quality variation with the broadcasting nature of the wireless channel itself disclosed a direction in the research of wireless networking, namely, the utilization of diversity. In this article, we summarize the causes of channel diversity in wireless communications, and how it is perceived in different layers of multihop wireless networks. To promote new research innovations in this area, we concentrate on link-layer diversity and speculate on the challenges and potential of diversity schemes at the network layer.

INTRODUCTION

A multihop wireless network, mobile or stationary, poses a challenge in network protocol design. In particular, the error-prone communication links and the unstable network structure are two of the most critical aspects in networking. Numerous efforts have been exerted to address these issues so that a multihop wireless network could be *as good as* a wireline network. In contrast, interest is increasing in utilizing a wireless communication channel by harnessing its broadcasting nature directly. Indeed, it is this nature that separates wireless networks from the rest, and no requirement exists to turn wireless links into wired lines. Only by a direct approach can we make full use of these networks and make wireless networks *better than* wireline networks. Any real-world operating environment of a multihop wireless network inevitably causes different levels of variation in link quality. One salient feature of such random fluctuation is its finer time granularity when compared to the response of a global (end-to-end) solution. Therefore, this is an issue that is unique to this type of network, and localized and dynamic cooperation among relaying nodes opens up a way for us to address this issue. In this article, we speculate on the problem of how to utilize channel diversity at the link layer and above. By reviewing the typical approaches in the literature and focusing on two recent explorations, we investigate the challenges involved and describe-existing solutions.

DIVERSITY IN WIRELESS NETWORKING

Diversity in wireless networking, sometimes called *channel diversity* or *link diversity*, refers to the phenomenon where transmissions at different channels, for example, frequency band, time slot, and so on, possess different reception conditions. A diversity scheme utilizes such a phenomenon for more reliable transmission. Fundamentally, the complex of electro-magnetic wave propagation generally can be attributed to such mechanisms as reflection, diffraction, and scattering.

Considering the basis, treatments, and effective scope, we review the primary forms of diversity schemes in wireless communications as follows: at the physical layer (the first three), at the link layer (the fourth), and a network-layer effect (the last).

Time Diversity — A wireless communication system inevitably is operated in a dynamic environment due to the mobility of both the transceiving parties and any obstacle. Thus, the channel gain is a stochastic process centered at a mean value. That is, instances of transmission at different times may have significantly varying levels of attenuation even if the transmitter and receiver are both stationary. At the extreme, such variance can be observed even within a single transmission. To combat this, identical messages can be transmitted multiple times for better robustness. Alternatively, forward error correction (FEC) coding can be used to spread information over a longer period of transmission time. This is the first form of channel diversity utilized in communications.

Frequency Diversity — Propagation of signals at different frequencies experience differences in reflection, diffraction, and scattering, even at the same time and location. Therefore, practically

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any wireless channel is affected by *frequency-selective fading*, that is, channel gain varying with frequency. Countermeasures to this include simultaneous transmission over multiple subcarriers (e. g., orthogonal frequency division multiplexing [OFDM]) and spreading information in a wider frequency band (e. g., direct sequence-code division multiple access [DS-CDMA]).

Space Diversity — Typically, between a transmitter and a receiver, there are multiple paths for the signal to propagate, whether there is a line-of-sight (LOS) component or not. In addition, the composition of these propagation paths relies on the exact positions of the transmitter, receiver, and all obstacles. Thus, a small change of the position of any of them can vary the channel gain significantly, which is *small-scale fading* in the spatial sense. In contrast, time diversity is a temporal sense of small-scale fading. To utilize space diversity, we can employ multiple transmitters (i.e., transmitter diversity) or multiple receivers (i.e., receiver diversity) for joint transmission of the same message. Multiple-input and multiple-output (MIMO) and space-time coding (STC) are examples using this technique. Depending on the distances among the transmitters or those among the receivers, relative to the wave length of the signal carrier, space diversity can be further classified as *microdiversity* and *macrodiversity*.

Multi-User Diversity — In a wireless network of multiple downlinks or multiple uplinks, or multiple transmitter-receiver pairs in general, scheduling and channel selection can be executed such that the users of the "best instances" are favored to best exploit the channel variation. Thus, the overall system throughput increases with the number of users and channel gain variance. A consequence of utilizing multi-user diversity is that the interface queues are not first-in-first-out (FIFO) any more.

Multipath Diversity — In a multihop wireless network, a given pair of source and destination can be connected through multiple (network-layer sense) paths in the network. The properties of these paths vary in many ways, such as hop length, bandwidth, total delay, queuing delay, expected transmission count, and so on. They are further induced and synthesized from the diversity of the links among these paths. In general, multipath diversity bears a global notion, and it takes the network a longer time to react.

Perception of channel diversity can be made at the physical, data link, and network layers. As we have noted, time diversity, frequency diversity, and space diversity are physical layer notions; multi-user diversity is a link layer one; whereas multipath diversity is a network-wide effect. Thus, diversity as a scheme for data transportation can work at any one or at a combination of these layers.

Physical layer diversity schemes are specific for different causes and thus, usually are addressed more directly. Due to the relatively simple solutions, time and frequency diversity schemes were adopted widely. In contrast, space diversity, especially at the macro level, only recently attracted an increasing number of research activities, collectively referred to as *cooperative communication* [1]. A major reason behind this thrust is the enhancement of the digital signal-processing capabilities that mobile devices possess. At the link layer, link variation usually is induced by historical transmission statistics on the sender side or collected and fed back from the receiver side. A link-layer diversity scheme typically takes measures by regulating link-layer behaviors, for example, parameters for medium-sharing control. An advantage of link-layer diversity is its lower requirement for hardware capabilities. On the other hand, its inductive nature can make it less responsive and timely in decision making. At the network layer, where network-wide routes are accounted, a path metric is always a cumulative quantity and thus takes a longer time to collect and respond to. In addition, these quantities are changing dynamically with the composite link metrics. Therefore, it is generally perceived to be difficult to utilize multipath diversity in multihop networks lacking global information and central control authority, even though potentially, it can help us to improve network performance.

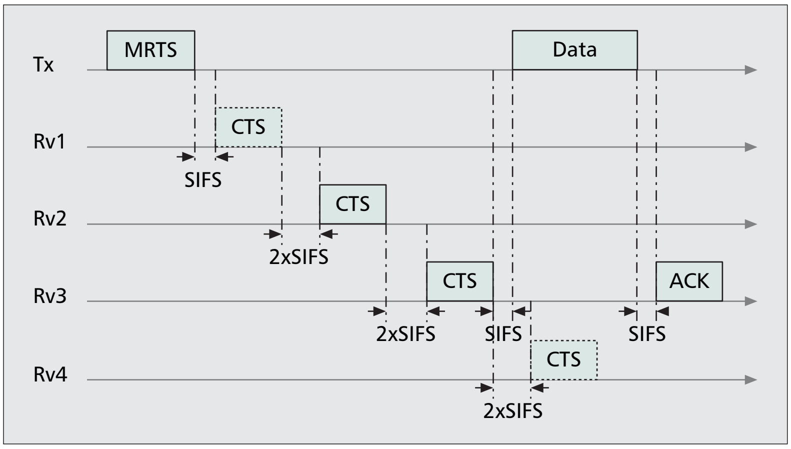
As a result, in this article, we focus on link-layer efforts — and through more explorative endeavors, on others, higher and above, based on the link layer — in exploiting channel diversity in the quest for diversity.

Space diversity, especially at the macro level, only recently attracted an increasing number of research activities, collectively referred to as cooperative communication. A major reason behind this thrust is the enhancement of the digital signal-processing capabilities that mobile devices possess.

Link-Layer Diversity in Multihop Wireless Networks

Here, we summarize existing link-layer diversity schemes in wireless networks. Because they were proposed in differing contexts, they may carry different names in the literature, such as selection diversity, multicast/group request-to-send (RTS), opportunistic scheduling, link-layer anycast, and so on.

Multi-user diversity first was addressed as a link-layer scheduling scheme by Knopp and Humblet [2] in cellular communication networks and later, was incorporated in CDMA systems. In such centralized systems, the channel-quality information is fed back from users in the cell through an uplink so that the base station can schedule transmissions to the favored users accordingly. In a multihop wireless network, where there is usually no central control authority at the link layer, it requires effective and efficient distributed coordination in transmission. Larsson [3] proposes an innovative handshake and selection diversity forwarding (SDF) to implement downstream forwarder selection in a multihop wireless network, where multiple paths are made available by the routing agent. In this case, a sender in the network dynamically can choose from a set of usable downstream neighbors that presents the lowest transient cost in forwarding the packet. For the sender to make the decision, the IEEE 802.11 distributed coordination function (DCF)-based DATA/ACK handshake is enhanced in two aspects. First, the receiver address (RA) field in the DATA frame is augmented to contain all eligible downstream neighbors. After the reception of DATA, these



■ **Figure 1.** *MRTS in as in [5].*

neighbors each respond with an ACK in the order prescribed by the RA field, interleaved by the short interframe space (SIFS) to avoid interruption. The ACK frame in this case also carries additional information such as link quality and queue length. Second, after collecting the ACKs from the downstream neighbors and selecting the best neighbor as forwarder, the sender transmits a forwarding order (FO) frame, addressed to that neighbor, which in turn responds with a forwarding order ACK (FOA) to confirm the order. Such a four-way handshake is the first explorative link-layer diversity scheme in multihop wireless networks.

Recently, the exploration of link-layer diversity in multihop wireless networks has attracted considerable research attention. In addition to multi-user diversity, it also was used to address such issues as head-of-line (HOL) blocking and opportunistic rate adaptation. These proposals are built upon the RTS/clear-to-send(CTS)/DATA/ACK four-way handshake of IEEE 802.11, given its predominance and availability in the area of multihop wireless networking, and are collectively referred to as multicast RTS (MRTS). Larsson and Johansson [4] refine SDF to accommodate packet forwarding for multiple flows in the network in their proposal of multi-user diversity forwarding (MDF). In MDF, a combination of data rate, forwarder, and flow is considered in the selection by the sender, thus the non-FIFO queuing. The implementation also adopts a preceding placement of the control frames that is more 802.11-compliant, as opposed to the trailing placement in SDF. Jain and Das [5] design a *link-layer anycast* to implement multipath routing, faithfully based on the IEEE 802.11 specifications. This is achieved by augmenting the standard RTS frame to MRTS that contains multiple RAs to poll them. Upon the reception of MRTS, the ith polled node backs off by $(2i - 1) \times T_{SIFS} + (i - 1) \times T_{CTS}$ before transmitting a CTS. After the sender has received a CTS, it unicasts a DATA frame after time SIFS (T_{SIFS}), a shorter time than what the next CTS requires to back off, to interrupt any additional CTSs from subsequent receivers. The receiver of DATA acknowledges it with an ACK after T_{SIFS} as well. Figure 1 provides an example of how it works. In the scenario, one node (Tx) has four downstream neighbors: Rv_1, Rv_2, Rv_3, and Rv_4. Assume that the MRTS is intended for all neighbors but was received correctly only by Rv_2, Rv_3, and Rv_4. After $3T_{SIFS} + T_{CTS}$, Rv_2 replies with a CTS that is garbled when Tx receives it. After $5T_{SIFS}$ + 2TCTS, Rv_3 replies with a CTS. After correct reception and a back off of T_{SIFS}, Tx transmits DATA. This cancels the CTS reply from Rv_4. The transmission is completed by the Rv_4 ACK after T_{SIFS} receives DATA. In a simultaneous investigation, Wang, Zhai, and Fang [6] specify an opportunistic packet scheduling and media-access control (OSMA) protocol to address the HOL blocking problem. HOL blocking occurs when the frame that is currently at the head of the interface queue at the sender's link layer cannot be transmitted successfully, for example, due to the temporary unavailability of the receiver. One salient feature of this protocol is the shorter back-off time of CTS transmission as a result of receiver-carrier-sensing capability. Thus, after the reception of MRTS, the ith polled receiver backs off by a shorter time of $T_{SIFS} + (i - 1) \times T_{slot}$ before transmitting a CTS. In this proposal, only one CTS is in fact transmitted, that is, by the first receiver in the ordered list that is able to respond; all remaining receivers yield to the upcoming DATA frame despite the fact that it may have received the MRTS successfully. In the context of rate adaptation in wireless LANs, Ji et al. [7] present an MRTS-based opportunistic scheduling with packet concatenation, called medium-access diversity (MAD). In MAD, when a high data rate is selected by the access point (AP), the AP can concatenate multiple frames into a longer DATA frame that lasts for approximately the same duration of sending a single frame at the basic rate. This effectively brings down the overhead-to-payload ratio at the link layer. On the other hand, the polling time in MAD is longer because the AP must wait for $n \times (T_{CTS} + T_{SIFS}) + T_{SIFS}$ before it transmits the DATA frame, where n is the number of receiver addresses in the MRTS. Zhang, Chen, and Marsic [8] improve MRTS to address HOL blocking by further reducing its operation overhead. In particular, a sender intelligently composes a shorter list of receivers for the MRTS using a learning module. The learning module selects a subset of the eligible downstream nodes that has the least correlation in channel condition, based on its recorded transmission history. A shorter list, and thus a smaller overhead, is shown to have the same likelihood of having at least one receiver available as a list that includes all eligible nodes. This approach is more efficient than the somewhat "blind" inclusion of all eligible receivers and thus, carries the exploitation of link-layer diversity one step further.

EFFICIENT COORDINATION

The implementation of link-layer diversity for the layer above the physical layer requires more sophisticated and efficient coordination. It is particularly challenging for multihop wireless networks that are void of centralized control authorities. Multi-user diversity, as a form of link layer diversity, is effective and became feasible because cell base stations can provide central intelligence and control. In contrast, a multihop wireless network operates in a more flexible setting, which introduces an entire spectrum of networking issues. The most predominant medium access control (MAC) for such networks is the IEEE 802.11 DCF. This type of MAC protocol essentially is a carrier-sensing multiple access with collision avoidance (CSMA/CA) scheme. IEEE 802.11-based networks usually operate with a simple two-way handshake of DATA/ACK frames between the sender and receiver. The optional RTS/CTS control frames are used to precede a DATA/ACK to address the *hidden terminal problem*, where two transmitters are out of carrier-sensing range of each other. The reason that such a four-way handshake usually is not preferred is its high overhead. Although these optional frames are short, they must be transmitted at the basic data rate to be robust. On the other hand, the physical layer module of an 802.11-compliant device is capable of transmitting the DATA frame at different data rates using different coding and modulation schemes, with the highest being many times faster than the basic rate. As a result, these control frames, along with the inter-frame spaces, impose a significant amount of communication overhead. The higher the data rate used to transmit the DATA frame, the lower the payload-to-overhead ratio is. Observe that all the MRTS-based link-diversity schemes are in fact extending the optional RTS/CTS to a mandatory poll-and-select paradigm. Despite their very explorative nature, these MRTS-based protocols should be further improved for better system performance.

Of course, this is easier said than done. In a unicast routing protocol for multihop wireless networks, a sequence of relaying nodes are enlisted by the routing module to forward the packet. A multipath routing protocol makes multiple paths available for a given pair of source and destination, mostly for better utilization of link-layer diversity. However, any given packet still follows a single path among these candidate paths though this type of path can vary from packet to packet for the same source-destination pair. The challenge is to ensure that a packet is forwarded by exactly one of the five eligible downstream relaying nodes with a minimum of extra system overhead. In addition, the original function of link-layer reliability still should be guaranteed. Link-layer diversity is built on the broadcasting nature of wireless channels. Thus, any eligible relay (e. g., as prescribed by the routing module) with good transient channel quality could potentially forward the packet. Without poll-and-select, the multitude of the relays with good channel conditions must coordinate with each other such that exactly one of them forwards the packet. And this is to be accomplished without introducing additional control frames. Apparently, this is looking one layer up, to the network layer, for help.

Link Layer and Above

When seeking diversity above the link layer, it is critical that the solution be sufficiently agile in response to the very fine time granularity of link variation. Thus, a dynamic and localized cooperation mechanism is imperative. The coordination at link layer and above was studied in a few different ways recently. We review Ex opportunistic routing (ExOR) [9] and [10] and two examples of innovation in quest for network performance. ExOR is a cross-layer protocol and blends the scheduling functionality of the link layer with the route selection functionality of the network layer. BEND is a lightweight link-layer solution that peeks at network layer information in its diversity-driven forwarder selection.

ExOR: Cross-Layer Opportunistic Forwarding

ExOR is an explorative cross-layer opportunistic forwarding technique in multihop wireless networks by Biswas and Morris. It fuses the MAC and network layers so that the MAC layer can determine the actual next-hop forwarder *after* transmission depending on the transient channel conditions at all eligible downstream nodes. Nodes are enabled to overhear all packets transmitted in the channel, whether intended for it or not. A multitude of forwarders can potentially forward a packet as long as it is included on the forwarder list carried by the packet. Thus, if a packet is heard by a listed forwarder closer to the destination with a good reception condition, this long-haul transmission should be utilized. Otherwise, shorter and thus more robust transmissions always can be used to guarantee reliable progress. The challenge is to ensure that exactly one of the listed forwarders relays the packet that is likely to be the closest to the destination at the same time. This is addressed by prioritized scheduling among the listed forwarders according to their distance to the destination. ExOR was tested on a 38-node mesh testbed, called MIT Roofnet, and shows significant performance gain compared to conventional packet transportation.

Route calculation in ExOR is essentially link-state-based source routing, where every node has global topology information. Each link between a node pair is associated with a quasi-static weight. Based on the cost of the links, each node executes a shortest path algorithm to obtain the "distances" to all other nodes in the network. The distance information is utilized later by a node as source to determine the priorities among intermediate nodes in helping to forward packets to a destination. ExOR operates in batches, where a set of packets from the source are processed collectively and cooperatively en route to achieve a small amortized per-packet overhead. For a given batch, 90 percent of packets are transported by opportunistic forwarding; whereas the remaining 10 percent are transported using conventional routing to clean up. In ExOR, the six MAC and network layers are tightly coupled, in that the forwarders as routing entities participate in packet scheduling directly. A higher-priority node in a batch backs off by a shorter delay than a lower-priority node before transmitting what it has overheard for the batch. Whenever a packet is forwarded, it carries a *batch map*. This map describes, for each packet in the

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Essentially, BEND considers the union of all of the interface queue contents at the nodes within a neighborhood, that is, a "neighborhood" coding repository; whereas traditional mixing methods only process "individual" coding repositories at separate nodes.

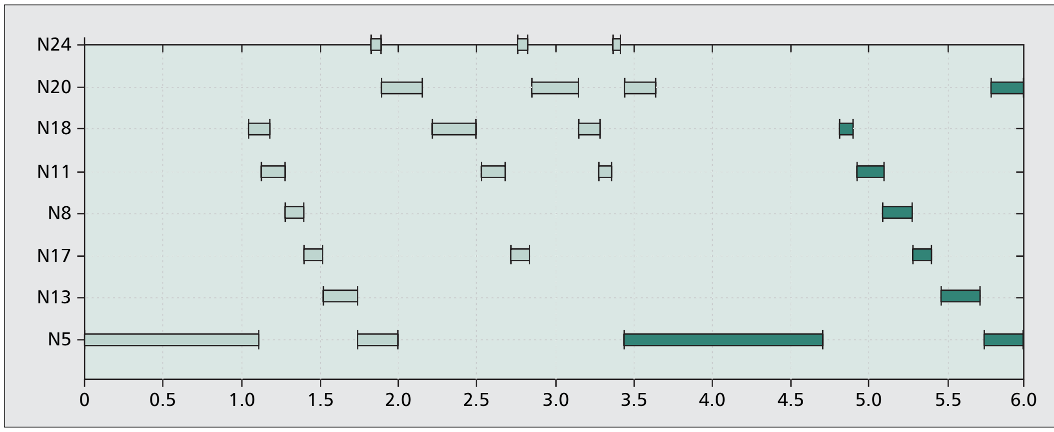


Figure 2. *ExOR.*

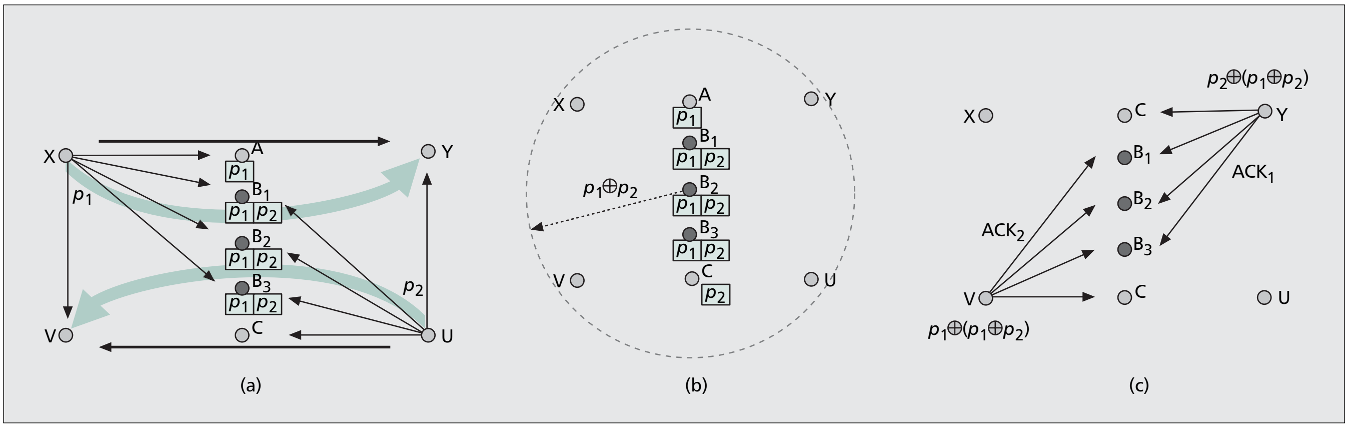
same batch, the highest-priority forwarder that this packet has reached, as a progress indicator, to the best of the forwarder's knowledge. Whenever a packet is overheard by a listed forwarder, its batch map is used to update the forwarder. Consequently, forwarding a packet downstream also serves as acknowledgment upstream. Figure 2 [9] depicts a timeline for transporting a batch of packets, followed by a partial second batch using opportunistic forwarding. In this scenario, node N_5 has packets to send to N_{24}. Nodes N_{24}, N_{20}, N_{18}, N_{11}, N_8, N_{17}, and N_{13} are the forwarders listed by the source N_5, as indicated by their relative positions on the y-axis. The complete first batch is indicated by the bars with a lighter shade and the partial second batch by the darker shade. The horizontal length of each bar corresponds to the number of packets transmitted by the node. At the beginning, node N_5 transmits the entire batch, some of which can reach as far as node N_{18}, with others falling short at closer nodes. Each of these forwarders relays packets that it has overheard but that have not been relayed by a higher-priority forwarder in the order dictated by the forwarder list. It takes node N_{24} three acknowledgments, that is, about 3.5 seconds, to finish transporting 90 percent of the batch. The remaining are forwarded using conventional routing, which is not depicted in the figure.

ExOR is more efficient overall than any of the MRTS-like protocols, not only because it does not use any additional control frames but also because the acknowledgment is piggybacked by the batch map carried by DATA frames. In this sense it is very innovative and has been shown to be very effective in a practical sense. To implement it, the packets must carry the forwarder list and batch map, which introduces communication overhead. When the network becomes large, this overhead inevitably increases to provide a full path with sufficient redundancy. After all, it is a source-routing protocol. In addition, the timing in delaying forwarding packets among the listed forwarder should be carefully engineered, which can be particularly hard in channels with significant variation. That said, it is even harder for a network to accommodate multiple simultaneous flows. Its cross-layer scheme gives the designer much more control to implement the above, but also blurs the boundary between the link and network layers. As a result, it will be hard to use ExOR in different device platforms. Nevertheless, ExOR is a definite eye-opener for a new communication paradigm in multihop wireless networks.

BEND — Proactive Packet Mixing

BEND is a MAC layer solution to practical network coding, originally proposed by Zhang, Chen, and Marsic to enhance the likelihood of the coding of packets in different flows in the proximity of a multihop wireless network. It is an exploration of the broadcasting nature of wireless channels to proactively capture more coding opportunities. In BEND, any node can code and forward a packet even when the node is not the intended MAC receiver of the packet if the node senses that in doing so it can lead the packet to its ultimate destination. Essentially, BEND considers the union of all of the interface queue contents at the nodes within a neighborhood, that is, a "neighborhood" coding repository; whereas traditional mixing methods only process "individual" coding repositories at separate nodes. Moreover, BEND is designed such that, if there is no network coding possible among multiple flows, it still can use multiple helping forwarders to utilize link-layer diversity effectively. It is light-weight, IEEE 802.11-compliant, and can support different routing protocols. It works because the ray of light bends in the presence of a gravitational field and thus, derives its name.

The basic operation of BEND is illustrated by a simple example in Fig. 3 although BEND works under more general conditions. In Fig. 3a, node X has packet p_1 for node Y that is two hops away, and U has p_2 for V, also two hops away. The forwarders determined by the routing protocol are nodes A and C, respectively. We assume that three other nodes, B_1, B_2, and B_3, are also in the range of X, Y, U, and V. When a packet, for example, p_1 or p_2, is handed from the network layer down to the MAC layer, its header is enhanced to include not only the address of the next-hop node but also that of the following-hop node. Such information can be obtained from the routing module. After the packets, p_1 of node X and p_2 of node



■ **Figure 3.** *BEND.*

U are transmitted, p_1 is received by nodes *A* (intended forwarder), B_1, B_2, B_3, and *V*, and p_2 is received by B_1, B_2, B_3, *C* (intended forwarder), and Y. Packet p_1 is placed in the queues of nodes *A*, B_1, B_2, and B_3 because they are all neighbors of the p1 second-next hop (node *Y*) as indicated by the packet header. It is, otherwise, buffered by *V* for future decoding. Similarly, p_2 is queued at nodes B_1, B_2, B_3, and *C* and buffered at *Y*. Nodes B_1, B_2, and B_3 can choose to transmit $p_1 \otimes p_2$ if they determine that the coded packets can be correctly decoded by their second-next-hop neighbors. All of the intermediate nodes *A*, B_1, B_2, B_3, and *C* could forward the packet(s) in their queues, coded or not. To expedite the packet forwarding, coded packets are transmitted with a higher priority without starving uncoded packets. This is achieved by assigning a different back-off time to forwarders. Assume that node B_2 wins the channel and transmits $p_1 \otimes p_2$ (Fig. 3b). The second-next-hop nodes *V* and *Y* receive the XORed packets and are able to decode them using the packets stored in their buffers. Then, they reply instantly with an ACK in a "distributed bursty" way in the order specified by the enhanced MAC header, separated by a SIFS. Such a reliable link-layer broadcasting mechanism also helps to remove the packets queued at the intermediate nodes to avoid packet duplication (Fig. 3c). When no coding is applicable, any intermediate node that has a good channel condition to receive the packet can forward it to the second-next hop opportunistically. Because all helpers are neighbors of the sender, it is likely they are within carrier-sensing range of each other. As a result, the CSMA-CA mechanism plus the trailing ACK can ensure that exactly one of them forwards the packet.

BEND was tested using computer simulation with a lossy physical-layer model and displays superb capabilities, simultaneously in traffic mixing for network coding when applicable, and in traffic dispersing for link diversity without network coding. The need for traffic concentration for network coding and the need for traffic separation to approximate the network capacity were in conflict until the BEND solution. Because the selection of a forwarder is a per-hop and per-flow decision, BEND is especially suitable for link-layer diversity. With a clear separation between the MAC and network layers, it can be ported easily to support a wide spectrum of routing protocols. On the other hand, being a completely link-layer solution, its current design is limited to two hops in forwarding assistance, and its extension to more hops has yet to be explored.

CONCLUSION

In wireless networking research, the focus is switching from making wireless channels as good as wireline channels to direct utilization of some of the inherent characteristics of wireless channels. Channel diversity is one such example. Although it can be perceived as a physical data link, or network 8 layer effect, it reveals more potential to further boost the performance of multihop wireless networks at the link layer and above. A challenge, as well as an opportunity, in doing so is the stringent requirement of dynamic and localized response mechanisms because of the finer time granularity than that which a traditional end-to-end solution can effect. Yet, as we have noticed, a link-layer-and-above scheme has indicated possibilities of further performance improvement despite its difficulty and overhead in implementation. Nevertheless, this opens up a new vision in exploration of channel diversity. For example, when reception diversity is studied as one way to look at the problem, can interference diversity also be investigated? That is, the interference level also can vary across a short distance or over a short period of time and as a result, the carrier-sensing behavior of transmitting nodes also presents diversity. Are they two sides of the same coin, or will it introduce new issues? Either for reception or interference diversity, there is always (positive or negative) correlation among links in a neighborhood. Is this something we can rely on for decision making? One step ahead, when power control is carefully exercised, multiple transmissions can happen simultaneously without interfering with each other. The space and frequency distribution of such parallel flows will change over time. Efficient coordination to enable this to occur will be beneficial but complex. How do we approach this? We will see for ourselves.

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Biographies

Biographies of the authors were not available when this issue went to press.

Distributed Cooperative MAC for Multihop Wireless Networks

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ABSTRACT

This article investigates distributed cooperative medium access control protocol design for multihop wireless networks. Cooperative communication has been proposed recently as an effective way to mitigate channel impairments. With cooperation, single-antenna mobile terminals in a multi-user environment share antennas from other mobiles to generate a virtual multiple-antenna system that achieves more reliable communication with a higher diversity gain. However, more mobiles conscribed for one communication inevitably induces complex medium access interactions, especially in multihop wireless ad hoc networks. To improve the network throughput and diversity gain simultaneously, we investigate the issues and challenges in designing an efficient MAC scheme for such networks. Furthermore, based on the IEEE 802.11 DCF, a cross-layer designed cooperative MAC protocol is proposed. The MAC scheme adapts to the channel condition and payload length.

INTRODUCTION

The limited radio spectrum and channel impairments are two key challenges in wireless communications. Although multiple-input multiple-output (MIMO) antenna systems can improve the capacity and reliability of wireless communications by utilizing multiplexing gain and diversity gain, respectively, packing multiple antennas on a small mobile terminal poses implementation difficulty. The cooperative communication approach [1, 2] provides a design alternative, where mobile nodes share their information and transmit cooperatively as a virtual antenna array, thus providing diversity without the requirement of additional antennas at each node. The main advantages of cooperative communications include:

- Increasing the communication reliability over a time-varying channel
- Increasing the transmission rate and decreasing communication delay across the network
- Reducing transmit power, decreasing interference, and improving spatial frequency reuse
- Enlarging transmission range and extending network coverage

Most existing work on cooperative communications focuses on various issues at the physical layer, and the advantages are often demonstrated by analyzing signaling strategies based on information theory. The signaling strategies typically involve coordinated transmission by multiple network nodes; but despite recent progress, there are still significant barriers to applying these results to the development of practical network protocols [3]. For example, many information theoretical results are based on asymptotically large data block length, and usually ignore the overhead need to set up and maintain coordinated transmissions. In practice, however, the payload length is always limited due to error control and may change with various applications, and overhead from each protocol layer is not negligible, especially when the payload length is short. Thus, the cooperation gain may disappear if higher-layer protocols are not appropriately designed. Moreover, the higher-layer protocols should operate according to the time-varying channel status due to user mobility, as cooperation can be inefficient under certain network conditions such as very good channel quality [1].

Hence, a higher-layer protocol for cooperative wireless communications should be not only payload-oriented (or application-oriented as the payload length may depend on applications), but also channel-adaptive. For efficient cooperative communication, operations at the physical layer should be coupled with those at higher layers of the protocol stack, in particular the medium access control (MAC) and network layers. In this article we focus on how physical-layer cooperation can influence and be integrated with the MAC layer for higher throughput and more reliable communication, rather than advantages of cooperation at the physical layer.

The remainder of this article is organized as follows. We first review related work on cooperative MAC. Next, we discuss various issues and challenges on designing an efficient cooperative MAC protocol, and then propose a novel cross-layer cooperative MAC based on the IEEE 802.11 distributed coordination function (DCF). Finally, we conclude the article with some remarks on further research on cooperative MAC design.

Related Work

While fairly extensive research has been carried out for the physical layer of cooperative communication networks [3], to the best of our knowledge, only a handful of papers [4–9] have considered relevant MAC design. Cooperative communications require many unique features in MAC, which should be distributed and cooperative for a multipoint-to-multipoint environment.

The existing cooperative MAC protocols can be classified into proactive schemes [4–7] and reactive schemes [8, 9]. In the former, the cooperation of the partner(s) is always provided by either the prearranged optimal [7, 10] or the random [4–6] helper(s) before the acknowledgment (ACK) from the receiver; while in the latter, the help from the partner(s) is initiated only when the negative acknowledgment (NACK) is received/detected.

In [4, 5], two similar protocols (called CoopMAC and *r*DCF) based on the IEEE 802.11 DCF are proposed to mitigate the throughput bottleneck caused by low-data-rate nodes. A high-rate node is allowed to help a low-rate node through two-hop transmission. With joint routing and cooperation, a cross-layer approach is introduced in [6]. Clusters of nodes near each transmitter form virtual multiple-input single-output (VMISO) link to a receiver on the routing table and as far as possible to the transmitter. Space-time codes are utilized to support transmission over a long distance, thus reducing the number of transmission hops and improving communication reliability. In [7] we propose a busy-tone-based cross-layer cooperative MAC (CTBTMA) protocol. Adaptive modulation and coding (AMC) and multimode transmission are scheduled together according to the channel condition to improve the network throughput. The use of busy tones helps to solve collisions in a cooperation scenario and to address the optimal helper selection problem. Reactive schemes [8, 9] have a similar strategy, which let neighbor(s) (overhearing the packet) retransmit the packet instead of the source node when the NACK is detected.

Issues and Challenges in Multihop Communications

When cooperative diversity is adopted in multihop wireless networks, a cooperation-based MAC scheme needs to be carefully designed. Some questions need to be answered, such as:

- Cooperate or not cooperate
- If cooperate, who the helper(s) should be and how to do the selection
- How to solve the new hidden and exposed terminal problem in cooperation scenarios
- Rate maximization or interference minimization

Indeed, all the questions are related to cooperative MAC, as discussed in more detail in the following.

Cooperate or Not Cooperate

For the first question, information theoretical analysis provides some indication whether or not cooperation outperforms noncooperation. A thorough comparison can be done from a diversity-multiplexing trade-off point view [3]. However, in practice, inefficiency will be inevitably introduced in communications by protocol overhead, and limited payload length will reduce the cooperation gain obtained under an asymptotically large block length. As cooperation introduces complexity, a MAC protocol should be carefully designed to prevent unnecessary cooperation, which means that an appropriate cooperative MAC protocol must have the ability to adapt to the payload length and the channel condition simultaneously. Cross-layer design between the physical and MAC layers is required.

The two MAC protocols, CoopMAC [4] and *r*DCF [5], have some ability to address the problem. Enquiries are sent out to one selected potential helper to check whether it can improve the source-destination single hop rate by high-rate two-hop transmission; however, the helper selection is not optimal as it is based on the observation of historical transmissions. Furthermore, the required information exchanges or waiting for an unresponsive helping request may result in inefficiency. The CTBTMA protocol [7] using instantaneous throughput maximization as a criterion can answer this first question. Nevertheless, the optimal helper selection needs to be refined, as discussed next.

Who the Helper(s) Should Be and How to Select

When cooperation is beneficial, which node should be the helper(s), and how should the node(s) be selected? There may be a number of helpers that can potentially improve the transmission quality (e.g., resulting in higher throughput and lower bit error rate) from a source to a destination. Without a central controller, how to find the optimal one(s) effectively and efficiently is vital to a practical MAC protocol.

First, let us look into the relation between the helper number and the cooperation gain. In general, from an information theory point of view, the more the helper number, the larger the diversity gain, which means that the reliability of communications is enhanced with an increased helper number. However, due to the half-duplex constraint or orthogonal transmission constraint, without complex physical-layer techniques such as distributed space-time codes, the data rate will certainly decrease. The essential dilemma may be traced back to the diversity-multiplexing trade-off in multiple-antenna channels, and the constraint of distributed cooperation further deteriorates this phenomenon.

From a MAC layer point of view, when many helpers are conscribed for one communication, two issues are well worth our attention. First, interference range will be enlarged proportionally to the helper number, and it may affect the spatial frequency reuse in wireless networks. A comparison of the interference range for multi-helper and single-helper cooperation is shown in Fig. 1. Second, more control overhead is required in multi-helper scenarios. When multi-helper cooperation fails in finding a sufficient number of helpers in a high traffic load situa-

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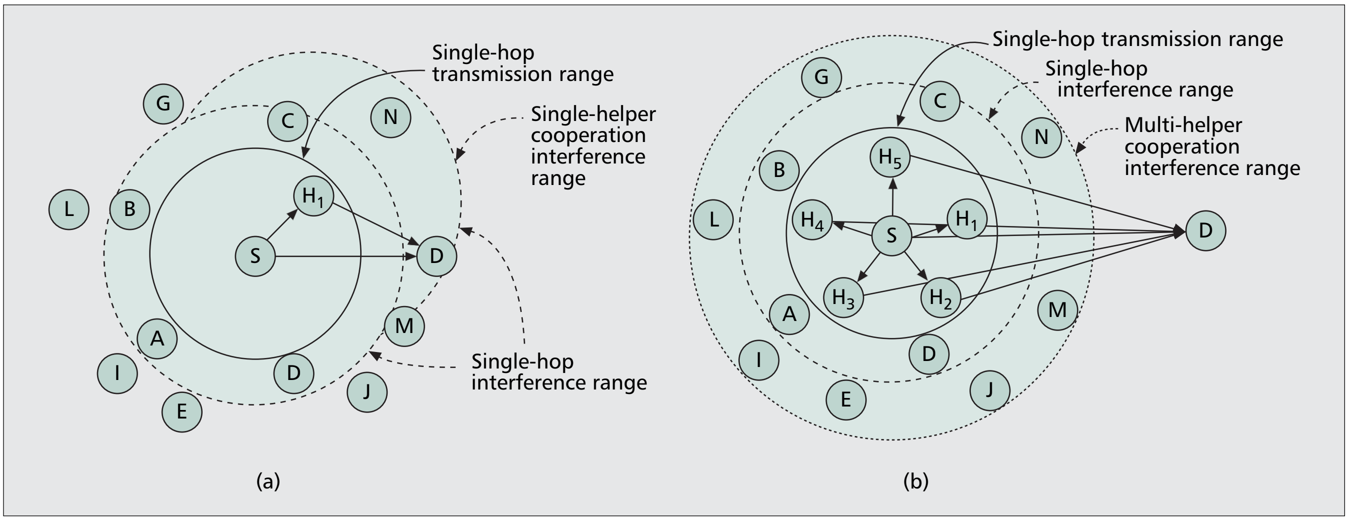


■ **Figure 1.** *a) Interference range in single-helper cooperation; b) interference range in multi-helper cooperation.*

tion, radio resources are wasted in information exchanges and unsuccessful transmissions. The VMISO protocol in [6] faces this problem. Some trade-off has to be made between the performance improvements by successful multi-helper cooperation and radio resource waste otherwise.

As to the helper selection, the CTBTMA protocol [7] uses a busy tone to select the optimal helper. The helper sends the longest busy tone to win the helper selection. However, the busy-tone duration consumes resources in terms of time and spectrum. It is desired to let the optimal helper win the channel as soon as possible. The backoff scheme for service differentiation in IEEE 802.11e can be imitated to overcome the problem. The better the helper, the shorter the backoff time. A similar idea can be found in [10], where the instantaneous channel condition is used as a parameter related with the backoff time. Information theoretical analysis of outage probability shows that the backoff scheme in [10] achieves the same diversity-multiplexing trade-off as a multi-helper cooperative protocol, where coordination and distributed space-time coding are required. In the following, we focus on the case of a single helper.

Hidden and Exposed Terminal Problems

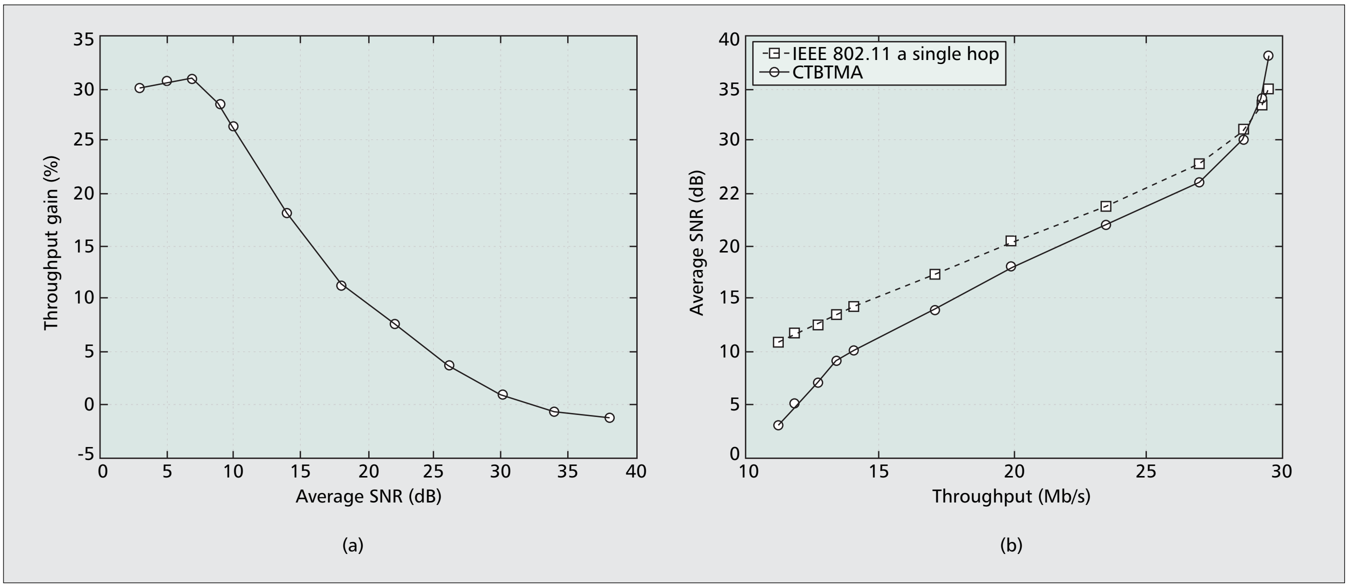
Cooperative communication introduces new aspects to the notorious hidden and exposed terminal problems in mobile ad hoc networks. A helper here not only receives packets from the source, but also transmits the packets to the destination. Thus, the transmissions from neighbors of the helper should also be carefully scheduled to avoid collisions. Otherwise, the cooperation gain can be reduced. Busy-tone-based protocols such as CTBTMA [7] can address the problem, at the cost of busy tones in both transmit power and spectrum and in implementation complexity (which is unavoidable in all busy-tone-based MAC schemes). IEEE 802.11 DCF-based protocols use request-to-send (RTS)/clear-to-send (CTS) handshake to alleviate the hidden and exposed terminal problem. Modifications to the handshake process and its setting of net allocation vector (NAV) strategy are all needed to accommodate the cooperative communication.

Rate Maximization or Interference Minimization

The trade-off between rate maximization and interference minimization comes from the merit of cooperative communication, diversity gain. With the diversity, throughput or transmission rate can be improved as shown in Fig. 2a, and the average signal-to-noise (SNR) requirement or transmit power can be decreased as shown in Fig. 2b [7].

With constant transmit power of all the senders, cooperative communication can increase transmission rate of an ongoing link, but also enlarge the interference range (as shown in Fig. 1), which is hostile to spatial frequency reuse. However, if the transmit power is set to the lowest requirement for a certain transmission rate, interference decreases, but the ongoing link may not have any rate increase. Thus, we need to balance the two aspects.

Most existing cooperative MAC protocols [4–9] use the strategy of rate maximization. One important consideration is that the control packets are always sent with fixed power. Taking IEEE 802.11 DCF as an example, the NAV setting is done using control packets, and no transmission is initiated if NAV is blocked and/or the channel is sensed busy at the physical layer. That is, decreasing transmit power in sending data packets is not only harmful to the node's ongoing transmission rate, but can also allow the nodes in the original interference range (rather than in the single-hop transmission range) to initiate new transmissions (e.g., sending control packets with fixed power), which spoils the ongoing transmission. Although it seems that interference minimization is inferior to rate maximization, cooperative communication gives us a good chance to make use of spatial frequency reuse. Thus, a large throughput gain is still possible to achieve by an interference minimization strategy.



■ **Figure 2.** *a) Throughput gain of CTBTMA with IEEE 802.11a for a single-hop transmission; b) average SNR vs. throughput of CTBT-MA and IEEE 802.11a for a single-hop transmission [7].*

CROSS-LAYER MAC DESIGN

In this section we propose a cross-layer cooperative MAC scheme, taking into account the preceding issues and challenges. The proposed scheme is based on IEEE 802.11 DCF and is capable of addressing the issues associated with cooperative communications. We first assume that cooperation is always needed and design the MAC protocol; then we investigate the impact of the protocol on the value of cooperation so as to let the protocol intelligently determine whether cooperation is worthwhile. We use a utility-based optimization problem on the protocol parameters and cooperation gain to achieve the goal and further optimize the performance of the protocol.

HELPER SELECTION AND HANDSHAKE PROCESS

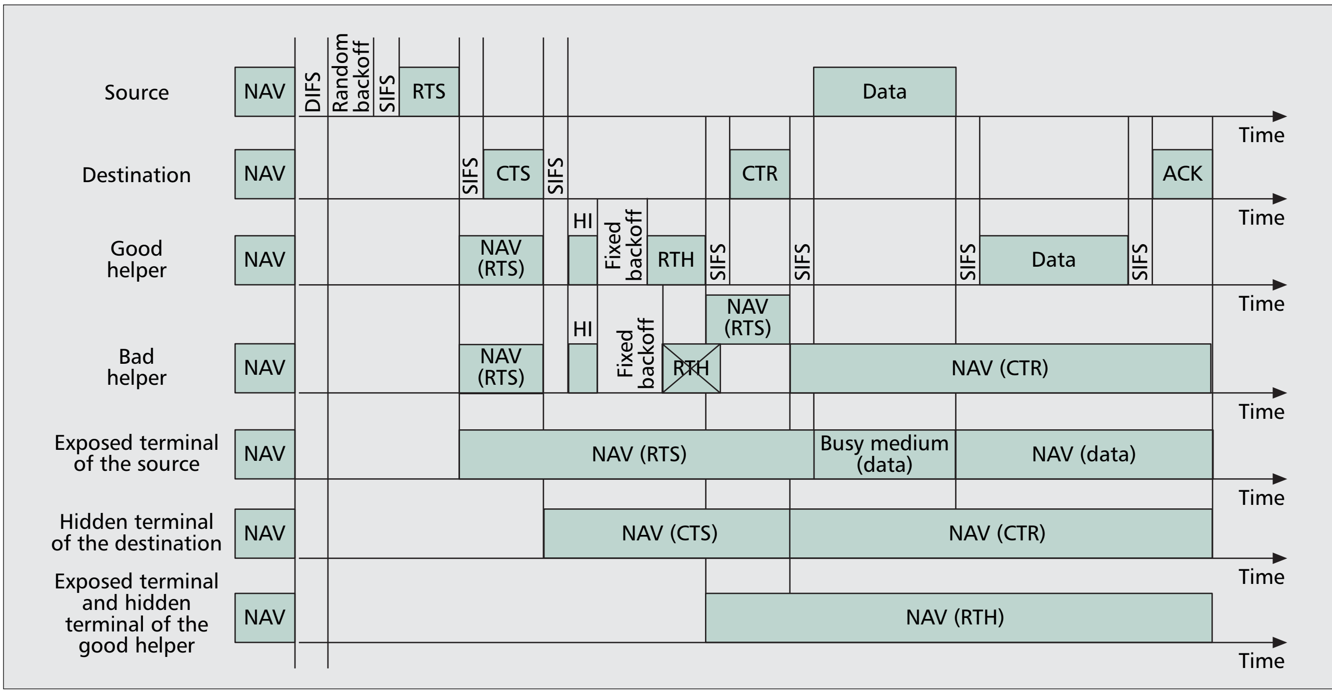
We design our cooperative MAC scheme based on the proactive approach, and the reactive approach can be automatically integrated with the protocol when we consider packet retransmissions later. Under the assumption of cooperative communication, we first devise a helper selection method.

We adopt the helper selection method proposed in [10], where the helper nodes monitor instantaneous channel conditions toward the source and destination via the RTS and CTS packets, and then decide in a distributed fashion which node has the strongest path for information relaying by letting the stronger path holder send a flag packet earlier. In [10] only one constant transmission rate is considered. Since the handshake between the pair of source and destination nodes has already taken place via exchanging the RTS/CTS packets, data transmission should be successful most of the time. Hence, using a helper means losing half the transmission rate. For uncoded proactive cooperation, due to the half-duplex constraint for orthogonal transmission, the overall data rate with cooperation is half that without cooperation. Coded cooperation based on channel coding has the ability to reduce the rate loss to a certain degree. Coding for different channel conditions under a specific rate requirement can be a challenge, as the overall code rate for coded cooperation should be less than half because half of the symbols will be erased [3], while the most common coding rates in popular communication protocols are all at least half, which means that rate loss can still happen. Thus, it is desirable to have multirate transmissions to compensate for the rate loss.

To extend the relay selection method in [10] to a scenario of multiple rates, we let a helper that can support a *higher source-destination link rate* send a flag packet *earlier*, similar to [7]. A mapping relationship between the received signal-to-noise-and-interference ratio (SINR) and rate setting of the IEEE 802.11 MAC protocol should be explored to facilitate the multirate setting, under the assumption that a packet is transmitted successfully if the received SINR is above the corresponding threshold for the packet length. The relation simplifies calculation of the effective transmission rate, relieving mobile nodes of the complex calculation of packet error as in [7]. The SINR threshold for a known payload length for required transmission accuracy can be acquired by analysis or simulation.

We use Fig. 3 to explain the proposed IEEE 802.11 DCF-based cooperation MAC. A source node initiates its transmission by sending an RTS packet to its destination node after finishing its backoff. The destination node that is idle responds to the source node with a CTS packet including the estimated SINR. Each of the common neighbors (including both good and bad helpers) of the source and destination nodes hears both the RTS and CTS packets, and finds out the potential maximal cooperative source-destination transmission rate. If a neighbor is capable of increasing the transmission rate, it contends to be the helper as follows.

First, the neighbor sends out a helper indica-



■ **Figure 3.** *An illustration of IEEE 802.11 DCF-based cooperative MAC.*

tion (HI), similar to busy-tone-based helper selection methods; but the purpose of HIs is to make the source and destination nodes aware of the willingness and existence of the helpers rather than to determine which neighbor is the optimal helper. As there may not be any helper, the HI is vital. If there is no HI, the source starts sending the data packet immediately, which means our MAC has the ability to quickly switch between cooperation and non-cooperation modes.

Second, the optimal helper should be determined as soon as possible. After sending an HI, each competing helper sends out a ready-to-help (RTH) packet (i.e., the flag packet in [10]), after a backoff time. The backoff time is equal to $\tau(M - i)$, where M is the helper rate number, i is the index of the achieved maximal rate, and τ is a constant unit time. The variable M is a design parameter to be optimized based on channel condition and payload length. The value of τ can be set to the symbol duration [10].

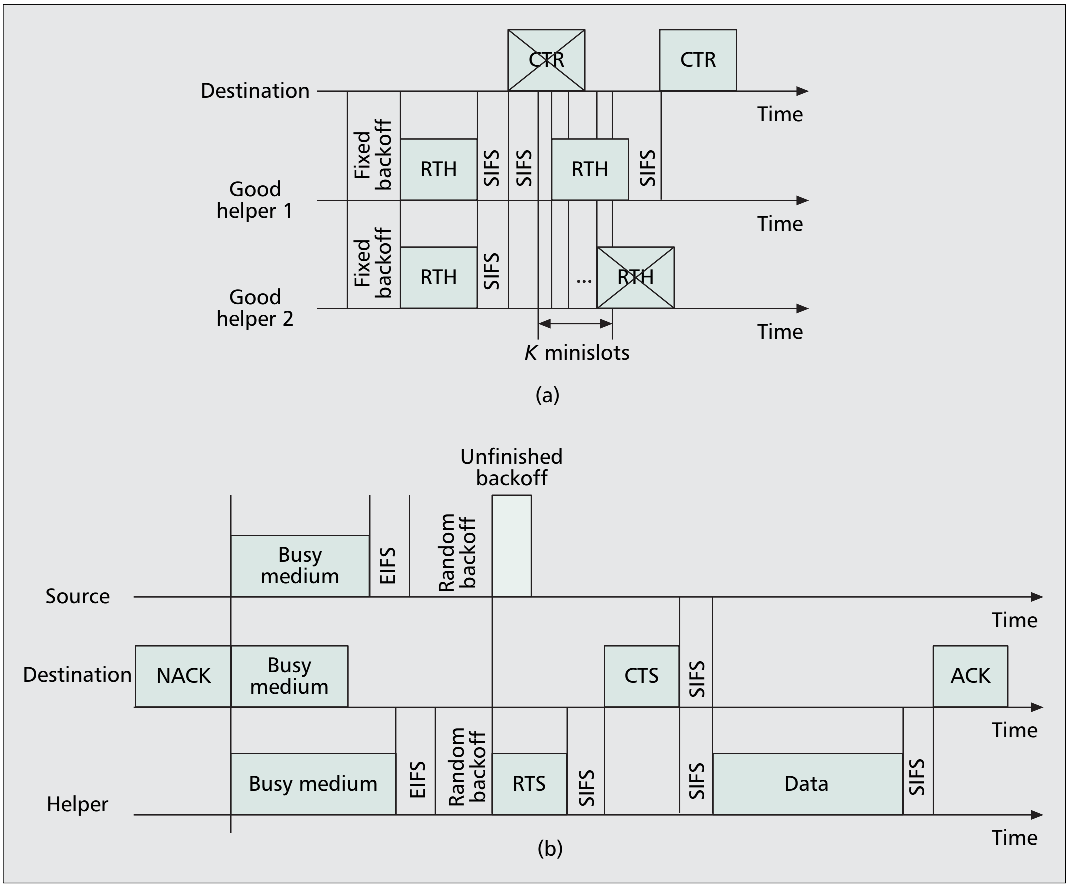
As in Fig. 3, the good helper ends the backoff process earlier than the bad helper; thus, it sends out an RTH packet first. The bad helper gives up contention and sets up its NAV after hearing the RTH packet from the good helper. After receiving the RTH packet, the destination node sends a clear-to-receive (CTR) packet (i.e., the broadcast message in [5]) to notify all helpers to stop contention (the helpers may be hidden to each other) and inform the source node to send data.

When there are multiple good helpers, the destination node does not send out the CTR packet, which will be detected by the good helpers after two short interframe spaces (SIFS), as shown in Fig. 4a. Then the good helpers resend their RTH packets in a random selected minislot from K minislots. The duration of a minislot is set equal to τ. The probability of RTH packet collision depends on the number of minislots. In this case the destination node sends out a CTR packet whenever an RTH packet collision happens, which has a one-bit stop label to notify all helpers to give up contention and invite the source node to start transmission without cooperation.

Retransmission Mechanism

Another issue that needs to be tackled is packet retransmission when the destination node responds with a NACK packet. We schedule the medium access for retransmission as shown in Fig. 4b. Usually, the helper has a better channel condition than the source node (otherwise, the helper should not be selected); hence, retransmission by the helper should be a better choice. Note that it is possible that user mobility can result in the helper being far away from the destination node after the medium becomes idle again. To deal with this situation, one approach is to let both the source and helper nodes contend to retransmit the data packet, with the helper increasing the contention window (CW) only by a step within [1, 2) and the source node doubling its CW. In Fig. 4b the helper accesses the medium earlier than the source node and sends the RTS packet to the destination node. To avoid retransmissions by different helper nodes, the retransmission is limited only by the prearranged helper or source node, without further cooperation in the retransmission; however, the destination node can jointly decode the newly received data packet and the previously received packet in error for improved accuracy.

For the reactive approach, since help from a relay node is activated only when the NACK packet is received, the mechanism is similar to the retransmission process of the proactive



■ **Figure 4.** *a) Solution to RTH packet collision by contention over* K *minislots; b) retransmission scheduling in cooperative transmission.*

approach. Thus, the reactive approach is incorporated in the proposed MAC protocol inherently.

COOPERATION DECISION AND PROTOCOL OPTIMIZATION

Next, we present how the MAC protocol can determine whether cooperation is worthwhile. As discussed earlier, channel condition and payload length are the most important parameters that govern the optimal transmission mode. Cooperation is helpful only if the following condition is satisfied:

$$\max\{R_2, R_{coop}\} > R_1, \quad (1)$$

where R_2 is the effective payload transmission rate from the source to the destination with two-hop transmission, R_{coop} is the rate with cooperative communication, and R_1 is the rate in one-hop transmission without cooperation, taking into account the transmission overhead in the MAC protocols. The transmission rates can be determined based on channel conditions for the source-destination, source-helper, and helper-destination links. Note that without cooperative diversity, the rate of one- and two-hop transmissions depends on the single link, while the transmission rate of cooperative communication depends on all the links from the source to the destination. Taking the scenario in Fig. 1a as an example, the transmission rate of two-hop transmission from S to H_1 is only decided by link S-H_1, and the transmission rate of cooperative communication from S to D is based on all three links, S-H_1, S-D, and H_1-D.

The condition of inequality 1 can be equivalently represented by

$$\min\{T_2, T_{coop}\} < T_1, \quad (2)$$

where T_2, T_{coop}, and T_1 are the time duration

Payload (bytes)	Source-destination link rate (Mb/s)							
	6	9	12	18	24	36	48	54
500	15	6	0	0	0	0	0	0
1000	19	11	6	0	0	0	0	0
1500	20	12	8	1	0	0	0	0
2000	20	12	8	3	0	0	0	0

■ **Table 1.** *The optimal backoff length* M *for different payload length and source-destination link rate.*

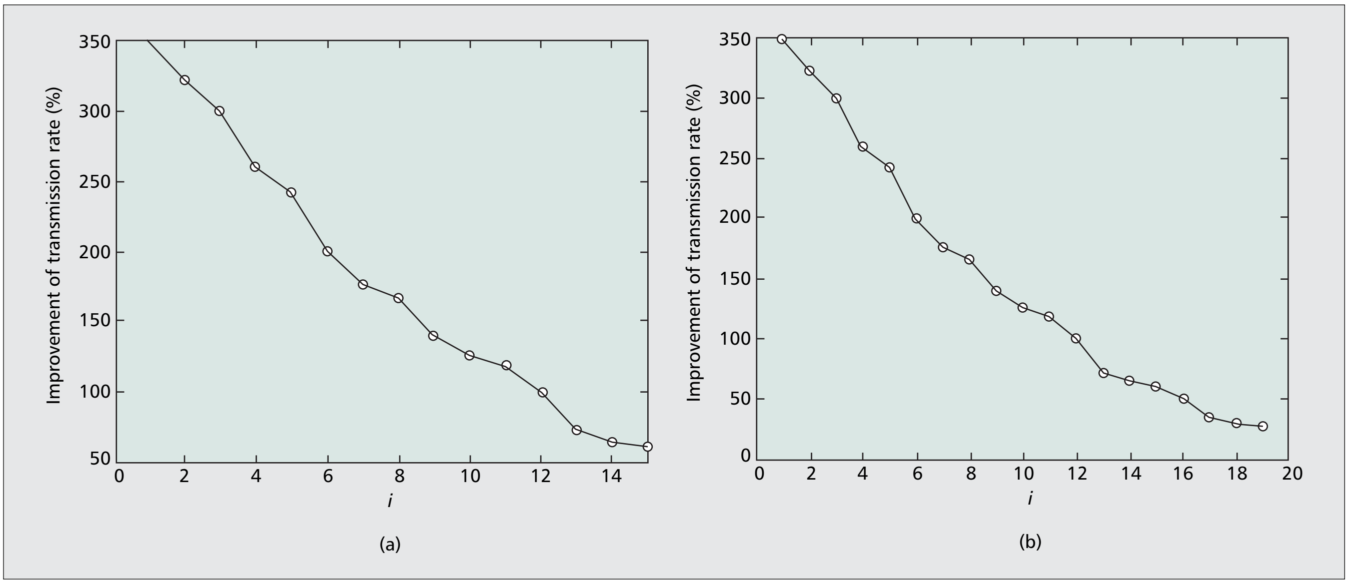


Figure 5. *The improvement of transmission rate with a payload of a) 500 bytes; b) 1000 bytes.*

required to send the same payload from the source to the destination, respectively, taking into account the overhead time in the MAC protocols.

A careful examination of inequality 2 reveals a relationship among the payload length, communication overhead, and the value of cooperation. The cooperation is meaningful only when the source-destination link has a low transmission rate and/or the payload length is sufficiently large compared with the communication overhead.

The condition of inequality 2 also gives us a straightforward way to find out whether a relay is needed under a specific channel condition and payload length. To find out whether a helper is useful or not, we only need to use the maximal supportable rate according to the SINR-rate mapping table to check whether or not inequality 2 is satisfied.

In addition, inequality 2 can help maximize the performance of the proposed MAC protocol, in terms of an optimal value for parameter M, the maximal fixed backoff length of the helper selection process. Using system parameters as in IEEE 802.11a and the length of RTH and CTR packets equal to the size of ACK packet, Table 1 shows computer exhaustive searching results of the optimal M values for four payload lengths, 500, 1000, 1500, and 2000 bytes, under eight channel conditions represented by the source-destination link rates. Figure 5 shows two examples of the improvement of transmission rate using the optimized M value, the source-destination link rate of 6 Mb/s, and payload of 500 and 1000 bytes, respectively. The numerical results are consistent with the relationship from inequality 2 among payload length, channel condition, and cooperation chances:

- Cooperation chances increase with the payload length.
- A meaningful cooperation happens when the source-destination channel can only support a low transmission rate.

With an optimal M value, the proposed cooperative MAC protocol always achieves higher throughput than that without cooperation for the cooperative links.

Concluding Remarks

Cooperative communication for wireless systems has recently attracted significant attention in the research community. Unlike conventional point-to-point communications, cooperative communications offers tremendous advantages such as allowing users (or nodes) to share resources to create collaboration through distributed transmission and processing. This new form of distributed spatial diversity can offer more reliable communications, increased network capacity, extended coverage area, and more energy-efficient transmissions. However, the higher-layer protocols of cooperative communication networks must be properly designed to take advantage.

We investigate how physical-layer cooperation can influence and be integrated with the MAC layer for higher throughput and more reliable communication. Several MAC layer issues and challenges are discussed for cooperative communication. A cross-layer design for cooperative MAC is presented to address the issues and meet the challenges. With channel and payload length adaptation, the proposed cooperative MAC protocol supports multiple transmission rates and transmission modes, and outperforms the traditional noncooperative MAC.

There are many open research issues in network protocol design for cooperative communications. For example, how to route a packet should be revisited. With cooperative communication, each node in the network assumes an additional new role: the helper. A routing protocol should balance traffic load in the presence of helpers. Also, as a helper uses its resources (e.g., transmit power) to assist transmission of other users, fairness in medium

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access should be studied. Furthermore, cooperation should not be limited to only transmission over links, but extended to a network-level approach to maximize the performance of the overall network.

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BIOGRAPHIES

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This new form of distributed spatial diversity can offer more reliable communications, increased network capacity, extended coverage area, and more energy efficient transmissions. However, the higher-layer protocols of cooperative communication networks must be properly designed to realize the advantages.

Cooperative Network Implementation Using Open-Source Platforms

Thanasis Korakis, Michael Knox, Elza Erkip, and Shivendra Panwar, Polytechnic Institute of NYU

ABSTRACT

Cooperative networking, by leveraging the broadcast nature of the wireless channel, significantly improves system performance and constitutes a promising technology for next-generation wireless networks. Although there is a large body of literature on cooperative communications, most of the work is limited to theoretical or simulation studies. To impact the next generation of wireless technologies and standards, it is essential to demonstrate that cooperative techniques indeed work in practice. This article describes two programmable cooperative communication testbeds built at Polytechnic Institute of NYU to achieve this goal. The testbeds are based on open-source platforms and enable implementation of cooperative networking protocols in both the physical and the medium access control layer. Extensive experiments carried out using the testbeds suggest not only that cooperative communication techniques can be integrated into current wireless technologies, but also that significant benefits of cooperation can be observed in terms of network throughput, delay, and video quality in real applications.

INTRODUCTION

Wireless will be the dominant mode of Internet access for end users in the near future. Technologies such as WiFi and WiMAX attempt to provide broadband wireless access. However, the bandwidth limitations of the wireless channel, interference from multiple users operating in the same band, and channel variations due to fading become bottlenecks for typical multimedia applications that require high bandwidth and an error-resilient communication medium.

Cooperation among users, by enabling wireless terminals to assist each other in transmitting information to their desired destinations, provides a good solution to the problems that current wireless technologies face. At the physical (PHY) layer, terminals overhear one another's signals, processing and retransmitting them to form a virtual antenna array. Through cooperation, it is then possible to obtain the spatial diversity benefits of multi-input multi-output (MIMO) systems without necessarily having a physical antenna array at each terminal. Cooperative communication techniques can adapt easily to a changing environment by opportunistically redistributing network resources such as energy and bandwidth. Incorporating the notion of cooperation at the medium access control (MAC) layer extends the benefits to large networks resulting in high throughput, low delay, reduced interference, low transmitted power, and extended coverage. Using a cross-layer design from the application layer down to the physical layer enables high quality multimedia transmission over cooperative wireless links.

Cooperative communications is a vibrant research area. There is extensive work in the literature on the study of cooperative schemes in the PHY layer [1–3] and to a more limited extent, in the MAC layer [4]. Almost all of this work is based on theory and simulations. The theoretical analysis and the simulation of a specific protocol or technique can give important information about performance in terms of throughput, delay, or power consumption. However, to have analytically tractable models, several simplifications of the real world environment must be made. Although simulations have the ability to incorporate more general models, the evaluation still is limited by the complexity of the simulation software and the simplification of the wireless environment. Some specific limitations of the simulation approach in depicting a real wireless network include inaccurate representation of the wireless medium, simplification of synchronization issues that occur in wireless terminals, and ignoring several aspects such as the computational overhead. Therefore, there is a significant justification for moving one step further than analysis/simulation and implementing cooperative protocols in a real wireless platform for deeper understanding of proposed schemes.

This article outlines the implementation efforts at Polytechnic Institute of NYU in building two programmable cooperative networking testbeds. The goal is to illustrate the feasibility of cooperation and to provide platforms over which different cooperative protocols can be tested. Our testbeds incorporate physical layer cooperation and a cooperative MAC layer and are amenable to cross-layer design to enable applications like cooperative

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video communications. The first testbed is based on open-source drivers, whereas the second one is built using software-defined radios (SDRs), thus providing different characteristics and abilities for flexible implementations. This article describes the functional characteristics of each platform and highlights the advantages, as well as the limitations of each approach. It also presents the details of the PHY and MAC layer implementations and our experimental results.

Overall, our testbeds represent the first fully functional, cross-layer experimental effort on cooperative networking. The open-source nature of the platforms enables further investigation and experimentation by other research teams. Our results indicate the feasibility of cooperative networking in practice and also suggest that the theoretically predicted gains in error rates, network throughput, delay, and multimedia signal quality apply to practical implementations as well. In the next section, we discuss the details of the two testbeds. Implementation of cooperative MAC protocols on both testbeds is covered in the following section, and cooperative PHY layer is discussed in the final section.

Description of the Cooperative Networking Testbeds

In setting up the cooperative networking testbeds, our focus was to use commercially available open-source platforms to enable future participation and contribution from other researchers. However, there is currently no system that simultaneously can accommodate the requirements of both PHY and MAC layer cooperation protocols, as well as cross-layer design. To combine the benefits of different platforms, we decided to build two separate, yet complementary testbeds, as shown in Fig. 1. The basic structure of each testbed along with its advantages and disadvantages is summarized in the following subsections.

Open-Source Drivers Cooperative Testbed

The goal of our first testbed is to implement cooperative wireless protocols focusing on the functionalities of the MAC and network layer (routing). The nodes are based on commercially available WiFi cards that have a fixed PHY layer. The MAC and network layer functionality is implemented in software based on open-source wireless drivers based on 802.11 protocol.

- *Advantages*:
  –The implementation is backward compatible with current WiFi products. This enables us to develop protocols based on a realistic detailed implementation of IEEE 802.11 and opens up the possibility of impacting WiFi standards in the near future.
  –The performance of the implemented cooperative protocols can be compared directly with commercial 802.11 solutions.
- *Disadvantages*:
  –We have access only to a portion of the MAC layer functionality. We cannot change time-sensitive functions.
  –There is no access to the PHY layer, and thus we cannot build PHY/MAC cross-layer algorithms and fully exploit the notion of user cooperation.

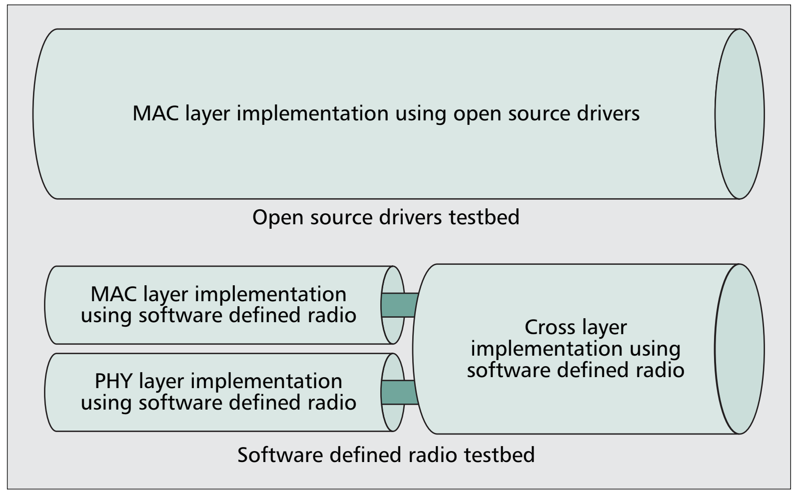

Figure 1. *Illustration of open source drivers and software defined radio cooperative testbeds.*

In the open-source drivers testbed, even though PHY layer cooperation is not exploited, cooperation at higher layers still results in many benefits for individual users and the network as shown in the next section.

A wireless driver that does not involve any time-sensitive issues (e.g., sending of an acknowledgment [ACK] after a short interframe space [SIFS] period) typically controls the functionality of the MAC layer. Thus, by changing the driver, one can change a significant part of the MAC layer and to some extent, one can build new protocols. There are three open-source Linux drivers available today: the MadWiFi driver that is based on Atheros chipsets; the HostAP driver, based on the Intersil Prism 2, 2.5, or 3 chipset; and the Intel PRO/Wireless 2100/2200/3945 drivers, based on the Intel chipset.

In a typical driver-card architecture, all the features specified in IEEE 802.11 MAC protocol are logically partitioned into two modules, according to the time criticality of each task. The lower module, which usually operates on the wireless card as a part of firmware, fulfills the time-critical functions, such as the generation and exchange of request to send/clear to send (RTS/CTS) control messages, transmission of ACK packets, execution of random back off, and so on. The other module, which normally assumes the form of the system driver, is responsible for more delay-tolerant control-plane functions, such as the management of MAC layer queue(s), the formation of the MAC layer header, fragmentation, authentication, association, and so on. For the MAC implementation, ideally we would like to have access to the firmware of the card as well and thus, have the ability to change the time-critical functionalities of the protocol. Unfortunately, the firmware is not accessible because it is proprietary. Thus, the only option is to change the part of the MAC functionality that is controlled by the driver. The basic wireless stack architecture of a typical

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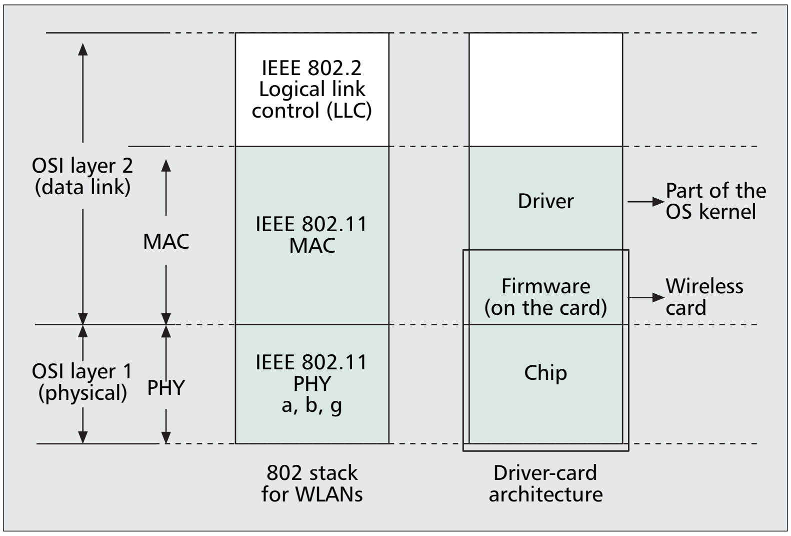


■ **Figure 2.** *The driver-chipset architecture for a typical IEEE 802.11 card.*

chipset is depicted in Fig. 2. Intersil and Intel chipsets follow this approach, whereas Atheros follows a slightly different architecture.

The open-source cooperative testbed is housed at the Wireless Implementation Testbed Laboratory (WITest Lab) [5] at the Polytechnic Institute of NYU. It currently consists of 20 nodes, and the network is managed by one server. Each node has a motherboard with a 1 GHz Pentium processor, 512 MB RAM, 40 GB of local disc, and the appropriate input-output interfaces. It has two mini peripheral-component interconnect (PCI) slots, where mini PCI wireless cards, based on one of the chipsets mentioned above, can be inserted.

SOFTWARE DEFINED RADIO COOPERATIVE TESTBED

The goal of the second testbed is to build an open-source architecture for rapid prototyping of PHY and MAC layers by leveraging existing open-source radio platforms. The testbed is developed by first modifying existing reference designs at the PHY and MAC layers separately. We then use the developed algorithms as a starting point for designing a joint cooperative PHY/MAC layer.

- *Advantages*:
  - –We have the flexibility to implement a cooperative PHY layer and build PHY/MAC cross-layer protocols, thus obtaining a more complete cooperative implementation.
  - –The benefits of cooperation at different settings, independent of a particular environment or standard, can be established.
- *Disadvantages*:
  - –We can build only simplified versions of MAC protocols used in the standards because no software exists with detailed implementation of any standard.
  - –The hardware places specific limitations on the system performance. For example, some of the SDR platforms that we tested limit the minimum time between two sequential transmissions to a period longer than the one in commercial 802.11 cards. Therefore, it is not possible to compare our protocols directly with commercial solutions. To provide comparisons with standard MAC protocols such as 802.11, we must build emulations of the standards, based on the same hardware architecture.

To accelerate deployment, we leverage two existing SDR platforms, namely the wireless open-access radio platform (WARP) from Rice University [6] and the GNU/universal software radio peripheral (USRP) [7]. The SDR testbed is housed within the Wireless Information Systems Laboratory (WISL) at the Polytechnic Institute of NYU and consists of six nodes of each technology.

The WARP system uses a field programmable gate array (FPGA) to process all symbol-rate and bit-rate data. This approach requires coding all of the modulation, demodulation, and communication algorithms directly inside the FPGA. The platform can process signals in excess of 20 MHz of radio bandwidth. The WARP system includes a baseband processing board, radio frequency (RF) radio board, and the open-source code. WARP nodes are based on a Xilinx Virtex-4 FPGA that has two embedded PowerPC processor cores. The Virtex-4 provides dedicated digital signal processing (DSP) slices, hardware blocks designed specifically for high-speed multiply-accumulate and other DSP operations. The WARP FPGA board provides four daughter-card slots, each wired to a large number of dedicated FPGA input/output (I/O) pins. These slots can house peripheral wireless cards. The four slots are functionally identical, enabling users to mount the combination of peripheral cards that best suits their application. The PHY layer radio platform, based on WARP nodes, provides a wideband radio front-end covering the unlicensed frequency bands at 2.4 GHz and 5.6 GHz. An advantage of the WARP system is that high symbol rates can be achieved while performing all signal processing inside the FPGA. However, the disadvantage of WARP is a steeper learning curve requiring knowledge of MATLAB Simulink and Xilinx System Generator. WARP also provides a framework called WARPLAB that enables signal processing and waveform generation in MATLAB and uses WARP boards to transmit and receive only these waveforms. Although WARPLAB is useful for rapid prototype design and over-the-air testing, the simplicity of implementing the PHY layer in MATLAB comes at the expense of low transmission rates.

GNU Radio [7] is an open-source software toolkit for building software radios, generally independent of the hardware. The GNU software is easily configured with the USRP, available from Ettus Research [8]. We chose to use GNU radio due to its popularity and ease of programming. However, GNU radio does not support 802.11 due to slow data transfer over the USB port to a personal computer. Hence, in this article, we focus on WARP, which can handle high data rates, rapid automatic gain control (AGC) for the received packets, and provides a good framework to integrate the PHY and the MAC layers.

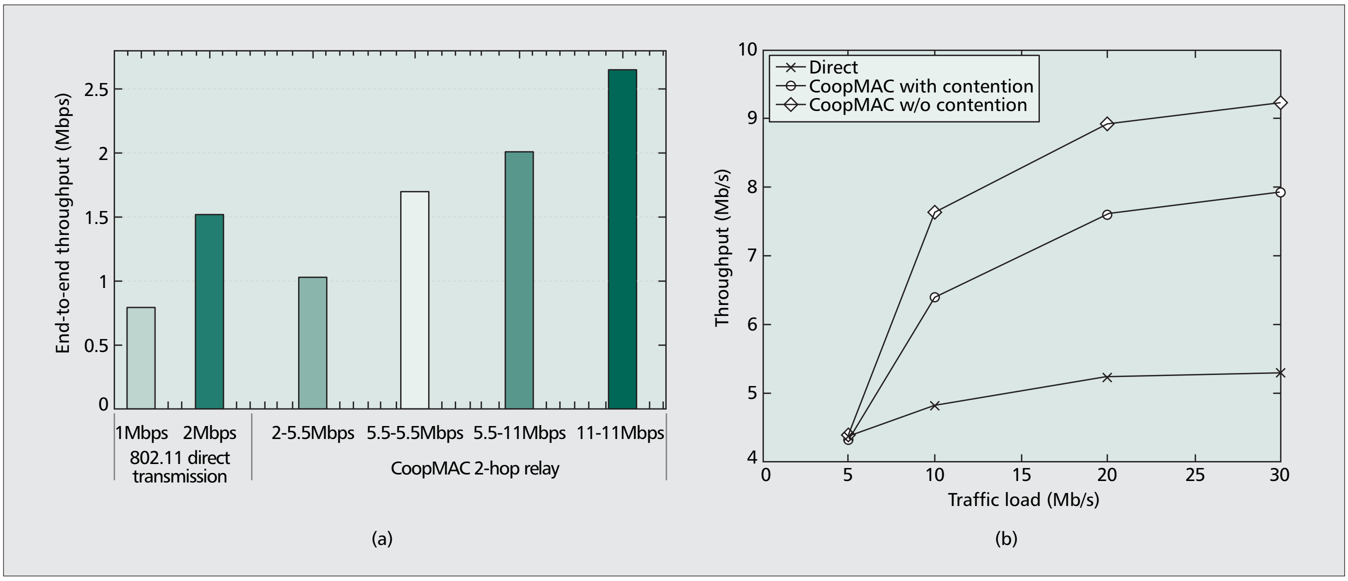


Figure 3. *Throughput comparison for UDP traffic: a) open source drivers platform; b) SDR platform.*

Implementation of MAC Layer Cooperative Protocols

Our MAC layer implementations include several cooperative MAC schemes both in the open-source drivers testbed, as well as in the SDR testbed. In the open-source drivers platform, we implement cooperative MAC protocols for unicast and multicast applications based on both HostAP and MadWiFi. Although the implementations are realistic and fully functional, some of the details of the protocols must be omitted due to platform limitations. Therefore, we also proceed with the SDR platform, in particular the WARP nodes. The details and the codes for all of our MAC layer implementations and our experimental results can be found at the WiTest Lab Web site [5]. In this article we focus on the implementation of a unicast cooperative MAC protocol called *CoopMAC* [9] in both testbeds.

The Cooperative MAC Protocol

A cooperative MAC protocol called *CoopMAC* developed in [9] enables participation of a third node (called the relay or the helper) to facilitate communication between a source and a destination. In conventional wireless networks, when a source experiences a bad channel toward its destination, it lowers its modulation scheme and coding rate to maintain a desired level of reliability. In CoopMAC, the source can use an intermediate relay that experiences a relatively good channel with both the source and the intended destination. Instead of sending its packets directly to the destination at a low transmission rate, the source transmits at a high rate to the relay, and then the relay forwards the packet to the destination again at a high rate. By using a two-hop alternative path via the relay, which collectively is faster than the original direct link, the protocol can take advantage of the spatial diversity between the three nodes.

The basic functionality of CoopMAC is described as follows. The source chooses a suitable relay, based on the two-hop sustainable rate, and this decision is broadcast through the RTS in the control plane. The relay indicates its availability to participate by transmitting a new control packet called helper ready-to-help (HTS). The destination completes the three-way handshake by sending a CTS packet. In the data plane, the source transmits the packet in the first hop, and the relay retransmits the packet over the second hop. Each node maintains a table called a *CoopTable*, containing information about available relays and the maximum supported rates for the two-hop transmission toward a destination. For further details, see [9].

Implementation of CoopMAC using Open-Source Drivers

We discuss here only our implementation using HostAP; MadWiFi efforts can be found at the WiTest Lab Web site [5]. Because the implementation of CoopMAC requires changes to both time-critical and delay-tolerant functions, unfortunately, the inaccessibility to firmware forces some compromises and alternative approaches in implementation. For illustrative purposes, three main circumventions are outlined below. An implication of these circumventions is that a faithful implementation of CoopMAC potentially outperforms the one demonstrated in this section.

- *Suspension of three-way handshake*: The strict sequence of RTS and CTS packets was hardwired in the firmware of 802.11 cards; therefore, an insertion of HTS as required by CoopMAC becomes impossible at the driver level. As an alternative, we suppress the use of control packets before data transmission.
- *Unnecessary channel contention for relayed packet*: After the channel access has been allocated to the source, the CoopMAC protocol suggests that the relay should forward the packet a *SIFS* time after its reception, without any additional channel contention.

As anticipated, the user perception is poor for video transmission in the 802.11 network because noticeable freezes and distortions occur frequently. Meanwhile, the video is smooth and artifact-free when CoopMAC is enabled.

(a) Legacy 802.11 (b) CoopMAC

Figure 4. *Open source drivers demo snapshot: a) legacy 802.11; b) CoopMAC.*

However, the ability to do this is controlled by the firmware and cannot be changed. As a result, we compromised on this approach by inserting channel contention for the relayed packet on the second hop.

• *Duplicate ACK*: Each successful data exchange in the original CoopMAC protocol involves only one ACK message, which is sent from the destination to the source directly. Because the acknowledgment mechanism is an integral function of firmware, it is impossible to suppress the unnecessary ACK message generated by the relay for each packet it forwards on behalf of the source. Therefore, the unwanted ACK from the relay must be tolerated instead of being eliminated.

Based on the CoopMAC implementation described above, we ran extensive experiments under different network topologies and different traffic loads. Experimental results in different topologies, with different number of nodes (up to nine) clearly show significant improvement in terms of throughput, delay, and jitter over legacy 802.11. As an example, in Fig. 3a, we show a throughput comparison between CoopMAC and IEEE 802.11 for a topology that consists of three nodes: a source, a destination, and a potential relay. We generate traffic from the source to the destination using *iperf*. Different first- and second-hop relay rates are shown on the horizontal axis. The figure suggests that CoopMAC performs efficiently in a real implementation and can give up to three-fold throughput improvements compared to IEEE 802.11. Further details of the implementation and the experimental results can be found in [10].

The above results are obtained in experiments that rely on large file-transfer traffic patterns. To obtain more insights into the performance of CoopMAC, we also consider video applications. To this end, we transmit a video clip in a testbed that consists of a source, a destination, and a relay. A video server is placed at the source and constantly streams a commercial video clip, while the destination plays the video. We assume that the source has a bad channel with the destination, whereas the relay has a good channel with both nodes. Therefore, when 802.11 is used, the direct transmission rate is set to 1 Mb/s, whereas the transmission rates between the source and the relay and the relay and the destination are both 11 Mb/s.

As anticipated, the user perception is poor for video transmission in the 802.11 network because noticeable freezes and distortions occur frequently. Meanwhile, the video is smooth and artifact-free when CoopMAC is enabled. Figure 4a and Fig. 4b provide snapshots of the video seen at the destination for 802.11 and CoopMAC, respectively. These two snapshots are typical and reveal the substantial difference between the video quality that these two different protocols can deliver.

CoopMAC Using Software-Defined Radio Approach

Whereas the open-source driver testbed enables backward compatibility with the IEEE 802.11 standard, the SDR testbed offers an environment where we can overcome the limitations discussed earlier. Furthermore, the SDR testbed enables us to design MAC and PHY cross-layer schemes jointly.

To study the performance of CoopMAC in the WARP implementation, we conducted several experiments measuring the total number of successful packets (throughput), as well as the average delay per packet. Details, as well as extensive experimental results, can be found in [11]. Here we describe the basic scenario that gives a clear indication of the performance gains. We compare the implemented CoopMAC protocol with two protocols, the first being a carrier-sense multiple-access (CSMA) approach that emulates the IEEE 802.11 MAC. We call this scheme *direct transmission*. The second protocol is an emulation of our implementation in the open-source drivers, in which we implement contention for the second-hop transmission, as well as two ACK packets, one for each hop. We call this scheme *CoopMAC with contention*. The accurate implementation of the CoopMAC mechanism is called *CoopMAC without contention*.

In Fig. 3b, we give the throughput comparisons of the above three schemes for a simple network of a source, a destination, and a relay, as well as for User Datagram Protocol (UDP) traffic. We assume that the source-destination channel is poor, and the relay is located in between. Direct transmission occurs at a data rate of 6 Mb/s (binary-phased shift keying [BPSK]), and the transmission through the relay for both hops is fixed at 24 Mb/s (16-QAM). The figure validates the open-source driver results by showing large throughput improvement of CoopMAC over IEEE 802.11. Moreover, it shows that the WARP implementation further improves the performance over open-source drivers significantly, by eliminating the overhead generated due to contention in the second hop and the double ACK per transmission.

Implementation of PHY-Layer Cooperative Protocols

The spatial diversity provided by user cooperation can be exploited at the PHY layer by enabling the destination to combine signals coming from the source and the relay. This results in higher data rates and a more robust communication system [1–3]. It has been shown, in theory and simulations, that system performance can be further improved when the attributes of PHY- and MAC-layer protocols are combined in a cooperative configuration [4].

This section summarizes some of our PHY-layer implementation results carried out using the WARP SDR testbed. These results provide the basis for our next phase of research involving a combined PHY-MAC implementation. We emphasize that our SDR testbed constitutes one of the few cooperative PHY-layer implementation efforts that goes beyond techniques that require the simple selection of a relay node [12] and provides a platform where source and relay-signal combination at the destination is possible.

The two most popular PHY-layer cooperative protocols are amplify-and-forward and decode-and-forward [3]. These two techniques form the building blocks for most of the known cooperative schemes. Whereas the amplify-and-forward approach was investigated from an implementation perspective in [13], we focused our implementation efforts on a type of decode-and-forward technique, also known as *cooperative coding*, [14, 15], which fits more naturally into a cross-layer perspective.

Cooperative Coding

A rate k/n channel encoder generates n coded bits for every k information bits. In a standard communication system, all coded bits are transmitted directly by the source. In cooperative coding [15], transmission is divided into two slots. In the first slot, the source punctures the code and transmits only a portion of the coded bits. These bits are received both by the relay and the destination. The destination temporarily stores the received data from the source. The relay attempts to decode the source information (successful decoding is possible, as long as the rate of the punctured code is at least one) and then re-encodes to obtain the coded bits. In the second time slot, the relay transmits only the coded bits that the source left off. The destination multiplexes the two received data streams into a single stream and passes through the channel decoder.

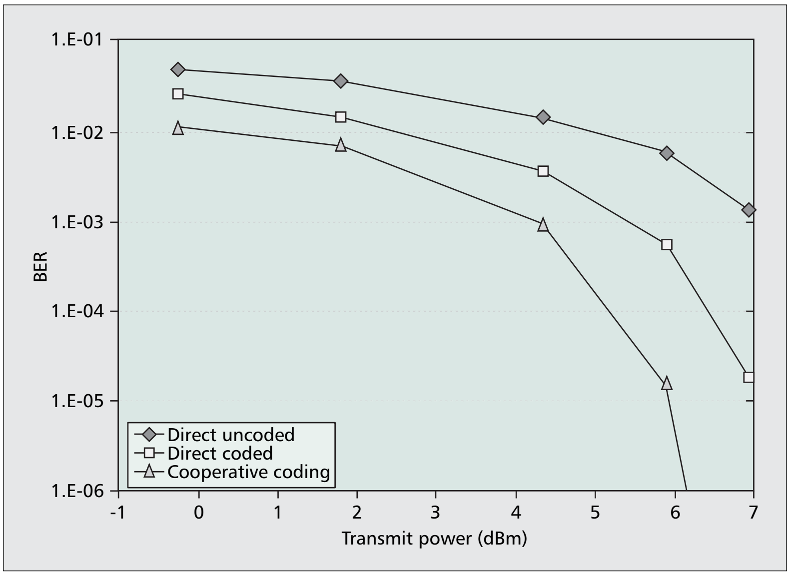
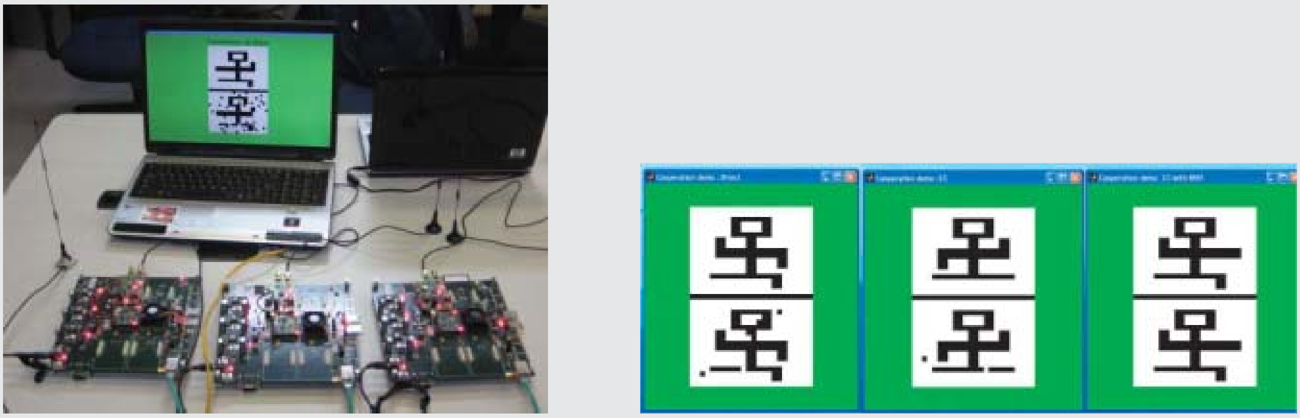


Figure 5. *Measured BER for a three-node cooperative coded system.*

Compared with direct communication from the source to the destination, in the cooperative system the total time and frequency resources remain unchanged. When cooperative coding is employed, the destination still receives the same number of coded bits; however, now, part comes from the source; the remaining part comes from the relay, thereby resulting in spatial diversity. The resulting diversity and code design criteria are discussed in [15].

Implementation of Cooperative Coding Using Software-Defined Radio

The operation of the three-node cooperative-coded system was verified using the WARP hardware and associated WARPLAB software running under Mathworks MATLAB R2006b. A channel code of rate 1/2 with a constraint length of 5 and a generator polynomial matrix of (37 33) was used. The feedback connection polynomial of the encoder is 37 [16]. The destination uses a hard-decision decoding algorithm that enables the use of the basic WARPLAB reference design without substantial modification. The bit error rate (BER) performance is measured on the WARP platform for three different test cases; a single uncoded link between source-to-destination, a single-coded link between the source-to-destination, and a cooperative-coded system using the relay node as described above. The same channel encoder/decoder is used for the two-coded configurations. Figure 5 shows the measured BER for the three system configurations. For the cooperative case, the transmit power level of the relay is set equal to the transmit power of the source. The transmit power levels are adjusted over the range of 0 dBm to +7 dBm, and the BER is measured for each configuration. In all cases, the receive gain was fixed at each node. The relay node is physically positioned between the source and destination nodes, resulting in very high-quality links between the source-to-relay and the relay-to-destination paths. For example, the measured BER



(a) (b)

■ **Figure 6.** *PHY layer demo on WARP platform: a) demo setup; b) demo snapshots from phases 1, 2, and 3, respectively.*

from the source-to-relay was less than 10^{-7}, and the relay-to-destination was less than 10^{-4}. As shown in the figure, the measured BER for the cooperative-coded system outperforms the system that has a single-coded link from the source-to-destination. As expected, the uncoded system operating directly from source-to-destination shows the lowest performance of all three test cases. The measured BER performance trends exhibit a remarkable similarity to the ones found by analysis and simulations [15]. Note that the measured performance for this cooperative system includes the effects of forwarding errors in the relay link.

To further demonstrate the advantage of cooperation in the PHY layer, we built a set up for transmission of a MATLAB video clip over the cooperative-coded system described above [17]. The demo set up is shown in Fig. 6a. The source continuously transmits frames of a video clip over the air. The destination receives the frames and displays the clip in MATLAB. The demo consists of three sequential phases. After all three phases are completed, the cycle starts from the beginning. To automate the phase transition, a MATLAB script is written that switches sequentially from mode to mode every 30 seconds. The phases can be summarized as follows, and the results are illustrated in Fig. 6b:

- **Phase 1 (direct mode)**: Non-cooperative network — source directly communicates with the destination. The quality of the received video is bad; noticeable distortions occur frequently.
- **Phase 2 (multihop)**: Relay is used for forwarding, but only the relay signal is used for decoding at the destination. The quality of the video is good.
- **Phase 3 (cooperative mode)**: The destination combines the signals from the source and the relay; the received signal strength indicator (RSSI) weighting is used in decoding. As expected, this results in the best quality of video.

The reported BER measurements and the above demo were based on hard-decision decoding; our next step is to implement optimum soft-decision decoding algorithms. We also plan to migrate WARPLAB implementations to the WARP FPGA to enable operation at high data rates.

Conclusions

In this article, we describe the implementation of cooperative wireless networking using two testbeds, as well as the results of the experiments we performed on the testbeds. We discuss the challenges that arose in implementation and the solutions we devised to address them. In addition to the MAC layer implementation, we present one of the first efforts on the implementation of cooperative-coding schemes in the PHY layer. Through the development of these schemes in a real environment, we show clearly the significant benefits of cooperation in wireless networks. Our ongoing work includes combining the MAC- and PHY-layer implementations into a unified cross-layer scheme for multi-node cooperative networks.

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BIOGRAPHIES

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COOPERATIVE AND RELAY COMMUNICATIONS

Context-Aware Protocol Engines for Ad Hoc Networks

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ABSTRACT

The protocols and architectures of the ad hoc networks of today reflect the severe memory and processing constraints that were imposed on computing equipment that was dedicated to communication tasks 40 years ago. As a result, most protocols are decoupled from the physical medium, each protocol layer operates independently of others, and processing and storage "inside" the network is kept to a minimum. We present the design and performance of a new approach to packet switching for MANETs, which we call a context-aware protocol engine. With a CAPE, nodes disseminate information in the network by means of context-aware packet switching, which enables the statistical multiplexing of bandwidth and the processing and storage of resources using integrated signaling that covers channel access, routing, and other functions, to share and store the context within which information is disseminated. Data packet headers consist of simple pointers to their context, and elections and opportunistic reservations integrated with routing are used to attain high throughput and low channel-access delay.

INTRODUCTION

The protocols and architectures used in mobile ad hoc networks (MANETs) today [1] still reflect the severe memory and processing constraints imposed on computing equipment dedicated to communication tasks 40 years ago [2]. These constraints forced the Advanced Research Projects Agency Network (ARPANET) protocols to be organized into a stack where they were decoupled from the physical medium. Additionally, protocol layers operated independently, and in-network processing and storage was kept to a minimum. This approach led to *in-band*, *in-packet* signaling where routers (packet switches) maintain the minimum amount of information required to forward packets (e.g., a next hop). Hosts attach headers that contain all of the control information for each layer in the protocol stack to data payloads. Each packet (datagram) is switched independently from other packets and also from the intent or payload type of the packet. All network content storage and processing occurs at hosts on the edge of the network.

Given the success of the Internet Protocol (IP) Internet, datagram switching, based on protocol stacks, was arguably the most sensible approach at the time the ARPANET was created. Furthermore, it is still adequate for wired segments of the Internet where link over-provisioning is feasible. However, the *stacked datagram* approach is not ideal for current MANETs with node mobility, radio channel characteristics, and a relative scarcity of bandwidth (compared to wired). Today, the in-network processing and storage power available even in small mobile nodes (e.g., personal digital assistants [PDAs], cellular phones) are orders of magnitude larger than what was available inside a network more than 40 years ago. Although there were compelling cost reasons for a division of labor between hosts and routers, today devices should not be subject to the constraints of the past. Wireless portions of the Internet must be ubiquitous and multi-functioned, acting as both network router and host to provide the content a user requests. This does not necessarily translate high network performance to that of the utilization on a given link. Interestingly, despite hardware advances, new approaches to packet switching in MANETs adhere to at least some of the original approach to packet switching introduced in the ARPANET.

In this article, we demonstrate that context-aware statistical multiplexing of network resources is far more effective (in some cases, by orders of magnitude) than implementing protocol stack-datagram switching. We introduce the context-aware protocol engine (CAPE) as an instantiation of context-aware packet switching for MANETs. The approach we advocate in CAPE is the exact opposite of current protocol architectures. A CAPE is based on nodes storing the entire packet-switching context, where packets contain a payload and a [[*pointer*]] for binding to the stored context. A CAPE employs *in-band*, *off-packet* integrated signaling, where signaling and control information (context) is sent before the corresponding data packets on the same link are sent. This context contains all of the control information required to integrate scheduling, routing, and congestion control using a single protocol. A key advantage of integrating the signaling for the operation of scheduling and routing is that shared channel-access schedules incorporate the constraints imposed by multiple-hop flows and network-level knowledge that neighbors of a node should receive the transmission. A CAPE utilizes the context-aware schedule access (CASA) protocol for signaling and channel access. CASA channel access combines distributed fair elections and a reservation mechanism. The elections determine which nodes are allowed to use or bid for available time slots but remove the requirement for direct node-to-node coordination. Reservations provide channel access-time guarantees to those nodes that successfully accessed the channel and must persist in using the same time slots.

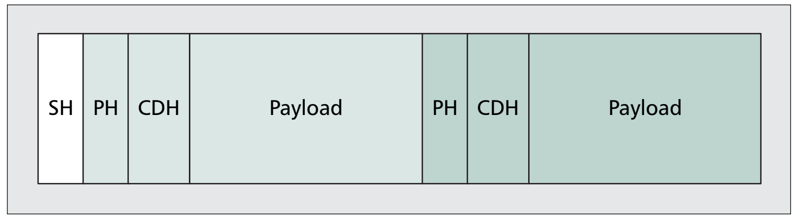
The impact of stacked-datagram design in MANETs is most evident in the low shared-bandwidth efficiency and large overhead incurred with context-free packet switching. Accordingly, although a CAPE spans all the layers of a traditional protocol stack, this article focuses primarily on the benefits obtained with context-aware channel access and integrated signaling. We compare the performance of CASA and the improvements in overhead that are attributed to integrated signaling against the use of traditional protocol stacks based on the IEEE 802.11 distributed coordination function (DCF). The results clearly show that the use of a contention-based channel-access discipline, which in essence attempts to emulate "Ethernets in the sky," is simply not applicable to MANETs with voice and data traffic. The context-aware channel access used in a CAPE provides a solid foundation for the support of multiple media in MANETs. The overhead incurred with datagram switching, implemented with layer-independent signaling, can be substantial when either the payload per packet is small or the duration of an end-to-end flow is relatively long (involving many packets). Our results indicate that the integrated signaling in a CAPE provides reduced overhead even when the signaling associated with route maintenance and end-to-end transport is not considered.

Improving the Transition to a Wireless Medium

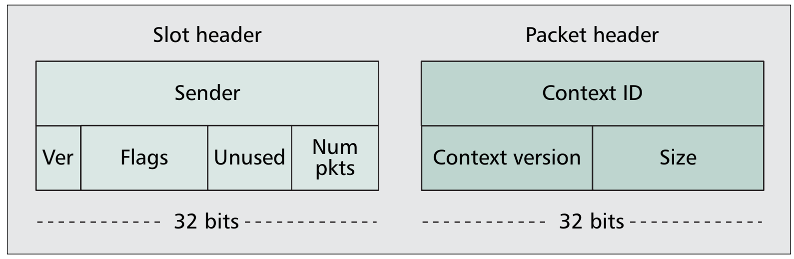
Considerable work in the past attempted to improve ad hoc network efficiency and scalability. We organize this work into protocol-header compaction or compression, cross-layer optimizations, and new protocol architectures.

Protocol-header compaction proposals focus on reducing the headers required for specific protocols without incurring any information loss, whereas compression proposals allow some loss of information. Header-compression or -compaction approaches are based on at least one of the following observations:

- Many header fields of packets in the packet stream of an end-to-end session are the same (e.g., source and destination address, port, version, protocol, flow label, hop limit).



■ **Figure 1.** *CASA packet.*



■ **Figure 2.** *CASA packet header.*

- Nodes can use local identifiers instead of globally unique identifiers, provided that they maintain a mapping for them.
- Protocol headers unnecessarily carry all the fields that the processing of a packet may require.

Most of the attention was given to the overhead of Transmission Control Protocol (TCP), User Datagram Protocol (UDP), and IP in ad hoc networks [3, 4]. What is most striking about all the approaches to date is the continued assumption of a layered-protocol architecture based on in-band, in-packet signaling in which one layer encapsulates the higher layer, and all protocol headers are included in each packet. Simply, the goal is to attempt to reduce the overhead of specific protocols defined within that architecture. Consequently, the benefits of cross-layer interaction are not fully exploited.

Because the characteristics of the physical layer impact the performance of the entire protocol stack in an ad hoc network, recent work attempts to increase efficiency by integrating the operation of multiple layers. To date, cross-layer optimization schemes have focused on exploiting advances at the physical layer and have addressed the following: using multiple channels at the link and network layers, taking advantage of more sophisticated receivers, and utilizing relaying nodes for information processing [5, 6]. These prior results demonstrate the benefits from dynamic approaches to channel division (in time, frequency, and space) and the requirement for concurrent transmission scheduling around receivers, based on their characteristics and nearby channel state, to fully take advantage of multi-packet reception (MPR). Hence, truly scalable protocol architectures for ad hoc networks must consider the use of scheduled channel access.

Very few new architectures were proposed to improve the performance of ad hoc networks. The majority of the architectures proposed for ad hoc networks in the past focused on organiz-

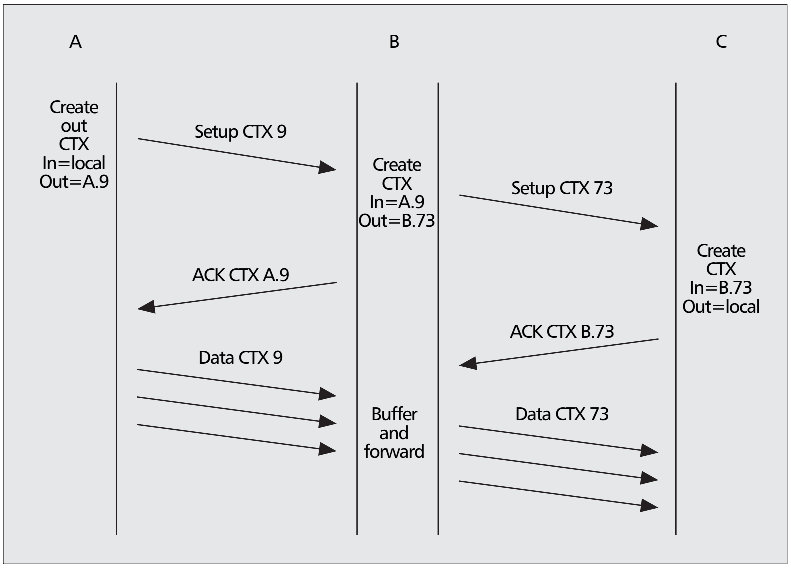


Figure 3. *Context setup exchange.*

ing nodes into clusters (identified either by a cluster identifier or a node identifier) or into a connected backbone that links all nodes to reduce signaling overhead. Recently, however, Ramanathan [7] proposed an architecture based on three layers: a relay-oriented physical layer, a path-access-control (PAC) layer, and a collaborative transport layer. While Ramanathan's proposal captures many of our goals in a CAPE, it has some limitations. The PAC layer is much more subject to unfair access to resources than the current 802.11 DCF. This imbalance results from the required physical-layer resource-reservation handshake that does not incorporate any information about the context of the transmitted flows. In addition, although cooperation among nodes is important, it may not be possible for the transmitters to have accurate channel-state information with quickly moving nodes. In a CAPE, fast moving nodes could be handled by using an election contention area scaled by relative velocity, such that node IDs of fast movers are disseminated over a larger subset of the network. In tactical networks with geo-coordinates, the increased flooding could be scoped to the likely trajectory of a node.

Context-Aware Protocol Engine

In contrast to a wired network, scheduling, routing, congestion control, and retransmission control are very interrelated in a MANET. A transmission schedule, in effect, defines a "link" between a transmitter and a receiver. Establishing and using a route impacts scheduling by maintaining some links and decaying others. Last, established routes determine link congestion, and changes to these routes or the allocation of their traffic impacts congestion. Therefore, establishing the context within which resources are shared and information is exchanged must occur together with channel access, and channel access must be performed jointly with the other network control functions.

Accordingly, in a CAPE, we substitute the traditional protocol stack with:

- A context database that stores the context within which user information is disseminated and network resources are shared.
- The CASA protocol, which is used to access the shared channel and disseminate all context and user information.
- A set of network-control algorithms.

The context database information includes the flows competing for shared bandwidth, the nodes capable of causing interference around receivers, link characteristics, node positions (if available), transmission schedules and routes, other characteristics of the environment that also can help define the context, and state information used by network-control algorithms.

CASA integrates the signaling required for channel access, routing, congestion control, and retransmission control. Instead of having a medium access control (MAC) protocol, a unicast routing protocol, a multicast routing protocol, and a congestion control protocol, where each carries its own independent signaling, nodes exchange their context with one another using CASA. This single version of the context includes all the control information required to attain integrated scheduling, routing, and congestion control. Although CASA supports signaling for multiple functions, the algorithms used to implement different network control functions are *not* integrated into a single algorithm, given that such an optimization problem would be much too complex.

To populate the context database, the CAPE nodes exchange packets to build local network state. These context packets are transmitted through other CASA-like packets and do not require a special out-of-band exchange mechanism. Environment context packets contain information about local neighbors and routes. They occur periodically and maintain routing tables and the neighborhood information required by the scheduler. Flow context packets relate specific data flow information. They are in charge of set up, update, and teardown of the context information required to forward the actual data.

CASA is a time-slotted MAC layer that uses a slot header and a per-packet packet header in aggregated slots. Both header types are 64 bits. The slot header identifies who the sender is, what version of CASA is used, control flags, and the number of packets in the slot. Packet headers contain the identifier of the context within which the packet should be processed, the version of the context identifier, and the size of the packet. This information determines the format of the context-dependent header (CDH) and how to process the packet. An example of data that might be part of the context dependent header is a *link sequence number*.

To establish a data flow, first a node must transmit context set-up packets to the next hop and receive a corresponding context acknowledgment (Fig. 3). In a CAPE, a source identifies a flow with a globally unique flow ID (FID) by appending a local counter to its node ID. The FID is propagated through the network as part of connection set up. The FID takes a role akin to an IP socket descriptor: it uniquely identifies a flow between end points. After a

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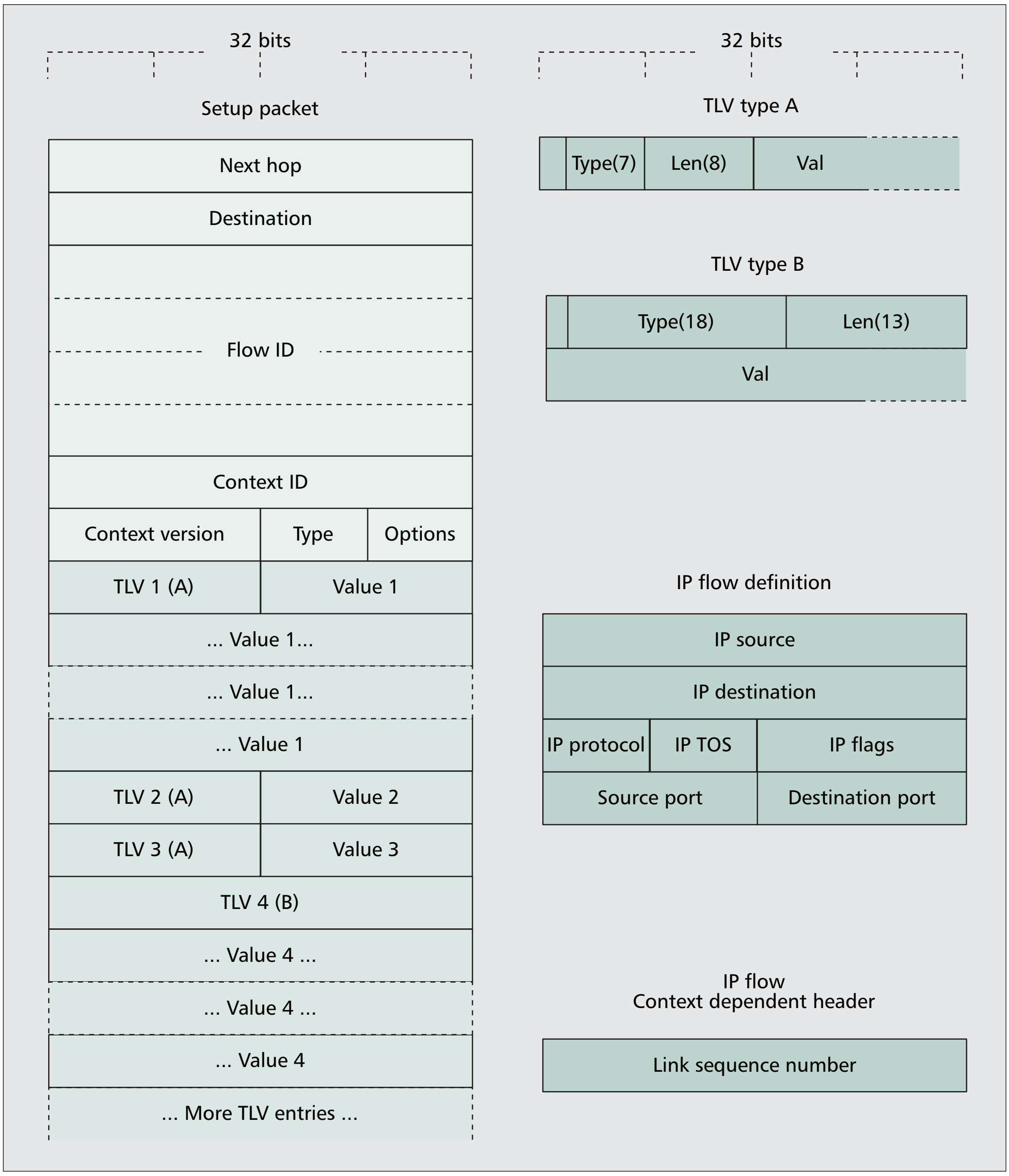


■ **Figure 4.** *CASA packets and fields.*

There are two types of TLV entries differentiated by the first bit. Type A, for most common options, uses seven bits for type and eight bits for length. Type B uses 18 bits for type and 13 bits for length to handle the extended options.

pair of neighbors along a flow path agree on the set up, the nodes switch the flow using the context ID. However, because nodes know the FID associated with specific peer connections, nodes can multiplex a flow over multiple paths. The cost of path repairs in a CAPE is minimized because nodes can reroute a flow locally using the FID.

The context id used for switching flow maps to a stored state from the set-up packet (Fig. 4), including the next hop for the packets, flow destination, FID, context version, flow type (e.g., encapsulated IP or native CAPE), and a set of type-length-value (TLV) entries used to define the flow and create context. There are two types of TLV entries differentiated by the first bit. Type A, for most common options, uses seven bits for type and eight bits for length. Type B uses 18 bits for type and 13 bits for length to handle the extended options. Most set-up packets contain a flow definition (the IP flow definition can be seen in Fig. 4).

The context defines how to interpret the context-dependent header and includes information that must be processed specifically for a given flow. In the case of an IP flow, the header includes a field called *Link Sequence Number*. This is a sequence number for the packets of this flow over the specific link in which they are sent. It is used for link retransmissions and flow control.

Efficiently Accessing the Channel through Cooperation

CASA provides access to a shared channel by means of elections and reservations based on the context information exchanged among nodes. Nodes implement a distributed election

Terrain	2500 m × 1000 m
PHY	802.11a at 12 Mb/s
802.11e	AC3 voice, AC0 TCP
Radio model	Statistical propagation, two-ray ground pathloss, constant shadowing, –111 dB propagation limit
Voice data	56 bytes CBR (17.6 kb/s codec), 30 s exp. turnaround
Bulk data	50 HTTP sessions
CASA frame	400 slots at 0.5 ms/slot (650-byte MTU)
Contention area	4 hops
Guard interval	10 µs
Reservations	300 out of 400 slots reservable
Res timeouts	1601 ms local, 2001 ms neighbor
Res rate	At most 4 new reservations/node/frame

■ **Table 1.** *Simulation parameters.*

algorithm to select which node is allowed to use the time slots that have not been reserved. There is no requirement to configure anything in the schedule other than the number of time slots used in each frame. Nodes that win slot elections can reserve the slots and use them over time to meet channel-access time guarantees. Nodes regularly broadcast their context consisting of neighborhood and reservation information to calculate schedules and propagate reservations. Nodes utilize an in-band time synchronization protocol that does not rely on external time sources, such as generalized processor sharing (GPS).

Neighbor maintenance is used to populate an N-hop neighbor table, which is the basis for the dynamic election of time slots that have not been reserved. A neighbor update consists of a neighbor node ID and the hop count (distance) to the neighbor. When sending context, all neighbors of distance less than *N* (for an *N*-hop neighborhood) are sent at once in one long array of neighbor updates, although an incremental approach is also possible. Neighbor entries time out if not refreshed.

Time-Slot Scheduling

Many of the previous dynamic scheduling protocols [8] offer high channel utilization even at high load, but still have problems with real-time data traffic because their randomized slot assignments do not provide tight enough bounds on channel-access times. Our approach maintains high utilization without sacrificing real-time data flows.

A major difference in the time-slot elections in CASA compared to previous election-based scheduling schemes is that prior work has focused on running an election for each individual time slot, whereas elections in CASA run for an entire frame. The CASA election algorithm is based on generating a pseudo-random permutation of time slots for each node. Then these permutations are compared in slot rank order to determine which node wins the right to transmit in each time slot. Although the permutations are frame-length, the algorithm can be run in amortized time over a frame. For each frame number t, we obtain a frame key $K(t)$ using a globally known hash function $K()$. The key is used as the random number seed for generating the random permutation vectors.

Each node determines the schedule of winning node IDs given the set of random permutation vectors in the matrix $P[node][slot]$ and the set of node weights for tie-breaking. After a node generates the random permutation vectors, it scans them such that the node with the minimum rank for a time slot wins the slot. The weights could be random numbers, or they could be a deterministic value based on node ID. When used with reservations, the reservation for a slot always takes precedence over the election results.

Reservations

The present reservation scheme is designed to support voice calls and, as such, is a "hold until done" strategy for keeping a reservation until it is no longer required. When a node has data to send and is below the reservation limits, it may reserve any slot for which it wins the election. CASA limits a node in the rate of slot reservations and in the maximum number of reservable slots. When a neighbor receives a slot reservation, it propagates the reservation information over the N-hop contention area so the participating nodes can use it in the distributed election. If a reservation is not refreshed, neighbors time out stale reservations using a soft-state approach.

A node may reserve a time slot if several conditions are met. The total reservations held by a node must be less than the global maximum. There is a maximum number of new reservations a node can make per frame. Nodes can keep their slot reservation based on their traffic type and the backlog at the node. To reserve a slot, a node sets the "R" flag in the slot header *flags* field. The node continues to set the "R" flag in the time slot as long as the traffic conditions are still met. A node implicitly releases its reservation after not setting the "R" flag for *Res timeout* (Table 1). It can no longer use the slot without winning it again in an election.

When a node receives a slot header with the "R" flag, it records the slot-header node identifier, the slot number, the current time, and the last-sent timestamp (initialized to 0). As part of a context-control message, every node transmits a list of the known reservations that were updated within a threshold. The reservations are ordered by their last-sent time such that the newest (initialized to 0) and least-recently sent reservations go first. Periodically, for example, on slot 0 of each frame, a node runs a maintenance routine that purges the reservation cache of any expired reservations, based on a time threshold.

A node can hear about multiple reservations for the same time slot. It should track all such reservations because they are not necessarily in conflict. However, if the node is also one of the reserving nodes, it must perform conflict resolution, such that no two nodes in a contention area hold the reservation. We use a random tie-break based on the hash of the node ID and slot number. There are no explicit reservation acknowledgments. Reservations are assumed to be valid unless the node receives a conflicting reservation and determines it lost the tie-break.

TIME SYNCHRONIZATION

We chose the clock-sampling mutual network synchronization (CSMNS) protocol [9] for time synchronization in CASA because of its simplicity. The basis of CSMNS is the use of a proportional controller to drive the clock at each node to a common time. In each slot header, a node transmits its timestamp as a 64-bit integer. To aid convergence, we also adopt the rule that if a node has never received a timestamp before, it adjusts its offset such that the local clock equals the received timestamp.

PERFORMANCE EVALUATION

In this section, we present our evaluation of a CAPE and compare its performance to 802.11e. The simulations are performed for a 50-node static mesh environment, and they evaluate how a CAPE handles a mix of HTTP and voice traffic. All simulations use the optimized link state routing (OLSR) protocol in the Qualnet simulator [10] with the parameters shown in Table 1. All traffic is peer to peer between the 50 mesh nodes. The HTTP traffic uses the built-in Qualnet HTTPlib traffic generator that models page load and user think times. Each HTTP session lasts the entire simulation time between the same two nodes. Each voice call is modeled as constant bit rate (CBR) traffic with a 30-s exponential talk time in one direction; then the direction reverses for another talk spurt. Each pair of distinct CBR end points are chosen uniformly. Some nodes may participate in more than one call.

We consider the metrics of delivery, latency, and HTTP bytes delivered. We show that a CAPE has comparable voice delivery to 802.11e with a few flows, but it is over three times better with many flows. The CAPE scheduling maintains low latency that is over an order of magnitude less delay than 802.11e while delivering more elastic TCP traffic. The graphs show the average of five independent simulation runs and the 95 percent confidence interval (assumed normal).

CASA has few collisions due to being a contention-free MAC protocol, but packets still might be dropped due to fading channels, errors due to additive noise, or collisions in the infrequent case of stale context. All CASA experiments are run with an acknowledgment scheme called *TcpAcks*. This scheme only acknowledges and retransmits TCP traffic; all voice (CBR) and routing traffic is strictly best effort. TcpAck utilizes a basic acknowledgment scheme that transmits an ACK for each TCP packet. Packets are retransmitted after 50 ms if no ACK is received by the sender. Packets are retransmitted a maximum of three times.

Figure 5 shows the delivery ratio of CBR packets. The delivery ratio is the total number of in-order datagrams received by a destination divided by the total number of datagrams sent by all senders. The 802.11e plots use standard 802.11 RTS/CTS/ACK with retransmissions. The CAPE plot using CASA does not send ACKs or retransmit any data. Despite this lack of link automatic repeat-reQuest (ARQ), the CAPE has a fairly flat delivery ratio over all loads. CASA and 802.11 have statistically equivalent delivery ratios when there are only five CBR flows. Beyond five CBR flows, the 802.11 delivery ratio drops quickly below the CASA performance.

Figure 6 shows the end-to-end CBR latency. CASA has a significantly lower delay than

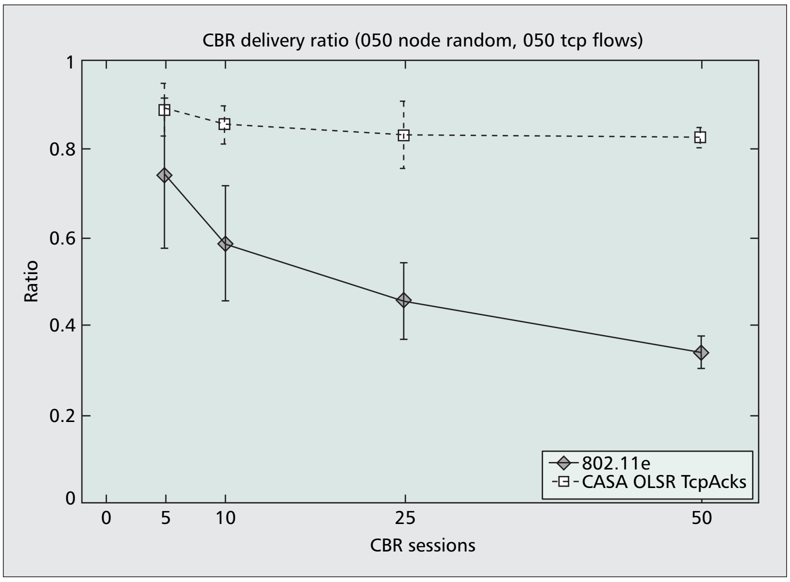


Figure 5. *Delivery ratio.*

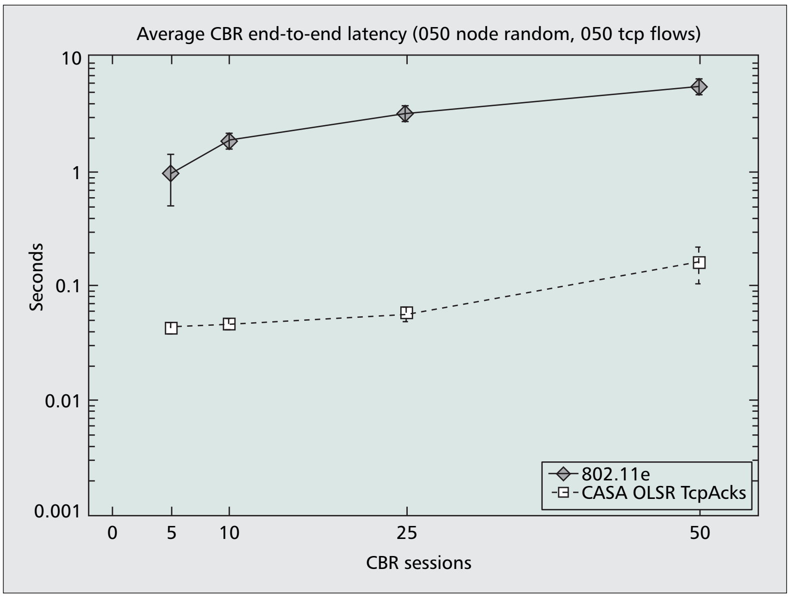
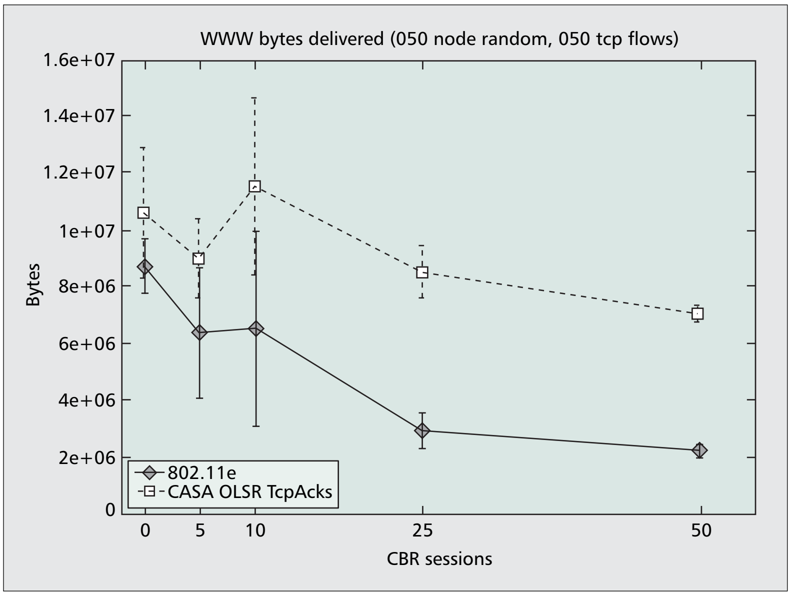


Figure 6. *CBR latency.*



■ **Figure 7.** *HTTP bytes.*

802.11e in all cases. The 802.11e delay ranges from 1 s to 6 s. The CASA delay is between 40 msec to 60 msec for up to 25 CBR flows and then up to 160 msec for 50 CBR flows. This means that even with each node having a 16 kb/s conversation with another node and each node browsing Web pages, CASA maintains a high CBR delivery ratio and low CBR delay, whereas 802.11e breaks down after about five CBR flows.

Figure 7 shows the total HTTP bytes delivered over TCP. In these scenarios, CASA uses the TcpAck mechanism to use link ARQ on TCP packets. 802.11e and CASA deliver statistically equivalent bytes at low load (up to 10 CBR sessions), but at 25 and 50 sessions, CASA delivers about 3× the bytes of 802.11e.

Conclusions and Future Work

We introduced the CAPE as an example of context-aware packet switching in MANETs. A CAPE is based on nodes storing the entire context within which packets are to be switched and having each data packet consisting only of its payload and a pointer to bind it to the stored context. All signaling required in CAPE is exchanged by means of a single protocol, the CASA protocol. We showed that CASA, the protocol used in a CAPE for integrated signaling and channel access, greatly improves channel utilization under heavy loads. It also performs very well in mesh traffic scenarios, enabling the multiple flows to cooperate and share the channel. To make the CAPE a reality, future work must be undertaken and includes the integration of routing with the scheduling and reservations described for channel access in CASA and the introduction of hop-by-hop congestion and retransmission control. Another important future extension of the CAPE is to incorporate fast moving nodes. Because the election scheme requires only a node ID, fast movers could inject their ID along their trajectory, ensuring correct operation.

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Biographies

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Marc Mosko (mmosko@parc.com) received his Bachelor's degree in physics (1989) and a Master's degree in history (1991) from the University of California, Davis, and his doctorate in computer engineering from UCSC in 2004. He is a member of research staff at the Palo Alto Research Center (PARC). From 1992 to 2005, he worked as a network design and security consultant for Fortune 500 corporations and other companies. In 2003 while at UC Santa Cruz, he developed a routing protocol in cooperation with Nokia. His research in ad hoc wireless networks includes analysis and protocols for MAC-layer broadcasting in contention channels and on-demand routing. His MAC layer research includes several statistical and flow-based models of multihop broadcast distribution and a method for statistical optimization of multihop broadcast distribution over a contention channel. At the routing layer, he developed several loop-free on-demand routing protocols with proofs-of-loop freedom and lockout freedom. He currently works on TDMA MAC protocols for multihop ad hoc networks. He has published 14 papers and articles, and filed over 19 U.S. patent applications. He was Local Chair for MobiCom 2008, served on the program committee for INFOCOM 2007, and was a reviewer for other conferences and journals.

IGNACIO SOLIS (isolis@parc.com) received his Ph.D. in computer engineering from UCSC. He is currently a member of research staff at PARC. His experience includes work on flexible header protocols for heterogeneous networks and prototyping architectures for protocol design. He also worked on minimizing delay for data aggregation and optimized mapping in sensor networks. With expertise in ad hoc networking, multicast routing, and collision-free wireless MAC protocols, currently he is focusing on content-centric, disruption-tolerant, and wireless-optimized network architectures.

REBECCA BRAYNARD (rbraynar@parc.com) completed her Ph.D. in computer science at Duke University, Durham, North California. She is a member of research staff at PARC. She also worked in the IBM WebSphere Technology Institute under IBM Fellow Jerry Cuomo. Her expertise is in distributed systems and networking from protocol design to experimental evaluation. Her interests include adaptive, energy-efficient, and resource-constrained protocols, overlay networks, content-based routing, and medium access control. Her thesis describes a novel approach to energy management for ad hoc sensor networks through an asynchronous, asymmetric, and adaptive duty-cycling medium access control protocol. She also collaborated with database researchers to achieve energy management goals for a sensor network through optimized application layer algorithms.

RUMI GHOSH (rumi@ieee.org) is pursuing her Ph.D. degree in computer engineering at UCSC. She received her M.S. in computer engineering from UCSC. Her current research interest is routing in wireless ad hoc networks. She worked as a member of research staff at PARC. She has 10 years of telecommunications industry experience in the areas of 3G, VoIP, and GPRS in several companies in the United States and India.

SERIES EDITORIAL

ANOTHER YEAR IN OPTICS…

Hideo Kuwahara *Jim Theodoras*

As 2008 ends and 2009 begins, one cannot help but reflect on the state of optical communications. All around us, world economies are shaky and futures uncertain. But the long slog of optical technologies continues. Sure, guidance of optical companies has been lowered, yet as of this writing, the growth in Internet traffic continues. One can only wonder at what point people may withdraw from their now firmly ingrained broadband habits. While quad-play and video have been overhyped in the last few years, to say the least, the data actually backs up the projections. Video does in fact consume a magnitude of order more bandwidth, and it has been taxing existing delivery networks, and this has led to continuous infrastructure investments by service providers. Just as MP3 music players forever changed the music distribution model, likewise, broadband has changed the video content distribution model. Perhaps this will be the saving grace of optical communications, at least limiting the impact of the downturn on companies in the optical chain, and at most allowing continuing growth.

This month, we start with an ECOC '08 conference report from Peter Van Daele, General Chair. ECOC continues to be wildly successful on many fronts, including consecutive years, breadth of sponsorship, attendance, and quality of programs.

One of the differences in video content from traditional data traffic is that video is typically transmitted and routed as flows, that is, packets joined together into a stream. The traditional versions of Ethernet and IP are challenged by these video flows, and thus extensions have been added to operations, administration, and maintenance (OAM) protocols to improve the situation. The article by Reddy and Lisle discusses OAM and protection issues encountered in the aggregation and transportation of Ethernet.

Today, one can satisfy their bandwidth cravings through a variety of means, although these can often be divided into two categories, wireless and wired. Traditionally these two approaches have been relatively isolated from one another, although even the best wireless connection must terminate to a wired connection eventually. However, clearly there could be system-level synergies in combining the two more intelligently. In our next article, Ghazisaidi, Assi, and Maier provide an up-to-date survey of hybrid fiber-wireless access networks, sometimes referred to as FiWi networks.

As already mentioned, video content stresses networks and is typically based on packet flows. The majority of these flows eventually reach the core/backbone network, running over a multiprotocol label switching (MPLS) protocol. The next article discusses the first multi-area MPLS and generalized MPLS (GMPLS) interoperability trial over a network including reconfigurable optical add-drop multiplexers (ROADMs) and optical crossconnects (OXCs), with some interesting conclusions reached.

Once again this year, we will be publishing an extra fifth "quarterly" issue of the Optical Communications Series (OCS) timed to coincide with OFC/NFOEC 2009. As in past years, this issue will focus on topics germane to the show, with contributions from leading industry leaders. So be sure to check back next month for the special March edition of OCS.

BIOGRAPHIES

HIDEO KUWAHARA [F] (kuwahara.hideo@jp.fujitsu.com) joined Fujitsu in 1974, and has been engaged for more than 30 years in R&D of optical communications technologies, including high-speed TDM systems, coherent optical transmission systems, EDFA, terrestrial and submarine WDM systems, and related optical components. His current responsibility is to lead photonics technology as a Fellow of Fujitsu Laboratories Ltd. in Japan. He stayed in the United States from 2000 to 2003 as a senior vice president at Fujitsu Network Communications, Inc., and Fujitsu Laboratories of America, Richardson, Texas. He belongs to LEOS and ComSoc. He is a co-Editor of *IEEE Communications Magazine*'s Optical Communications Series. He is currently a member of the International Advisory Committee of the European Conference on Optical Communications, and chairs the Steering Committee of CLEO Pacific Rim. He is a Fellow of the Institute of Electronics, Information and Communications Engineers (IEICE) of Japan. He has co-chaired several conferences, including Optoelectronics and Communications Conference (OECC) 2007. He received an Achievement Award from IEICE of Japan in 1998 for the experimental realization of optical terabit transmission. He received the Sakurai Memorial Award from the Optoelectronic Industry and Technology Development Association (OITDA) of Japan in 1990 for research on coherent optical communication.

JIM THEODORAS (jtheodoras@advaoptical.com) is currently director of technical marketing at ADVA Optical Networking, working on Optical + Ethernet transport products. He has over 20 years of industry experience in optical communication, spanning a wide range of diverse topics. Prior to ADVA, he was a senior hardware manager and technical leader at Cisco Systems, where he managed Ethernet switch development on the Catalyst series product. At Cisco, he also worked on optical multiservice, switching, and transport products and related technologies such as MEMs, electronic compensation, forward error correction, and alternative modulation formats, and was fortunate enough to participate in the "pluggable optics" revolution. Prior to acquisition by Cisco, he worked at Monterey Networks, responsible for optics and 10G hardware development. He also worked at Alcatel Networks during the buildup to the telecom bubble on DWDM long-haul transport systems. Prior to DWDM and EDFAs, he worked at Clarostat on sensors and controls, IMRA America on a wide range of research topics from automotive LIDAR to femtosecond fiber lasers, and Texas Instruments on a variety of military electro-optical programs. He earned an M.S.E.E from the University of Texas at Dallas and a B.S.E.E. from the University of Dayton. He has 15 patents granted or pending.

ECOC CONFERENCE REPORT

REPORT FROM ECOC 2008

BY PETER VAN DAELE, GENERAL CHAIR

In the field of optical communications technology, the European Conference on Optical Communication (ECOC) is the largest event in Europe. This year ECOC 2008 was held in Brussels, Belgium, on September 21–25, 2008, and chaired by Prof. Peter van Daele of Ghent University and Prof. Djan Khoe of the Technical University of Eindhoven. The Technical Program Committee was chaired by Prof. Piet Deemester, Ghent University, and Prof. Ton Koonen, TU Eindhoven. More than 1200 attendees gathered, with approximately 55 percent from Europe.

Topics were divided into six categories: (1) Fibers, Fiber Devices, and Amplifiers; (2) Waveguide and Optoelectronics Devices, (3) Subsystems and Network Elements for Optical Networks; (4) Transmission Systems; (5) Backbone and Core Networks; (6) Access Networks and LAN. 811 papers were submitted with 267 accepted for oral presentation and 129 for poster. This yields an acceptance ratio of 34 percent for oral only, and 50 percent if posters are included. As for post deadline papers, 71 papers were submitted; 25 accepted was a 35 percent acceptance ratio.

ECOC held two symposia on (1) Fiber Radio Convergence and (2) Network Solutions to Reduce the Energy Footprint of ICT, as well as nine Workshops on (1) Multitone Transmission Technologies for Optical Networks, (2) Optical Grids, Drivers, and Applications for High-Performance Optical Networks, (3) Short-Range Optical Networks, (4) Optical Fiber Sensors — Where Are We and What's to Come?, (5) All-Optical vs. OEO Networks, (6) ROADM in NG Optical Transport Networks, (7) Towards Foundries for Photonic Components and ICs, (8) 2nd Workshop on Future Internet Design, and (9) Everything Converged: Today, Tomorrow and After Tomorrow. These symposia and workshops were very timely, covering themes relevant to optical communications today

The conference was accompanied by an exposition with some 330 exhibitors and about 3400 attendees. The exposition floor also held several workshops with titles "Phorce21 (Photonics Research Coodination Europe — Photonics21) — Towards a Bright Future for Europe," "How to Get Access to Pre-Competitive Photonic Components for Free," and "ePIXfab Silicon Photonics Prototyping Services."

Topics at ECOC 2008 covered a wide range of technologies including 100G transmission, coherent detection, OFDM, fiber radio convergence, silicon photonics, and WDM-PON. Multilevel modulation for higher spectral efficiency is a major trend, and this year participants appeared much more conscious of the importance of power efficiency.

Next year ECOC2009 will be held on 20–24 September 2009 in Vienna, Austria.

Ethernet Aggregation and Transport Infrastructure OAM and Protection Issues

Pasula Reddy and Sam Lisle, Fujitsu Network Communications

ABSTRACT

As Ethernet traffic increases rapidly in carrier networks, service providers have an increasing need for Ethernet infrastructure networking in order to scale their Ethernet services. Ethernet infrastructure transports Ethernet traffic over distance, protects the traffic from link and nodal failures, and aggregates Ethernet connections from a large number of low-speed ports onto a much smaller number of high-speed ports. An Ethernet infrastructure allows service providers to inexpensively interconnect end user locations with one another for private line services, and to interconnect end users with the VPLS and IP/MPLS service edges for enterprise, residential, and mobility services. OAM and protection capabilities are critical to enable Ethernet to be deployed as an infrastructure technology. This article reviews several important developments in Ethernet OAM and protection standards, and discusses how those capabilities are vital for the creation of an effective Ethernet infrastructure.

INTRODUCTION

Ethernet traffic is entering North American provider networks at unprecedented rates. For example, retail enterprise Ethernet ports are projected to grow at a 40 percent compound annual growth rate between 2007 and 2012 [1]. Introduced initially in provider networks as a disruptive metro enterprise service, Ethernet is now expanding to inter-metro virtual private network (VPN) services as a layer 2 alternative to multiprotocol label switching (MPLS) VPN services, and has become the overwhelmingly dominant backhaul interface for residential broadband and triple play services delivery. Ethernet is also poised to become the dominant backhaul interface for mobile services with the pending deployment of fourth-generation (4G) technology.

As the volume of Ethernet traffic has increased and the applications for Ethernet have broadened, service provider Ethernet architectures have evolved. For example, initial provider Ethernet networks that provided intra-metro enterprise switched services consisted largely of Ethernet switching platforms in the metro core running 802.1ad provider bridging protocol (or switch/routers running both Ethernet bridging and MPLS protocols), accompanied by simple media conversion or demarcation devices at the customer location. These bridged/routed switched services networks existed separately from networks providing layer 1 Ethernet port transport services that utilized synchronous optical network (SONET) or wavelength-division multiplexing (WDM) point-to-point connections.

These early switched services networks typically utilized no underlying transport among the switch/router service elements or between the switches/routers and the demarcation devices, so every customer port appeared as a port on the service element, and all interconnections between elements used dedicated optical fiber runs and were unprotected at the physical layer. As these networks grew, service providers deployed Ethernet port transport infrastructure that provided layer 1 protection and enabled multiple Ethernet ports among the switches or between customer locations and the service edge to interconnect over the same fiber. However, this transport infrastructure provided no Ethernet aggregation; therefore, each port at the customer location still appeared as a port on the switch/router service element. Similarly, as enterprises began to use Ethernet for Internet access and IP business services access, the transport infrastructure would typically carry each customer port to a dedicated port on the service edge element.

As service providers deployed residential broadband services using platforms that utilized Ethernet as the network interface, it became absolutely critical to not only transport Ethernet ports effectively, but to provide significant Ethernet aggregation back toward the service edge (Fig. 1). Provider networks often complemented layer 1 transport by using separate switch/router service elements to provide the aggregation and protection functions as Ethernet traffic was backhauled to the service edge. While these switch/router platforms often provided the essential aggregation and protection functionality, the operations, administration, and maintenance (OAM) and protection protocols involved were often more suited to routed network functionality rather than classic aggregation network functionality; therefore, the performance, scalability, and cost of the solution were sometimes compromised.

Hence, as the volume of Ethernet traffic grows, Ethernet extends into additional applications such as mobile backhaul, and the need for higher-quality Ethernet private line services emerges, the OAM and protection capabilities of Ethernet reviewed in this article will become more and more vital for smooth operation and scaling of these large networks. It is these capabilities, along with efficient integration into the WDM-based transport layer, that will allow service providers to deploy integrated Ethernet aggregation and transport infrastructures that provide private-line-equivalent E-LINE services and unidirectional multipoint services, and cost-effectively bring large volumes of Ethernet traffic from end customer locations to the switched/routed service edge for E-LAN and IP services delivery.

Ethernet OAM Tools

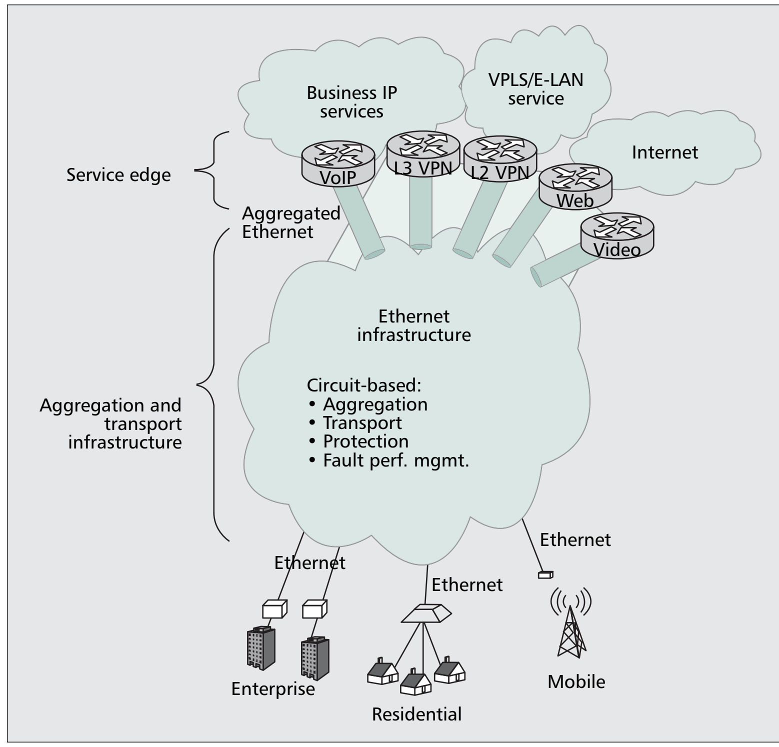
As current and next-generation services migrate to Ethernet, it becomes imperative for Ethernet to support a wide variety of OAM tools that enable providers to capitalize on the simplicity and flexibility of Ethernet, while enabling providers to precisely manage large Ethernet infrastructures. Since many important services are currently being delivered over SONET infrastructure, the Ethernet OAM toolkit needs to provide at least comparable functionality. In addition, these tools must enable providers to offer more measurable, yet granular and stringent service level agreements (SLAs) to their customers.

OAM protocols typically comprise the following four components:
- Configuration and service provisioning
- Fault indication
- Diagnostic functions
- Performance monitoring

There are a number of OAM tools available for the Ethernet aggregation and transport infrastructure that can be used effectively to discover network elements, bring up and tear down services, monitor services individually, measure the performance against the SLA contract, and troubleshoot at the network, nodal, link, and per-service levels.

Work on packet-based OAM was largely pioneered for asynchronous transfer mode (ATM) technology, where a variety of mechanisms for fault and performance management, loopbacks, and other functions were developed [2]. Building on this conceptual foundation, the IEEE, International Telecommunication Union — Telecommunication Standardization Sector (ITU-T), and Metro Ethernet Forum (MEF) standards bodies have developed the following OAM protocols that can be used effectively in any network that uses an underlying Ethernet transport infrastructure to deliver services:
- IEEE 802.1AB [3]
- IEEE 802.3ah [4]
- IEEE 802.1ag [5]
- ITU-T Y.1731 [6]
- MEF 16 E-LMI [7]

These protocols enable provisioning, monitoring, and troubleshooting of E-LINE, E-LAN, and E-TREE services that are delivered completely within the Ethernet transport network. In addition, they can also be used in Ethernet transport networks that are used to aggregate traffic from customer locations and hand it off to the service edge network elements (which are predominantly IP/MPLS-based) or backhaul mobile traffic from a base station at a cell tower to the mobile switching center.



■ **Figure 1.** *Ethernet aggregation and transport infrastructure.*

Service	E-LMI	Y.1731
Network	802.1ag	
Transport	802.3ah	
Discovery	802.1AB	

■ **Figure 2.** *Ethernet OAM protocols.*

These protocols operate at different layers within the Ethernet stack, as shown in Fig. 2, and serve different purposes, as discussed in the following paragraphs.

Discovery Layer

The discovery layer is not directly tied to the transport, network, and service layers. The protocols within the discovery layer assist in dynamically discovering attributes of physical links on network elements. This information is typically exported to the network management systems, and used for creation of topological maps and assisting in end-to-end path computation. IEEE 802.1AB (Link Layer Discovery Protocol) is used at this layer to discover physical links on network elements.

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The transport layer is the physical or link layer within the Ethernet stack. IEEE 802.3ah Link Level OAM protocol is used at this layer for monitoring and isolating faults. In addition, IEEE 802.1ag can also be used at this layer for monitoring and fault-detection purposes.

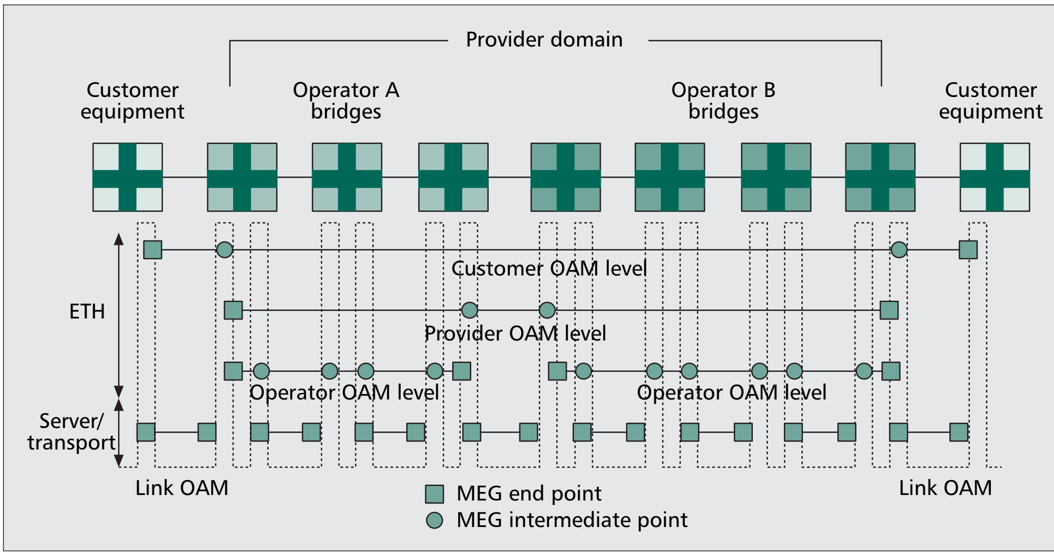

Figure 3. *802.1ag operating over multiple operator domains.*

TRANSPORT LAYER

The transport layer is the physical or link layer within the Ethernet stack. IEEE 802.3ah link-level OAM protocol is used at this layer for monitoring and isolating faults. In addition, IEEE 802.1ag can also be used at this layer for monitoring and fault -detection purposes.

IEEE 802.3ah operates on point-to-point links between Ethernet devices. Since it operates at the link or physical level, 802.3ah does not have any *service awareness*. 802.3ah relies on OAM protocol data units (PDUs) exchanged between the two Ethernet devices at either end of the point-to-point link. These OAM PDUs conform to the slow protocol exchange rates (maximum rate of 10 frames/s). As a result, the 802.3ah OAM PDUs can be generated and processed in software.

802.3ah supports the following functions:

- OAM discovery: Discover OAM capabilities on a peer device.
- Link monitoring: Event notification when error thresholds on the link exceed pre-set values.
- Remote failure indication: Notifies peer that the receive path is down or the link is slowly degrading in quality.
- Remote loopback: Puts the peer in intrusive loopback state to test the link and the peer. Statistics can also be collected while testing the link. These loopback messages are initiated on operator command.

The link events supported by 802.3ah include:

- Errored symbol period event: This event is triggered when the number of errored symbols exceed a preconfigured threshold within a window (measured in number of symbols).
- Errored frame event: This event is triggered when the number of errored frames exceed a preconfigured threshold within a time period (measured in 100 ms time intervals).
- Errored frame period event: This event is triggered when the number of errored frames exceed a preconfigured threshold within a window (measured in number of received frames).
- Errored frame seconds summary event: This event is triggered when the number of errored frame seconds exceed a preconfigured threshold within a window (measured in 100 ms time intervals).

Ethernet devices running the 802.3ah protocol can be in geographically disparate locations, enabling providers to monitor and isolate faults remotely without a truck roll.

NETWORK LAYER

The network layer deals with the forwarding of Ethernet frames based on tunnel identifiers within the frame such as VLAN tags. This layer could be used as the aggregation component for the service layer. For example, multiple EVPL services could be aggregated into the same Ethernet tunnel, where the tunnel is represented by the S-Tag in 802.1ad/point-to-point Q-in-Q networks or a B-Tag in 802.1Qay (PBB-TE) networks. (Individual EVPL service instances embedded within the tunnel are represented by the C-Tags in Q-in-Q networks or I-SIDs in 802.1Qay networks.) The IEEE 802.1ag connectivity fault management protocol is used at this tunnel layer for fault detection, network monitoring, and fault isolation.

The IEEE 802.1ag protocol enables providers to detect faults within milliseconds from the time they occur and also provides tools for isolating the faults. 802.1ag is a flexible hierarchical protocol that can be enabled at multiple levels and multiple layers [8]. It allows providers to partition their networks into multiple operational domains and end-to-end services to span multiple domains/carriers. For example, a service provider could provide an end-to-end EVPL service that spans two different operator networks, as shown in Fig. 3. 802.1ag allows each of the two operators to enable 802.1ag functionality independently within their networks, while also allowing the service provider and even the customer to enable end-to-end 802.1ag functionality that spans multiple operator networks.

Each node participating in an 802.1ag session is either a maintenance endpoint (MEP) or a maintenance intermediate point (MIP). As the names suggest, these represent the ingress/egress nodes and transit nodes within a maintenance domain.

802.1ag protocol supports the following management functions:
- Continuity check messages (CCMs): These are exchanged among MEPs to detect loss of continuity or incorrect network connections. These messages contain Remote Defect Indication flags to report faults to other MEPs. CCMs can be sent every 3.3 ms, thereby ensuring that faults are detected within milliseconds from the time they occur.
- Loopback messages: These can be used to verify connectivity to remote MEPs and MIPs. Loopback messages are typically initiated by operator command as an in-service operation. These can also be used as an out-of-service diagnostic test.
- Linktrace messages: These messages are typically initiated by operator command and can be used to trace the path to remote MEPs and MIPs.

Providers can use 802.1ag management functions to monitor, detect, and isolate faults at the network layer. In Ethernet aggregation and transport networks, where the network layer could potentially be aggregating thousands of services into the same Ethernet tunnel, CCMs can be run for the tunnel at either 3.3 or 10 ms granularity rather than running CCMs on individual EVC services. This allows providers to monitor the Ethernet transport tunnels very aggressively and scales extremely well, since the total number of transport tunnels in an Ethernet aggregation and transport network is significantly lower than the number of services carried within those tunnels. In addition, these tunnel CCM messages can trigger protection switching immediately after the detection of faults, thereby allowing providers to meet 50 ms protection switching SLA requirements equivalent to SONET private line requirements.

Service Layer

The service layer deals with individual service instances, where a service instance represents a unique customer/subscriber and/or a unique flow within a customer/subscriber traffic stream. IEEE 802.1ag, MEF 16 E-LMI (Ethernet — Local Management Interface), and ITU-T Y.1731 are the protocols used at this layer for service provisioning, fault detection, and isolation within the scope of the service, service performance monitoring, and measurement.

E-LMI helps operators turn up services rapidly by automating the provisioning of Ethernet service attributes on attached customer premises equipment (CPE). It is an asymmetric protocol that allows the user–network interface (UNI)-N device to communicate relevant service related attributes to the CPE as shown in Fig. 4.

A few examples of some of these EVC attributes are:
- EVC state on the provider's Ethernet NE.
- UNI status: Conveys the UNI-N status and other service attributes of the UNI.
- C-VLAN ID to EVC mapping: This is used to convey information on how the CE-VLAN IDs are mapped to specific EVCs.

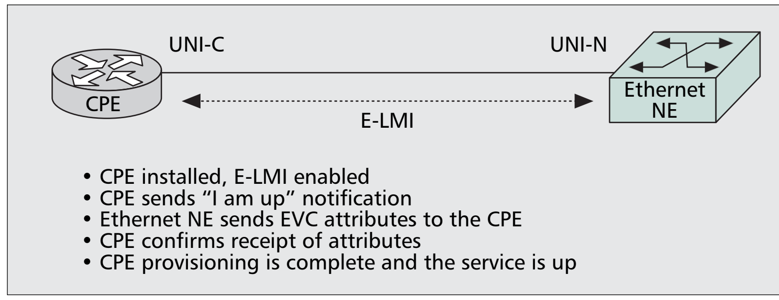

Figure 4. *E-LMI operation.*

- BW Profiles: Conveys bandwidth attributes such as CIR, CBS, EIR, EBS. These attributes can be used by the CPE to ensure that traffic originating from it conforms to the ingress bandwidth profiles agreed on in the SLAs.

IEEE 802.1ag can also be used at the service layer to monitor and troubleshoot individual service instances. Since protection switching is typically bound to the network layer, the CCM timers for the 802.1ag sessions running at the service layer are typically less aggressive, in the seconds/minutes interval granularity. 802.1ag sessions at the service layer can also be enabled on demand to monitor any connectivity problems associated with individual service instances that are carried within the Ethernet transport tunnels (which operate at the network layer).

ITU-T Y.1731 also supports OAM functionality at the service layer. This protocol offers a few additional features on top of the IEEE 802.1ag protocol. It supports alarm indication signals (AISs), which can be used to propagate defect detection from a lower maintenance level to a higher maintenance level. When network elements receive AIS frames, the receiving network element records the AIS conditions, but does not generate loss of continuity alarms with peer MEPs at the local service layer. This benefits operators by suppressing potentially thousands of unwanted alarms on individual services that may be caused by underlying network or transport layer faults. Y.1731supports measurements of performance parameters (frame loss ratio, one-/two-way frame delay, and frame delay variation) at the service instance granularity. These can serve as important tools for providers to measure their network performance and also document these measurements to prove to end customers that SLAs are being met. Typical implementations require specialized hardware assistance to measure the performance accurately. This can potentially translate to an increase in the cost of equipment that supports this functionality.

OAM Protocol Network Application

Having reviewed the major Ethernet OAM protocols, it is instructive to examine how these protocols work together in a simple example network environment. Figure 5 shows an Ethernet aggregation and transport infrastructure composed of Ethernet network elements bringing traffic from a customer edge (CE) router to another CE router or provider edge (PE) router.

E-LMI runs at the service layer over the Ethernet UNI between the CE router and the Eth-

E-LMI runs at the service layer over the Ethernet UNI between the CE router and the Ethernet network. This allows providers to turn up new services or modify existing services without dispatching a technician.

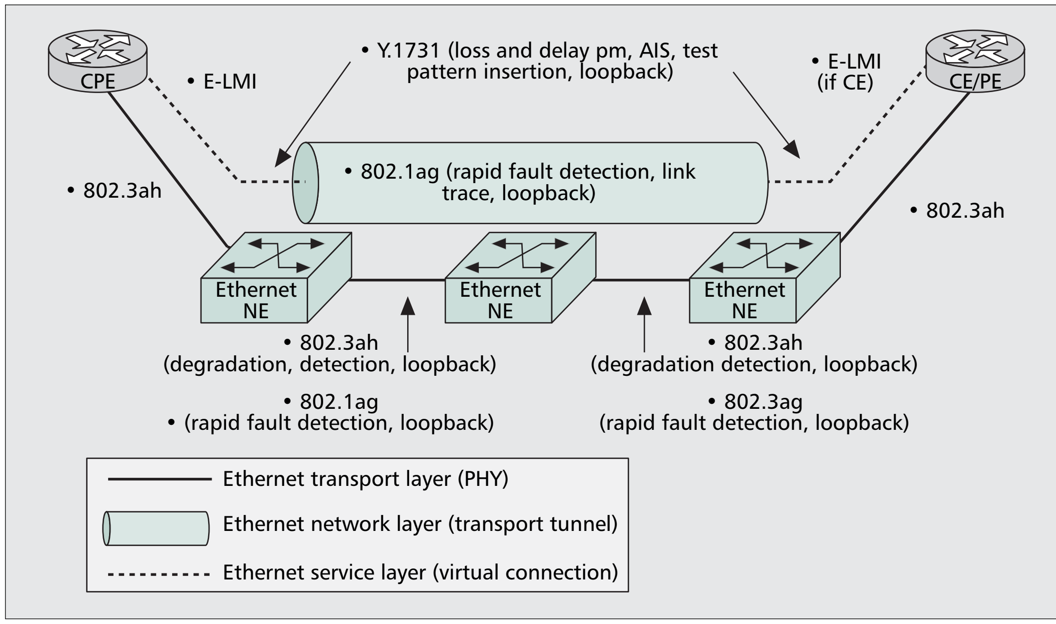

Figure 5. *Ethernet OAM protocol application summary.*

ernet network. This allows providers to turn up new services or modify existing services without dispatching a technician.

The 802.3ah link OAM protocol is used over all physical links in the network. This has particular value on UNI links to proactively monitor customer to network links and determine whether degradation defects exist within the provider network. 802.3ah can also be utilized on links within the network, but the slow protocol nature does not allow rapid detection of defects, so 802.1ag is also helpful on these links.

802.1ag operates on Ethernet transport tunnels at the network layer, enabling providers to monitor the Ethernet transport tunnel health, detect faults within the 10 ms benchmark established in SONET networks, and trigger protection switching onto a preprovisioned alternate path. Once faults have been detected (and traffic protected), the faults can be isolated using the loopback and linktrace messages.

802.1ag at the service layer monitors the connectivity at individual service instance granularity. The timers for the CC messages used at the service layer may be less aggressive than those for CC messages used in the network layer, because the network layer CC messages are used to trigger protection switching events on the transport tunnels.

Y.1731 can be used at the service layer to monitor connectivity, and also measure loss and delay performance at individual service instance granularity. Providers may choose to measure these PM characteristics for their high-touch customers with mission-critical applications.

Because the number of service instances can potentially run into hundreds of thousands within a large Ethernet aggregation and transport network, network providers need to consider the potential scalability implications if they enable very aggressive (millisecond granularity) 802.1ag or Y.1731 CC timers. In addition, if network providers do so, they also need to consider the amount of OAM traffic traversing their network and take into account the bandwidth used by this OAM traffic in their call admission control (CAC) algorithms.

Occasionally, there is some confusion regarding the relative roles of IEEE 802.1ag and ITU-T Y.1731. Y.1731 comprises both fault management and performance management. The fault management functionality specified within Y.1731 is very similar to the IEEE 802.1ag specification, the only major exception being additional AIS functionality, which is present in Y.1731 but not in 802.1ag. Performance management functions such as frame delay measurements are only specified in Y.1731. Carriers looking to support performance measurement functions may be better served using Y.1731 for both fault management and performance management.

Depending on the service and application requirements, one or more of these tools can be used. For example, a service that offers E-LINE services completely within the Ethernet transport infrastructure network may enable the OAM protocols at all the layers, while a service that is backhauling broadband traffic to a BRAS may only enable the OAM protocols at the transport and network layers. The key point to note is that Ethernet provides OAM tools comparable to *non-packet-based* transport infrastructure OAM tools and also to other *non-Ethernet-based* packet transport infrastructure (e.g., ATM) OAM. These OAM protocols enable providers to implement a faster migration plan to an Ethernet-based packet transport infrastructure for their current and next-generation services/applications.

Interlayer OAM Relationships — Importantly, the OAM protocols at different layers complement each other. Defects detected within the OAM protocols at the lower layers can be propagated up to the higher-layer OAM protocols on an as-needed basis so that the higher-layer protocols can take appropriate actions. Some faults such as link quality degradation may not necessarily be detected by higher-layer OAM protocols.

For example, 802.3ah link layer OAM at the transport layer may detect that the number of errored frames received within a time period has exceeded the configured threshold. This event needs to be propagated up to the network layer OAM protocol (802.1ag), which will enable it to set the Remote Defect Indication bit in the CCMs that it generates at the network layer. This allows the OAM protocol peers at the network layer to detect the fault. This information can be used as a protection switching trigger to switch the network layer transport tunnel to an alternative path, even before faults are explicitly detected using the triggers available natively at that layer. This could drastically minimize the service quality disruption times at the individual service instance granularity. It is important to note that, in this scenario, OAM protocols at the service layer will not declare any faults at that layer, preventing potentially thousands of unneeded alarms from being raised.

As discussed, 802.1ag CFM protocol can be instantiated at multiple layers. For example, 802.1ag can be enabled at the transport layer to detect loss of connectivity in tens of milliseconds to initiate a protection switching event. Although 802.3ah can also detect the loss of connectivity at the transport layer, there is a significant difference because 802.3ah link OAM is a slow protocol, which by definition implies that the failure detection times can only be on the order of hundreds of milliseconds at best. 802.1ag can also be used at the service layer to monitor and troubleshoot individual service instances, but it does not offer any PM capabilities. Y.1731, on the other hand, offers these PM capabilities, which can be used by providers to measure SLAs at service instance granularities.

Standards and Implementation Status — Although the ongoing Ethernet OAM specification efforts within the IEEE and ITU organizations are progressing well, as of this writing, there is a need to define additional specifications focusing on OAM interworking functions between the Ethernet and IP/MPLS layers. As carriers migrate to Ethernet-based access and aggregation networks that feed into MPLS-based service networks, interworking between Ethernet OAM and MPLS OAM becomes very critical.

Since most services are typically individually identified across the inter-metro core at the MPLS layer and encapsulated in pseudowires, interworking between 802.1ag (or Y.1731 fault management functions) and VCCV is required to be able to propagate faults from one network domain to another, and also to provide end-to-end service traceability. If an MPLS network only relies on the Ethernet layer for providing layer 1/layer 2 connectivity between various MPLS-enabled network elements, the only required functions are monitoring and fault detection capabilities. 802.3ah and 802.1ag can be deployed to meet these requirements.

Concerning implementation status, as of this writing IEEE 802.3ah is being deployed by a large number of carriers for link monitoring functions. IEEE 802.1ag (or the fault management functionality within ITU-T Y.1731) is increasingly being considered by various carriers as a mechanism to allow better service visibility to their customers. The performance management/measurement functionality specified within ITU-T Y.1731 is also being closely looked at by a large number of carriers, especially as they begin to offer new types of services that require meeting very stringent SLAs. Interoperability between different vendor implementations is an issue of concern to carriers, because they are forced to use the *lowest common denominator* capabilities in their multivendor networks. In some cases, this limits carriers from being able to measure and guarantee very granular SLAs. The same concerns apply to services that span multiple-carrier networks

Ethernet Protection Protocols

Service availability (uptime) and protection switching speed are critical requirements for Ethernet services and Ethernet-based access to IP services. As end users migrate mission-critical services away from TDM onto Ethernet, it becomes crucial for the Ethernet aggregation and transport infrastructure to offer availability and protection switching performance equivalent to that of SONET networks. Therefore, the Ethernet aggregation and transport network must be able to recover from link and/or nodal failures within 60 ms of a failure (10 ms for detection and 50 ms for switching). Ethernet spanning tree protocol (STP) variations are not capable of this level of performance.

There are two Ethernet forwarding mechanisms that provide Ethernet aggregation and transport infrastructure:

- Ethernet aggregation and transport over point-to-point VLAN(s)
- Ethernet-based packet aggregation and transport over IEEE 802.1Qay (PBB-TE) [9]

802.1Qay is a more scalable technology, but service providers have a deployed base of VLAN-based aggregation and a desire to transition toward 802.1Qay over a period of time.

The point-to-point VLAN model aggregates and transports traffic from the customer edge to the service edge using VLAN tag(s). Customer traffic is isolated using a combination of VLAN tags, typically using Q-in-Q encapsulations. In the simplest case, services can be delivered over S-Tags. To increase the scalability, services can also be delivered over S- and C-Tagged connections. In this model both the S-Tag and C-Tag have significance within the provider network, enabling them to scale the number of service instances much more efficiently.

802.1Qay provides a more scalable forwarding mechanism that meets all the functional and availability requirements of Ethernet-based packet aggregation and transport networks. 802.1Qay uses an extended service identifier (I-SID) to embed individual services instances that exist at the service layer into a backbone tag (B-Tag) that exists at the network layer. It offers a highly scalable and efficient protection mechanism by ensuring that all the service instances between a given pair of ingress and egress nodes fully *fate-share* with the associated transport tunnel at the network layer. This allows the protection scheme to be completely implemented within the network layer, thereby reducing the number of protected sessions in the network drastically.

As end users migrate mission critical services away from TDM onto Ethernet it becomes crucial for the Ethernet aggregation and transport infrastructure to offer availability and protection switching performance equivalent to that of SONET networks.

The G.8031 protocol allows for 10ms fault detection and 50ms automatic protection switching that can operate at the service or network layers. It can be used at the network layer to provide protection for point to point VLAN based forwarding mechanisms and can optionally be applied to 802.1Qay networks.

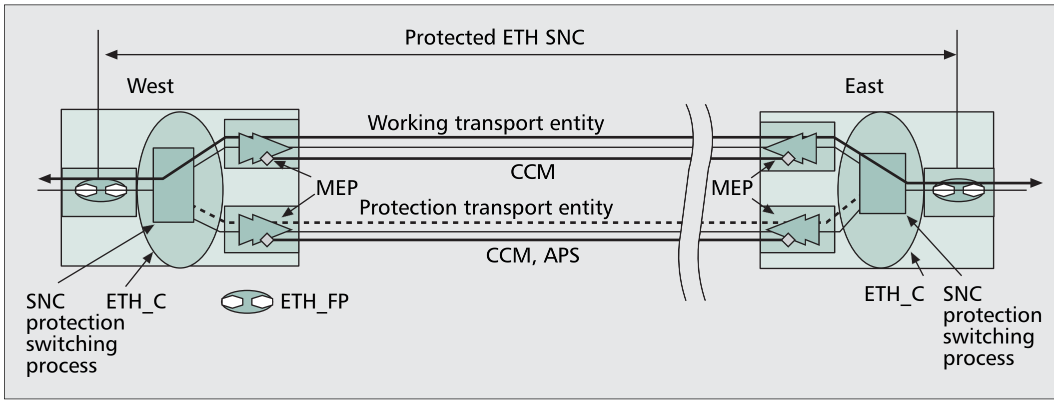

Figure 6. *G.8031 APS protocol.*

Both forwarding mechanisms create point-to-point Ethernet networks with no MAC learning and flooding functions enabled. Therefore, protection mechanisms in these networks are not bound by xSTP reconvergence times while recovering from failures. Each forwarding mechanism has a protection switching capability that allows the transport infrastructure to deliver 50 ms automatic protection switching.

The G.8031 protocol [10] is a robust ITU standard that allows for 10 ms fault detection and 50 ms automatic protection switching that can operate at the service or network layers. This protocol can be used at the network layer to provide protection for point-to-point virtual LAN (VLAN)-based forwarding mechanisms and optionally be applied to 802.1Qay networks.

G.8031 supports the following protection modes:
- 1+1 bidirectional protection switching
- 1:1 bidirectional protection switching
- 1+1 unidirectional protection switching

G.8031 uses an APS PDU to signal to the far end that a protection switching event needs to be triggered. This is required to ensure that traffic is *co-routed* (carried along the same physical paths) in both the forward and reverse directions (for bidirectional service connectivity). Working and protect paths are preprovisioned, and both can be actively monitored using the CC messages (running at 3.3 or 10 ms granularity) in the 802.1ag CFM protocol. It is to be noted that the APS PDUs are only sent on the protect path as described in Fig. 6.

G.8031 supports a variety of triggers for protection switching. Some of the triggers are:
- Signal fail: This can either be detected locally or using 802.1ag CC messages.
- Signal degradation: This can be detected either locally or using any of the OAM protocols described in the previous section.
- Operator initiated switchovers: Operators can initiate switchovers manually to perform hardware upgrades and so on.

In addition, the G.8031 protocol supports both *revertive* and *non-revertive* modes of operation and options to control the time interval before a revertive switching event can be initiated.

The G.8031 APS function can be enabled at either the service instance granularity or on a group of service instances that fate-share the path between a given pair of ingress and egress network elements, using the *test trail* functionality. Invoking the protocol at the network layer offers much higher scalability than invoking the protocol on a per service instance basis.

PBB-TE can optionally utilize G.8031, but also supports a simplified scheme that allows for load sharing. Working and protect paths are preprovisioned between pairs of ingress and egress network elements within the 802.1Qay network. Like G.8031, this mechanism also assumes that the paths in the forward and reverse directions are co-routed. The working and protect paths are monitored at the network layer using 802.1ag CC messages (running at 3.3 or 10 ms granularity). Once a failure is detected by the tail end node, it signals to the head-end using the RDI bit in the 802.1ag CC messages. This indicates to the head end that it needs to initiate a protection switching event. An APS PDU is not used in 802.1Qay protection switching. The additional flexibility 802.1Qay offers is load sharing support between the working and protect paths/tunnels. Like G.8031, 802.1Qay supports revertive and non-revertive modes of operation with options to control how long to wait before reverting to the working path. 802.1Qay supports only a 1:1 bidirectional protection switching mechanism

Since failures are detected within tens of milliseconds from when they occur and the protection switching event is triggered as soon as the failures are detected, 802.1Qay networks recover from link and/or nodal failures within 50 ms. In addition, since the protection switching state machine runs at the network layer, it does not suffer from any of the potential scalability concerns from which other schemes that rely on protection switching state machines running at individual service instance granularity do.

As we can see, Ethernet offers two simple yet efficient protection protocols to meet aggregation and transport infrastructure requirements that could not be addressed by spanning tree protocols. This allows providers to migrate services from a SONET-based transport infrastructure to an Ethernet infrastructure without compromising protection switching speed.

Conclusions

As Ethernet applications continue to proliferate and traffic continues to grow, service providers must scale their deployments by constructing

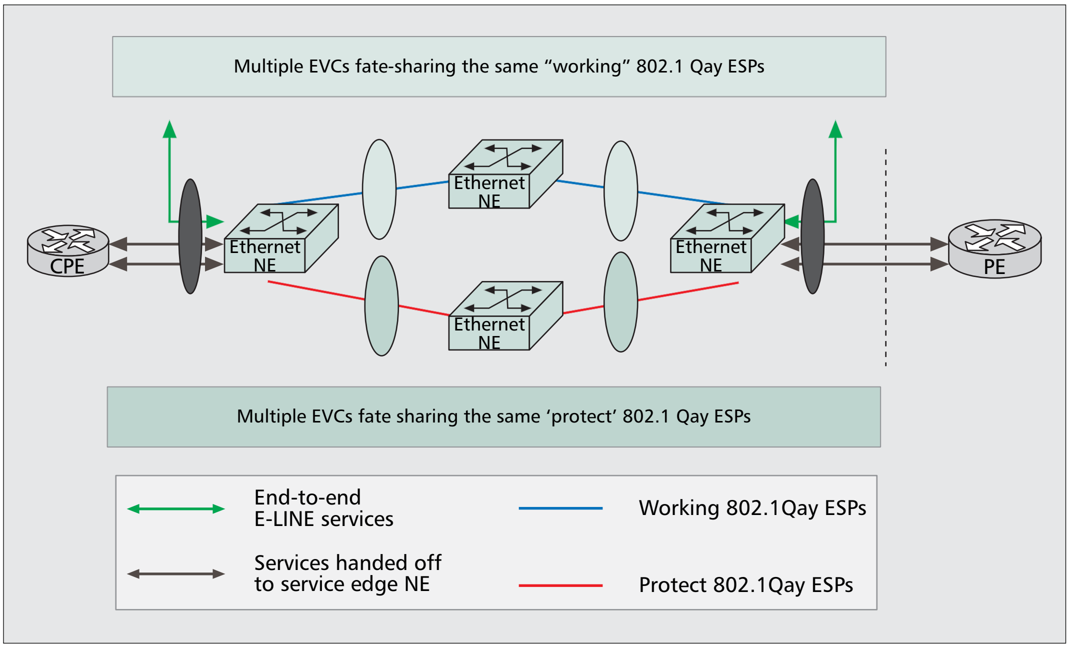

Figure 7. *Protection in 802.1Qay.*

Of particular importance is the ability to operate OAM and protection protocols at the Ethernet network layer on aggregated tunnels where a small number of protocol instantiations provide fault detection and sectionalization and protection switching for a large number of Ethernet services.

Ethernet aggregation and transport infrastructure networks. These infrastructure networks cost-effectively aggregate, transport, and protect point-to-point Ethernet connections between end user locations providing native E-LINE services, and between end user locations and IP/MPLS/VPLS service edges. Just as SONET infrastructure has provided precision fault sectionalization, performance management, and rapid automatic protection switching, Ethernet infrastructure must provide similar functionality to support the many mission-critical applications of enterprise, wholesale, and mobility users in particular.

This article has reviewed key Ethernet OAM and protection switching protocols that have been standardized by the IEEE, ITU, and MEF. These protocols are essential enhancements to evolve Ethernet beyond a simple switched metro enterprise service. These protocols operate at various Ethernet layers, including transport, network, and service layers. The protection switching protocols enable dedicated 50 ms protection switching, which is identical in switching speed performance to SONET networks that have set the industry benchmark for protection performance. Of particular importance is the ability to operate OAM and protection protocols at the Ethernet network layer on aggregated tunnels where a small number of protocol instantiations provide fault detection and sectionalization and protection switching for a large number of Ethernet services.

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TOPICS IN OPTICAL COMMUNICATIONS

Fiber-Wireless (FiWi) Access Networks: A Survey

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ABSTRACT

This article provides an up-to-date survey of hybrid fiber-wireless (FiWi) access networks that leverage on the respective strengths of optical and wireless technologies and converge them seamlessly. FiWi networks become rapidly mature and give rise to new powerful access network solutions and paradigms. The survey first overviews the state of the art, enabling technologies, and future developments of wireless and optical access networks, respectively, paying particular attention to wireless mesh networks and fiber to the home networks. After briefly reviewing some generic integration approaches of EPON and WiMAX networks, several recently proposed FiWi architectures based on different optical network topologies and WiFi technology are described. Finally, technological challenges toward the realization and commercial adoption of future FiWi access networks are identified.

INTRODUCTION

The ultimate goal of the Internet and communication networks in general is to provide access to information when we need it, where we need it, and in whatever format we need it in. To achieve this goal, wireless and optical technologies play a key role. Wireless and optical access networks can be thought of as complementary. Optical fiber does not go everywhere, but where it does go, it provides a huge amount of available bandwidth. Wireless access networks, on the other hand, potentially go almost everywhere, but provide a highly bandwidth-constrained transmission channel susceptible to a variety of impairments. Clearly, as providers need to satisfy users with continuously increasing bandwidth demands, future broadband access networks must leverage on both technologies and converge them seamlessly, giving rise to *fiber-wireless* (FiWi) access networks.

Passive optical networks (PONs) might be viewed as the final frontier of optical fiber to the home (FTTH) or close to it (FTTX) networks, where they interface with a number of wireless access technologies. One interesting approach to integrate optical fiber networks and wireless networks are so-called radio-over-fiber (RoF) networks. RoF networks are attractive since they provide transparency against modulation technique,s and are able to support various digital formats and wireless standards in a cost-effective manner, for example, wideband code-division multiple access (WCDMA), IEEE 802.11 wireless local area network (WLAN), personal handyphone system (PHS), and Global System for Mobile Communications (GSM) [1]. To realize future multiservice access networks, the seamless integration of RoF systems with existing and emerging optical access networks is important, such as FTTX and wavelength-division multiplexing (WDM) PON networks. RoF networks are also well suited to avoid frequent handovers of fast-moving users in cellular networks. An interesting approach to avoid handovers for train passengers is the use of an optical fiber WDM ring-based RoF network installed along the rail tracks in combination with the moving cell concept, as recently proposed in [2]. The concept of moving cells enables a cell pattern and a train to move along on the same radio frequency during the whole connection in a synchronous fashion without requiring handovers.

In this article we assume that optical fiber paves all the way to and penetrates into the homes of residential and business customers. Arguing that, due to its unique properties, optical fiber is likely to entirely replace copper wires in the near to midterm, we elaborate on the final frontier of optical networks: convergence with their wireless counterparts. Optical and wireless technologies are expected to coexist over the next decades. Future broadband access networks will be bimodal, capitalizing on the respective strengths of both technologies and smartly merging them in order to realize future-proof FiWi networks that strengthen our information society while avoiding its digital divide. By combining the capacity of optical fiber networks with the ubiquity and mobility of wireless networks, FiWi networks form a powerful platform for the support and creation of emerging as well as future unforeseen applications and services (e.g., telepresence). FiWi networks hold great promise to change the way we live and work by replacing commuting with teleworking. This not only provides more time for professional and personal

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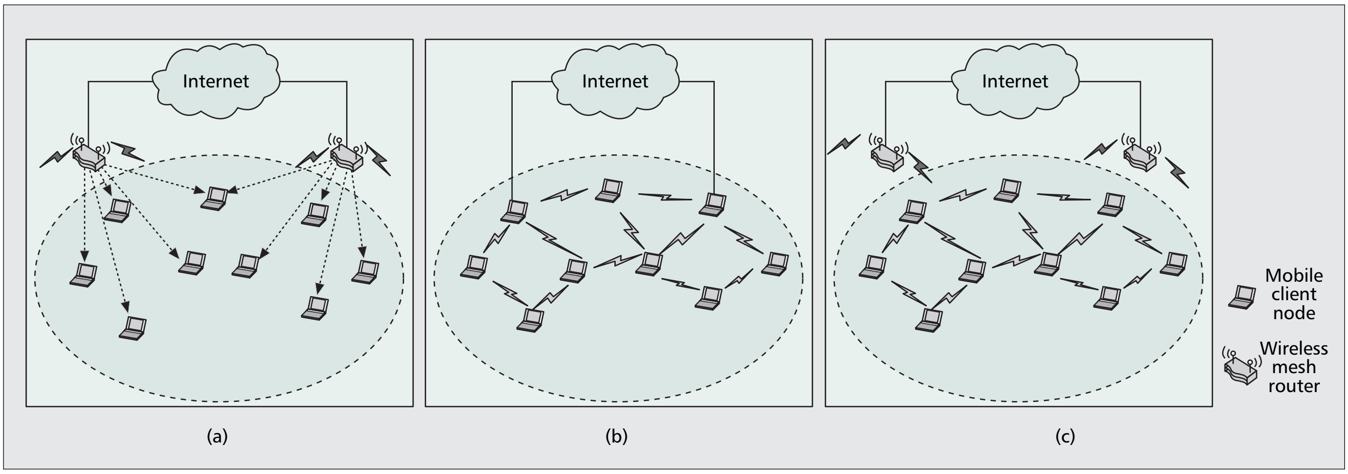


■ **Figure 1.** *Wireless mesh networks: a) infrastructure; b) client; c) hybrid.*

activities for corporate and personal benefit, but also helps reduce fuel consumption and protect the environment, issues that are becoming increasingly important in our lives [3].

In this article we provide an up-to-date survey of FiWi access networks. After reviewing state-of-the-art wireless and optical access networks and briefly highlighting future developments in both areas, we focus on enabling technologies and elaborate on emerging FiWi architectures, and also discuss their future challenges. The remainder of the article is structured as follows. The next two sections overview the state of the art of wireless and optical (wired) access networks, respectively. We then describe enabling technologies and various FiWi network architectures. In the following section we address future challenges of emerging FiWi networks. The final section concludes the article.

Wireless Access Networks: State of the Art

Wireless Mesh Networks

Recent advances in wireless communications technology have led to significant innovations that have enabled cost-effective and flexible wireless Internet access, and provided incentives for building efficient multihop wireless networks. A wireless ad hoc network precludes the use of a wired infrastructure and allows hosts to communicate either directly or indirectly over radio channels without requiring any prior deployment of network infrastructure.

Wireless mesh networks (WMNs), on the other hand, are networks employing multihop communications to forward traffic en route to and from wired Internet entry points [4]. In contrast to conventional WLANs and mobile ad hoc networks (MANETs), WMNs promise greater flexibility, increased reliability, and improved performance. WMNs can be categorized into infrastructure, client, and hybrid WMNs (Fig. 1). A router in an infrastructure WMN has no mobility and performs more functions than a normal wireless router. Among others, a router performs mesh functions (routing and configuration) and acts as a gateway. In a client WMN, clients perform mesh and gateway functions themselves. Efficient routing protocols provide paths through the wireless mesh and react to dynamic changes in the topology, so mesh nodes can communicate with each other even if they are not in direct wireless range. Intermediate nodes on the path forward packets to the final destination. Due to the similarities between WMNs and MANETs, WMNs can apply ad hoc routing protocols (e.g., ad hoc on-demand distance vector [AODV] and dynamic source routing [DSR], among others).

Enabling Technologies

New technologies and protocols in the physical (PHY) layer, medium access control (MAC) protocols, and routing protocols are required to optimize the performance of WMNs. In the PHY layer, smart antenna, multi-input multi-output (MIMO), ultra wideband (UWB), and multichannel interface systems are being explored to enhance network capacity and further enable wireless gigabit transmission. Recently, gigabit transmission resulting from a combination of MIMO and orthogonal frequency-division multiplexing (OFDM) has been demonstrated. MAC protocols based on distributed time-division multiple access (TDMA) and CDMA are expected to improve the bandwidth efficiency of carrier sense multiple access with collision avoidance (CSMA/CA) protocols [4].

Currently, IEEE 802.11 a/b/g (WiFi) technologies are widely exploited in commercial products and academic research of WMNs due to their low cost, technological maturity, and high product penetration [5]. However, since these protocols were originally designed for WLANs, they clearly are not optimized for WMNs. Proprietary wireless technologies and WiMAX have been proposed. Unlike WiFi, IEEE 802.16 allows for point-to-multipoint wireless connections with a transmission rate of 75 Mb/s and can be used for longer distances.

Additionally, orthogonal frequency-divisiom multiple access (OFDMA) and smart antenna technologies extend the scalability of WiMAX. These technologies are exploited to enhance the capacity, reliability, and mobility of WMNs.

Optical networks lend themselves well to offloading electronic equipment by means of optical bypassing as well as reducing their complexity, footprint, and power consumption significantly while providing optical transparency against modulation format, bit rate, and protocol.

Ultra-high-bandwidth standards such as IEEE 802.16m, which aims to provide 1 Gb/s and 100 Mb/s shared bandwidth, can be employed to further enhance the bandwidth and mobility of WMNs. Since packets are routed among mesh routers in the presence of interference, shadowing, and fading, a cross-layer design is required to optimize the routing in WMNs. For instance, DSR uses link quality source routing (LQSR) to select a routing path according to link quality metrics. LQSR includes three performance metrics: per-hop packet pair, per-hop round-trip time (RTT), and expected transmission count (ETX). ETX shows the best performance in networks with fixed nodes, while minimum hop count shows good performance in networks with mobile nodes.

Future Developments

Given the increased demand for mesh networks, a task group was formed in 2004 to define the Extended Service Set (ESS) mesh networking standard; its goal is the development of a flexible and extensible standard for WMNs based on IEEE 802.11. The IEEE 802.11s amendment can be split up into four major parts: multihop routing, MAC enhancements, security, and general topics. It also defines a new mesh data frame format that can be used for transmitting data within the WMN. Traffic in mesh networks is predominantly forwarded to and from wireline gateway nodes forming a logical tree structure. The 802.11s defines a default mandatory routing protocol (Hybrid Wireless Mesh Protocol [HWMP]) that uses hierarchical routing to exploit this tree-like logical structure and on demand routing protocols to address mobility; the on demand routing protocol is based on AODV, which uses a simple hop count routing metric. Alternatively, the standard allows vendors to operate using alternate protocols, one of which is described in the draft (Radio Aware Optimized Link State Routing [RA-OLSR]). RA-OLSR uses multipoint relays, a subset of nodes that flood a radio-aware link metric, thereby reducing control overhead on the routing protocol.

Other interesting developments are concerned with the integration of different access technologies; for instance, the authors of [6] presented an approach for integrating WiMAX and WiFi technologies, and discussed several issues pertaining to protocol adaptation and QoS support.

Optical Access Networks: State of the Art

Optical fiber provides unprecedented bandwidth potential far in excess of the wireless and any other known transmission medium. A single strand of fiber offers a total bandwidth of 25,000 GHz. More important, optical networks lend themselves well to offloading electronic equipment by means of optical bypassing as well as reducing their complexity, footprint, and power consumption significantly while providing optical transparency against modulation format, bit rate, and protocol.

FTTX Networks

FTTX networks are poised to become the next major success story for optical fiber communications. Not only must future FTTX access networks unleash the economic potential and societal benefit by opening up the first/last mile bandwidth bottleneck between bandwidth-hungry end users and high-speed backbone networks, but also enable the support of a wide range of new and emerging services and applications, such as triple play, video on demand, point-to-point (P2P) audio/video file sharing and streaming, multichannel HDTV, multimedia/multiparty online gaming, and telecommuting. Due to their longevity, low attenuation, and huge bandwidth, PONs are widely deployed to realize cost-effective FTTX access networks [7].

PONs

Typically, PONs are time-division multiplexing (TDM) single-channel systems, where the fiber infrastructure carries a single upstream wavelength channel (from subscribers to a central office) and a single downstream wavelength channel (from a central office to subscribers). IEEE 802.3ah Ethernet PON (EPON) with a symmetric line rate of 1.25 Gb/s, and International Telecommunication Union — Telecommunication Standardization Sector (ITU-T) G.984 Gigabit PON (GPON) with an upstream line rate of 1.244 Gb/s and a downstream line rate of 2.488 Gb/s represent current state-of-the-art commercially available and widely deployed TDM PON access networks, but standardization efforts have already been initiated in the IEEE 802.3av Task Force to specify 10 Gb/s EPON. GPON offers strong operation, administration, maintenance, and provisioning (OAMP) capabilities, and provides security at the protocol level for downstream traffic by means of encryption using Advanced Encryption Standards. Furthermore, GPON efficiently supports traffic mixes consisting not only of asynchronous transfer mode (ATM) cells but also TDM (voice) and variable-size packets by using the GPON encapsulation method (GEM). EPON aims at converging the low-cost equipment and simplicity of Ethernet and the low-cost infrastructure of PONs. Security and OAMP are not specified in the EPON standard IEEE 802.3ah, but may be implemented using the data over cable service interface specification (DOCSIS) OAMP service layer on top of the MAC and PHY layers of EPON. Given the fact that 95 percent of LANs use Ethernet, and most applications and services (e.g., video) are moving toward Ethernet, in conjunction with Ethernet's low cost and simplicity, EPON is expected to increasingly become the norm.

Both GPON and EPON are commonly perceived to carry a single wavelength channel in each direction. The majority of real-world PON deployments, however, use an additional downstream wavelength channel for video distribution according to the wavelength allocation in ITU-T Recommendation G.983.3, which specifies a so-called enhancement band from 1539 to 1565 nm plus an L-band reserved for future use. The enhancement band and L-band can be used to

Features	FSO	RoF
Connectivity	Point-to-point	Point-to-point and point-to-multipoint
Transmission mode	Full duplex	Full duplex
Scalability	High in terms of bandwidth Low in terms of user and service	Low in terms of bandwidth High in terms of user and service
Availability	Low in fog High in rain	High in fog Low in rain
Interference	Background sunlight	Electromagnetic signals
Spectrum licence	Not required	Required

Table 1. *Comparison between wireless segments of FSO and RoF.*

FSO is a type of direct line-of-sight (LOS) optical communications that provides point-to-point connections by modulating visible or infrared (IR) beams [9]. It offers high bandwidth and reliable communications over short distances.

enable additional services such as overlay of multiple PONs on a single fiber infrastructure or optical time domain reflectometry (OTDR) for testing and troubleshooting.

Future Developments

Adding the wavelength dimension to conventional TDM PONs leads to WDM PONs, which have several advantages. Among others, the wavelength dimension may be exploited to:

- Increase network capacity
- Improve network scalability by accommodating more end users
- Separate services
- Separate service providers [7]

An interesting approach to increasing split ratio (i.e., number of subscribers) and range is the so-called long-reach PON (LR-PON), which is currently receiving considerable attention from network operators in an attempt to optically bypass central offices and consolidate optical metro and access networks, resulting in major cost savings and simplified network operation. LR-PONs can also be interesting for new operators wishing only to connect the major geographically distributed business clients.

Most of the reported studies on advanced PON architectures have considered standalone PON access networks, with a particular focus on the design of dynamic bandwidth allocation (DBA) algorithms for quality of service (QoS) support and QoS protection by means of admission control [8].

FiWi Networks

Enabling Technologies

Currently, there are two technologies used to implement fiber-wireless (FiWi) networks:

- Free space optical (FSO), also known as optical wireless (OW)
- Radio over fiber (RoF)

FSO is a type of direct line-of-sight (LOS) optical communications that provides point-to-point connections by modulating visible or infrared (IR) beams [9]. It offers high bandwidth and reliable communications over short distances. The transmission carrier is generated by deploying either a high-power light emitting diode (LED) or a laser diode, while the receiver may deploy a simple photo detector. Current FSO systems operate in full-duplex mode at a transmission rate ranging from 100 Mb/s to 2.5 Gb/s, depending largely on weather conditions. Given a clear LOS between source and destination and enough transmitter power, FSO communications can work over distances of several kilometers. At both source and destination, optical fiber may be used to build high-speed LANs, such as Gigabit Ethernet (GbE).

RoF, on the other hand, allows an analog optical link to transmit a modulated radio frequency (RF) signal. There are different techniques available to realize RoF networks. Typically, an RoF transmitter deploys a Mach-Zehnder intensity (MZI) modulator in conjunction with an oscillator that generates the required optical carrier frequency, followed by an Erbium doped fiber amplifier (EDFA) in order to increase the transmission range. RoF networks provide both P2P and point-to-multipoint connections. Recently, a full-duplex RoF system providing 2.5 Gb/s data transmission over 40 km with less than 2 dB power attenuation was successfully demonstrated using the millimeter-wave band [10]. There are many cost-efficient optical approaches to mixing and upconverting millimeter wave signals.

Table 1 summarizes and compares the salient features of both enabling technologies of FiWi networks.

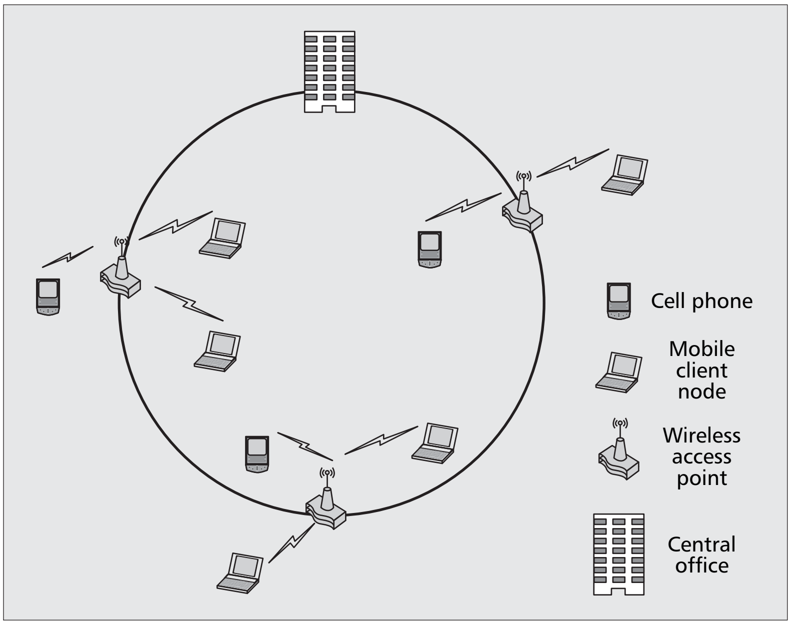
Architectures

We present in this section available architectures for enabling FiWi integration. For instance, the integration of EPON and WiMAX access networks can be done in several ways; according to [11], the following four architectures can be used.

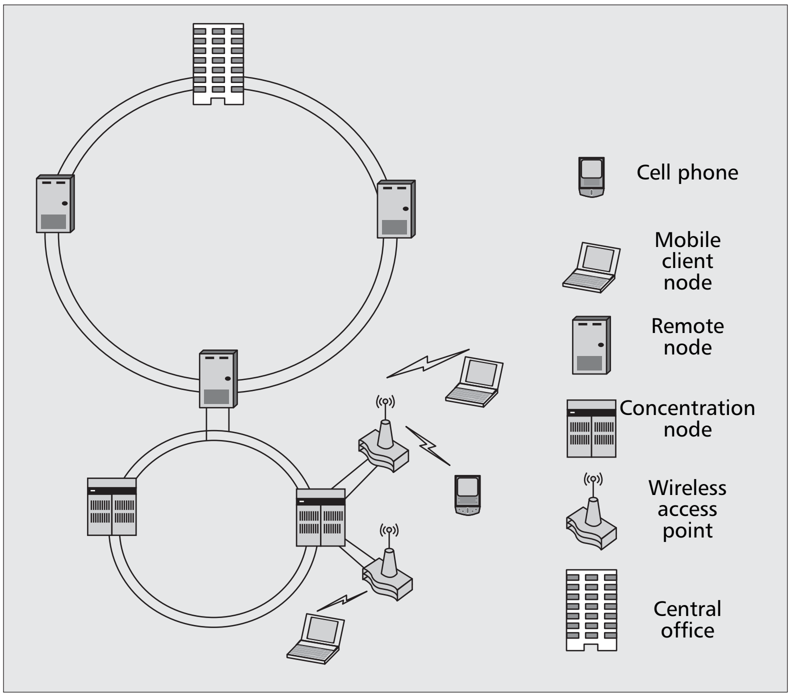
Independent Architecture — In this approach WiMAX base stations serving mobile client nodes are attached to an optical network unit (ONU) just like any other wired subscriber node, whereby an ONU denotes the EPON customer premises equipment. WiMAX and EPON networks are connected via a common standardized interface (e.g., Ethernet) and operate independent of each other.

Hybrid Architecture — This approach introduces an ONU-base station (ONU-BS) that integrates the EPON ONU and WiMAX BS in both hardware and software. The integrated ONU-BS controls the dynamic bandwidth allocation of both the ONU and BS.

Unified Connection-Oriented Architecture — Similar to the hybrid architecture, this approach deploys an integrated ONU-BS. But instead of carrying Ethernet frames, WiMAX MAC protocol data units (PDUs) containing multiple encapsulated Ethernet frames are used. By carrying WiMAX MAC PDUs, the unified architecture can be run like a WiMAX network with the ability to grant bandwidth finely using WiMAX's connection-oriented rather than EPON's queue-oriented bandwidth allocation.



■ **Figure 2.** *Optical unidirectional fiber ring interconnecting WiFi-based wireless access points.*



■ **Figure 3.** *Optical interconnected bidirectional fiber rings integrated with WiFi-based wireless access points.*

Microwave-over-Fiber Architecture — In this approach the WiMAX signal is modulated on a wireless carrier frequency, and is then multiplexed and modulated together with the baseband EPON signal onto a common optical frequency (wavelength) at the ONU-BS. The central node consists of a conventional EPON optical line terminal (OLT) and a central WiMAX BS, called a macro-BS. The OLT processes the baseband EPON signal, while the macro-BS processes data packets originating from multiple WiMAX BS units.

Besides the aforementioned generic integration approaches of EPON and WiMAX networks, several other FiWi architectures based on WiFi technology have been studied, as described in the following.

The network shown in Fig. 2 interconnects the central office (CO) with multiple WiFi-based wireless access points (WAPs) by means of an optical unidirectional fiber ring [12]. The CO is responsible for managing the transmission of information between mobile client nodes (MCNs) and their associated WAPs as well as acting as a gateway to other networks. Each WAP provides wireless access to MCNs within its range. All MCNs take part in the topology discovery, whereby each MCN periodically sends the information about the beacon power received from its neighbors to its associated WAP. In doing so, WAPs are able to estimate the distances between MCNs and compute routes. Multihop relaying is used to extend the range. To enhance the reliability of the wireless link, the CO sends information to two different WAPs (path diversity). The proposed implementation can support advanced path diversity techniques that use a combination of transmission via several WAPs and multihop relaying (e.g., cooperative diversity or multihop diversity). Consequently, the CO must be able to assign channels quickly and efficiently by using one or more wavelength channels on the fiber ring to accommodate multiple services such as WLAN and cellular radio network.

Figure 3 shows a two-level bidirectional path-protected ring (BPR) architecture for dense WDM (DWDM)/subcarrier multiplexing (SCM) broadband FiWi networks [13]. In this architecture the CO interconnects remote nodes (RNs) via a dual-fiber ring. Each RN cascades WAPs through concentration nodes (CNs), where each WAP offers services to MCNs. For protection, the CO is equipped with two sets of devices (normal and standby). Each RN consists of a protection unit and a bidirectional wavelength add-drop multiplexer based on a multilayer dielectric interference filter. Each CN contains a protection unit. The WAP comprises an optical transceiver, a protection unit, up/down RF con-

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verters, and a sleeve antenna. Each WAP provides channel bandwidth of at least 5 MHz and covers up to 16 MCNs by means of frequency-division multiplexing (FDM). Under normal operating conditions, the CO transmits downstream signals in the counter-clockwise direction via RNs and CNs to the WAPs. If a fiber cut occurs between two RNs or between two CNs, their associated controllers detect the failure by monitoring the received optical signal and then switch to the clockwise protection ring. If a failure happens at a WAP, the retransmitted signals are protection switched through other optical paths by throwing an optical switch inside the affected WAP. This architecture provides high reliability, flexibility, capacity, and self-healing properties.

Figure 4 depicts a hybrid FiWi architecture that combines optical star and ring networks [14]. Each fiber ring accommodates several WiFi-based WAPs, and is connected to the CO and two neighboring fiber rings via optical switches. The optical switches have full wavelength conversion capability, and interconnect the WAPs and CO by means of shared P2P lightpaths. The network is periodically monitored during prespecified intervals. At the end of each interval, the lightpaths may be dynamically reconfigured in response to varying traffic demands. When traffic increases and the utilization of the established lightpaths is low, the load on the existing lightpaths is increased by means of load balancing. Otherwise, if the established lightpaths are heavily loaded, new lightpaths need to be set up, provided enough capacity is available on the fiber links. In the event of one or more link failures, the affected lightpaths are dynamically reconfigured using the redundant fiber paths of the architecture.

The FiWi network proposed in [15] consists of an optical WDM backhaul ring with multiple single-channel or multichannel PONs attached to it, as shown in Fig. 5. More precisely, an optical add-drop multiplexer (OADM) is used to connect the OLT of each PON to the WDM ring. Wireless gateways are used to bridge PONs and WMNs. In the downstream direction, data packets are routed from the CO to the wireless gateways through the optical backhaul and then forwarded to the MCNs by wireless mesh routers. In the upstream direction, wireless mesh routers forward data packets to one of the wireless gateways, where they are then transmitted to the CO on one of the wavelength channels of the optical backhaul WDM ring, as each PON operates on a separate dynamically allocated wavelength channel. Since the optical backhaul and WMN use different technologies, an interface is defined between each ONU and the corresponding wireless gateway in order to monitor the WMN and perform route computation taking the state of wireless links and average traffic rates into account. When the traffic demands surpass the available PON capacity, some of the TDM PONs may be upgraded to WDM PONs. If some PONs are heavily loaded and others have less traffic, some heavily loaded ONUs may be assigned to a lightly loaded PON by tuning their optical transceivers to the wavelength assigned to the lightly loaded PON. This architecture provides cost effectiveness, bandwidth efficiency, wide coverage, high flexibility, and scalability. In addition, the reconfigurable TDM/WDM optical backhaul helps reduce network congestion and average packet latency by means of load balancing. Moreover, the dynamic allocation of radio resources enables cost-effective and simple handovers.

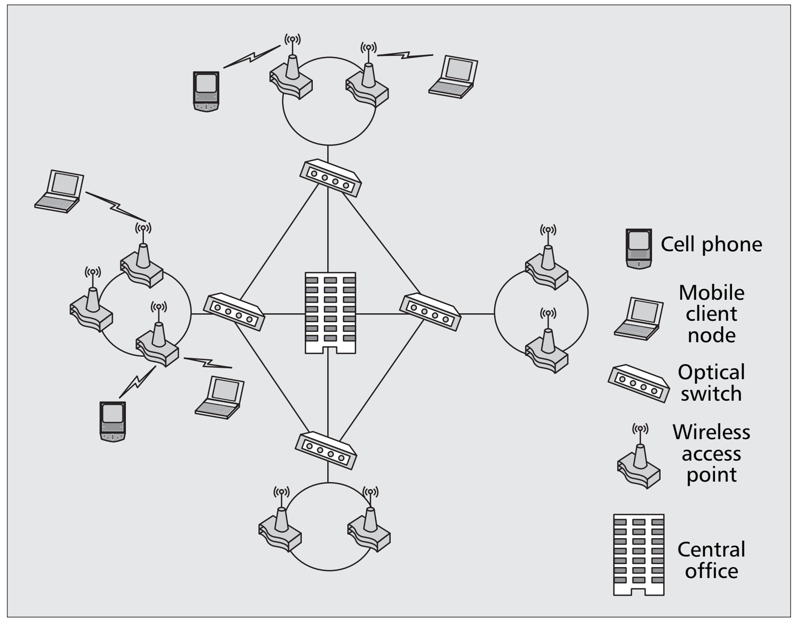

Figure 4. *Optical hybrid star-ring network integrated with WiFi-based wireless access points.*

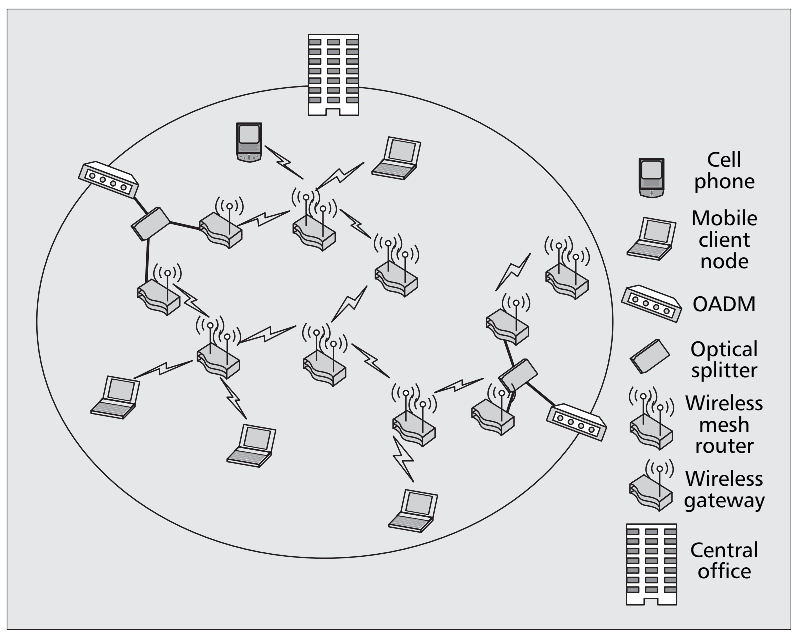

Figure 5. *Optical unidirectional WDM ring interconnecting multiple PONs integrated with a WiFi-based wireless mesh network.*

FUTURE CHALLENGES

We have seen that FiWi networks can be realized by deploying different architectures and several technologies. Toward commercial adoption, FiWi access networks still face a number of tech-

By seamlessly converging optical and wireless access technologies, hybrid FiWi access networks hold great promise to support a plethora of future and emerging broadband services and applications on the same infrastructure.

nological challenges. One of the most critical challenges is to determine a feasible, scalable, and resilient architecture along with the corresponding enabling technologies. As discussed earlier, future broadband access networks will undoubtedly be a combination of first/last mile optical fiber access solutions (i.e., FTTX) and heterogeneous broadband wireless networks providing connectivity to end users. One first challenge is to seamlessly integrate these technologies; while FTTX networks provide TDMA to wired ONUs, mobile client nodes in a WMN access the medium through enhanced distributed channel access (EDCA) and multihop routing used to forward their packets to wireless mesh gateways.

New approaches to exploit the huge bandwidth available in optical access networks for offloading bandwidth-limited wireless networks should be studied in greater detail. The design and evaluation of powerful load balancing and reconfiguration techniques to improve the bandwidth efficiency of future FiWi networks is another interesting research avenue, including reconfiguration techniques for unpredictable traffic. Routing in WMNs remains a critical issue, and designing efficient routing protocols that are aware of the bandwidth allocation on PON is more challenging; routing algorithms that exploit this large bandwidth potential to offer fair access to WMN nodes as well as load balancing across the mesh links are key for future FiWi networks. Additionally, these current access networks are designed to carry traffic with various QoS requirements. Various QoS bandwidth allocation algorithms for PONs have emerged; however, designing QoS-aware routing protocols in WMNs is still an open issue and is not addressed within the 802.11s standard. In general, applications have different QoS requirements. Research on powerful end-to-end resource allocation techniques in FiWi networks is necessary.

Resiliency against failures is another challenge of future FiWi networks. FiWi networks should allow WMN gateways to interconnect with the optical backhaul through multiple points in order to enable multipath routing and improve their survivability. Additionally, the optical backhaul should implement appropriate protection switching functions to deal with network element failures rapidly.

The 802.11s standard currently defines a new frame format for transmitting traffic over the WMN. However, most of today's deployed PON systems are based on EPON or BPON/GPON. Therefore, interfaces are needed to allow for protocol adaptation and enable network interoperability.

Finally, implementation simplicity will be key to the commercial success of FiWi networks. Reducing the installation and protection costs by means of transferring expensive devices and complex functions to the central office appears to be a promising approach to building cost-effective FiWi networks. In particular, cost-efficient and feasible modulation formats for optical/RF signal conversion are needed. Despite recent developments in RoF networks, more research on physical layer related issues is necessary due to the high atmospheric absorption in high-frequency bands such as the millimeter-wave band.

CONCLUSIONS

By seamlessly converging optical and wireless access technologies, hybrid FiWi access networks hold great promise to support a plethora of future and emerging broadband services and applications on the same infrastructure. We have observed that research and development of future FiWi network architectures and protocols have made significant progress, but many open issues mostly related to the design of low-cost components, integrated routing, end-to-end service differentiation, and resiliency must be solved in order to render FiWi access solutions commercially viable. By simultaneously providing wired and wireless services over the same infrastructure, FiWi networks are able to consolidate (optical) wired and wireless access networks that are usually run independent of each other, thus potentially leading to major cost savings. An interesting future research avenue would be the techno-economic comparison of different FiWi network architectures.

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A Multi-Area MPLS/GMPLS Interoperability Trial over ROADM/OXC Network

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ABSTRACT

This article describes the first multi-area multiprotocol label switching and generalized MPLS interoperability trial over a reconfigurable optical add/drop multiplexer and optical cross-connect network. The interoperability trial demonstrated the routing of label switched paths over a multi-area GMPLS controlled ROADM/OXC network and the control of Ethernet over MPLS transport service on top of the GMPLS network. The trial was conducted using various network elements provided by 14 institutions and was carried out in Tokyo and Virginia. This article introduces the motivation for the trial, technical issues related to controlling multi-area MPLS/GMPLS networks, test network topology, and experimental results. The results show that the interior gateway routing protocol-based multi-area routing architecture is a promising solution for the nationwide deployment of GMPLS networks within a carrier domain. In addition, this article discusses the technical issues of routing constraints in ROADM/OXC networks and the limit of multi-area routing without the Path Computation Element Protocol.

INTRODUCTION

The generalized multiprotocol label switching (GMPLS) technology [1] is a framework that unifies network control of various types of network elements (NEs) across multiple network layers. This framework not only enables network operators to simplify the development of network control functionality in their network management systems, but also provides a foundation for the deployment of resilient and reliable networks. To gain such key benefits, the inter-operability of GMPLS protocols across NEs is of critical interest. Many network operators and vendors have expended significant effort and have conducted a number of MPLS/GMPLS interoperability trials since the initiation of GMPLS standardization activity in the Internet Engineering Task Force (IETF). Due to these efforts, the feasibility of the GMPLS control architecture was proven, and the interoperability of current GMPLS protocols was significantly improved with the help of the GMPLS addressing draft [2], which was created through the experience of these interoperability testing activities [3,4]. As a result,

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GMPLS technology has matured to the point of more realistic deployment and operational scenarios, such as the integration of GMPLS networks into existing Internet Protocol (IP)/MPLS networks and GMPLS control over intra/inter-carrier multiple routing domains.

This article introduces an interoperability trial of a multi-area MPLS/GMPLS network employing the Interior Gateway routing protocol (IGP). This functional evaluation is quite important in overcoming the scalability limitations of a single-area routing architecture and the operation of hundreds of NEs within a nationwide carrier domain. The primary motivation of this trial was to evaluate key functionality for the control of inter-area label-switched paths (LSPs) across nationwide GMPLS networks, including an evaluation of the limit of the per-domain path calculation (PDPC) [5] solution. An additional motivation included the evaluation of the GMPLS control plane integrity with state-of-the-art optical technologies and services that can use the MPLS/GMPLS infrastructure such as the Ethernet over MPLS (EoMPLS) transport service. The authors believe that this trial is unlike any previous trial because it conducts single-carrier and multiple-area routing architecture in an optical network domain, which is different from the previous trials that employed single-area [3, 4], hierarchical [6], and inter-autonomous system (AS) routing architecture [7]. The testbed network, which comprised network elements from 14 institutions, including (G)MPLS test equipment, IP/MPLS routers, time-division multiplexing cross connects(TDM-XCs), and optical network elements of reconfigurable optical add/drop multiplexer (ROADMs) and optical cross connects (OXCs) with multiple-area routing architecture, was constructed over a transpacific control network between the Toyo Corporation in Tokyo and Isocore in Virginia.

Technical Issues Facing Multi-Area MPLS/GMPLS Networks

The technical difficulty related to the inter-area routing of LSPs originates from the specification of traffic engineering (TE) extensions to the existing IGPs, considering scalability limitations [8]. In the case of the Open Shortest Path First (OSPF) protocol, the advertisement of TE information is limited to the local area scope to reduce the volume of the advertised TE link information. This functional specification is quite important from the view point of operational stability. Traditionally, carriers have separated routing areas into a reasonable size or adequate operational domain, which is quite effective in preventing failure propagation across networks and in minimizing the impact on commercial networks. Consequently, a GMPLS-controlled NE is not capable of calculating a full end-to-end route for the LSPs. To cope with this issue, the current IETF proposal includes the employment of the:

- PDPC scheme [9]
- Signaling of loosely routed paths [10]

Per-Domain-Path Calculation

A routing strategy of MPLS/GMPLS protocols widely employed in current vendor implementations is a "source-routing" strategy. Namely, the route calculation is performed at the ingress node of the LSP, based on the traffic engineering database (TED) and the Constraint-based Shortest Path First (CSPF) algorithm stored in the node. In the case of route calculation of inter-area LSP, the ingress node must discover the area border node to route the LSP toward the egress node and then perform a CSPF calculation to the area border node. The application of Path Computation Element Protocol (PCEP) [5] currently discussed in the IETF PCE-working group (WG) includes automatic discovery of the area border nodes to perform full automatic inter-area calculation. This benefit also exists in the case of route calculation and discovery of autonomous system border nodes for inter-domain LSPs.

It also is possible to apply a gateway routing strategy. In this scenario, the assignment of the area border node is performed statistically or manually, based on the operational policy of network service providers. Specifically, for the service providers employing a ring-based network topology, this design, which requires a partly manual LSP establishment and assignment of area border nodes, may be an acceptable operational burden. This article assumes such an operational scenario. The target of this trial is to evaluate the PDCP solution, namely the evaluation of the PDPC function utilizing statistically or manually assigned area border node information on the route of the inter-area LSPs.

Signaling of Loosely Routed Paths

Figure 1 outlines the procedure of signaling to create a loosely routed LSP based on the gateway routing strategy. The head-end node of the LSP performs the CSPF calculation toward the area border node and inserts the route information obtained by the CSPF calculation into the signaling message. The signaling message is transmitted node by node along the LSP. After receiving the signaling message, the area border node performs a CSPF calculation toward the destination node of the LSP to be created. If the destination node is outside of the area to which the area border node belongs, the area border node performs a CSPF calculation toward the next area border node. In the gateway routing strategy, the next-hop area border node is statically configured according to the destination node in the area border node. The area border node then sends the signaling message toward the destination node. The process iterates with the next-hop area border node performing a similar procedure to create the inter-area LSP.

Thus, the target of this testing includes the evaluation of the interoperability of ReSource reserVation Protocol with Traffic Engineering (RSVP-TE) to create loosely routed paths and the combined operation of the RSVP-TE and the PDPC at the area border OXCs (ABR-OXCs).

A routing strategy of MPLS/GMPLS protocols widely employed in current vendor implementations is a "source-routing" strategy. Namely, the route calculation is performed at the ingress node of the LSP, based on the TED and the CSPF algorithm stored in the node.

The ABR-OXC performs the PDPC, based on the TE link information within the area to which it belongs. After receiving an RSVP-TE signaling message from the upstream node, the ABR-OXC processes the explicit route object (ERO) within the messages.

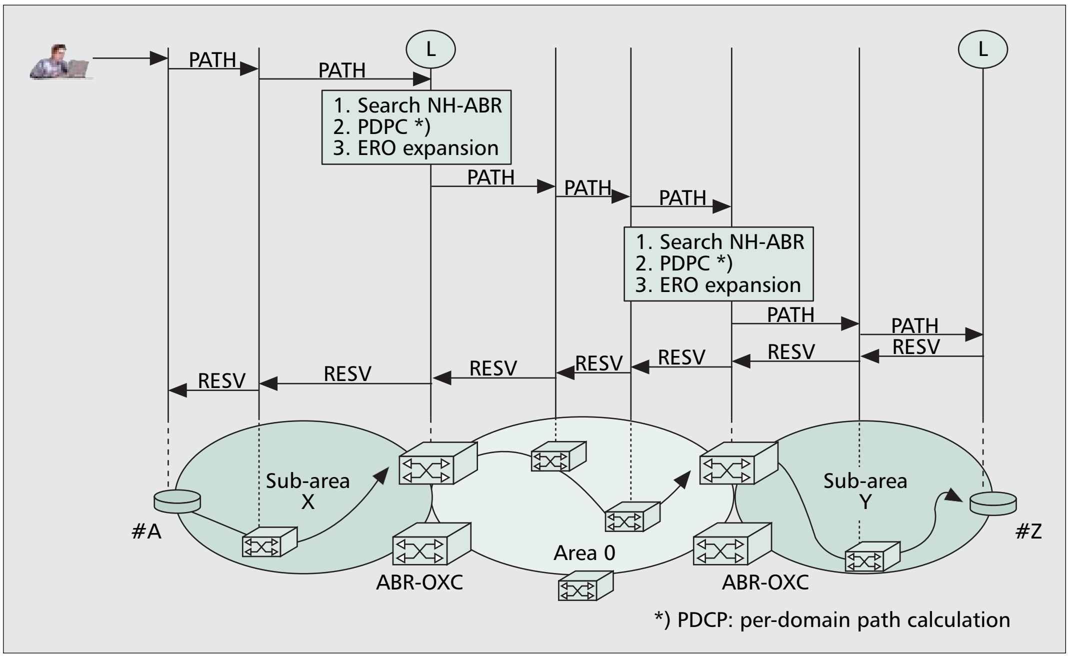

Figure 1. *Procedure of the per-area hop route calculation and the signaling of a "loosely routed path."*

Multi-Area MPLS/GMPLS Testbed

Figure 2 shows an overview of the IGP routing architecture in the MPLS/GMPLS testbed. The routing area of the MPLS and GMPLS layers were isolated. The MPLS network was constructed with an OSPF backbone area (Area 0). The GMPLS layer was comprised of a backbone area and three sub-areas making up the overall testbed network. The backbone area (Area 0) of the GMPLS domain was allocated to the OXC network area. On the other hand, the sub-area numbers area 1, 2, and 3 of the GMPLS domain were allocated to the TDM-XC network area in Virginia (Isocore site) and two ROADM network areas in Tokyo (Toyo Corporation site), respectively. All of MPLS/GMPLS border routers were located in the sub-areas of the GMPLS domain.

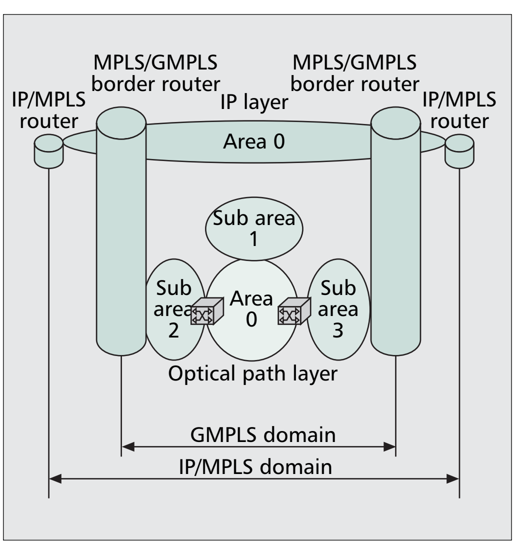

Figure 2. *Overview of routing area architecture.*

Table 1 shows the detailed list of network elements equipped for the interoperability trial. The network comprises mainly four types of switching capabilities, namely, IP/MPLS/GMPLS testers, MPLS routers, MPLS/GMPLS border routers as packet-switch capable (PSC), ROADMs as lambda-switch capable (LSC), and TDM-XCs and OXCs used as fiber-switch capable (FSC), and a total of 25 NEs from 14 institutions.

Figure 3 shows a detailed configuration of the network, constructed in Tokyo and Virginia. The network comprises synchronous transport module (STM)-16, optical Gigabit Ethernet (GbE), and STM-16/GbE multirate optical links. Here, the STM-16/GbE multirate links were constructed in the section between peer OXCs. Data-plane links between the ABR-OXCs in Tokyo and the area 1 network in Virginia were configured using virtual STM-16 links. Namely, the synchronous digital hierarchy/synchronous optical network (SDH/SONET) interfaces between them set loopback to virtually activate the transpacific data link. Additionally, four out-of-fiber signaling control networks (SCNs) were constructed according to routing area separation. A transpacific Internet Protocol security (IPSec) tunnel also was established over the public Internet to connect the ABR-OXCs at the Tokyo site into the SCN in Virginia. Generic routing encapsulation (GRE) tunnels also were created over some SCNs to form virtual point-to-point control channels between peer GMPLS capable nodes.

Area Border OXC

The ABR-OXC performs the PDPC, based on the TE link information within the area to which it belongs. After receiving an RSVP-TE signaling message from the upstream node, the ABR-OXC processes the explicit route object (ERO) within the messages. The RSVP-TE signaling instance in the ABR-OXC requests the CSPF instance to execute a route calculation toward the next-hop area border node considering the destination node information of the LSP. The RSVP-TE signaling instance then inserts a new ERO into the RSVP-TE message to assign a detailed LSP route within the area, based on the result of the CSPF calculation. In other words, the ABR-OXC executes a so-called ERO expansion, taking responsibility to determine the route of the inter-area LSP downstream of the routing area to which the ABR-OXC belongs. The ABR-OXCs are designed to search next-hop ABR-OXCs dynamically by using the "Summary link state attribute (LSA)" of the OSPF-TE protocol as a temporal solution if there is no next-hop ABR-OXC to reach destination nodes in its static routing tables. Each ingress node selects the default ABR-OXC in its sub-area to create the inter-area LSPs.

MPLS/GMPLS Border Router

The MPLS/GMPLS border router also acts as an area border node. MPLS/GMPLS border routers support MPLS LSP hierarchy and virtualize the LSPs created in the GMPLS domain as forwarding adjacencies (FAs) in the MPLS domain, which enables autonomous routing of MPLS LSPs within the MPLS domain. In the MPLS domain, the interoperability test of MPLS RSVP-TE also was conducted using four routers from two vendors to evaluate the functionality of Ethernet over MPLS (EoMPLS) transport over the GMPLS network. A video stream was transported over the Ethernet pseudo-wire established among MPLS routers and MPLS/GMPLS border routers.

Experiments and Results

In the first step of this testing, the interoperability of the OSPF-TE protocol was evaluated. At the initial stage of the testing, the ABR-OXC did not have the capability to advertise router address type-length-value (TLV) [8] for the multiple routing areas to which the ABR-OXC belonged. After resolving this problem, we successfully achieved multi-area operation of the OSPF-TE protocol. The ABR-OXCs were designed to search next-hop ABR-OXCs dynamically by using the Summary LSA of the OSPF-TE protocol; however, the ABR-OXCs failed to create the routing table to assign next-hop ABR-OXCs to reach destination nodes outside their area. This was because some GMPLS routers inactivate advertising functionality of reachability information toward their node IDs (node ID advertisement as stub area in router LSAs). Therefore, for this experiment, we manually set next-hop ABR-OXCs to reach destination nodes in the static routing tables of the CSPF instance within each ABR-OXC. Hereafter, this article explains the results of:

- GMPLS signaling interoperability to control strictly routed inter-area LSPs
- GMPLS signaling interoperability to control loosely routed inter-area LSPs
- MPLS signaling interoperability to initiate Ethernet transport service over the MPLS/GMPLS network

Vendor	Network element type
A	IP/MPLS routers
B	IP/MPLS/GMPLS routers
C	IP/MPLS/GMPLS routers
D	IP/GMPLS routers
E	IP/MPLS/GMPLS testers
F	IP/MPLS/GMPLS testers
G	TDM-XC(STM-16c/OC-48c XC)
H	TDM-XC (STM- 16c/OC-48c XC)
I	OXC
J	OXC
K	OXC
L	OXC
M	ROADM
N	ROADM

Table 1. *List of evaluated NEs.*

GMPLS Signaling Interoperability of Strictly Routed Paths

The interoperability testing of the GMPLS RSVP-TE signaling was conducted assuming a scenario where the operators manually design the route of inter-area LSPs. In this operational scenario, the RSVP-TE signaling message includes the ERO to strictly assign the route of LSPs. The interoperability testing evaluated two types of LSPs:

- SDH
- Ethernet encoding

Type was assigned in the generalized label request object of the Path message to create STM-16 and GbE LSPs, respectively. Some OXCs accommodating STM-16/GbE multirate links inactivated the encoding type check and generalized payload identifier (G-PID) in the generalized label request object assigning the type of LSPs and SENDER traffic specification (TSPEC)/FLOW SPEC objects assigning bandwidth of LSPs.

Table 2 describes the LSP routes and the round trip time (RTT) of RSVP PATH/RESV two-way signaling messages to create each LSP. The RTT of the RSVP PATH/RESV message ranged from 64 msec to 11.6 sec —performance that is good enough to realize fast provisioning in service providers' networks. Most scenarios that exceeded a one-second RTT included transit NEs that control the STM-16 interfaces. The time to initiate IP-packet forwarding typically requires eight to nine seconds in addition to the

The MPLS/GMPLS border router also acts as an area border node. MPLS/GMPLS border routers support MPLS LSP hierarchy and virtualize the LSPs created in the GMPLS domain as forwarding adjacencies in the MPLS domain, which enables autonomous routing of MPLS LSPs within the MPLS domain.

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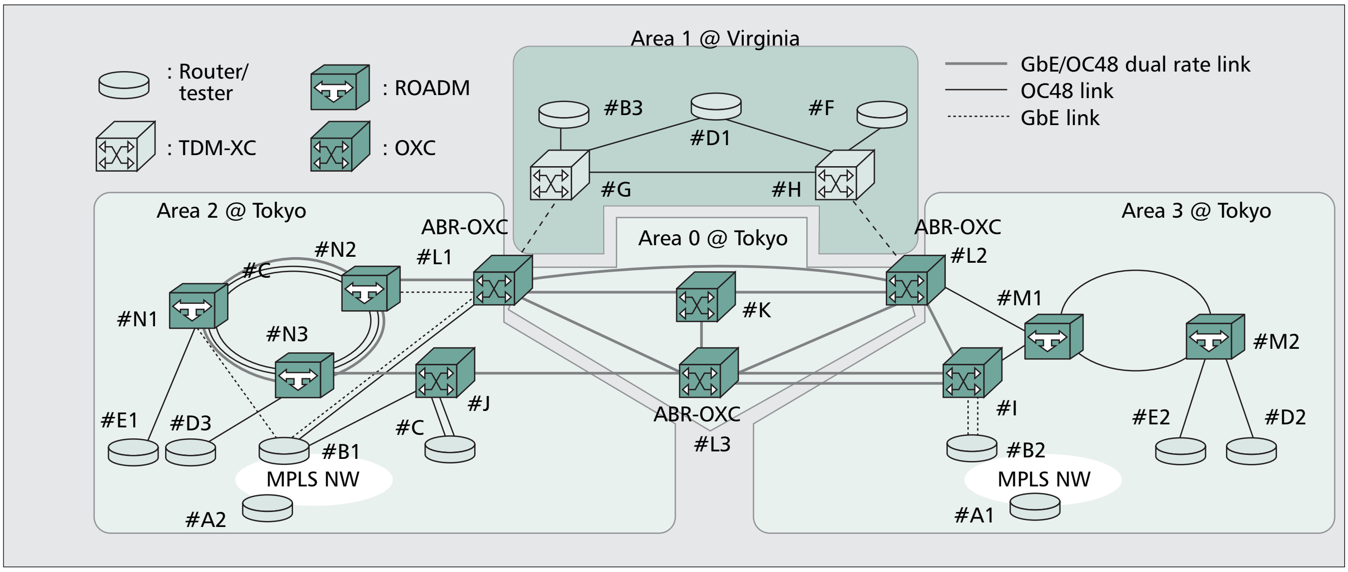


Figure 3. *Configuration of the multi-area MPLS/GMPLS interoperability testbed.*

RTT. This time includes the processing time for changing IP forwarding tables in the MPLS/GMPLS border routers. In the case of GbE interface, the negotiation of Address Resolution Protocol (ARP) also is executed between the pair of the MPLS/GMPLS border routers before initiating IP-packet forwarding. The IP-packet forwarding time was 17.8 sec after transmitting the RSVP PATH message from the ingress node, when the LSP traversed seven switches in two ROADM sub-areas and the backbone OXC area (scenario 1 in Table 2).

GMPLS Signaling Interoperability of Loosely Routed Paths

Next, the interoperability testing of the GMPLS RSVP-TE signaling was conducted assuming the scenario that the operators manually assign only node IDs of the ABR-OXC attached to the source area and destination node of inter-area LSPs. In this operational scenario, the RSVP-TE signaling message includes an ERO that loosely assigns these two nodes. Specifically, we evaluated the PDPC function of the ABR-OXCs that belonged to the backbone area or destination area.

Table 2 also describes the LSP routes and the RTT of RSVP PATH/RESV two-way signaling messages in this operational scenario. The dominant factor of the inter-area LSP is not the route calculation time in each ABR-OXC, but the interface control of the NEs. The PDPC function was successfully performed in less than 50 to 100 milliseconds in the ABR-OXCs in three LSP creation scenarios. Each ABR-OXC successfully inserted a new ERO into the RSVP-TE message to assign a detailed route in the transit area and destination area. For example, node L1 inserts node K and L2 into the ERO to assign a detailed route of the transit area, and node L2 inserts node I to assign a detailed route of the destination area in the case that an inter-area LSP is created between node B1 and B2.

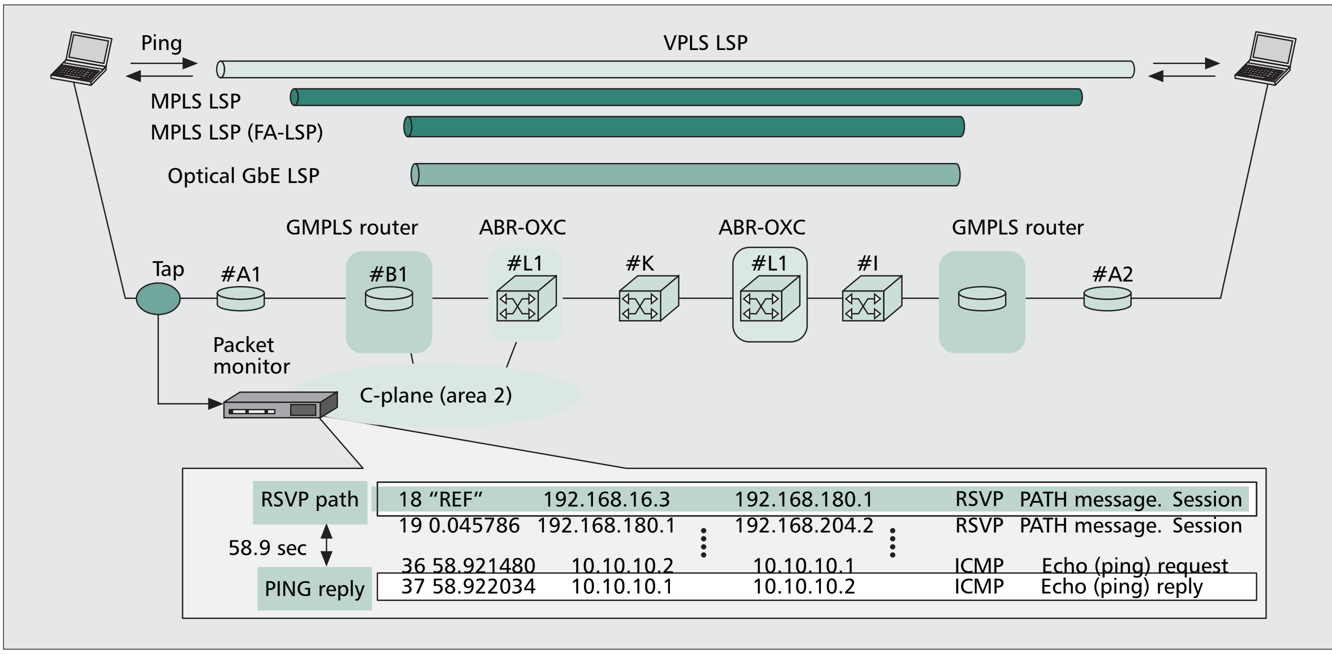
On the other hand, we must comment on failed scenarios — specifically, the LSP creation scenarios wherein the LSP traverses a ROADM ring in the destination area. Because the ABR-OXCs do not have the ability to understand constraints such as the asymmetric switch architecture of ROADMs, the ABR-OXC failed to calculate a precise route for inter-area LSPs in the ROADM/OXC hybrid destination area. These issues are addressed in the Discussion section below.

	Type	LSP route	Time (ms)
1	OC48	E1-N1-N2-L1-L2-I-M1-M2-E2	9,209
2	OC48	E2-M2-M1-I-L2-K-L1- N2-N1-E1	6,928
3	OC48	D3-N3-N2-L1-K-L2-M1-M2-D2	11,653
4	OC48	B1-J-C	64
5	OC48	B1- L1-G-B3 (Transpacific)	890
6	OC48	D2-M2-M1-L2-H-F (Transpacific)	7,027
7	GbE	B1-N1-N2-L1-L3-I-B2 (per area route calc.)	5,216
8	GbE	B1-L1-L2-I-B2 (per area route calc.)	583
9	GbE	B1-L1-K-L2-I-B2 (per area route calc.)	694
10	MPLS	A1-B2-(LSP 7 or LSP 8)-B1-A2	

Table 2. *Ten successful scenarios and round-trip time of RSVP PATH/RESV messages to create LSPs.*

Ethernet Pseudo-Wire Over MPLS/GMPLS Network

On top of the GMPLS layer, an Ethernet pseudo-wire also was established by employing MPLS LSP hierarchy. The MPLS domain consists of three layers in this trial as depicted in



■ **Figure 4.** *Layer architecture of EoMPLS transport and its service activation time.*

Fig. 4. The lowest MPLS layer acts as an FA-LSP between the MPLS/GMPLS border routers. The FA-LSP can provide a protection capability to cope with the failures in the GMPLS domain, if the MPLS FA-LSP is established with a primary and secondary path. The FA-LSP is advertised into the MPLS domains as a TE link by the MPLS/GMPLS border router. The middle MPLS layer is an LSP controlled by the RSVP-TE protocol session between provider edge (PE) routers. Finally, the upper MPLS layer acts as a virtual private LAN service (VPLS) layer, which is controlled by a Label Distribution Protocol (LDP) session. The PE routers build a medium access control (MAC) address table from the Ethernet traffic transiting those devices and destined to the remote computation elements (CEs).

In the experiment, we connected a movie camera and two personal computers with IP addresses of 10.10.10.1/24 and 10.10.10.2/24 to the PE routers. It took 58.9 sec before the first Ethernet frames were forwarded and available on the edge MPLS routers after transmitting the RSVP-TE PATH message from the GMPLS routers to create an LSP in the lowest GMPLS layer. This time includes the process of the GMPLS LSP creation, the MPLS base FA-LSP creation, the re-establishment of MPLS LSP between the PE routers, and the ARP process to associate IP/MAC addresses pairs. After investigation, it appeared that advertising the FA into the IP/MPLS network required some cycle. This cycle is expected performance in re-optimizing the routes of MPLS LSPs to transit over the newly created GMPLS LSP. At this stage, this performance seems sufficient for the purpose of MPLS LSP re-optimization, although it is necessary to conduct further evaluation and assess the dependency on number of NEs, MPLS LSPs, and so on.

Discussion

The results of the experiment demonstrate that the interoperability of basic GMPLS protocol suites has become almost stable. We encountered only a few problems in the experiment related to the interpretation and implementation of basic GMPLS extensions in the IETF RSVP-TE protocol documents. We encountered problems of routing mainly related to two aspects:

- Routing signaling packets in the control plane
- Routing LSPs, considering constraints in the data plane

Through the experiment, we found that the stability and performance of the control plane functionality of ABR-OXCs greatly impacts the overall operability of GMPLS networks. With the routing architecture established in this experiment, all end-to-end signaling packets such as RSVP-TE notify messages or RSVP-TE signaling messages stitched or nested to the optical LSPs transit the control plane module of the ABR-OXCs. Also, explicit requirements to advertise IP reachability information are quite important for exchanging such end-to-end signaling packets, which is also valid for the PCE architecture [5] in discovering "next-hop PCE" in the route calculation of inter-area/AS LSPs.

Furthermore, the extension of the GMPLS traffic engineering specification is required to cope with the advent of new optical switches such as ROADMs and transparent OXCs. In the transparent optical network with the ROADMs and OXCs, each LSP traverses in single wavelength over optical multiplex sections (OMSs) to satisfy so-called wavelength continuity constraint. Namely, the GMPLS traffic engineering specification must incorporate the information of the resource status of wavelength space in fibers forming OMSs so as to take into account the wavelength continuity constraint. In other words, the information should include transparent opti-

On top of the GMPLS layer, we successfully confirmed service activation of the Ethernet transport service over a multi-vendor MPLS network. We successfully demonstrated sophisticated Ethernet transport service activation over a multi-area GMPLS network.

cal network domain-wide unique encoding to represent the wavelength space over each OMS. In addition, the traffic engineering information should properly represent not only the asymmetric selectivity of optical switches but also the capability of wavelength conversion in ROADMs and OXCs.

Furthermore, the optical transport network can accommodate various types of optical LSPs having different data rates. We employed commercially available regenerators capable of regenerating optical signals with an intensity modulation and direct detection (IM-DD) format of up to 2.4 Gb/sec data rate. However, there is no adequate specification to advertise the status and capability of the fiber links, taking into account the bandwidth of optical band-pass filters inserted at both ends of the fiber links, the range of the supportive data range, and the modulation format of transmitter and receivers attached to both ends of the fiber links, and so on.

Currently, these problems are addressed in the IETF draft of [11]. The representation called *lambda labels* provides an effective way to understand the resource status of wavelength space in each OMS. Also, this IETF draft proposes the incorporation of connection matrix and new link attributes to represent constraints in the optical domain, which provides a solution to perform the CSPF calculation to create LSPs over the transparent optical network. Thus, continuous standardization activity and evaluation processes still are required to realize nationwide deployment of the GMPLS control plane technology. Specifically for incumbent service providers employing photonic backbone networks, the transparent optical network is expected to be a unique service platform to support both legacy services, such as SDH/SONET-based private line services, and IP/Ethernet services with various types of maintenance grades, reliability, quality of service, and so on. The PCE architecture for transparent optical network that is discussed in [11] also is helpful not only to overcome the IGP limitation on more optical information required for end-to-end path computation, but also to realize carrier-specific policy-based control in conformance to the policies of these services.

Conclusions

This article discussed the evaluation of MPLS/GMPLS control-plane technology mainly from three view points:

- The key functionality to control inter-area LSPs, namely, the PDPC function and the signaling to control loosely routed paths
- The integrity of the GMPLS control plane with state-of-the-art wavelength switching technologies
- The feasibility evaluation of services making use of the MPLS/GMPLS infrastructure such as Ethernet transport service

Due to the standardized definition of GMPLS, GMPLS-based networks are certain to be configured in a multi-vendor fashion. This article demonstrated a multi-area MPLS/GMPLS interoperability trial using various types of NEs from 14 institutions, that is, IP/MPLS/GMPLS testers from two vendors; an MPLS router from one vendor; MPLS/GMPLS border routers from two institutions; a GMPLS router from one institution; TDM-XCs and ROADMs, respectively, from two institutions; and OXCs from four vendors and institutions. To our knowledge, this trial was the largest interoperability trial based on a homogeneous GMPLS control-plane architecture, based on the number of participant vendors. The article also discussed technical issues related to the control of transparent optical networks with ROADMs and OXCs. On top of the GMPLS layer, we successfully confirmed service activation of the Ethernet transport service over a multi-vendor MPLS network. We successfully demonstrated sophisticated Ethernet transport service activation over a multi-area GMPLS network.

Finally, the authors note that this activity did not address interworking among various types of control-plane technologies other than IETF GMPLS although the authors believe that this activity can help enhance the scalability of an automatically switched optical network (ASON). Currently, the interworking among heterogeneous control planes is addressed actively by the Optical Internetworking Forum (OIF) [6], and the OIF specifications are the most feasible solution for that case.

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