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CONTINUING with the series of columns on new technology areas started in April, with a column on the IoT, followed by a column on 5G in May and a column on Fog Computing in July, the focus of this issue is the area of Network Softwarization. Network Softwarization is expected to revolutionize the way networks and computing infrastructures are designed and operated to deliver services and applications in an agile and cost-effective way. The Communications Society has been playing a leading role in establishing and promoting this important area through its technical activities, conferences, and publications. In this article, Raouf Boutaba, ComSoc’s Director of Magazines, describes some of ComSoc’s ongoing activities and future plans in this important area.

Raouf Boutaba received the M.Sc. and Ph.D. degrees in computer science from the University Pierre and Marie Curie, Paris, in 1990 and 1994, respectively. He is currently a Professor of Computer Science at the University of Waterloo (Canada) and Associate Dean Research of the Faculty of Mathematics. He is the Founding Editor-in-Chief of the IEEE Transactions on Network and Service Management (2007–2010). He has also served on the ComSoc Board of Governors in various Director roles. He has received recognition and honors, including the Premier’s Research Excellence Award, the IEEE ComSoc Hal Sobel Award (2007), the Fred W. Ellersick Prize (2008), the Joe LociCero and the Dan Stokesbury awards (2009), the Salah Aidarous Award (2012), and the McNaughton Gold Medal (2014). He is a fellow of the IEEE, the Engineering Institute of Canada, and the Canadian Academy of Engineering.

The networking industry is undergoing a major change similar to that of the computer industry in the 1980s. The computing industry in the early ’80s was dominated by closed, proprietary, and vertically integrated mainframe computers provided by a handful of vendors with specialized hardware, specialized software, and specialized applications. That landscape completely changed in the late ’80s with the advent of microprocessor technology and open interfaces. Many operating systems were developed and applications proliferated. As a result, a computing industry that was closed, proprietary, and vertically integrated became huge with rapid innovations and a multitude of players in all hardware, operating system, and application markets. For the past few decades, the networking industry has been in a situation similar to that of the mainframe industry. Network owners and operators have been relying on a handful of vendors that provide proprietary and vertically integrated hardware, operating systems, and control features with limited or no programmability. The lack of network programmability, control inflexibility, and the need for network operators to adapt tedious and error-prone manual configuration methods to provision and manage network services have led to increased operational complexity and prolonged time to market for new services. The recent trend toward Network Softwarization is transforming the networking industry into an open ecosystem by separating the hardware on which network functions/services run and the software that realizes and controls the network functions/services. It promises to empower network owners and operators, to increase the pace of innovation, diversify the supply chain for networking hardware and software, and drive the transformation of networks into a highly capable platform in supporting emerging IoT and data science applications, among others.

SDN, NFV, and the Broader Network Softwarization

Software-Defined Networking (SDN) and Network Function Virtualization (NFV) could be seen as different expressions of the overall transformation trend toward network softwarization, which is deeply impacting and bridging the Telecom and IT industries.

SDN involves three principles: separation of control and packet forwarding; centralization of control; the ability to program behavior of the network through well-defined interfaces. In contrast to traditional networks, where control is distributed and embedded into network devices (switches and routers), the SDN control plane is implemented in software in the form of a logically centralized controller. The controller runs on a single or cluster of servers, has a global view of the network, and makes traffic management decisions according to operational policies. Packet forwarding (the data plane) is much simpler compared to that in traditional networks; it is provided by generic switching equipment that is built using cheap merchant silicon chiefly for the efficient forwarding of traffic. The treatment of traffic flows by the switches is programmed by SDN controllers through well-defined interfaces such as OpenFlow, OpFlex, or OnePK. This ability to program the network enables faster innovation, leading to greater responsiveness, security, efficiency, and cost effectiveness.

NFV virtualizes network functions such as load balancers, firewalls, intrusion detection systems, and signaling systems that were previously provided by special-purpose hardware and that will now be implemented by software running on virtual machines running in a cloud computing infrastructure. As such, NFV reduces capital expenditures through the replacement of special-purpose dedicated hardware by software running on commodity hardware. It also reduces operational expenditures by leveraging the efficiencies that derive from virtualization in cloud computing such as the economies of scale, flexibility, customization, and elasticity.

SDN and NFV together drive the softwarization of networks toward a paradigm where software controls the treatment of flows in the network, adds value to these flows by software processing, and orchestrates the dynamic allocation of resources to meet the needs of customer applications while also promoting energy efficiency through the right-sizing and optimal placement of packet processing in a converged network and cloud infrastructure.
The long-term impact of SDN and NFV is that most (if not all) network and service functions could be virtualized, dynamically placed on the underlying physical infrastructure, and accessible through open control and management interfaces. This will shift the current CAPEX-based business model to an OPEX-based model. It will also lower the barrier to entry for new network equipment manufacturers and operators. The exploitation of high levels of automation, increased flexibility and programmability allows reinventing future network architectures, accelerating service deployment, and facilitating infrastructure management. This new environment is impacting core and edge networks alike and is forever changing the way public and enterprise communications are enabled. However, SDN and NFV are only a part of the wider systemic trend of Network Softwarization, which will impact not only the network and service platforms in operator and data center networks, but also application development, industrial and environmental sensors, to name a few. These changes will have a broader impact on society, including aspects of regulation, policy, social impacts, and new business models.

For our networking industry, the replacement of existing networking equipment and the introduction of a new framework for traffic and network management will entail extensive changes in management processes, practices, and organization, and hence major changes in the training and preparation of personnel. An entirely new set of skills from the engineers that design and operate networks is required, personnel with the training that specifically encompasses converged IT and communication engineering along with enabling technologies such as softwarization, virtualization, and cloud/edge/fog computing. In this respect, ComSoc has a major role to play.

COMSOC’S LEADERSHIP ROLE

As previously mentioned, ComSoc has been playing a leading role in establishing and promoting the emerging area of network softwarization through its technical activities, conferences, and publications. For instance, ComSoc’s Emerging Technologies Committee, which is responsible for identifying and nurturing new technology directions, formed the SDN-NFV Technical Sub-committee to focus on exploring next generation networking technologies enabling software defined service delivery, network virtualization, and network function virtualization. ComSoc has also launched a new conference on SDN and NFV. The first IEEE SDN-NFV conference took place on 18-21 November 2015 in San Francisco, California, USA; the second edition is scheduled to take place on 7-9 November 2016 in Palo Alto, California, USA (see: http://nfvsn2016.ieee-nfvsdn.org/).

In addition, SDN and NFV have been featured in various ways such as tutorials, workshops, keynotes, business forums, and technical sessions, as part of ComSoc’s flagship conferences including Globecom, ICC, INFOCOM, IM, and NOMS. Similarly, a number of special issues of technical journals and magazines on SDN and NFV have been organized in the last two years as part of ComSoc’s publications, including the IEEE Transactions on Network and Service Management, IEEE Communications Magazine, IEEE Network Magazine, and IEEE Wireless Communications Magazine. ComSoc’s Standards Activities Council has also been active examining standardization opportunities in SDN, NFV, and related areas. Several other technical activities related to SDN and NFV have been conducted in the last few years with ComSoc’s extensive participation and contribution. The IEEE SDN Initiative is perhaps the most comprehensive effort within the IEEE to promote the fast growing area of network softwarization and in which ComSoc is playing a leading role.

THE IEEE SDN INITIATIVE

In 2014, the IEEE SDN Initiative (http://sdn.ieee.org/) was launched by the IEEE Future Directions Committee, with the consensus of all IEEE societies and council presidents, as a cross-Society IEEE worldwide program addressing the main techno-economic aspects concerning SDN and NFV. The IEEE SDN Initiative is today comprised of seven committees: Conference, Education, Publications, Publicity, Standards, Pre-industrial, and Outreach, addressing specific stakes and challenges raised by Network Softwarization that go beyond technical issues to also encompass skill development and economics. With approximately 50 volunteers, the IEEE SDN initiative has been building a large technical community, counting today more than 1,000 experts worldwide. Within IEEE a number of societies have been supporting in various ways the creation and operations of the SDN initiative, triggering links with various technical committees in these societies, namely the Communications Society, the Computer Society, the Signal Processing Society, the Consumer Electronics Society, and the Reliability Society.

Among its realizations so far, the IEEE SDN initiative launched the International Conference on Network Softwarization (NetSoft) as its flagship event and primary IEEE forum for publication and technical exchange of the latest research results and innovations in this area. The first IEEE NetSoft conference took place in London (UK) on 13-17 April 2015 with the theme “Software-Defined Infrastructures for Networks, Clouds and Services.” NetSoft 2015 attracted 165 participants with a paper acceptance ratio of 19.1%, and featured three keynotes, three workshops, three tutorials, and seven demonstrations. The 2nd edition of the IEEE NetSoft conference (NetSoft’16) took place in June in Seoul (South-Korea) with the theme “Software-driven Internet of Things” with the following highlights: an acceptance ratio of 18.6%; 215 participants (a 30 percent increase from the previous year); many top-level keynotes, plenary sessions, parallel technical sessions, an industry exhibition from major industry players, demonstrations, and posters; broad industry participation; and eight hands-on tutorials and four workshops. The next edition of the conference (IEEE NetSoft 2017) will take place on 3-7 July 2017, in Bologna (Italy) with the theme “Softwarewarization Sustaining a Hyper-Connected World: en route to 5G!”. The deadline for submitting papers to NetSoft 2017 is 1 December 2017. In addition to its flagship conference, the IEEE SDN initiative has also been sponsoring a number of events featuring network softwarization such as the IEEE SDN-NFV conference, the IEEE Cloud Networking Conference (CloudNet), the International Conference on Network and Service Management (CNSM), IM, NOMS, and the European Workshop on SDN (EWSDN), to name a few. In addition to conference organization, the IEEE SDN initiative has been active on many other fronts, including the following.

Publications: In November 2015 a newsletter was launched, which now appears bi-monthly covering the latest developments in SDN and NFV related technologies in an easy to digest format (see: http://sdn.ieee.org/newsletter). The substantial and rapidly growing interest in the topic domain can be seen from the tremendous increase in the number of subscribers to the newsletter and the monthly page views in just a few months. To provide different publication outlets in the area of network softwarization, the SDN Initiative is currently in the process of launching a new IEEE magazine on Networking Software covering technical innovation, open source projects, standardization, research and developments in networking software, and softwarized networks, as well as aspects of regulation, policy, social impacts, and new business models. A letter of intent for creating this new magazine was well received by
the periodical review committee at its IEEE Technical Activities Board meeting in June 2016. The next step is to submit a fully developed proposal for review at the IEEE TAB meeting in November 2016. ComSoc is the main financial sponsor and managing society of the new magazine, with several other societies potentially as financial or technical co-sponsors, including the Computer Society, Reliability Society, Signal Processing Society, Consumer Electronics Society, and the Vehicular Technology Society. If approved, the new magazine will start appearing in the first quarter of 2018. To cover more in depth research and archival quality papers in the area, the SDN initiative also intends to submit a proposal for a new IEEE transactions journal in the same area. In the meantime, multiple special issues dedicated to SDN and NFV have been successfully organized within the IEEE Transactions on Network and Service Management, attracting more than 50 submissions.

**Industry Outreach through Pre-Industrial Activities:** The IEEE SDN initiative organized the first industry workshop focused on Mobile Edge Cloud, held on 16 November 2015 at IEEE Headquarters in Piscataway, NJ, USA. The second edition of the workshop was recently held in Venice, Italy, hosted by Telecom Italia Mobile, Future Centre on 15-16 June 2016. The workshop brought together visionaries and thought leaders from industry and academia to address various approaches around the Mobile Edge Cloud and its applications (e.g., supporting 5G networks) in order to build a consensus on its overall architectural framework.

**Standards:** The IEEE SDN initiative has made efforts to advance SDN standardization driving the functionality, capabilities, and interoperability of SDN and NFV. In particular, the Standards Activities Council of the IEEE Communications Society has established two research groups and two study groups to examine standardization opportunities in SDN, NFV, and related areas. In this perspective, Rapid Reaction Standardization Activity (RRSA) sessions are organized to solicit contributions from stakeholders, including ComSoc relevant Technical Committees. The next RRSA session is planned for 1 November 2016 in Berlin in conjunction with the 2016 IEEE Conference on Standards for Communications & Networking (CSCN) conference, and will be featuring SDN/NFV and 5G standardization.

**Education:** The IEEE SDN initiative provides links to a selection of online educational materials pertaining to SDN and NFV such as Webinars, keynote talks, courses, market oriented events, and blogs.

**CALL FOR PARTICIPATION**

After many years of the status quo, the networking industry is now in the middle of a major transformation with immense potential for growth driven by the increasing reliance on Commodity Off the Shelf (COTS) hardware, the growing availability of open source software solutions, and the current movement toward network softwarization. At ComSoc, we are privileged to be part of this industry, and at the forefront of its transformation with tremendous opportunities for contributions and impact. If not already, we hope that you will join and become involved.

“The best way to predict the future is to invent it.”

-Alan Kay
CALL FOR PAPERS AND PROPOSALS

IEEE ICC 2017 will be held at Palais des Congrès - Porte Maillot, Paris, France, 21-25 May 2017. Located in the heart of the City of Lights, IEEE ICC 2017 will exhibit an exciting technical program, complete with 13 Symposia highlighting recent progress in all major areas of communications. IEEE ICC 2017 will also feature high-quality Tutorials and Workshops, Industry Panels and Exhibitions, as well as Keynotes from prominent research and industry leaders.

Prospective authors are invited to submit high-quality original technical contributions for presentation at the conference and publication in the IEEE ICC 2017 Proceedings and IEEE Xplore. Proposals for Tutorials, Workshops, and Forums are also invited. Visit http://icc2017.ieee-icc.org for more details.

TECHNICAL SYMPOSIA

SELECTED AREAS IN COMMUNICATIONS

Access Systems and Networks
Steven Hranilovic, McMaster University, Hamilton, Canada

Big Data Networking
Shu Yu, Deakin University, Australia

Cloud Communications and Networking
Ioannis Pappas and Papadopoulos, Purdue University, USA

Communications for the Smart Grid
Deepa Kundur, University of Toronto, Canada

Data Storage
Onur Ozcan Kayaboglu, University of Arizona, USA

E-Health
Jaime I. Peralta, Polytechnic University of Valencia, Spain

Internet of Things
Antonio Jara, University of Applied Sciences Western Switzerland (HES-SO), Switzerland

Molecular, Biological, and Multi-Scale Communications
Urbaishi Mitra, University of Southern California, USA

Satellite and Space Communications
Igor Blak, University of Oviedo, Italy

Social Networking
Darla Turpil, University of Central Florida, USA

Software Defined Networking & Network Function Virtualization
Laszlo Liddle, Luxembourg University, Luxembourg

Ad-Hoc and Sensor Networking
Shibin He, Zhejiang University, China

Aden Kantini, University of Rennes, France

Cheng Li, Memorial University of Newfoundland, Canada

Cognitive Radio and Networks
Oliver Holland, King’s College London, UK
Wald Saad, Virginia Tech, USA

Communication and Information Systems Security
Peter Mueller, ETH, Switzerland
Cong Wang, University of Hong Kong, China

Communications Software, Services and Multimedia Applications
Maria G. Martin, King’s College London, UK
Said Hoseini, University of Paris 12, France
Aida Khan Shami, Western University, Canada

Communication Theory
Frieder Ayanoglu, University of California, Irvine, USA
Fulvio Babin, University of Trieste, Italy
Steven Weber, Drexel University, USA

Green Communications Systems and Networks
Ciek Cavdar, Royal Institute of Technology Stockholm, Sweden
Michela Moe, Politecnico di Torino, Italy

Internet of Things
Antonio Jara, University of Applied Sciences Western Switzerland (HES-SO), Switzerland

Next Generation Networking and Internet
Shivren Moe, Auburn University, USA
Mahesh K. Marin, University of Edinburgh, UK
Sidi-Mohammed Senouci, University of Bourgogne, France

Optical Networks and Systems
Grzegorz Danielewicz, Poznan University of Technology, Poland
George Roukas, North Carolina State University, Raleigh, USA

Wireless Communications
Mohamad Assaad, CentraleSupelec, France
Azeddine Bouchetche, University of Ottawa, Canada
David Davey, University of Bologna, Italy
Hiromi Sato, Aalto University, Finland
Yang Song, Missouri University of Science and Technology, USA

Mobile and Wireless Networking
Mohammed Aloussari, University of Oklahoma, USA
Mendi Bennis, University of Oulu, Finland
Jael Ben-Othman, University of Paris 13, France
Shaojie Wang, Nanyang University, China

INDUSTRY FORUMS AND EXHIBITION PROGRAM

IEEE ICC 2017 will feature several prominent keynote speakers, major business and technology forums, and a large number of vendor exhibits. Submit your proposals to the IEEE Co-Chairs Luis M. Correia (luis.correia@etor.fr) and Jamshed Khan-Jush (khanjush@l.q.ualberta.ca).

TUTORIALS

Proposals are invited for half- or full-day tutorials in all communication and networking topics. For inquiries, please contact Tutorials Program Co-Chairs Hanna Bogucka (hanna.bogucka@psl.poznan.pl) and Luc Vandendorpe (luc.vandendorpe@ulgouv.be).

WORKSHOPS

Proposals are invited for half- or full-day workshops in all communication and networking topics. For inquiries, please contact the Workshops Program Co-Chairs Abbas Jamali (jamali@ieee.org) and Constantinos Papadias (cpp@ait.ac).

Full details of submission procedures are available at icc2017.ieee-icc.org
The 19th ICIN conference was held March 1–3 in Paris, France, with the support of Orange, Nokia, Deutsche Telekom, Mitel, and UPMC University. The conference attracted 186 delegates from 22 countries representing approximately 57 different organizations to discuss new technologies related to the Internet and the telecommunications industry. ICIN has a 27-year history of anticipating key trends in communications networks and services, and showcasing technologies and architectures that become vital elements of delivering services.

The acceptance rate of the conference was 24 percent for full papers. Each paper was evaluated by at least four reviewers with an average of six reviews per paper.

The conference was structured around four tracks: Network IT-isation and 5G (chaired by Prosper Chemouil of Orange Labs, France, who is an IEEE Fellow); Big Data Insights for Networking (chaired by Françoise Soulié Fogelman, Tianjin University, China); Real-Time Communication Platforms and Services (chaired by Axel Küpper, Deutsche Telekom, Germany); and Internet of Things (chaired by Payam Barnaghi University of Surrey, UK). There was also a special session on Digital Platform Economics and Regulation (chaired by Alain Vallée, Innovation & Regulation in Digital Services Chair, France), and a workshop on Green Communication Systems (chaired by Noel Crespi, Institut Mines-Telecom, France). The conference included a demonstration track (co-chaired by Jean-Christophe Schiel, Airbus Defence and Space, France, and Max Michel, Orange, France), which included five demonstrations with poster presentations. Seventeen research articles were presented, plus 16 poster presentations.

The conference was opened by Emmanuel Bertin (Orange Labs, France), Chair of the ICIN 2016 Technical Program Committee, and Noel Crespi, Chair of the ICIN International Advisory Board. A series of keynote speeches were delivered around the theme for the conference. Raouf Boutaba, (a professor at the University of Waterloo, Canada, and IEEE Fellow) presented a view that service architecture changes with generations of technology roughly every 25 years. He also offered insights from his research on service deployment and management in clouds. Nicolas Demassieux (Senior VP of Research, Orange Labs) presented insights on Orange’s research in networks, highlighting the need for multiple methods of access, with the view that software is increasingly becoming critical to the real world, which creates new opportunities for network operators. Ina Minie (Google, USA) presented a view of Google’s cloud network environment, highlighting the challenges of managing services and servers in a global network, as well as their work toward improving the state of the art. The remainder of the first day was spent in invited talks for the first track (Network IT-isation and 5G) highlighting the need for service support at the network edge to support the real-time response demanded by many applications, and the first of two poster and demonstration sessions.

The second day of the conference began with a Keynote from Marcus Weldon (President of Nokia Bell Labs and CTO of Nokia), which outlined the need for near real-time response from networks in support of the “things” connected to the network, demonstrating that physical laws demand that processing be close to endpoints to meet that need, presenting a unique opportunity for network operators. A session from the Big Data Insights for Network Track followed, presenting insights in privacy, network management, and the challenges of the next generation of databases and data analysis. The afternoon featured a workshop on green communications systems and a session on real time communications platforms and services, which represented a revival in research interest in this topic. The workshop also presented insights on global identity management and reachability, as well as the role of WebRTC in supporting next generation communication services. The day concluded with a special session on platform economics and regulation, and a second evening of posters and demonstrations.

The final day began with a keynote from Volker Ziegler, (Chief Architect of Technology and Innovation, Nokia Networks, Germany). He presented the challenges of 5G networking and focused on the opportunities for network operators in a unique position to support the demands to optimize transport to access services in order to meet bandwidth and latency requirements. The rest of the day featured sessions from the conference track on Network IT-isation and 5G. The IoT track presented new results in securing access, optimizing applications considering device power and communication limits, and challenges for network operators in supporting the massive expansion of endpoints entailed by the IoT.

ICIN has consistently previewed trends in networks and services, and this year was no exception. Both the keynotes and papers highlighted the significant changes coming from the connection of massive numbers of machine endpoints to the network and the associated requirements for bandwidth and real-time response. These factors are driving a new generation of access technologies, as well as fundamentally changing the architectures required to deliver services, driving more functions to the network edge to overcome transport delays and meet response requirements. Other key insights were presented in the area of interworking and identity management, and services based on RTC. One other key focus was in the area of network function virtualization, which has been presented in several past ICIN conferences, but has now become reality and is reducing costs and increasing flexibility for network operators. Virtualization is being used for an increasing number of functions in networks, even extending to functions now performed by custom hardware on network flows. The trend is expected to continue.

Planning for ICIN 2017 is already underway under the leadership of Antonio Manzalini (Telecom Italia), Technical Program Chair for ICIN 2017. The conference will take place in Paris on 7–9 March 2017. For more information please visit www.icin.co.uk
IEEE North Jersey Section Recognizes Society Members Awardees

By Amit Patel, IEEE North Jersey Section Communications Society Chapter Chair

At the IEEE North Jersey Section’s Annual Awards Reception, held May 1, 2016 at the Birchwood Manor, Whippany, New Jersey, USA, several section Communications Society members were recognized for their outstanding contributions.

The reception honors numerous local members from different technical society chapters and committees for their achievements. Awardees and their guests celebrate many different accolades covering section, regional, international, and society awards as well as Fellow grade elevations.

The section had the pleasure of recognizing three of its Communications Society members this year. The honorees were for a Fellow elevation, a Society prize, and IEEE Region 1 recognition.

The IEEE North Jersey Section is very active and has numerous opportunities for both members and visitors to attend meetings on both technical and professional topics. For a listing of local meetings, events, conferences, and publicity coverage, visit the IEEE North Jersey Section website at: http://sites.ieee.org/north-jersey/.

Emad Farag is LTE Modem Software Technical Manager at Nokia–Bell Labs in Murray Hill, New Jersey, USA. He received his Ph.D. from the University of Waterloo in 1997, and his Master of Science from Ain Shams University in 1994. Prior to his time at Bell Labs he was a research assistant professor at the University of Waterloo. He is also currently a volunteer with the North Jersey Section, and he chairs the Instrumentation Measurement Society Chapter.

With more than 18 years at Nokia he has authored many patents and he has held roles of increasing responsibility managing numerous engineers on different teams. He was received the IEEE Region 1 Technological Innovation Award for outstanding leadership and significant innovations in wireless modem architectures.

Thomas Marzetta is the originator of Massive MIMO, the most promising technology available to address the ever increasing demand for wireless throughput. He is group leader of Large Scale Antenna Systems at Nokia–Bell Labs, and co-head of their FutureX Massive MIMO project.

Dr. Marzetta received a Ph.D. and B.S. in electrical engineering from Massachusetts Institute of Technology in 1978 and 1972, respectively, and the M.S. in systems engineering from the University of Pennsylvania in 1973. In 1995 he joined Bell Labs, where he served as the director of the Communications and Statistical Sciences Department within the former Math Center.

He received the 2015 IEEE Communications Society Stephan O. Rice Prize for the paper co-authored with Hien Quoc Ngo and Erik G. Larsson entitled “Energy and Spectral Efficiency of Very Large Multiuser MIMO Systems,” published in the IEEE Transactions on Communications, vol. 61, no. 4, pp. 1436-1449, April 2013.

Osvaldo Simeone received the M.Sc. degree (with honors) and the Ph.D. degree in information engineering from Politecnico di Milano (Continued on Newsletter page 4)
Highlights of Internet of Things (IoT) Conferences in Silicon Valley

By Alan J Weissberger, IEEE Sr. Life Member and GCN NA Correspondent, USA

With extraordinary potential, impact, and influence, the Internet of Things (IoT) promises to be a big revenue generator and improve efficiencies for many different industries and companies. Gartner Group has forecast "$1.9 trillion value-add IoT revenue across industry sectors in 2020." Also from Gartner: "IoT endpoints will grow at a 31.7 percent CAGR from 2013 through 2020, reaching an installed base of 20.8 billion units."

While IoT is one of the hottest tech topics trending now, it is marked by tremendous hype, confusion, and chaos. This article reviews three recent IoT conferences in Santa Clara, CA (the center of Silicon Valley) that were aimed at cutting though the confusion, enumerating the challenges and opportunities, and discussing the current status of the IoT.


This IEEE conference focused on the different IoT challenge areas that are of concern to various industries, such as: power/energy, aviation/transportation, automotive, industrial control, medical, and wearables.

Keynote speakers included Ken Caviasca from Intel, who discussed IoT Technologies, and Phil Kelly, Chief Scientist at Emergous, who delivered a keynote on IoT Wearables. Key use cases, adoption patterns, and techno-socio-economic challenges were discussed, along with security (which continues to be the number one challenge for most IoT applications).

There was an informative presentation on "The Transformative Role of IoT on the Future of Education," and a lively panel focused on venture capital and the IoT areas that are attracting investments. Not surprisingly, most investments are in start-ups that are building IoT software platforms that run in the cloud, or in big data that analyzes data collected by "things."

A panel on an IEEE standard for an IoT architectural framework provided an overview of that ongoing work in progress. The architectural framework defined in the P2413 IEEE standard is intended to promote cross-domain interaction, aid system interoperability and functional compatibility, and fuel the growth of the IoT market. The adoption of a unified approach to the development of IoT systems will reduce industry fragmentation and create a critical mass of stakeholders around the world. The architectural framework for IoT provides a reference model that defines relationships among various IoT verticals (e.g., transportation, healthcare, etc.) and common architecture elements. It also provides a blueprint for data abstraction and the quality "quadruple" trust that includes protection, security, privacy, and safety. Also, this standard provides a reference architecture that builds upon the reference model. The reference architecture covers the definition of basic architectural building blocks and their ability to be integrated into multi-tiered systems.

INTERNET OF THINGS DEVELOPERS CONFERENCE – MAY 25-26, 2016

This conference and trade show focused specifically on the IoT product developer with in-depth technical presentations and panel sessions. Hands-on demonstrations helped attendees learn about a vendor’s IoT platform.

Maarten Bron of Underwriters Laboratories presented a very refreshing session on IoT security. Bron stated that "as our connected society continues to expand, we are rapidly approaching the practical limits of classical IT security certification. Since nobody certifies the Internet, we must make sure that the Internet of Things becomes a reliable and trustworthy place to be in."

There were two key takeaways:
• IoT security should be as foolproof as an unpickable lock.
• Each vendor’s IoT security capabilities should be STAR rated, by either an independent certification agency or the vendor, after executing several (to be defined) test procedures.

One of the more interesting panel sessions was titled “Sorting Through the Myriad of IoT Connectivity Options.” This included both wireless LANs and new narrow bandwidth wireless WANs. These are often referred to as NB-IoT and/or LPWA (or LPWA). There are many choices here, but the main ones are:
• LTE Category 1, M1, M2 (Verizon, many other LTE operators).
• LoRa WAN (Orange, SK Telecom).
• SIGFOX (new 2G type of wireless WAN being deployed in France, Belgium, the U.S., etc.).
• Wi-Fi HaLow™ (WiFi Alliance designation for products incorporating IEEE 802.11ah technology operating <1GHz).

 Needless to say, there are a plethora of standards bodies and forums working on new connectivity standards or tweaking existing standards for IoT applications. This is illustrated in the figure below, courtesy of Ericsson.

IOT WORLD – MAY 10-12, 2016

This was the largest IoT conference in the world with more than 10,100 registered attendees, as well as the biggest IoT trade show in terms of exhibit space. Every square inch of the Santa Clara Convention Center was packed with exhibits and vendor presentation or showcase areas.

IoT industry verticals like smart homes, connected cars, wearables (fitness trackers, medical monitors, etc.), industrial IoT, manufacturing/factory floor, smart building/smart cities, were all represented on the show floor, conference sessions, and vision theater. There were also sessions and exhibits on IoT security, IoT cloud, big data/analytics, smart cities, wearables, and many other hot topics broadly related to IoT.

The main theme of the May 10 executive keynotes, “Disrupt, Innovate, and Monetize,” with new business models, improved productivity and other benefits, greatly added to the hype, while providing little or no substance to resolve the critical issues that have plagued IoT since the name was coined (replacing M2M communications).

(Continued on Newsletter page 4)
Activities of the IEEE ComSoc Kerala Chapter
By Sreevas Sahasranamam, Secretary of the IEEE Communications Society Kerala Chapter

INTERNET OF THINGS (IoT) WORKSHOP

The IEEE Communications Society (ComSoc) Kerala Section chapter organized a one day workshop on the contemporary and interdisciplinary engineering topic Internet of Things (IoT) on 10 December 2015 in Trivandrum. This workshop was co-organized with the third edition of the International Conference on Recent Advances in Computational Systems (RAICS) 2015. This workshop was supported by $US800 in funding from the Region 10 ComSoc Chapter. The event was attended by 40 people, which included a mix of students, academics, and industry professionals.

The event started at 9:30 am with a welcome speech by R. Ananthalakshmi Ammal, the ComSoc Kerala Section Chair. The workshop led by Dr. Srikant (Chief Knowledge Officer, Nanocell Networks) was scheduled into four modules. The first module provided an overview of IoT applications and current trends in IoT. The second module discussed the technologies enabling IoT, such as sensors and actuators, wireless, IoT platforms, big data, and cloud. In the first half of the afternoon session, real time demos of IoT applications such as remote home monitoring and automated moisture control for soil were demonstrated by Mr. Vignesh Pai (Senior Technical Trainer, Nanocell Networks). The final session focused on protocols and standards of IoT applications.

DISTINGUISHED LECTURE PROGRAM

The IEEE Communications Society Kerala Section organized two Distinguished Lecture Programs (DLPs) during the month of March 2016 on the topic “Cooperative Service Management in the Healthcare Sector: Emerging Trends and Future Challenges.” The first session was held on 30 March 2016 at the CDAC Trivandrum from 5:30 to 6:30 pm, with participation of approximately 35 people. The second session was held on 31 March 2016 at the Rajagiri School of Engineering and Technology (RSET), Kochi from 12 noon to 1:00 pm, with participation of more than 50 people. The sessions were presented by Professor Pradeep Ray from the University of New South Wales, Australia. Professor Ray is also the Director of the WHO Collaborating Centre on eHealth. His talk extensively discussed the role of technology in e-health and m-health initiatives, and its impact on rural areas. The sessions were organized in collaboration with the IEEE Engineering in Medicine and Biology Society (EMBS) and the IEEE Society on Social Implications of Technology (SSIT).

College of Engineering Trivandrum
ComSoc Student Branch Chapter

The ComSoc Student Branch at the College of Engineering Trivandrum conducted a ham radio event involving Morse code.

From the Republic of Macedonia
Chapter: Another Successful FutureComNet Edition

By Vladimir Atanasovski, IEEE R. Macedonia ComSoc Chapter Chair

FutureComNet is an annual conference organized by the Institute of Telecommunications at the Faculty of Electrical Engineering and Information Technologies in Skopje. The main idea is to celebrate the World Telecommunications and Information Society Day (WTISD), on 17 May. Moreover, this year (2016) marks two noteworthy milestones: the 100th anniversary of the birth of Claude Shannon, and the 60th anniversary of the birth of Nikola Tesla. Therefore, the IEEE R. Macedonia ComSoc chapter decided to support FutureComNet 2016 in order to celebrate these important milestones and increase the visibility of the chapter and its wider dissemination. Prof. Liljana Gavrilovska is the General Chair of FutureComNet 2016 and the past president of the IEEE R. Macedonia Com-
IOT CONFERENCES/Continued from page 2

The Executive Keynotes (often sponsored talks) were from Hitachi, SAP, ADT, Microsoft, Silver Springs Networks, HP Enterprise, Schneider Electric, and JCI. One standout was the HP Enterprise announcement of their Universal IoT Platform.

The new functionality in the HP Enterprise Universal IoT Platform was said to be a driving force in building the infrastructure that will enable and sustain the growth of IoT. The HPE universal IoT platform is aligned with the oneM2M industry standard and is designed to be industry and vendor-agnostic, enabling IoT operators to simultaneously manage heterogeneous sets of sensors, operate vertical applications on machine-to-machine (M2M) devices, as well as process, analyze, and monetize collected data in a single secure cloud platform.

The HPE Universal IoT Platform provides increased support for long range, low power connectivity, ensuring that LoRa® and SIGFOX deployments can be supported alongside other connectivity protocols, including cellular, radio, Wi-Fi, and Bluetooth.

In a May 11 technical session titled “Navigating the IoT Connectivity Landscape,” Alex Kengen of Dialog Semiconductor stated that there is a “sea of confusion” when it comes to IoT connectivity standards for LANs and PANs, such as WiFi, Bluetooth, ZigBee, Thread, and DECT ULE. That confusion also extends to new IoTity standards for LANs and PANs, such as WiFi, Bluetooth, ZigBee, etc.

Continuing the conversation, the session concluded with the idea that confusion about connectivity options and standards will mean that IoT systems will need to employ a variety of communication options, including cellular, radio, Wi-Fi, and Bluetooth.

KERRIL CONSIDING CONCERNS

• We wonder how “universal” the HP and other IoT platforms really are and what truly differentiates them from more than 350 other IoT platforms that have been announced.

• Connectivity is a critical issue for multi-vendor interoperability. IoT PANs, LANs, and WANS. However, it is just the first step for IoT interoperability and may not be IP based (e.g., a different packet format and addressing is used by ZigBee). Many other protocol-related issues are still unresolved, including: message format above the transport layer, authentication, security protocols and countermeasures, failover/protection/restoration, OAM&P, etc.

• There needs to be much more discussion on how much of the “thing” data, control, and status signals should be sent to (or from) the internet vs. being handled by a local access controller. Of course, there will likely not be any local controller for heavy industrial equipment in the field or a cargo container moving through the ocean. But what about a connected home, car, or factory floor?

• Security and privacy remain huge challenges for IoT. That has remained the case with little observable progress over the past six years.

NORTH JERSEY SECTION/Continued from page 1

c/o di Milano, Milan, Italy, in 2001 and 2005, respectively. He is currently with the Center for Wireless Communications and Signal Processing Research (CWCSPR), New Jersey Institute of Technology (NJIT), Newark, NJ, where he is an associate professor. His research interests are in wireless communications, information theory, optimization, and machine learning.

In July 2015 he received the IEEE Communication Society Best Tutorial Paper Award for the paper “Multi-Cell MIMO Cooperative Networks: A New Look at Interference,” published in IEEE Journal on Selected Areas in Communications in Dec. 2010. While on sabbatical at Imperial College, London in late 2015, he was elevated to IEEE Fellow for contributions to cooperative cellular systems and cognitive radio networks.

KERALA CHAPTER/Continued from page 3

practice. An interactive session was first conducted with members of a ham radio club at the Kerala State Science and Technology Museum, Thiruvananthapuram on 21 July 2015.

The event started with a welcome speech from the ComSoc student branch chair. This was followed by a session by Joseph Daniel, who is a member of the Radio Amateur Society of Ananthapuri (RASA). He discussed the basics of ham radio, Morse code, and the relevance of ham radio in the modern world where a variety of communication options are available. A live demonstration was carried out and we were able to contact a ham operator living in Attingal. Fifteen students participated in the session, including those who appeared for the examination on 31 July 2015. The session was concluded with a vote of thanks by the secretary of the ComSoc Student Branch at the College of Engineering Trivandrum.

The ham radio examination was conducted at the Kerala State Science and Technology Museum PMG on 31 July 2015. Seventeen students from the College of Engineering Trivandrum attended the examination, and all of them successfully passed the examination.
### Conference Calendar

**2016**

**SEPTEMBER**

Valencia, Spain
http://www.ieee-pimrc.org/

Palma de Mallorca, Spain
http://www.asmsconference.org/

Vienna, Austria
http://edoc2016.univie.ac.at/

**ITC28** 2016 — Int’l. Teletraffic Conference, 12–16 Sept.
Würzburg, Germany
http://itc28.org/

Munich, Germany
http://ieehealthcom2016.com/

**IEEE 37th Sarnoff Symposium** 19–21 Sept.
Newark, NJ
http://sites.ieee.org/sarnoff2016/

Poznan, Poland
http://iswcs2016.org/

Jaipur, India
http://icacci-conference.org/2016/home


Montreal, Canada
http://networks2016.etsmtl.ca/

Aachen, Germany
http://www.ti.rwth-aachen.de/ WiSE2016/index.html

**OCTOBER**

Pisa, Italy
http://cloudnet2016.ieee-cloudnet.org/

Kaiserslautern, Germany
http://www.icmu.org/icmu2016/

Kanazawa, Japan
http://www.ieice.org/~icm/apnoms2016/

Hanoi, Vietnam
http://atc-conf.org/

Nicosia, Cyprus
http://honet-ict.org/

Yangzhou, China
http://wcsp2016.org/

Philadelphia, PA
http://cns2016.ieee-cns.org/

Jeju Island, Korea
http://www.ictc2016.org/main/

Porto, Portugal
http://www.giis-conf.org/

Toronto, Canada

Fez, Morocco
http://www.wincom-conf.conf.org/

Berlin, Germany
http://cscn2016.ieee-cscn.org/

Montreal, Canada
http://www.cnsm-conf.org/2016/

**NOVEMBER**

**MILCOM 2016** — Military Communications Conference, 1–3 Oct.
Baltimore, MD
http://events.afcea.org/milcom16/Public/enter.aspx

**IEEE SmartGridComm** — IEEE Int’l. Conference on Smart Grid Communications, 6–9 Nov.
Sydney, Australia
http://sgc2016.ieee-smartgridcomm.org/
The driving forces behind the exponential growth in mobile cellular network traffic have fundamentally shifted from the steady increase in demand for conventional “connection-centric” communications, such as phone calls and text messages, to the explosion of “content-centric” communications, such as video streaming and content sharing. The Cisco Visual Networking Index projects that video traffic will amount to 72 percent of global mobile data traffic by 2019. The mobile cellular network architectures of today are, however, still designed with a connection-centric communication mindset. Moreover, the myriad technological advances proposed for beyond fourth generation (4G) and 5G mobile networks still mostly focus on capacity increase, which is fundamentally constrained by the limited radio spectrum resources as well as the diminishing investment efficiency for operators, and therefore will always lag behind the growth rate of mobile traffic. It can be argued that the logjam in cellular networks cannot be addressed by improving connection capability alone, but instead must be tackled by fundamentally addressing the underlying ineffectiveness of the current communication architecture for massive content delivery.

To cope with the shift to content-centric mobile cellular networks, a new design paradigm beyond the current connection-centric communication architecture is needed. In today’s mobile networks, caching and computing capabilities are already ubiquitous, both at the base stations and on user devices themselves. How to effectively utilize these existing capabilities for massive content distribution is a fundamental question current and future research must address. It is envisioned that the leveraging of ubiquitous caching and computing at the wireless network edge can induce physical layer transmission cooperation and hence enjoy spectrum efficiency gain. The article “Game Theoretic Approaches for Wireless Proactive Caching,” on the other hand, introduces game theoretical approaches to modeling the cooperative and non-cooperative behaviors of participants in a wireless system with proactive caching.

The next two articles leverage cloud computing in content-centric mobile networks. In particular, the article “Harnessing Cloud and Edge Synergies: Toward an Information Theory of Fog Radio Access Networks” lays theoretical foundations for the main trade-offs between system performance (i.e., worst-case delivery latency) and system resources (i.e., caching and fronthaul capacities) in a fog radio access network architecture. The article “The Role of Cloud Computing in Content-Centric Mobile Networking” brings cloud computing into both core network and RAN caching with detailed elaboration on cloud content delivery on the core network side and cloud RAN with caching as a service (CaaS) on the RAN side.

The seventh article, “Exploring Synergy between Communications, Caching, and Computing in 5G-Grade Deploy-
ments,” provides a detailed analysis of the synergy between communications, caching, and computing in next-generation wireless deployments with supporting full-fledged trial results on a live cellular network.

The remaining five articles address various issues on content delivery in specific mobile networking scenarios. In particular, the article “Enhancement for Content Delivery with Proximity Communications: Architecture and Challenges” presents architectural issues and content delivery strategies for combining small cell base station caching with proximity-based device-to-device caching for video delivery. The article “Mobility-Aware Caching for Content-Centric Wireless Networks: Modeling and Methodology” explores efficient content caching strategies at the wireless edge, via exploitation of available information regarding user mobility. By exploiting the common interest of mobile users and activating their opportunistic communications, the novel integrated network presented in the article “Socially Aware Integrated Centralized Infrastructure and Opportunistic Networking: A Powerful Content Dissemination Catalyst” demonstrates superior content delivery efficiency over classic centralized infrastructure networks. The article “NMRTS: Content Name-Based Mobile Real-Time Streaming” presents a novel transport protocol to support seamless handover for real-time video streaming in content-centric mobile networks. The last article, “SAVING - Socially Aware Vehicular Information-centric Networking,” proposes a new solution of leveraging smart vehicles with their caching, computing, and communication capabilities to facilitate content delivery in urban environments.

We would like to thank the large number of people who significantly contributed to this FT, including the authors, reviewers, and IEEE Communications Magazine publications staff. We hope that the readers enjoy this FT and that the selected articles stimulate new ideas and innovations in future content-centric mobile communication networks.

**Biographies**

Meixia Tao (mxtao@sjtu.edu.cn) is a professor in the Department of Electronic Engineering, Shanghai Jiao Tong University (SJTU), China. She received her Ph.D. from Hong Kong University of Science and Technology in 2003. Prior to joining SJTU, she was with National University of Singapore as an assistant professor. She currently serves as an Executive Editor of IEEE Transactions on Wireless Communications and an Editor of IEEE Transactions on Communications. She received the IEEE Heinrich Hertz Award for Best Communications Letters in 2013 and the IEEE ComSoc Asia-Pacific Outstanding Young Researcher Award in 2009.

Wei Yu (F) (weiyu@ece.utoronto.ca) is a professor and Canada Research Chair in Information Theory and Wireless Communications at the University of Toronto, Canada. He obtained his Ph.D. from Stanford University in 2002. He currently serves on the IEEE Information Theory Society Board of Governors, and is an IEEE Communications Society Distinguished Lecturer. He received a Steacie Memorial Fellowship, IEEE Communications Society Best Tutorial Paper Award in 2015, and an IEEE Signal Processing Society Best Paper Award in 2008. He is a Highly Cited Researcher according to Thomson Reuters.

Wei (Andrew) Tan (andrew.tan@huawei.com) is a senior research scientist and architect at Central Research Institute, Huawei Technologies, Co. Ltd Shanghai, China. He received his Ph.D. in 2008 from Harbin Institute of Technology. He has published over 20 papers in refereed journals and conferences proceedings and more than 20 patents. He has been engaged in wireless communication technology R&D work, including LTE, ad hoc networks, and 5G networks. He is also very active in global standards for 5G, including 3GPP and IMT-2020.

Suant Roy (F) (sroy@u.washington.edu) is presently the Integrated Systems Professor of Electrical Engineering, University of Washington, Seattle, where his research interests include analysis/design of communication networks with a diverse emphasis: next-generation wireless LANs (802.11) and emerging beyond 4G cellular standards, multi-standard wireless internetworking/coexistence, cognitive radio platforms for spectrum sharing, and sensor, vehicular, and underwater networks. He was elevated to IEEE Fellow by IEEE ComSoc for his “contributions to multi-user communications theory and cross-layer design of wireless networking standards.” He has served ComSoc as Associate Editor for IEEE Transactions on Communications and IEEE Transactions on Wireless Communications, and as a Distinguished Lecturer (2014–2015).
Communications, Caching, and Computing for Content-Centric Mobile Networks

Wireless Caching: Technical Misconceptions and Business Barriers

Georgios Paschos, Ejder Baştuğ, Ingmar Land, Giuseppe Caire, and Mérouane Debbah

ABSTRACT

Caching is a hot research topic and poised to develop into a key technology for the upcoming 5G wireless networks. However, the successful implementation of caching techniques crucially depends on joint research developments in different scientific domains such as networking, information theory, machine learning, and wireless communications. Moreover, there are business barriers related to the complex interactions between the involved stakeholders: users, cellular operators, and Internet content providers. In this article we discuss several technical misconceptions with the aim of uncovering enabling research directions for caching in wireless systems. Ultimately, we make a speculative stakeholder analysis for wireless caching in 5G.

INTRODUCTION

Caching is a mature idea from the domains of web caching, content delivery networks, and memory optimization in operating systems. Why is caching still an active topic of discussion? In the 1990s, the traffic in the web exploded, leading its inventor, Sir Tim Berners-Lee, to declare network congestion as one of the main challenges for the Internet of the future. The congestion was caused by the dotcom boom, specifically due to the client-server model of connectivity, whereby a web page was downloaded from the same network server by every Internet user in the world. The challenge was ultimately resolved by the invention of content delivery networks (CDNs) and the exploitation of web caching. The latter replicates popular content in many geographical areas and saves bandwidth by avoiding unnecessary multihop retransmissions. As a byproduct, it also decreases access time (latency) by decreasing the distance between two communicating entities.

Today, 30 years later, we are reviving the same challenge in the wireless domain. The latest report of Cisco [1] predicts a massive increase of Internet devices connected through wireless access, and warns of a steep increase in mobile traffic, which is expected to reach roughly 60 percent of total network traffic by 2018, the majority of which will be video. Wireless system designers strive to fortify fifth generation (5G) wireless networks with higher access rates on one hand and increased densification of network infrastructure on the other. Over the last three decades, these two approaches have been responsible for the majority of network capacity upgrade per unit area, successfully absorbing the wireless traffic growth. However, with the explosion of access rates and number of base stations, the backhaul of wireless networks will also become congested [2, 3], which motivates further use of caching: storing popular reusable information at base stations to reduce the load at the backhaul. Furthermore, a recent technique [4] combined caching with coding and revolutionized how goodput scales in bandwidth-limited networks. Therefore, caching has the potential to become the third key technology for wireless system sustainability.

The research community is converging to an enabling architecture as shown in Fig. 1. In the network of the future, memory units can be installed in gateway routers between the wireless network and the Internet (e.g., in 4G this is called the serving gateway, S-GW), in base stations of different sizes (small or regular size cells), and in end-user devices (mobile phones, laptops, routers, etc.). In this article, we discuss important topics such as:

- The characteristics of cacheable content and how this affects caching technologies in wireless
- Where to install memory
- The differences between wireless caching and legacy caching techniques

Last, we focus on business barriers that must be overcome for the successful adoption of wireless caching by the industry.

DEALING WITH MASSIVE CONTENT

Not all network traffic is cacheable. Interactive applications, gaming, voice calls, and remote control signals are examples of information objects that are not reusable and hence cannot be cached. Nevertheless, most network traffic today (an estimated 60 percent [1]) is deemed cacheable. We refer to cacheable information objects as content in this article. Since the performance of caching is inherently connected to the specifics of contents, this section is dedicated to the understanding of these specifics.

In particular, we focus on the following misconceptions:

- The static IRM model is sufficient for experimentation.

Georgios Paschos, Ingmar Land, Mérouane Debbah are with Huawei Technologies; Ejder Baştuğ is with Université Paris-Saclay; Giuseppe Caire is with Technische Universität Berlin
• User information cannot be used for popularity estimation due to the vast number of users.
• Security issues preclude caching at the edge.

INSUFFICIENCY OF STATIC POPULARITY MODELS

The standard approach to designing and analyzing caching systems involves generating content requests to replace the actual request traces — this approach is often several orders of magnitude faster.

The de facto model for performance analysis of web caching is the independence reference model (IRM): content \( N \) is requested according to an independent Poisson process with rate \( \lambda \), where \( p_n \) refers to the content popularity modeled by a power law (i.e., \( p_n \propto n^{-\alpha}, \alpha > 0 \)). This well established model thrives due to its simplicity; it only has two parameters: \( \lambda \) to control the rate of requests, and \( \alpha \) to control the skewness of the popularity.

Numerous studies fit the IRM to real traffic with satisfactory results [5], so why do we need to change it? The IRM assumes that the content popularity is static, which of course is not true. Trending tweets, breaking news, and the next episode of Game of Thrones are examples of ephemeral content with rapidly changing popularity; they appear, become increasingly popular, and they gradually become unpopular again. In fact, [6] considers large YouTube and video on demand (VoD) datasets and discovers that time-varying models are more accurate than the IRM with respect to caching performance analysis; Fig. 2 reproduces the comparison when fitting YouTube data and shows the superiority of modeling the popularity as time-varying. In the inhomogeneous Poisson model proposed in [6], each content is associated with a “pulse” the duration of which reflects the content life span and the height of which denotes its instantaneous popularity. The model is called the shot noise model (SNM), mirroring the Poisson noise from electronics. While the shape of the pulse is not important, the study observes strong correlations between popularity and duration; apparently, popular contents prosper longer. Finally, a class-based model [6] can conveniently capture spatio-temporal correlations while allowing analytical tractability. Mobile users are especially keen on downloading ephemeral content; thus, it is expected that in the case of wireless content, the improvement in modeling accuracy will be even greater.

To optimize a cache one needs to track the changes in content popularity. For example, the classical web caching systems adopt dynamic eviction policies like least recently used (LRU) in order to combat time-varying content popularity in a heuristic manner. However, the joint consideration of popularity variations with wireless systems reveals a new challenge that renders LRU policies inefficient. While a typical CDN cache normally receives 50 requests/content/day, the corresponding figure for base station cache may be as low as 0.1 requests/content/day. With such a small number of requests, fast variations of popularity become very difficult to track, and classical LRU schemes fail.

This development motivates novel caching techniques that employ learning methodologies to accurately track the evolution of content popularity over time. A recent study [7] analyzes the SNM model and gives the optimal policy for joint caching and popularity estimation. Additionally, [8] proposes as an alternative solution the use of LRU with prefilters.

HOW TO TRACK POPULARITY VARIATIONS

Since content popularity is time-varying, caching operations can only be optimized if a fresh view of the system is maintained. This requires massive data collection and processing, and statistical inference from this data, which by itself is a complex task to handle. Additionally, user privacy is a concern that can limit the potential of collecting such information. So, can we promptly gather all this information in a wireless network?

Consider \( K \) users subscribed to telecom operator and \( L \) caches placed in the network (e.g., in

*Figure 1. An illustration of caching in future wireless networks. Contents available in the origin server are cached at the base stations and user devices for offloading the backhaul and the wireless links.*
Hit probability comparison between best fit of the IRM, SNM, and Figure 2.

cache size (number of videos)

<table>
<thead>
<tr>
<th>Cache size (number of videos)</th>
<th>Hit probability</th>
</tr>
</thead>
<tbody>
<tr>
<td>$10^2$</td>
<td>0.05</td>
</tr>
<tr>
<td>$10^3$</td>
<td>0.05</td>
</tr>
<tr>
<td>$10^4$</td>
<td>0.05</td>
</tr>
<tr>
<td>$10^5$</td>
<td>0.05</td>
</tr>
</tbody>
</table>

instance, low-rank matrix factorization methods, unfortunately, popularity correlation induces such low spectral dimension, and the system can be continuously estimated in order to enable the correct cache decisions at the base stations. These representations are factor matrices, which can be employed to construct the r-rank version of the matrix, using the fact that users’ interests are correlated and predictable when r is small. This additionally allows the collected statistics to be stored in a more compact way. As a result, a big data platform installed in the operator network can provide efficient collection and processing of user access patterns from several locations, as evidenced in [9].

Further development of novel machine learning tools, such as clustering techniques, are needed to improve the estimation of the time-evolving content popularity matrix (i.e., $P(t)$ for base station $L$), which may differ from base station to base station.

It is worth noting that a caching system has requirements similar to those of a recommendation system. For example, the well-known Netflix movie recommendation system exploits information of a user’s past activity in order to predict which movie is likely to be scored high by the user. Similarly, a caching system exploits the request sequence to predict what contents are popular enough to be cached. In this context, user privacy regulations may affect the collection of these valuable data. A key topic of research in this direction is privacy-preserving mechanisms that can enable sufficient sampling of the time-evolving and location-dependent popularity matrix $P(t)$ without compromising user privacy.

**SECURITY IS A KIND OF DEATH**

A common anti-caching argument relates to the operation of caching in a secure environment. The secure counterpart of HTTP, called HTTPS, was originally used to provide end-to-end (e2e) encryption for securing sensitive information like online banking transactions and authentication. Due to the recent adoption from traffic giants Netflix and YouTube, the HTTPS protocol is growing to soon exceed 50 percent of total network traffic. Content encryption poses an unsurmountable obstacle to in-network operations, including caching. Since encrypting the data makes them unique and not reusable, caching, or even statistically processing encrypted content, is impossible. Ironically, Tennessee Williams’ statement “security is a kind of death” seems to squarely apply to wireless caching.

Security is definitely a precious good everyone welcomes. Although securing a video stream might seem an excessive measure, in some cases it may be well justified. Unfortunately, e2e encryption is clearly not in the Berners-Lee spirit since it prevents operators from optimizing their networks and reanimates the server-client ghost of congestion, a reality that equally no one can overlook. In fact, modern CDN systems resolve this issue by having “representatives” of the content provider at the edge of the Internet. These representatives are trusted entities that hold the user keys and are able to decrypt the requests and perform standard caching operations. Ultimately, this methodology is neither entirely secure for the user nor efficient for the network [10]. The need to make the system sustainable finally overrules the need for e2e encryption, which is an argument against HTTPS for video delivery. Given this situation, however, how can we realistically push caching deeper into the wireless access?

Currently, content providers install their own caching boxes in the operator network and intercept the related encrypted content requests deeper in the wireless access network. In this approach, the boxes are not controlled by the operator, which leads to several limitations:

- The caching boxes cannot perform complex tasks.
- It is difficult to apply learning techniques without context information from the operator.
- The caching approach is similar to CDNs and therefore does not exploit the performance opportunities specific to wireless caching, as we discuss below.

New security protocols have been proposed to enable operators to perform caching on encrypted requests [10]. This leads to an interesting research direction: to combine user security and privacy with facilitation of the network management operations, which are crucial for the sustainability of future wireless systems.

**TOWARD A UNIFIED NETWORK MEMORY**

The proposition of information-centric networking (ICN) as a candidate for the future Internet has also raised the subject of where to install net-
work memory [11]. The ICN approach proposes to equip routers with caches, and to allow content replication everywhere in the network. A recent work [12] came up with striking conclusions about the ICN approach: most of the caching benefits of ICN can be obtained by caching at the edges of the network using existing CDNs, and any extra caching in the core network brings only negligible improvements at very high costs. In the wireless domain, however, the question remains relevant: does it make sense to cache even closer to the user than CDN?

It is commonly believed that caching is very inefficient near the user, and thus should be done at CDN. Below we explain the main causes of inefficiency and argue that they can be overcome.

**Caching Deeper than CDN**

Mitigating backhaul and wireless link overload requires going beyond CDN and caching at the base stations and mobile users. However, the efficient operation of such caches is very challenging.

In particular, there are two main challenges:

a. Caches used in wireless networks are typically smaller compared to CDN caches.

b. The popularity profile of traffic is highly unpredictable when non-aggregated.

To understand point a, consider that the effectiveness of a cache is measured with the hit probability (i.e., the fraction of requests found in the cache). This can be upper bounded by the survival function of the sum of i.i.d. Bernoulli random variables, where is the ordered popularity distribution with \( p_j \) denoting the probability of requesting the most popular file.

For power-law popularity the sum can further be approximated by \((M/N)^{1-\alpha}\), where \( \alpha < 1 \) is the power-law exponent. A very small ratio \( M/N \) means that the hit probability becomes vanishingly small. For example, if we are caching Netflix (12.5 PB) in a mobile phone (10 GB), \( M/N \approx 10^{-4} \), \( \alpha \approx 0.8 \), and the hit probability is less than 10 percent. However, base stations equipped with a disk array (40 TB) can be extremely effective when caching contents for a mobile video on demand (VoD) application. In this context there are three promising research directions:

- Restrict caching operations to a subset of the catalog while maintaining network neutrality.
- Store only parts of the content using partial caching techniques.
- Install massive memory at the edge in the form of small-sized data centers.

The third option will be realized by the fog computing paradigm. Table 1 provides some indicative numbers for the memory types available and the catalog sizes of reasonable applications.

<table>
<thead>
<tr>
<th>Data Type</th>
<th>Catalog Size</th>
<th>Hit Probability</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disk</td>
<td>~0.1%</td>
<td>100%</td>
</tr>
<tr>
<td>Disk array</td>
<td>~0.5%</td>
<td>2%</td>
</tr>
<tr>
<td>Data center</td>
<td>50%</td>
<td>100%</td>
</tr>
</tbody>
</table>

Table 1. Typical data size values for normalized cache size \( M/N \) taken from the study of [8]. In practice, it is anticipated that wireless traffic is an 80–20 mix of torrent-like traffic and live VoD traffic tailored to wireless device capabilities.

To understand the unpredictable nature of sparse requests (formulated as challenge b above), consider as an example the delivery of breaking e-news in a city served by a single CDN node. Most users will download the news only once. The CDN system can quickly detect the rising popularity of the news, since it will receive many requests in a short timeframe. From the point of view of a mobile user, however, the detection of the popularity of the trending news becomes very difficult because the news is requested only once by a single user. This example shows that detection efficiency depends on the number of requests aggregated at the popularity learner. To illustrate this, Fig. 3 shows the optimal hit probability in a hierarchy of \( L \) base stations. Learning at the global CDN cache is shown to detect variations that are \( L \) times faster than those at local caches. To remedy the situation, it is possible to use an architecture that combines information obtained at different aggregation layers [7].

**Memory is Cheap but Not Free**

Although the cost of a small cache is dwarfed by the base station cost, the total amount of installed memory in a mobile network can be considerable; therefore, deciding to install wireless caching requires a careful cost analysis [8]. To compute the optimal size of memory to install at each location, one needs to know:

- The cost coefficients
- The skewness of content popularity
- The local traffic distribution in cells

Predicting how a and b will evolve is quite challenging, but as in [8] a survey may determine a good set of parameters at any given time.

For c, the literature is extensively based on grid models, which in the case of future wireless networks might be off for a significant factor. More accurate models have recently been introduced from the field of stochastic geometry, where the cache-enabled base stations are distributed according to a spatial point process (often chosen to be 2D Poisson), thus enabling the problem to be handled analytically. The validity of such modeling compared to regular cellular models has been verified using extensive simulations. Additional insights for the deployment of cache-enabled base stations can be obtained by analytically characterizing the performance metrics, such as the outage probability and average delivery rate, for a given set of parameters such as given number of base stations, storage size, skewness of the distribution, transmit power, and target signal-to-interference-plus-noise ratio (SINR) [13]. Therefore, although storage units become increasingly cheaper, the question of how much storage we should place at each location should be studied together with realistic topological models.

**Wireless ≠ Wired**

Web caching has traditionally been studied by the networking community. A common misconception says that caching is a network layer tech-
nique, and hence the web caching approaches are sufficient for wireless caching as well.

However, following the fundamental work of Maddah-Ali and Niesen [4], the idea of caching has penetrated the information theory community with a new twist called coded caching, which promises unprecedented gains. In the following, we discuss the differences between wired and wireless caching.

**WIRELESS CACHING LIENS AT BOTH THE NETWORK AND PHY LAYERS**

Suppose that a base station wants to deliver information to \( K \) users at a rate of 1 Mb/s each for streaming a video. If the video is the same for all users (broadcast video), this might be possible for an arbitrarily large number of users. For example, the base station could use an omnidirectional antenna, exploit the broadcast characteristic of the wireless medium, and transmit at 1 Mb/s to all users simultaneously. When the videos are different, this is clearly not possible: the base station needs to multiplex the users over frequency, time, or codes, where each such resource block is then associated with a single user. Since the resource blocks are finite, ultimately the base station can serve 1 Mb/s videos up to a maximum number of users \( K_{\text{max}} \); this can be verified by taking the limit of users \( K \) as the resource blocks does not increase with the number of users \( K \); this can be verified by taking the limit \( K \to \infty \) whereby the above quantity converges to a constant. The result is summarized in Fig. 4.

![Figure 3. Optimal hit probability comparison between observing the aggregate request process at the CDN level (global) and observing the individual request process at each base station cache (local), when refreshing the catalog. The hit probability performance depends on how fast the time-varying popularities can be learned: global is faster than local.](image)

**Figure 3.** Optimal hit probability comparison between observing the aggregate request process at the CDN level (global) and observing the individual request process at each base station cache (local), when refreshing the catalog. The hit probability performance depends on how fast the time-varying popularities can be learned: global is faster than local.

From the implementation point of view, promising research directions include extensions to capture system aspects such as:

- Popularity skewness
- Asynchronous requests
- Cache sizes that scale faster than \( \sqrt{N} \)
- Asynchronous requests
- Cache sizes that scale slower than \( N \)

Assuming that these practical challenges are resolved, caching for wireless systems will become intertwined with physical layer techniques employed at the base station and handheld devices.

**ONE CACHE ANALYSIS IS NOT SUFFICIENT**

A contemporary mobile receives the signals of more than 10 base stations simultaneously. In future densified cellular networks, the mobile will be connected to several femto-, pico-, or nanocells. The phenomenon of wireless multi-access opens a new horizon in caching exploitation [14].

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Footnote:

2 In fact, finding the optimal index code is a very difficult problem, and hence the proposed approach resorts to efficient heuristics.
Since a user can retrieve the requested content from many network endpoints, neighboring caches should cooperate and avoid storing the same objects multiple times.

Content placement optimizations of wireless caching typically boil down to a set cover problem in a bipartite graph connecting the users to the reachable caches. Therefore, finding what contents to store at each cache is a difficult problem even if the popularities are assumed known [14]. It is possible to relax the problem to convex optimization by the use of distributed storage codes, where each cache stores coded combinations of contents [14], or obtaining a fractional placement by time sharing different integral placements. These ideas lead to several interesting algorithms in the literature of cooperative caching.

What is the gain from these approaches? Cooperative caching typically saves space in the cache by avoiding caching the same popular contents in neighboring caches. Equivalently, we may think of multiplying the cache size $M$ by a small number, at best a gain of 3–5. With respect to hit probability, this can correspond to very different levels of gain, depending on the value of $M/N$. Due to the skewness of the popularity distribution, marginal hit probability gain is high when $M/N$ is small, and very small when $M/N$ is large. Since in wireless we expect the former, high gains are expected from cooperative wireless caching.

The current proposals on cooperative caching assume static popularity, and therefore a promising direction of research along these lines is to design caching schemes that combine cooperation with learning of the time-varying popularity. The time to search and retrieve the content from a nearby cache may also be significant; hence, intelligent hash-based filtering and routing schemes are required [15].

**A Stakeholder Analysis for Wireless Caching**

The business of wireless caching involves three key stakeholders that together form a complex ecosystem.

**The users** of telecommunication services are primarily the customers and consumers of the content, but in the case of wireless caching they are also active stakeholders. Users might be requested to help in the form of contributing with their own resource (e.g., in the case of coded caching it will be memory and processing, or in device-to-device, D2D, caching it will also be relaying transmissions), and they will end up spending energy for the benefit of better performance. On the other hand, one could envision users employing D2D technology to enable caching without the participation of other stakeholders. Due to the complexities mentioned above, however, efficient wireless caching will require heavy coordination and extensive monitoring/processing. Hence, D2D approaches will be limited to restricted environments.

**The operators** of telecommunication networks are well placed for wireless caching. Due to the particularities of coded caching and multi-access caching, operators are in a unique position to implement new protocols in base stations, affect the standards for new mobile devices, and develop big data processing infrastructure that can realize wireless caching. Nevertheless, for reasons related to encryption, privacy, and global popularity estimation, operators might not be able to install these technologies without the cooperation of the other two stakeholders.

**The providers** of Internet content are champions of trust from the user community. Apart from the security keys, they also hold extensive expertise in implementing caching techniques in core networks. From this advantageous position, they can positively affect the progressive evolution of caching in wireless networks. On the other hand, content-provider-only solutions cannot unleash the full potential of wireless caching, since they are limited to alienated boxes in the operator network that can perform caching only with legacy CDN techniques. The deeper the caches go into the wireless network, the less efficient they will be if they stick to legacy CDN techniques.

We summarize what each stakeholder offers and needs in Fig. 5. What are the prospects of the required collaboration among the stakeholders? Operators and content providers seek a “best friends forever” union in order to mutually harvest benefits in the digital value chain while keeping their users happy. This is a favorable environment for the scenario of wireless caching. In fact, if telecom operators enable caching capabilities at the edge of their networks, their infrastructure will become sustainable while they gain access to new business models. On the other hand, content providers can benefit from a caching collaboration since:

- Traffic will be intercepted earlier and the content transport cost will be reduced.
- User demand will not be held back by sustainability issues.
- Costs associated with the deployment of large memory units will be avoided.
- They will be able to reach closer to their users and extend computing infrastructures to the fog paradigm.

Lastly, it is foreseeable that in some situations
Figure 5. A stakeholder analysis.

the roles of the content provider and the wireless operator may converge.

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BIographies

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ABSTRACT

Caching is an essential technique to improve throughput and latency in a vast variety of applications. The core idea is to duplicate content in memories distributed across the network, which can then be exploited to deliver requested content with less congestion and delay. The traditional role of cache memories is to deliver the maximal amount of requested content locally rather than from a remote server. While this approach is optimal for single-cache systems, it has recently been shown to be significantly suboptimal for systems with multiple caches (i.e., cache networks). Instead, cache memories should be used to enable a coded multicasting gain. In this article, we survey these recent developments. We discuss both the fundamental performance limits of cache networks and the practical challenges that need to be overcome in real-life scenarios.

INTRODUCTION

Over the last decade, the composition of cellular traffic has shifted from being mainly voice to being mainly content. Unlike voice, which is generated in real time just ahead of transmission, content is typically generated well ahead of transmission. This pre-generation of content allows it to be cached throughout the network (either directly in mobile phones or in small cell base stations) during periods of low network utilization. During periods of high network utilization, this cached content can then be used to reduce network load. Such caching of content is expected to become a central component of future cellular network architectures [1].

Fundamental limits on the performance of systems with a single cache were developed in the computer science community mainly in the 1980s and ’90s. The main gain of caching in such single-cache systems derives from the local delivery of content from a nearby cache. However, this theory of single-cache systems does not apply to systems with multiple caches (i.e., cache networks). It is precisely such cache networks that are relevant in the context of next-generation cellular systems.

It is only very recently that a fundamental understanding of such cache networks has been obtained. This magazine article surveys these recent developments. We start by introducing a basic model for a cache network, consisting of a single server connected through a broadcast channel to a number of caches. An analysis of this network reveals that, in addition to the local delivery gain available in single-cache systems, cache networks allow a second, global, caching gain. This global caching gain is exploited through coded multicast transmissions from the server that are simultaneously useful for several users. Unlike the local gain, this global gain scales with the number of caches in the network. An information-theoretic argument shows that there are no other caching gains scaling with the system parameters. Thus, this global gain is a fundamental quantity for the basic cache network, and coded multicast transmission is a fundamental technique for this network.

These results demonstrate that coding plays a key role in the optimal operation of cache networks. However, their derivation relies on a number of stylized assumptions (e.g., simultaneous user demands, uniform content popularity, offline cache updating, and broadcast network topology). We discuss how these assumptions can be relaxed. We review a flexible decentralized coded caching approach that allows for caches to be populated independent of each other. Building on this decentralized coded caching scheme, we discuss how to handle scenarios with asynchronous user demands, nonuniform content popularity, and online cache updating. We also provide a brief overview of how these results extend to more general network topologies. Finally, we discuss a recent video streaming prototype using coded caching, demonstrating the applicability of these concepts in a real-life setting.

CODING FOR CACHING: FUNDAMENTAL LIMITS

To highlight the fundamental nature of coding for caching, we focus on a basic network model introduced in [2]. This network consists of a server connected through a shared bottleneck link to \( K \) users, as shown in Fig. 1. The server holds \( N \) files (e.g., movies) each of size \( F \) bits. Each user is equipped with an isolated cache of size \( MF \) bits for some real number \( M \in [0, N] \). For simplicity of exposition, we assume \( N \geq K \).

This setting can model a wireless network with a transmitter (e.g., a WiFi access point or a cellular base station) and several users (cell phones, laptops), all sharing the common wire-
less medium. The caching could take place directly in the end-user equipment. Alternatively, the caching could take place in femtocells close to the users. The basic cache network setting can also model a wired network with several caches connected to a common server. In this wired scenario the shared link models a bottleneck along the path between the server and the users. As we see later, the intuition developed for this basic cache network carries over to more complicated network topologies.

We assume that the system operates in two phases: a content placement phase and a content delivery phase. The placement phase is executed when traffic is low, and network resources are cheap and abundant (e.g., in the early morning). In this phase, each cache prefetches data from the central server subject to the memory limit of $MF$ bits. The key assumption is that the system is not aware of the users’ future demands, and therefore, the placement phase cannot be a function of those demands. The delivery phase occurs after the placement phase, when traffic is high and network resources are scarce and expensive (e.g., in the evening). At the beginning of this phase, each user reveals its request for one of the $K$ requested files. In response, the server sends $RF$ bits (or the equivalent of $R$ files) over the shared link. The number $R$ is called the rate of the server transmission. From the server transmission and its local cache content, each user needs to be able to recover its requested file.

We can design both the content placement and delivery phases. The objective is to minimize the rate $R$ with which every possible set of user demands can be satisfied. The constraints are the memory limit during content placement and the recovery requirement during content delivery.

As a baseline for future comparison, we start by reviewing a conventional uncoded caching scheme.

**Example 1 (uncoded caching).** In this solution, each user caches the first $M/N$ fraction of the bits of each file during the placement phase. Since there are a total of $N$ files, this respects the memory constraint at each user. The motivation for this content placement is that the system should be ready for any possible set of demands. Thus, each user should dedicate the same fraction of its memory to each file.

During the delivery phase, each user requests one of the $N$ files. Recall that each user already has a fraction $M/N$ of its requested file stored in its local cache. Thus, the server only needs to transmit the remaining $1 - M/N$ fraction of the requested file for the user to be able to decode. Since there are $K$ users in the system, the rate of this scheme adds up to

$$R_U(M) = K \cdot (1 - M/N). \tag{1}$$

For a specific example, consider the case with $K = N = 2$ users and files, and with memory size $M = 1$. Then $R_U(M) = 1$.

The rate-memory trade-off $R_U(M)$ for this caching scheme has two terms. The first term, $K$, is the rate of the network in the absence of caches. The second factor, $1 - M/N$, appears because an $M/N$ fraction of the requested file is available locally. We call this second factor in Eq. 1 the local caching gain.

Since neither the placement phase nor the delivery phase of this scheme uses coding, we refer to it as uncoded caching. In this uncoded scheme, the role of caching is to deliver part of the requested files locally.

We have selected the uncoded caching scheme in Example 1 for simplicity of exposition. There is a long list of other uncoded caching approaches developed for different applications and scenarios. Perhaps the best known of these are least recently used (LRU) and least frequently used (LFU).

All of these uncoded caching schemes share the following basic properties:

- The main role of caching is to deliver part of the content locally. As a consequence, maximizing the hit rate (i.e., the chance of local delivery) is often chosen as the objective.
- For isolated private caches, each user can only derive a caching gain from its own cache.
- Caches that observe statistically similar demands should cache similar content.

These common sense principles lead to efficient solutions for systems with a single cache. However, as we see next, these principles do not carry over to networks with multiple caches. Indeed, we show that local delivery achieves only a small fraction of the gain that cache networks can offer. Cache memories, if populated correctly, can offer additional gains that scale with the size of the network. We explain the main idea with a toy example from [2].

**Example 2 (coding caching, $K = N = 2$, $M = 1$).** Consider a system with $K = 2$ users, each with a cache large enough to store one file so that $M = 1$. Assume that the server has $N = 2$ files, $A$ and $B$. We split each file into two non-overlapping subfiles of equal size, $A = (A_1, A_2)$ and $B = (B_1, B_2)$. In the placement phase, instead of placing the same content in all caches, we place different content pieces at the users’ caches as shown in Fig. 2. In particular, user one caches $(A_1, B_1)$ and user two caches $(A_2, B_2)$.

For the delivery phase, let us consider a generic case in which user one requests file $A$ and
user two requests file $B$ as depicted in Fig. 2a. From the placement phase, user one already has subfile $A_1$ but is missing $A_2$, and user two already has subfile $B_2$ but is missing $B_1$. Thus, similar to the uncoded approach, half of the requested file can be delivered locally. The server could send the missing parts $A_2$ and $B_1$ with total rate of 1. However, as we shall see, the particular pattern of content placement creates an opportunity for reducing the transmission rate through coding.

We note that $A_2$ required at user one is available in cache two. Similarly, $B_1$ required at user two is available in cache one. Unfortunately, the two users are unable to exchange these file parts since their caches are isolated. Instead, the server can exploit this situation by transmitting $A_2 \oplus B_1$ over the shared link, where $\oplus$ denotes bitwise XOR. User one can recover $A_2$ from the received signal $A_2 \oplus B_1$ and its cache content $A_1$. Similarly, user two can recover $B_1$ from the received signal $A_2 \oplus B_1$ and its cache content $B_2$. This is shown in Fig. 2a. Since the coded signal $A_2 \oplus B_1$ is simultaneously useful for both users, the load of the shared link is reduced by a factor of 2 compared to the uncoded approach. Note that this multicasting opportunity is created for the two users even though they ask for different files.

As shown in Fig. 2, although the content placement does not depend on the future requested files, the above coded multicasting opportunity is available for all four possible demand tuples. The required rate is 0.5 in each case. Thus, the rate of the coded scheme in this example is $R_C(M) = 0.5$. This compares to the rate $R_U(M) = 1$ for the uncoded caching scheme from Example 1 with the same parameters.

Example 2 shows that the role of caching is not limited to local delivery. In particular, careful placement of content not only allows local delivery of part of the content but in addition creates multicasting opportunities through coding. In Example 2, this coded multicasting opportunity reduces the load of the shared bottleneck link by an additional factor of 2.

The rate-memory trade-off for coded caching for arbitrary number of files $N$, number of users $K$, and local cache size $M$ has been derived in [2]. For ease of exposition we state the result here for the case $K \leq N$. It is shown that for local cache size of $M \in \{0, N/K, 2N/K, \ldots, N\}$, coded caching achieves the rate

$$R_C(M) \leq K \cdot (1 - M/N) \frac{1}{1 + KM/N}, \quad (2)$$

For general $0 \leq M \leq N$, the lower convex envelope of these points is achievable.

Comparing the rate expression $R_C(M)$ in Eq. 2 for the coded scheme with the rate expression $R_U(M)$ in Eq. 1 for the uncoded scheme, we see that the first two factors are the same. This means that both caching schemes enjoy the gain of local delivery quantified by the factor $1 - M/N$. However, the coded scheme alone enjoys an additional gain of a factor

$$\frac{1}{1 + KM/N}.$$

This additional factor quantifies the gain of creating and exploiting coded multicasting opportunities.

\[ \text{Figure 2. (Centralized) coded caching strategy for } K = 2 \text{ users, } N = 2 \text{ files, and cache size } M = 1. \text{ Each file is split into two subfiles of size } 1/2 \text{ (e.g., } A = (A_1, A_2)), \text{ user 1 caches } (A_1, B_1) \text{ and user 2 caches } (A_2, B_2). \text{ The delivery phase uses coding to satisfy two user demands with a single transmission. This coded-multicasting opportunity is available simultaneously for all four possible demand tuples shown in the four subfigures: a) demand } (A, B); \text{ b) demand } (B, B); \text{ c) demand } (A, A); \text{ d) demand } (B, A). \]

This gain is enabled by deliberately placing content in order to allow the server to satisfy $1 + KM/N$ (partial) user demands with a single coded multicast transmission. Thus, these users enjoy the gain of multicasting even though they ask for different demands. This coded multicasting opportunity is available simultaneously for every one of the $N^K$ possible user demand vectors.

\textbf{Coding gain is global}

Unlike the local caching gain, which derives from the local cache at a single user, the coding gain inherently derives from several caches at several users. Motivated by this observation, we refer to the coding gain as the \textit{global} caching gain.

Observe that the local caching gain

$$1 - M/N = \frac{1}{1 + M/N}$$

is significant when the local cache size $M$ is comparable to the size of the entire content $N$. In contrast, the global caching gain

$$\frac{1}{1 + KM/N}$$

is significant when the cumulative cache size $KM$ is comparable to the size of the entire content $N$. Thus, by employing coding, we make approximately $K$ times better use of the available memory resources.

Coding also results in better use of the com-
Coding gain scales with network size

Unlike the gain of local delivery, the global caching gain due to coding scales with the number of caches in the network.

Figure 3 shows the rate-memory trade-off for $K = 20$ users for both the uncoded and coded schemes. To compare the two schemes, let us focus on the case where each user has space to cache half of the content (i.e., $M/N = 0.5$). In this case uncoded caching reduces the load of the shared link from 20 files down to the equivalent of 10 files. In contrast, for the (centralized) coded caching scheme, the load of the bottleneck link is less than the equivalent of just a single file.

It can be shown that the rate-memory trade-off $R_C(M)$ achieved by coded caching is within a constant factor of the information-theoretic optimum for all values of the problem parameters [2]. This implies that the local and global caching gains quantified above are fundamental: there are no other gains that scale with the system parameters.


Coding for Caching: Practical Challenges

The analysis of the basic cache network indicates that coding plays a crucial role in the operation of cache networks and in attaining their fundamental performance limits. To apply coded caching in real systems, several practical challenges need to be addressed. These include handling asynchronous demands, dealing with nonuniform content distributions, updating cache contents online, delivering delay-sensitive content such as video streaming, and dealing with more general network topologies, among others. These issues are currently subject to extensive research activity in both academia and industry. Here we present some of the solutions.

A key tool to deal with several of these challenges is a more flexible content placement scheme developed in [3]. This scheme uses decentralized content placement, in which the placement phase is independent of the number and identity of users. We explain the main idea using a toy example from [3]. We then outline how this decentralized scheme can help address some of the practical challenges raised above.

Example 3 (decentralized coded caching, $K = N = 3$). Consider the caching problem with $N = 3$ files, say $A, B, C$, and with $K = 3$ users, each with a cache of size $M$ bits. In the placement phase, each user caches $M/N$ bits of each file uniformly at random, independently of other caches. Clearly this placement satisfies the memory constraints. The most important feature of this placement scheme is that each cache is populated independent of the number and identity of the other users in the system.

In the delivery phase, consider a generic request tuple with users one, two, and three requesting files $A, B,$ and $C$, respectively. Let us partition file $A$ into eight subfiles,

$$A = (A_{12}, A_1, A_2, A_3, A_{12}, A_{13}, A_{23}, A_{123}),$$

where $A_i$ denotes the bits of file $A$ that are stored exclusively at users in the set $S_i$ and similarly for $B$ and $C$. Based on this partitioning, the content of each cache can be expressed as shown in Fig. 4. For delivery, the server uses greedy linear coding as follows. It first sends the coded packet $A_{12} \oplus B_{13} \oplus C_{12}$, which is simultaneously useful to all three users (as is easily verified by examining the cache contents). It then sends coded packets that are useful for two users, $A_2 \oplus B_1$ for users one and two, $A_3 \oplus C_1$ for users one and three, and $B_3 \oplus C_3$ for users two and three. Finally, the server sends $A_{g_5}, B_{g_5},$ and $C_{g_5}$, each useful to only one user.

Even though the placement phase is independent of the number of users $K$, the rate of this decentralized coded caching scheme can be shown to be within a constant factor of universally optimal for any number of users $K$ [3]. The rate of the decentralized coded caching scheme is also very close to the rate of the centralized coded caching scheme, as can be seen in Fig. 3.

The universality property of the decentral-
maintain efficiency, the cache contents should be adapted accordingly. This is usually done by employing online caching schemes, in which the cache contents are modified at the end of the delivery phase. However, the popular least recently used (LRU) cache update rule is approximately optimal for single-cache systems, it can be significantly suboptimal for cache networks [9]. In an alternative approach proposed in [9], the server has two delivery modes. It uses coded delivery for the requested files that are already partially cached. For the files that are not partially cached, the server uses uncoded delivery. Each cache makes use of such uncoded delivery by exciting some old file parts and replacing them with randomly chosen parts of the newly delivered file. Since the number of requests from the cached files are random, the universality of decentralized coded caching is again used here.

The ideas developed for the basic cache network topology can be extended to other networks such as tree networks [3], hierarchical cache networks [10], device-to-device communication [11], cache-aided interference channels [12, 13], and channels with erasures [14] (Fig. 5).

To demonstrate the gains of caching in a practical setting, [15] describes a prototype implementation applying coded caching to the problem of video streaming. A screenshot of this prototype is shown in Fig. 6a. In addition to the techniques described so far, several other issues arise in this context, such as protocol overhead, state synchronization, delay constraints, and computational limitations, as discussed next.

Protocol Overhead: Header information is required for the users to know which file bits are coded together in each received packet. In order for this protocol overhead to be manageable, each file is split into chunks of size 10 kB. For example, video3.flv[3689] in Fig. 6a refers to chunk 3689 of file video3.flv. The decentralized content placement and delivery schemes described earlier are then applied at the chunk level rather than at the bit level. For example, line 2 in Fig. 6a shows the decoding operation from chunk video3.flv[3691] XORed with chunk video2.flv[5452].

State Synchronization: In order to form the
correct coded transmissions, the server needs to know the state of the user caches. A convenient way to achieve this state synchronization is to use a pseudo-random number generator for the decentralized content placement. If a user with already populated cache connects to a server, it can simply communicate the seed number of the pseudo-random number generator used for the content placement to the server. From this seed, the server can then recompute the entire state of the user cache.

**Delay Constraints:** Video streaming applications have tight delay constraints, typically on the order of 30 s. To meet these constraints, instead of requesting entire files, users instead request individual file chunks. At the server, the requested chunks are kept in an ordered queue according to their delivery deadline. Coding opportunities are only exploited among the chunks in the queue. Whenever the deadline is up for a particular chunk, it gets coded with other chunks in the queue and delivered to the users.

**Computational Limitations:** Finding the optimal coding opportunities among the chunks in the queue is computationally intractable. Instead, a greedy coding scheme is used, which has complexity that is only linear in the queue length. A trace of the coding gain (defined as the ratio of the rate of uncoded and the rate of coded caching) corresponding to Fig. 6a is shown in Fig. 6b. In this trace, the number of users is increased from one to four with an interval of 30 s. Each time a new user joins, the coding gain increases as the system is able to exploit coding opportunities among a larger number of users. Also shown in dotted lines are the theoretical values of the coding gain. The figure indicates that the additional protocol overhead and other losses due to delay constraints and computational limitations are modest in this scenario.

**Open Problems**

The results presented so far demonstrate the promise of coded caching. There are many open problems left to be investigated.

**Sharpening of Approximations:** Characterizing the exact optimal rate-memory trade-off or at least sharpening the approximations is of interest. We have seen that uncoded content placement combined with linearly coded content delivery is sufficient for achieving a rate within a constant factor of optimal for the canonical broadcast cache network. One important open question is if nonlinear codes are needed to exactly achieve the optimal rate-memory trade-off. Another open question concerns the value of coded content placement. In [2], it is shown that placing coded content in the user caches can improve the rate for small cache sizes. However, it is still unknown if coded content placement is required to exactly achieve the rate-memory trade-off for large cache sizes. Conversely, it is known that the current bounds are not tight in general and need to be improved.

**General Cache Networks:** Characterizing (either exactly or approximately) the rate-memory trade-off for general cache networks is of great interest, but expected to be difficult. The reason is that the related multiple-unicast problem over general networks is known to be challenging; for example, nonlinear coding can offer unbounded performance improvement, and Shannon inequalities are insufficient to prove optimality. Alternatively, one may focus on specific topologies that are relevant in practical settings (e.g., layered networks) or restrict the class of solutions (e.g., only allowing linear codes or only allowing coding at some of the network nodes).

**Impact of Other Constraints:** As we have seen in the context of the video streaming prototype, in many applications we may encounter additional constraints. In particular, we have seen constraints on the protocol overhead, state synchronization overhead, delivery delay, and computational resources. Other such constraints are on the number of sub-packets per file and the number of disk reads at either the server or the caches. Here we have presented heuristic solutions to handle some of these constraints; deriving fundamental limits of caching under these additional constraints is an important, largely open, question.

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Figure 5. Other cache network topologies: a) hierarchical cache networks [10]; b) device-to-device [11]; c) cache-aided interference channels [12, 13].
Cache-Aided Wireless Communication: For the wireless broadcast channel, we have seen that caching can offer a coded multicasting gain. For other wireless scenarios, caching can offer additional gains. For the interference channel, caching at the transmitters offers interference cancellation and interference alignment gains [12]. When caches are also present at the receivers, we can in addition exploit the coded multicasting gain seen earlier [13]. Cataloguing what caching gains are available in other wireless scenarios is of interest. Characterizing the capacity region (or at least the degree of freedom region) of cache-aided multi-user wireless channels is also an important open question. System-level evaluations (either analytical or based on simulations) taking into account the backhaul load, queue stability, and control signal overhead could provide a better understanding of the performance of cache-aided wireless networks in practice.

Conclusions

In this article, we have argued that coding plays a crucial role in the efficient operation of cache networks. In particular, coding can offer an improvement of network load that scales with the network size. We have also demonstrated that this gain is achievable in the presence of practical system constraints and for different network topologies. Finally, we have presented numerous open problems, both theoretically and practically motivated, that need to be addressed.

References


Biographies

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The authors propose a PHY-caching scheme for 5G wireless networks to achieve spectral efficiency gain over conventional caching schemes, which do not induce physical layer cooperation. By properly caching some popular contents at the BSs, the proposed PHY-caching can opportunistically transform the topology of the RAN from an unfavorable topology (e.g., relay or interference topology) into a more favorable MIMO broadcast topology and enjoy spectral efficiency gain. Specifically, we first introduce a generic cache model and cache-assisted PHY transmission, and show that PHY-caching can significantly enhance the spectral efficiency of the wireless network by inducing dynamic side information at the BSs. Then we discuss the design challenges and solutions of PHY-caching. We introduce maximum-distance-separable-coded caching and online cache content placement design as a potential solution. As a case study, we analyze the performance trade-off between PHY-caching at the BS and CN-caching at the gateway under capacity-limited CN-fronthaul. We show that even though PHY-caching covers fewer users than CN-caching and results in a lower cache hit rate, it is still efficient to do PHY-caching at the BS.

INTRODUCTION

One of the key technologies to meet the 1000× increase in capacity demand in the fifth generation (5G) is small cell dense wireless networks, where the base station (BS) density is much larger than that of current wireless networks. Such a dense BS deployment essentially brings the network closer to mobile users and thus can significantly improve the spectral efficiency per unit area, as illustrated in Fig. 1. While there are huge potential opportunities associated with dense wireless networks, there are also two key technical challenges: the fronthaul/backhaul issue and the interference issue. In regard to the first issue, greater BS density, for instance, means that more high-speed payload fronthaul/backhaul is required per unit area, which will significantly increase the fronthaul/backhaul cost. In regard to the second issue, as the BS density increases, interference significantly limits the spectral efficiency gain in dense wireless networks. To overcome the interference issue, coordinated multipoint (CoMP) transmission has been proposed as one of the most important core technologies for Long Term Evolution (LTE+) and future 5G wireless networks. However, a conventional CoMP scheme is quite costly because it requires high-capacity core network (CN)-fronthaul (i.e., a wired connection between the CN gateway and BS) for payload exchange between BSs, and this poses a huge challenge for practical applications of CoMP, especially for small cell dense wireless networks.

Recently, wireless caching has been proposed as a cost-effective solution to partially address the above technical challenges. In [1–3], by caching contents at the BS, the packets requested by users can be obtained directly from the serving BS, and hence can alleviate the fronthaul loading issues in the wireless network. The fundamental limits of caching are studied in [4], which shows that it is possible to achieve a global caching gain that provides an order-wise improvement over the local caching gain by caching coded contents at the receivers. However, in all these works, the cache state (hit or miss) does not affect the underlying physical layer topology of the wireless network, and hence the role of caching in these works does not contribute to resolving the aforementioned interference issues.

In this article, we introduce a new wireless caching technique called physical layer (PHY)-caching [5, 6], which can exploit BS-level caching to mitigate interference and improve the degrees of freedom (DoF) in the radio interface of wireless networks. In PHY-caching, if the content accessed by several users exists simultaneously at nearby BSs/relay stations (RSs), it can indirectly induce dynamic side information at the BSs/RSs, and hence they can engage in CoMP to serve these users and enjoy CoMP (spatial multiplexing) gain without consuming CN-fronthaul bandwidth. In this way, we can opportunistically transform the interference/relay topology into a more favorable multiple-input multiple-output (MIMO) broadcast channel topology. As such, not only can PHY-caching reduce the CN-fronthaul cost, but it also has an interference mitigation benefit and spectral efficiency gains due to cooperative spatial multiplexing. The contributions of this article are summarized as follows.

General model for cached wireless networks: We introduce a generic cache model and a cache-assisted PHY transmission to embrace all...
possible caching strategies and PHY transmission strategies for exploiting the dynamic side information induced by the cache hits. Based on that, we can obtain a performance upper bound for cached wireless networks.

**Implementation challenges and solutions:** We elaborate the implementation challenges associated with realizing the advantages of PHY-caching, and introduce maximum distance separable (MDS)-based caching and online cache content placement design as a potential solution.

Case study on interplay trade-off between PHY-caching and CN-caching: We consider a case study to illustrate the benefits of PHY-caching, and the trade-off between CN-cache and PHY-cache in a small cell network. We show that the MDS-based caching scheme can achieve the optimal trade-off for some interesting special cases, and that the gap between the achievable trade-off region of MDS-based PHY-caching and the outer bound of the trade-off region is small in many operating regimes.

**GENERAL THEORY OF PHY-CACHING IN WIRELESS NETWORKS**

PHY-caching is a caching strategy to induce dynamic side information at the BS to promote MIMO cooperation between peer transmitters in wireless networks [5, 6]. In this section, we first introduce a generic cache model and a cache-assisted PHY transmission, and show that PHY-caching can enhance the spectral efficiency of the wireless network over traditional wireless networks without a cache. Consider a general cached wireless network consisting of $M$ BSs with $N_T$ antennas and $K$ users with $N_C$ antennas, as illustrated in Fig. 1. Each BS is equipped with a cache with $B_C$ bits. There are $L$ content files on the content server indexed by $l \in \{1, ..., L\}$. The size of the $l$th content file is $F_l$ bits. The index of the content file requested by the $K$th user is denoted by $\pi$. Define $\pi = [\pi_1, \pi_2, ..., \pi_K]$ as the user request vector (URV), which changes on a timescale much slower than the time slot.

**GENERIC CACHE MODEL**

In a cached wireless network, each BS has a cache that stores some of the content packets from the server. We assume the cache capacity is not large enough to store the entire content library. Collectively, the caches of all the BSs can be modeled by a generic black box model, as illustrated in Fig. 2. The input of the cache at the $M$th BS is the collective requests from the $K$ users given by URV $\pi$ at the beginning of a communication session. The output of the cache is the cache state vector $w = [w_{1,1}, ..., w_{M,K}] \in \{0, 1\}^{MK}$, where $w_{m,k} \in \{0, 1\}$ indicates whether the content requested by user $k$ is in the cache at the $m$th BS. The generic cache model has $B_C$ and $q$ as the model parameters, where $q = [q_{1,1}, ..., q_{1,M} ... q_{L,M}]$ is the cache placement vector, and $q_{l,m} \in [0, 1]$ denotes the portion of content $l$ stored in the cache of BS $n$ during the initialization. Hence, the total amount of information from the $l$th file stored in the cache of the $M$th BS is $q_{l,M}F_l$. Summing over all $L$ files, the total amount of information bits stored in the cache of the $m$th BS is $\sum_{l=1}^{L}q_{l,m}F_l$, which must be less than the total storage capacity of the cache $B_C$.

Note that there are a total of $N_C = 2^{MK}$ possible cache states, and let $w^{(i)}$ denote the $i$th cache state vector. The caches accept a user request vector $\pi$ at the beginning of a communication session, and the associated cache state at each time slot is given by the $i$th cache state $w^{(i)}$ with probability $p^{(i)}$. Denote $p^* = [p_1, ..., p_{2^{MK}}]$ as the cache state probability vector under URV $\pi$. For a given URV $\pi$, the cache state probability vector $p^*$ is constrained by the amount of information bits stored in the cache. Specifically, the amount of information bits transmitted from the $m$th BS cache to the $k$th user cannot exceed the total amount of information bits stored in the $m$th BS cache. Note that the cache model does not specify how the contents at the BSs store the content, and therefore, it embraces different possible caching schemes, including possibly random caching.

**CACHE-ASSISTED PHY TRANSMISSION MODEL**

The availability of side information at the BSs is determined by the cache state vector $w$. As a result, the PHY transmission mode (or the PHY topology) depends on the dynamic cache state, as illustrated in Fig. 3. The cache state vector $w$ in the cache induces some side information in the BSs, with which the BS in the wireless network can exploit and engage in partial cooperation...
When the caches of both BSs have full side information of the content requested by the users, they can serve the users cooperatively, delivering four streams of data to the users. On the other hand, when each BS only has the data requested by one of the users, the system can only deliver two streams of data to the users.

[7], resulting in an enhanced DoF in the radio interface. Specifically, the spectral efficiency gain induced by PHY-caching depends on the cache induced side information and the channel state information available at the transmitter (CSIT). When there is CSIT shared between transmitters, cooperative spatial multiplexing between concerned users becomes possible when the transmitters share side information of payload induced by PHY-caching. On the other hand, when CSIT is not available, we can focus on improving the reliability of wireless communications measured by the spatial diversity gain. For example, consider the cached interference channel in Fig. 3, with $M = 2$ BSs and $K = 2$ users. Each BS and each user has two antennas, and the global CSIT is available at the BS. When the caches of both BSs have full side information of the content requested by the users, they can serve the users cooperatively, delivering four streams of data to the users. On the other hand, when each BS only has the data requested by one of the users, the system can only deliver two streams of data to the users.

**ImplementAtion consIderAtIons And chAllenges**

Since the role of the PHY-cache is to induce side information to promote MIMO cooperation opportunity among peer transmitters, the design of PHY-caching is quite different from that of traditional caching in fixed line networks or content delivery networks (CDNs), which does not induce physical layer cooperation. We shall elaborate these design challenges below.

**ExponentIally smAll MIMO**

**COOPERATION PROBABILITY WITH ASYNCHRONOUS ACCESS**

The overall performance gain of the cached wireless network depends heavily on the probability of the cooperative MIMO transmission mode opportunity $p_c$. To engage in CoMP, the peer BSs need to have the payload packets requested by all the concerned users. Since user requests are asynchronous and packets need to be delivered to the users in sequence, it is very difficult to align the transmission timing of different requested packets from the peer BSs. As a result, using naive random caching, $p_c$ will be exponentially small even if a significant portion of the content files are stored at the BS cache.

**Example 1 (naive random caching with asynchronous access):** Consider a cached interference channel with $M = 2$ BSs, $K = 2$ users, and $L = 2$ content files with equal size $F$. Each file consists of 12 packets, and suppose every node stores the same half of each file (packets 1–3 and 7–9) without coding. User 1 requests file 1 at time slot $t_1$, and user 2 requests file 2 at time slot $t_2$. Without packets, the BSs have to be sent sequentially to the users from the first packet to the last packet, as illustrated in Fig. 4a. Note that the starting times of the transmission of file 1 and file 2 are random (since user requests are usually asynchronous with each other). As a result, the cooperative transmission probability is 0.5$^2$ (averaged over all random request timings $t_1$ and $t_2$).

This will be the case no matter how we select the packets to be cached at the BSs. Hence, the MIMO cooperation probability decreases exponentially w.r.t. the number of users. This problem may be solved if we add a unique sequence number for each information packet and schedule the transmission order to align the transmission timing of the cached packets. However, this will impose a large segmentation and reassembly overhead and delay at the user.

Note that even if the user requests happen to be synchronized, the cached bits still cannot always be aligned in time due to the different instantaneous transmission rates caused by the wireless fading channels, and hence the transmission timing of each packet will be different. Thus, the problem cannot be resolved by merely synchronizing the user requests. One solution to address this problem is to use MDS-based PHY-caching [5, 6]. Specifically, each file is divided into fixed segments and encoded with MDS-coding, as illustrated in Fig. 4. Every BS stores the same MDS-coded packets (parity bits) of each segment in the PHY-cache for MIMO cooperation. With MDS-based PHY-caching, packets within a segment do not need to be delivered in sequence, and there is no need for packet reassembly at the user. More importantly, this allows the BSs to schedule and align the transmission timing of the cached parity bits for different users (within a segment) so as to improve the probability of MIMO cooperation. This is illustrated in the following example.

**Example 2 (MDS-based PHY-caching with asynchronous access):** We use the same setup as in Example 1. Each file is divided into two segments, and each segment consists of six packets. Suppose the BS-cache stores the first three
PHY-CACHING INDUCED PARTIAL MIMO COOPERATION

The cache state in the PHY-cache induces some side information in the BSs with which the BSs can exploit and engage in partial cooperation [7], resulting in an enhanced DoF in the radio interface. However, there are exponentially large numbers of cache states and associated partial cooperation patterns (transmission modes), and thus a complicated MIMO precoding and transmission design is required to mitigate the interference under various side information patterns. To simplify the design of the PHY, we consider a binary PHY transmission mode, the cooperative MIMO mode (when $\gamma_{mn} = 1$), and coordinated MIMO mode (otherwise) [5, 6]. There are some works studying partial cooperation in cached wireless networks. For example, in [8], the authors propose a cache-added transmission scheme for a three-user interference channel, and show that DoF gain can be achieved by caching different parts of the same content file at the transmitters. However, this work only provides an achievable scheme for a three-BS-user-pair single-antenna interference channel. The caching and transmission scheme, and the upper bound of general cached MIMO wireless networks are still unknown.

CACHE CONTENT PLACEMENT

We assume the cache storage capacity at the BS is not large enough to store the entire content library. As a result, a critical component in the PHY-caching design is a cache content placement algorithm that determines the dynamic priority of caching one file over the others. The design of the cache content placement algorithm is a complex large-scale stochastic optimization problem with intrinsic coupling between the dynamic physical layer topology and the cache placement vector [5, 9]. Furthermore, the solution should be adaptive to the content access popularity as well as the channel statistics. However, in practice, such information is not easy to obtain and is also non-stationary over time. It is desirable but challenging to have a self-adaptive and self-learning algorithm without explicit knowledge of these statistics. To address the cache content placement issue, we can first exploit the timescale separations of optimization variables and stochastic decomposition techniques to decompose the two-timescale stochastic optimization problem into a short-term precoding problem and a long-term stochastic cache placement problem. Then stochastic learning algorithms such as stochastic subgradient or stochastic cutting plane methods can be used to solve the cache content placement problem without explicit knowledge of the popularity of the content files.

**Case Study: PHY-Caching vs. CN-Caching**

In this section, we consider the cellular network described earlier, where we may cache at the CN gateway or at the BS. We consider the situation where the capacity of each CN-fronthaul connecting to a BS is only sufficient to deliver contents requested by the users associated with this BS, and is not high enough to deliver the other payloads for MIMO cooperation. The role of CN-caching at the content gateway is to offload the CN-backhaul. On the other hand, the role of the proposed PHY-caching at the BS is to create MIMO cooperation opportunity and provide DoF gain without requiring high-speed CN-fronthaul. Under the above CN-fronthaul capacity constraint, there will be a trade-off between CN-backhaul offloading gain and DoF gain. Specifically, for a fixed total storage budget $C$, if we allocate 100 percent storage to CN-caching, we can achieve maximum CN-backhaul offloading gain at the expense of no improvement in the spectral efficiency of the radio interface. In contrast, if we allocate 100 percent storage to the PHY-cache at the BS, we can achieve maximum spectral efficiency gain at the radio interface due to CoMP gain at the expense of CN-backhaul offloading gain. In this section, we study the trade-off between CN-caching and PHY-caching, and the associated cache capacity allocation problem between them.

**System Setup**

Consider the cached cellular network described earlier with global CSIT available at each BS. The content files have an equal size of $F$ bits. The content file popularity distribution follows the Zipf distribution with non-negative parameter $\gamma$. Specifically, the probability that a user requests the $l$th file is given by $l^{-\gamma} \sum_{i=1}^{F} l^{-\gamma}$. A larger value of $\gamma$ means that the user requests are concentrated on a few popular files. Zipf popularity is widely used to model Internet traffic [10]. For practical systems, the typical range of $\gamma$ is 0.5 to 3 depending on the applications, and larger values of $\gamma$ are usually observed in mobile applications [10].

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1. However, CN-caching cannot replace the role of the high-speed CN-fronthaul required to support CoMP.
2. This is because the capacity of the CN-fronthaul is not big enough to deliver all payloads to all BSs, and hence the BS cannot engage in CoMP transmission, regardless of the cache state (hit or miss) at the CN-cache.
The trade-off between CN-backhaul offloading gain and spectral efficiency gain is realized by the cache capacity allocation between the PHY-cache and CN-cache in the network. We consider a high signal-to-noise ratio (SNR) asymptotic analysis, where the spectral efficiency of the RAN and the CN-backhaul offloading are both characterized in terms of DoF, which refers to the pre-log factor of the rate expression normalized with the log SNR for high SNR. The trade-off behavior between CN-caching and PHY-caching can be characterized by the ratio between the CN-caching efficiency (CN-backhaul offloading gain per unit cache size) and PHY-caching efficiency (DoF gain per unit cache size). In general, there are three types of trade-off behaviors depending on the system parameters:

- **Type I**: The CN-caching efficiency is much larger than the PHY-caching efficiency, such that the trade-off curve is almost a vertical line (the x-axis is the DoF gain, and the y-axis is the CN-backhaul offloading gain). It is more beneficial to allocate all storage to the CN-cache.
- **Type II**: The CN-caching efficiency is on the same order as the PHY-caching efficiency, such that the trade-off curve has a slope of constant order. Both PHY-caching and CN-caching are effective. The storage is allocated depending on the specific network requirements.
- **Type III**: The CN-caching efficiency is much smaller than the PHY-caching efficiency, such that the trade-off curve is almost a horizontal line. It is more beneficial to allocate all storage to the PHY-cache.

Since the CN-caching can cover all the $M$ BSs, one may expect the CN-caching to be much more efficient than the PHY-caching and the trade-off curve to belong to Type I as the number of
BSs $M$ becomes large. Surprisingly, the trade-off analysis in the next section shows that, in most cases, the result is the opposite.

**Trade-off Analysis and Discussion**

Using the generic cache model and cache-assisted PHY transmission model described above, the DoF outer bound in the radio interface of the cached wireless network for given PHY-cache state $\mathbf{w}$ can be obtained using the DoF bounds for general interference networks given in [11], and the PHY-cache state probability vector $p^{\mathbf{w}}$ is constrained by the amount of information bits stored in the cache. With the outer bound on the DoF, PHY-cache state probability, and CN-cache hit probability, we can obtain an outer bound of the trade-off region between the CN-backhaul offloading gain and DoF gain. It can be shown that MDS-based PHY-caching (with a binary PHY transmission mode) can achieve the optimal trade-off for the cached wireless networks with $M = K = 2$ and $N_T \geq N_R$ under any general content popularity distribution. The gap between the achievable trade-off region of MDS-based PHY-caching and the outer bound of the trade-off region for acached wireless network with $M = K = 2$, $N_T = 8$ users with $N_R = 1$, and $L = 1000$ content files is illustrated in Fig. 5a. As illustrated, the gap is small under different total storage budgets $C$.

In the following, we summarize the asymptotic trade-off scaling behavior of the MDS-based PHY-caching under large network sizes, which help capture first order insights for practical deployments. Specifically, we scale up the number of BSs $M$, but keep:

- A constant total cache storage size per BS, $C = \Theta(M)$
- A constant total cache size per total content size, $C = \Theta(L)$
- A sufficiently large number of users, $M = o(K \log(K))$

Supposing a total storage capacity of $C$ bits in the system, one may tend to allocate more storage to the CN-cache instead of the PHY-cache at the BSs since the CN-cache has a much higher cache hit rate. However, our results illustrate that this may not be preferable in general, and PHY-caching can play a vital role in improving the capacity of MIMO cellular networks despite the lower cache hit rate. There are two cases.

- **Super-critical case** ($\gamma > 0$): When $M \to \infty$ and $\gamma > 0$, the CN-backhaul offloading gain is in constant order, as illustrated in Fig. 5c. However, the DoF gain scales to infinity, as illustrated in Fig. 5d. Hence, the PHY-caching efficiency is much larger than the CN-caching efficiency. The trade-off behavior belongs to Type III, as illustrated in Fig. 5b, and thus we should allocate the whole cache capacity to the PHY-cache.

- **Critical case** ($\gamma = 0$): When $M \to \infty$ and $\gamma = 0$, the popularity of each file is equal. In this case, both the DoF gain and CN-backhaul offloading gain are in constant order, and the trade-off behavior belongs to Type II.

Note that in the above considered typical asymptotic regime, there is no sub-critical case where CN-caching is more efficient than PHY-caching. The intuition behind the above results are explained below. Note that the DoF gain is equal to the MIMO cooperation probability multiplied by the DoF gain of MIMO cooperation. The DoF gain of MIMO cooperation is in order $\Theta(M)$ as the number of BSs $M$ increases. The MIMO cooperation probability decreases in order $\Theta(M^{-1})$ for $0 \leq \gamma < 1$; $\Theta\left(\ln^{-1}M\right)$ for $\gamma = 1$; and $\Theta(1)$ for $\gamma > 1$, which depends on the popularity distribution parameter $\gamma$ because the per BS PHY-cache capacity is in constant order $\Theta(1)$, and the number of content files $L$ increases in order $\Theta(M)$. As a result, the scaling of the DoF gain is summarized in the aforementioned two cases. Note that a large number of users $K$ is needed to make sure that there is a sufficient number of users to exploit the MIMO cooperation gain; otherwise, there will be a DoF loss since the MIMO cooperation gain is also limited by the number of users. On the other hand, the CN-backhaul offloading gain is in constant order $\Theta(1)$ since the total cache capacity and the total number of content files increase in the same order. Although we need some asymptotic assumptions for rigorous analysis, the asymptotic trade-off results still capture first order insights for practical deployments, where the number of BSs involved in CoMP is relatively small and the number of users is not so large, as illustrated in Fig. 5a.

**Conclusion**

In this article, we propose PHY-caching to improve the spectral efficiency in wireless networks. After introducing a generic cache model and cache-assisted PHY transmission, we point out the design challenges of PHY-caching and provide corresponding solutions. We then compare the efficiency between CN-caching and PHY-caching in a case study. We show that even though PHY-caching covers fewer users than CN-caching, PHY-caching is still more efficient than CN-caching in most cases, and thus it is worthwhile to do PHY-caching at the BS. PHY-caching has shown great potential to enhance the spectral efficiency in future 5G wireless networks. Interesting future work would be exploiting flexible cooperation and joint optimization of the cache content placement among multiple cells. Moreover, the concept of PHY-caching is not limited to caching at the BSs. PHY-caching can also be extended to user devices so that several users may exploit the cached content to simultaneously transmit multiple data streams to other users. In this case, PHY-caching can also induce physical layer topology change in device-to-device communications.

**References**


**Biographies**

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ABSTRACT

Recently proposed wireless proactive caching has opened up a promising avenue for reducing the redundant traffic load of the future wireless network. In addition to infrastructure updating, caching schemes for different wireless caching scenarios are also essential problems of concern. Along with the benefits brought by proactive caching, the cost of proactive caching should also be taken into account, since caching schemes can be influenced by different interest relevant parties in wireless networks. In this article, we discuss several typical caching scenarios and apply corresponding game theoretical models to illustrate how the selfishness of different parties may influence the overall wireless proactive caching. Efficient caching strategies can be designed by carefully considering the relations and interactions among these parties by using game theory. We also outline possible future research directions in this emerging area.

INTRODUCTION

With recent improvements in mobile communication technologies, an increasing number of users are attracted to diversified online services when using mobile devices, which leads to explosive demands for mobile data. Recent studies have shown that mobile video streaming accounts for 50 percent of mobile data traffic, which is currently still growing rapidly. As a result, the excessive demand for data is draining the limited spectrum resources of wireless transmissions, especially the wireless link between base stations and users, and the wireless backhaul link between base stations and the core network.

To cope with this problem, a promising solution is to cache popular contents at the edge of mobile networks (e.g., base stations, access points, or even users’ mobile devices) [1]. In this way, the number of duplicate content transmissions can be greatly reduced, and real-time transmissions can be accelerated. Due to the recent development of learning techniques, the popularity of contents (i.e., the demand for different contents) can be predicted by tracking users’ requesting frequency and analyzing historical data [2]. This advantage improves caching efficiency by pre-downloading popular contents during off-peak times and serving predictable peak-hour demands, which is referred to as proactive caching [2]. With a proper proactive caching strategy, not only can the heavy traffic load be relieved at peak hours, but also the request latency can be decreased, which results in better user experience. Although dozens of techniques can be implemented to improve proactive caching, such as network coding [3] and multicasting [4], a systematic study on the interactions among different parties related to wireless proactive caching is still lacking.

In the architecture of wireless networks, multiple-interest relevant parties exist, which usually consist of the service providers (SPs) who provide contents to download, the mobile network operators (MNOs) who manage the facilities of the radio access network (RAN), and the mobile users who wish to enjoy different contents [5]. Different parties have their own benefits when applying a specific caching strategy, and their benefits could conflict with each other. For instance, caching more contents into the storage of base stations may bring more profit for users. But this could also jeopardize the benefit of the MNOs who manage the base stations, due to the cost of additional power consumption and occupying limited caching storage. Since each party only cares about its own profit, it is necessary to analyze the interactions among these parties and design proper solutions. To this end, game theory can be adopted as an effective tool to deal with such problems.

In this article, we apply game-theoretical approaches to model and analyze the problems that may arise in the typical scenarios of proactive caching in wireless networks. Different benefits and costs for different parties are taken into account, and the conflicts of interests among different parties are also considered. Specifically, we provide four representative scenarios of wireless proactive caching and discuss them with different game theoretical approaches.

Based on the structure of the considered wireless network, we further classify the scenarios of proactive caching into two categories: centralized wireless networks and distributed wireless networks. In the former category, the contents from SPs are cached into the storage of wireless facilities in RANs owned by MNOs. Either the SPs or the MNOs carefully design caching schemes based on the demands for contents in a central-
Typical Scenarios of Proactive Caching in Wireless Networks

In this section, we classify proactive caching scenarios into two categories based on the structure of the considered wireless network.

In each category, two typical scenarios are presented, which address the major concerns of the interactions among different parties or individuals.

Proactive Caching in Centralized Wireless Networks

In centralized wireless networks, the caching procedure is carried out by SPs or MNOs, and the caching strategies are usually designed in a centralized manner.

Considering different kinds of facilities in RANs that may be cache-enabled, we select two representative scenarios: SBS caching and RSU caching.

Small Cell Base Station Caching: Small cell base stations (SBSs) are considered as a promising infrastructure to deal with the rapidly increasing wireless traffic by achieving high-density spatial reuse of communication resources. However, the backhaul link capacity of SBSs is the major obstacle to providing satisfactory download speed. By caching popular contents into the storage of SBSs, duplicate transmissions can be reduced, and heavy traffic can be alleviated. At the same time, users can experience better QoS (e.g., lower delay or download time) as they request contents. The system model of SBS caching is shown in Fig. 1.

For each SP who possesses a certain set of contents, the QoS of its own users is the major concern. Therefore, each SP has intentions to cache its contents into SBSs to improve the expectation of user experience. According to the average user density covered by each SBS and the popularity distribution of the contents, a certain SP, in general, would like to cache its most popular contents into the most popular SBSs. However, due to the existence of multiple SPs and the finiteness of caching storages, competition among SPs is unavoidable. How to effectively cope with the competition and simultaneously guarantee high overall user experience is a major concern.

Roadside Unit Caching: With the application of high-end facilities on vehicles and the deployment of RSUs, OBU s can resort to more resource-intensive online services, which may drain the limited wireless resources. One possible solution is to equip the RSUs with caching abilities, in which way contents requested by OBUs can be obtained without the crowded backhaul links from RSUs to the Internet [6]. Fortunately, the driving routes of OBUs can be pre-determined with the help of navigation software on mobile devices or cars. Therefore, OBUs can ask for pre-caching services from the MNO who manages the RSUs based on their driving routes, as shown in Fig. 2.

In an ideal situation where the driving speed and download speed can be predicted precisely, the MNO only needs to divide the specific content into segments and cache them sequentially in the SBSs along the route. However, the practical situation makes it difficult to guarantee the continuity of downloading.
By dividing the content into overlapping segments instead of independent ones, the possibility of keeping continuity can be increased against the uncertainty of driving speed or download speed. But this also leads to a higher storage occupation cost for the MNO. Note that there might be different types of OBUs who have different QoS demands. The MNO can provide different caching schemes for different QoS demands. Therefore, how to design a proper pricing strategy for different types of OBUs to guarantee the MNO's utility becomes the major concern in this scenario.

**ProActive Caching In Distributed Wireless Networks**

In the distributed wireless networks, caching procedure is carried out directly by each user, and the caching strategies are usually designed in a distributed manner. One important difference between caching in distributed and centralized wireless networks is: The distributed one can reduce the duplicate transmissions at the wireless downlink of RAN, in addition to those at the backhaul link. In contrast, the centralized one can only reduce the duplicate transmissions at the backhaul link since contents are cached in RAN. In the distributed wireless networks, we select two representative caching scenarios based on whether the users are temporarily static: Device-to-Device Caching and Vehicle-to-Vehicle Caching.

**Device-to-Device Caching for Mobile Users:**

D2D communication is becoming more and more important in the future architecture of wireless communications, which provides direct high-speed short-range communications. Since the storage of mobile devices is becoming larger and cheaper, there is a possibility that mobile users can keep the contents that they have already watched/used/consumed and serve other nearby users who want the contents by D2D communications. In this way, cache-enabled user devices can be seen as distributed caching storage, which is an important supplement to the centralized proactive caching in cellular networks, as shown in Fig. 3.

Here, we consider users to be temporarily static (e.g., in an office or a cafe). Therefore, users are able to cooperatively download their commonly desired content, then cache and transmit it to each other through D2D communications. For each user, the download time can be shortened, and the power consumption of setting up cellular links can be reduced. However, the additional power consumption of caching and D2D communications still cannot be ignored. Whether the user can obtain a higher payoff by caching cooperatively depends on the utility of him/her joining each downloading group. Therefore, the major concern in this scenario is how to propose the strategy for users to form effective coalitions in a distributed way.

**Vehicle-to-Vehicle Caching for Onboard Users:**

Although RSUs provide OBUs with better Internet connectivity, the high speed of vehicles and the limited downlink capacity of RSUs both make it hard to download all of the desired contents just by utilizing infrastructure-to-vehicle (I2V) communications. With the help of V2V communications, OBUs can transmit cached contents to other OBUs, or obtain cached copies from other nearby OBUs.

Note that the major difference between D2D caching and V2V caching is the OBUs' mobile nature. Since the neighbors of a certain OBU may change stochastically with high frequency, cooperative caching and downloading are less effective than in the D2D caching scenario. Therefore, OBUs have to pre-determine whether to keep the contents after watching/using/consuming them. Whenever an OBU wants to download a content, the RSUs are able to help him/her by checking whether there is another nearby OBU who possesses the desired content.

However, free riders may exist in this scenario, that is, those who only ask for cached contents but never cache for others. A reasonable method to avoid the existence of free riders is to make a regulation like this: Only those users who participate in V2V caching can obtain cached contents from others, and the users who participate in V2V caching have to keep the contents in storage after watching/using/consuming them. This is actually an agreement that OBUs can decide whether to comply with, based on their own types (e.g., frequency of driving, frequency...
of requesting contents) and the price of RSUs’ help on notification of available caching.

The major concern is whether different types of OBUs are willing to participate in V2V caching. The system model is shown in Fig. 4.

**Figure 4.** The system model of vehicle-to-vehicle caching.

**Game Theoretic Approaches for Proactive Caching in Wireless Networks**

**Proactive Caching in Centralized Wireless Networks**

**Auction Game for Small Cell Base Station Caching:** The auction game is a branch of game theory that has been widely used in trading if the prices of the commodities are undetermined [7]. Since there are usually not enough commodities for all of the potential buyers, the trading prices are determined by the competition among these potential buyers during the bidding procedure in the auction.

In the SBS caching scenario, each SP is able to predict the popularity of its content by analyzing historical data. Based on the users’ requests from different SBSs, we also assume that the SP is aware of the user density within different coverage areas of SBSs. In this way, SPs can evaluate their contents while it is less possible for MNOs to know SPs’ valuations. Therefore, an auction game is suitable in this scenario to determine the trading price through the competition among SPs [8]. In this setting, the storage of SBSs is considered as objects to be auctioned. If there is only one MNO, the auctioneer (the one who is in charge of the auction procedure) could be the MNO itself. If there are multiple MNOs, the auctioneer would probably be another third party with enough credit. Since the caching cost is imposed on MNOs who own the SBSs, the price determined in the auction should be paid to MNOs by SPs who obtain corresponding caching storage.

Given a fixed auction mechanism, each rational bidder (SP) aims to maximize its own profit by bidding properly. This means that the bids for objects are not guaranteed to be the same as valuations (depending on the specific auction mechanism). The Nash equilibrium bidding strategy for a certain bidder is the strategy that is the best response to others’ current bidding strategies. Thus, the Nash equilibrium outcome can be seen as the outcome of a more practical situation for SBS caching. As depicted in the system model, the primary objective is to guarantee the caching efficiency (for better user experience), which means that a proper caching allocation should be accomplished by the auction mechanism.

**Contract Game for Road-Side Unit Caching:** In economics, the contract game studies how economic actors construct contractual arrangements, generally in the presence of asymmetric information [9]. One prominent application of the contract game is the design of optimal schemes for managerial compensation. For instance, the boss in a company does not know the types (e.g., hardworking or lazy) of his/her employees at first, and uniform treatment of different employees does not guarantee high workload for the boss. But by properly designing a workload–salary table, all rational employees are guaranteed to make their best choice, which can in turn maximize the total workload.

In RSU caching, OBUs who have pre-determined driving routes can ask for specific caching services. However, MNOs can provide different QoS of caching services, and different types of OBUs have different QoS demands. Although the exact type of each user is unknown to the MNO, the MNO is able to estimate the proportions of different types. Therefore, the contract game can be utilized by the MNO to design proper pricing strategies for different types of users in order to maximize its total payoff.

The main idea is to design a monopolist-dominated quality-price contract, which means that the MNO dominates the quality-price designing and wishes to maximize its own profit. To make the contract feasible, incentive compatibility and individual rationality are the two necessary and sufficient properties [10]. In this way, each OBU’s best choice is to pay for the caching service designed for his/her type, and no OBUs can obtain more utility by not choosing caching services. Based on the above two principles, a contract can be designed to maximize the profit of the MNO.

**Proactive Caching in Distributed Wireless Networks**

**Coalition Game for Device-to-Device Caching:** The coalition game is a cooperative game where some of the players have intentions to form cooperative groups (i.e., coalitions) [11]. A coalition represents an agreement among the players to act as a single entity to get a higher payoff for each participant.

In the D2D caching scenario, depending on the specific caching strategy, each user device will suffer higher power consumption by using caching storage (depending on the size of the cached contents) or serving others through D2D communications (depending on the communication range). Since each user is only able to estimate its own utility (perhaps with the help of an application on the mobile device), how to effectively form D2D caching groups should be considered as a distributed problem. By means of the coalition game, we can analyze and predict...
how users will cooperate with each other in a distributed manner based on different conditions.

For a coalition game, the primary objective is to find stable outcomes. In a stable outcome, multiple coalitions with different sizes are expected to be formed by users. If a user can obtain the highest payoff in the current coalition, then the state for him/her is stable, otherwise he/her has the intention to leave the current coalition and join another one. For the user who has a low caching cost (which depends on the mobile device), he/she is more likely to cooperatively cache with others. In addition, if adjacent users have similar preferences for the contents, then a coalition is also expected to form. Generally speaking, D2D caching with coalition formation is able to reduce average download cost for mobile users.

**Evolutionary Game for Vehicle-to-Vehicle Caching:** The evolutionary game is the application of game theory to analyze how the proportions of different strategies in the population change [12]. The evolutionary game differs from classical game theory by focusing more on the dynamics of strategy variation, which is influenced by both the quality and proportion of the competing strategies in the population.

In the V2V caching scenario, only users who participate in V2V caching can obtain cached contents from others, and users who participate in V2V caching have to keep the contents in storage after watching/using/consuming them. We denote caching population as the proportion of OBUs who participate in V2V caching. Whether an OBU should participate in V2V caching depends not only on its his/her own type, but also on the caching population. If more OBUs participate in V2V caching, it is easier for a certain OBU to obtain a cached content. In addition, OBUs may change strategy based on the current caching population he/she estimates. Therefore, the evolutionary game can be utilized to analyze the time-variant feature of OBUs’s strategies.

The major objective of applying the evolutionary game is to find evolutionary stable strategies. An evolutionary stable strategy is a mixed strategy for the whole population, such that the influence of a small proportion of other strategies will gradually disappear in the long-term evolution. For V2V caching, in particular, each type of OBU will have a certain possibility to choose whether he/she participates in V2V caching. Such a state is meaningful in both theoretical study and practical implementation. This is because the system can maintain stable caching performance against the turbulence induced by a small number of OBUs leaving or joining the system.

Table 1 summarizes the aforementioned four typical caching scenarios, their major problem descriptions, the information assumptions, and the corresponding kinds of game theory, along with the economic payment direction.

<table>
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<tr>
<th>Typical proactive caching scenario</th>
<th>Category</th>
<th>Information assumptions</th>
<th>Major problem description</th>
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<th>Payment</th>
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<td>SBS caching</td>
<td>Centralized</td>
<td>SPs know their own valuations, while the MNOs do not know SPs’ valuations.</td>
<td>Competition among multiple SPs for limited caching storage.</td>
<td>Auction game</td>
<td>SPs to MNOs</td>
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<tr>
<td>RSU caching</td>
<td>Centralized</td>
<td>OBUs know their own types, while the MNO knows the percentage of each type.</td>
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Table 1. Game theoretic approaches for typical scenarios in wireless proactive caching.

**One Example: Multi-Object Auction for Small Cell Base Station Caching**

In this section, we briefly introduce the multi-object auction, and then apply this method to solve the content storage allocation problem in SBS caching.

**Basis of Multi-Object Auction**

Auctions are being widely used in trading where the prices of commodities are determined by the competition among all the potential buyers during the bidding procedure. A multi-object auction is a special kind of auction that allocates multiple objects (instead of only one object) to multiple bidders in a single round of bidding. A multi-object auction can be classified into three categories, based on the relations of the objects in each round of auction [7]:

- **Independent objects:** The marginal utility of a certain object is not influenced by what objects the bidder has already owned.
- **Substitutable objects:** The marginal utility of a certain object is likely to decrease if another object is owned by the bidder.
- **Complementary objects:** The marginal utility of a certain object is likely to increase if another object is owned by the bidder.

We present below an example to show how a multi-object auction with substitutable objects can be applied to solve the caching problem in SBS caching.

**Small Cell Base Station Caching by Multi-Object Auctions**

As shown in Fig. 1, we study a small cell network which involves several MNOs that manage a number of overlapping SBSs and several SPs that own different sets of contents. If the content requested by a certain user is cached in one of its nearby SBSs, the request can be served by the SBS, which leads to a lower delay. Otherwise,
any one of the nearby SBSs can serve its request by setting up backhaul connections to the core network and downloading the content from servers. In a real-world situation, the average loads of SBSs are different at different hours of a day, and the speed-limited backhaul link of SBSs may suffer congestion during peak hours. The main objective is to minimize the overall average delay of content requests from users at each hour by properly designing the caching scheme. However, due to the overlapping regions among SBSs and the finiteness of caching storage, the optimal solution for this problem is NP-hard, indicating that the problem cannot be resolved by any algorithm with low time complexity.

The proposed caching scheme is carried out by holding a series of multi-object auctions, where the storage blocks of SBSs are abstracted as objects and the contents of SPs are abstracted as bidders. Specifically, the caching allocation in each hour is determined by holding a series of multi-object auctions sequentially, where the \( i^\text{th} \) auction is the auction on SBSs that are auctioned off in the \( i^\text{th} \) auction. This process is essentially to auction the storage of all SBSs concurrently in multiple steps, as shown in Fig. 5.

The valuation of each bidder for each object is determined by the marginal utility of caching, that is, how much the average delay can be decreased by caching the content into the given SBS. Before each auction, the valuations of bidders for objects have to be updated according to the current caching allocation, since the marginal utility of each object for each bidder is not constant due to the overlapping of SBSs. In other words, the storage sizes of different SBSs are partly substitutable objects. In reality, SPs submit bids on behalf of their own contents based on their marginal utilities. After the auction is done, each SP obtains a certain amount of caching spaces of SBSs. Then contents can be cached into SBSs according to the auction result (which can be done automatically by its server).

For each multi-object auction, the market matching algorithm is adopted, which takes the valuations as input, matches the bidders and objects, and outputs the allocation results and trading prices [13]. This algorithm starts by adding virtual objects to equalize the number of bidders and the number of objects. Then it tries to find a perfect matching between bidders and objects based on the preferences of bidders. If the perfect matching cannot be found, some of the prices of the objects will be increased in order to change the bidders’ preferences. The algorithm is guaranteed to end with a perfect matching that shows the allocation of bidders (contents of SPs) and objects (storage blocks of SBSs).

To evaluate the performance, the proposed mechanism is carried out by simulations. Figure 6a shows how the average delay changes in a day. The outcome of the proposed mechanism greatly reduces average delay and also surpasses the highest popularity algorithm (which only caches the most popular contents in each SBS and is used as a benchmark here). Figure 6b shows the impact of the capacity of SBSs and the total number of contents. It can be observed that the same amount of storage capacity makes greater difference in a low-storage-capacity situation. In addition, a greater amount of contents makes it more difficult to achieve low latency.

**Future Outlook**

**Proactive Caching in Heterogeneous Wireless Networks**

One typical heterogeneous wireless network can be a macrocell base station (MBS) along with its underlaid SBSs. Users can download content from the SBSs’ caching storage directly (with the least delay), download content from the MBS’s caching storage through the SBSs’ backhaul (with medium delay), or in the worst case, download content from the servers of the SPs (with the highest delay). This heterogeneous architecture makes the caching problem difficult to analyze. Furthermore, if the MBS and SBSs have different owners, the interactions among SPs and MNOs become far more complicated. Since the profits brought by caching are divided among multiple parties, a bargaining game can be adopted to balance the benefits of different parties and achieve a stable outcome [14].

**Proactive Caching in Wireless Relay Networks**

In cache-enabled wireless relay networks, contents can be cached in the storage of wireless relays [15]. Relays can transmit data to users simultaneously with the base station; thus, the diversity of wireless transmissions can be improved. However, the base stations and relays could belong to different MNOs. The auction game can still be applied in this scenario since multiple SPs are willing to compete for finite caching storage of both base stations and relays. However, evaluation of contents should be carefully considered due to the complicated topology of the network.

**Conclusion**

In this article, game theoretical approaches have been introduced to model and analyze the cooperative and non-cooperative behaviors of different parties in wireless proactive caching. In centralized wireless networks, the auction game is used to resolve the competition among multiple SPs in the scenario of SBS caching, and the contract game is adopted to design the proper pricing strategy in the scenario of RSU caching. In distributed wireless networks, the coalition game is applied to analyze the possibility of cooperative caching in the scenario.
of D2D caching, and the evolutionary game is considered as a promising tool to analyze the behavior of users in V2V caching. A more detailed multi-object auction-based solution for the small cell caching scenario is presented. Possible future research directions of game-theory-based wireless proactive caching are also outlined.

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Harnessing Cloud and Edge Synergies: Toward an Information Theory of Fog Radio Access Networks

Ravi Tandon and Osvaldo Simeone

ABSTRACT

Edge caching and the centralization of baseband processing by means of the C-RAN architecture are among the most promising and transformative trends in the evolution of wireless networks. A key advantage of C-RAN is the possibility to perform cooperative transmission across multiple edge nodes, such as small cell base stations, thanks to centralized cloud processing. Cloud processing, however, comes at the cost of the potentially large delay entailed by fronthaul transmission between edge and cloud. In contrast, edge caching enables the low-latency transmission of popular multimedia content, but at the cost of constraining the operation of the edge nodes to decentralized transmission strategies with limited interference management capabilities. In order to accommodate the broad range of quality of service requirements of mobile broadband communication, in terms of spectral efficiency and latency, that are envisioned to be within the scope of 5G systems and beyond, this article considers a hybrid architecture, referred to as fog RAN (F-RAN), that harnesses the benefits of, and the synergies between, edge caching and C-RAN. In an F-RAN, edge nodes may be endowed with caching capabilities, while at the same time being controllable from a central cloud processor as in a C-RAN.

INTRODUCTION

Edge processing and cloudification are among the most promising trends in the evolution of wireless network architectures toward the specification of fifth generation (5G) systems and beyond. Edge processing refers to the placement of storage and computing resources at the network edge, that is, closer to the users. This localization of content and computing eases low-latency or location-based applications, as well as to multimedia transmission with local content reuse [1]. Cloudification amounts to the complementary trend toward the decoupling of physical network elements, such as base stations, from the control and processing logic that is implemented centrally at a cloud processor. The resulting sharing of the control and processing resources of the cloud across multiple network elements yields significant gains in terms of capital and operating expenses, flexibility in ownership models, statistical multiplexing, and interference management [2].

A network architecture based on edge processing is illustrated in Fig. 1a. Here, edge nodes (ENs), such as base stations or eNBs in Long Term Evolution (LTE), are equipped with local caches that can be used to store popular content, most notably multimedia files, with the aim of reducing the delivery latency and the overhead on the backhaul connections to the content server. Edge processing via caching provides an ideal solution for data traffic classes, such as video, characterized by high local content reuse [1]. A scenario with cloudification of the functionalities of the ENs, also known as the cloud radio access network (C-RAN), is depicted in Fig. 1b. In this architecture, the ENs are connected to the cloud processor by so-called fronthaul links. Due to the enhanced interference management capabilities afforded by centralized baseband processing at the cloud, which can operate jointly across all connected ENs, C-RANs are particularly well suited to enhance the spectral and cost efficiency of interference-limited dense deployments with less stringent delay constraints [2].

To summarize, while C-RAN provides high spectral efficiencies thanks to cooperative cloud-based transmission, but at potentially large latencies due to fronthaul transmission, edge caching enables the low-latency delivery of popular content, but with limited interference management capabilities because of decentralized baseband processing at the ENs. Recognizing that modern wireless networks, including 5G systems, are expected to cater to a broad range of quality of service requirements for mobile broadband communication, in terms of spectral efficiency and latency, a hybrid architecture was recently advocated [3, 4], which is illustrated in Fig. 2 and referred to as fog RAN (F-RAN). In an F-RAN, ENs may be endowed with caching capabilities, while at the same time being controllable from a central cloud processor. As such, the F-RAN architecture captures the key benefits of central-
ized baseband processing and low-latency delivery of C-RAN and edge caching, respectively. It should, however, be noted that the F-RAN architecture does not retain the important C-RAN feature of a reduced deployment cost for the ENs [2], which, unlike for a C-RAN, need to be provided with baseband processing capabilities so as to be able to locally process the cached content.

The main goal of this article is to lay the theoretical foundations for the study of the optimal operation of an F-RAN architecture. Optimal design requires edge caching, fronthaul, and wireless transmission to be jointly designed so as to leverage the discussed synergistic and complementary features of edge processing and virtualization. The resulting design problem is extremely challenging, as it includes the joint optimization of caching, fronthaul, and transmission policies, making a brute force approach prohibitive. To overcome these challenges, in this article, we propose and analyze a novel information-theoretic framework with the aims of illuminating the main trade-offs between the system performance in terms of latency on one hand, and the resources available for caching, fronthaul, and wireless transmission on the other, as well as revealing design guidelines for the optimal design of F-RAN via analytical arguments. Examples are offered to illustrate the merits of the proposed approach. References [5, 6] provide the technical details that are omitted here in order to focus on the key ideas.

The rest of the article is organized as follows. We present the proposed information-theoretic model and performance metrics, along with the design space for an F-RAN, which encompasses caching, fronthaul, and transmission policies. We present two case studies that exemplify the analysis afforded by the proposed framework. Generalizations are discussed that concern the impact of imperfect channel state information (CSI) and of the network topology. Finally, we present some concluding remarks and an outlook on open problems and research issues.

**INFORMATION-THEORETIC MODEL AND DESIGN SPACE**

As illustrated in Fig. 3, we consider an F-RAN architecture with $M$ edge nodes (ENs), which can serve a set of $K$ users over a shared wireless channel. The ENs are connected to the cloud by means of fronthaul links of capacity $C_F$ bits per symbol (of the edge wireless channel) for each EN. The capacity $C_F$ is assumed to be fixed, reflecting conventional scenarios in which fronthaul links correspond to dedicated wired connections [2]. Each EN is equipped with a cache of limited size.

We assume the presence of a library of $N$ files, each of length $L$ bits, which represent the content that may be requested by users. As in the majority of related analyses [1, 7, 8], this library of popular files is assumed to remain unchanged during the period of time over which the content of the ENs' caches is not refreshed. For instance, caches may be updated in the early morning, when the traffic load is at a minimum, and kept unmodified for the rest of the day. The period of time in which caches and library are assumed to be fixed encompasses multiple transmission intervals, which are identified by an index $t = 1, 2, \ldots$.

![Figure 1.](image)

We focus here on the scenario in which no popularity distribution is available to describe the relative likelihood that one of the files is selected by a user, so all files are equivalent and may potentially be requested [7, 8]. The cache of each EN can store $\mu N L$ bits for some fractional cache size $0 \leq \mu \leq 1$.

At each transmission interval $t$, users issue a vector of requests. We make no assumption on the nature of the time variability of the demands made by users. The collective time-varying wireless CSI $H(t)$ at transmission interval $t$ collects all the channel coefficients that characterize the propagation between all the ENs and the $k$th user. These coefficients describe the channel profile in the frequency and/or time domain for the given spectral and temporal resources allocated at transmission interval $t$ to a pair of EN and user. For the sake of illustration, following a conventional modeling choice [9], we focus here on the setting in which the channel coefficients are generated independent from an identical continuous distribution. Besides fading, the channel model for the wireless segment includes additive Gaussian noise. Topological constraints are discussed later.

In order to avoid the explicit dependence on the bandwidth of the edge wireless channel, throughout, rates are measured in bits per symbol of the wireless channel, and time metrics are mea-
The design problem entails the optimization of the joint caching-fronthaul-transmission policy. Here we summarize the design space under the assumption of full CSI at the ENs and at the cloud. We discuss the impact of imperfect CSI later.

**Cache storage policy.** The caching policy operates at the discussed timescale over which the set of popular files is expected to remain constant (e.g., one day), which contains many transmission intervals (indexed by $t$). The caching policy is defined by a function that decides the cache content of each EN. The latter must satisfy the cache capacity constraint, that is, the size of the content stored at each EN cannot exceed $\mu_{NL}$ bits (Fig. 3). We note that the cache of each EN is populated based solely on the library of files, without knowledge of the instantaneous users’ demands as well as without CSI, which vary across transmission intervals $t$.

A general approach to cache storage is to split each file into a number of fragments of a certain size and to adopt one of the following classes of policies: uncoded caching and coded caching. For **uncoded caching**, each EN stores a subset of the fragments depending on the normalized cache size $\mu$. Uncoded caching policies can create virtual and overlapping clusters of collaborative ENs, where cooperative transmission can be carried out over shared file fragments. For **coded caching**, one can allow for both intra-file coding (i.e., coding within the fragments of a file) and inter-file coding (i.e., coding across different files). Note that coded fragments could also be replicated across ENs to benefit from cooperative transmission as in [10].

**Fronthaul policy.** The fronthaul policy, as well as the transmission policy, operate separately over each transmission interval $t$ as a function of the instantaneous demands of the users as well as of the CSI of the shared wireless medium. Accordingly, the fronthaul policy is defined by a function of the instantaneous demands and of the CSI that determine the duration $T_{tf}$ of the fronthaul transmission (measured by normalizing with respect to the duration of a symbol of the edge wireless channel). The fronthaul message cannot exceed $T_{tf} C_{F}$ bits, where $C_{F}$ denotes the fronthaul capacity as seen above. Note that the fronthaul policy can hence control the fronthaul duration within the given transmission interval.

We can identify two main approaches to the design of the fronthaul policy:

- **Hard-transfer mode**, whereby fragments — coded or uncoded following the classification of caching policies mentioned above — are transferred to the ENs.
- **Soft-transfer mode**, whereby the cloud directly encodes the files, producing baseband signals that are quantized and sent over the fronthaul links on the ENs following the C-RAN principle [2].

The design of fronthaul policies in the hard-transfer mode, including the selection of coded or uncoded strategies, follows the guidelines discussed above for caching policies, with the important caveat that the fronthaul policy can adapt the choice of fragments to be sent to ENs to the users’ current demands.

To illustrate the design space for soft-transfer mode, consider first the scenario with no caches (i.e., a C-RAN system). Here, the optimization of fronthaul policies is equivalent to that of the transmission over a broadcast channel in which the set of ENs form a multi-antenna transmitter, with the limitation that the encoded baseband signals are subject to the distortion caused by fronthaul quantization. In a more general F-RAN, the design space acquires novel degrees of freedom due to the interplay between coding at the cloud, as in a C-RAN, and coding at the ENs, based on the locally cached content. For instance, each EN can transmit a superposition of the quantized baseband signal received on the fronthaul link and of a function of the cached content. Since the latter is not subject to any quantization noise, this can potentially enhance the performance.

**Edge transmission policy.** The edge transmission policy, or transmission policy for short, operates on each transmission interval and selects the codewords sent on the wireless channel by all the ENs, and hence also their duration $T_{tf}$, under an average power constraint given by the parameter $P$. Note that the codeword transmitted by each EN can depend on the local cache, the received fronthaul message, the instantaneous demands, and the CSI. As elaborated above, the design of transmission policies is strongly interdependent with the caching and fronthaul policies. For instance, with uncoded caching and hard-transfer operation of the fronthaul, the transmission policy design amounts to the problem of coding over a single-hop interference network with arbitrary sets of messages at the ENs.

**Latency Metric: Normalized Delivery Time.**

In order to compare different design choices and to enable system optimization, we adopt the performance metric of the delivery time, that is, the time required by the system to satisfy arbitrary users’ requests in a given transmission interval. Neglecting the time needed for the cloud and the ENs to register the users’ requests, delivery latency is generally affected by the time required for transmission on the two segments of fronthaul-
Different assumptions can be made regarding the level of pipelining possible between transmissions on the two segments. For example, ENs may immediately start transmitting on the wireless channel while at the same time receiving information on the fronthaul links, which can be causally encoded into the wireless transmission. In this work, we focus on a baseline scenario in which no pipelining is possible, in the sense that fronthaul transmission is followed by wireless transmission.

To elaborate, we first define the delivery time per bit $\Delta(\mu, CF, P) = (T_E + T_F)/L$, which measures the latency within each transmission interval for the worst case users’ request vector, as normalized by the size of the file $L$. Following the standard Shannon-theoretic framework, the file size $L$ and the blocklengths $T_E$ and $T_F$ are allowed to be arbitrarily large so as to satisfy any desired level of probability of error (see also [11]). The optimal latency performance is in principle obtained by minimizing the delivery time per bit $\Delta(\mu, CF, P)$ over all possible caching-fronthaul-transmission policies. This optimization is generally prohibitive and is also dependent on all parameters $(\mu, CF, P)$.

With the aim of obtaining analytical insights, we propose a novel tractable metric that retains the key dependence of latency on cache size and fronthaul capacity while adopting a high-signal-to-noise ratio (SNR) approximation in the vein of the by now standard degrees of freedom (DoF) analysis of interference networks [9]. To this end, we let the fronthaul capacity scale with the SNR parameter $P$ as $CF = r \log(P)$, where $r$ is a parameter that measures the capacity scaling of the fronthaul links’ capacity as compared to the wireless channel.

The key idea is to evaluate the relative latency between the F-RAN system under study and that of a baseline system with no interference and unlimited caching, in which each user can be served by a dedicated EN that has all files. The delivery time per bit of this ideal system is well known to be $1/\log(P)$, and hence we define the normalized delivery time (NDT) $\delta(\mu, r)$ as the limit of the ratio $\Delta(\mu, CF, P)/(\mu \log(P))$ for large SNR $P$ on the wireless channel. As such, an NDT of $\delta$ indicates that the worst case time required to serve any possible request is $\delta$ times larger than the time that would be needed by the baseline system. Optimizing over all possible policies yields the minimum NDT $\delta^*(\mu, r)$.

Based on the definitions above, in the proposed framework, the goal of the analysis is the characterization of the novel metric NDT $\delta^*(\mu, r)$ that captures the interplay between latency and resources, that is, the normalized cache storage $\mu$ and the fronthaul multiplexing gain $r$.

## Case Studies

### Case Study 1: Edge Caching in Interference-Limited Scenarios

We first consider systems with edge caching operating over interference-limited channels with no fronthaul as illustrated in Fig. 1a. We note that the conventional design of cache-aided wireless systems abstracts the contribution of the physical layer by assuming fixed coverage areas and implicitly assumes uncoordinated ENs (see, e.g., [12]). In contrast, it has been recently recognized that, in the presence of shared content in the ENs’ caches, the ENs are enabled to use more sophisticated transmission schemes, including coordinated beamforming and precoding. The
With the aim of obtaining analytical insights, we propose a novel tractable metric that retains the key dependence of latency on cache size and fronthaul capacity while adopting a high-SNR approximation in the vein of the by now standard degrees of freedom (DoF) analysis of interference networks.

Interplay between caching and cooperative transmission was first studied in [10], in which it is proposed to store the same erasure-coded packets at all ENs in order to allow for joint beamforming across all ENs. These works are based on dynamic optimization arguments and signal processing. In [8], instead, the cache allocation problem was studied under an information-theoretic framework, from the point of view of DoF analysis, for a scenario with three ENs and users, under the assumption that all the requested files are cached at ENs.

To illustrate the insights afforded by the NDT analysis, we consider the setup in Fig. 3a in which two ENs, labeled EN1 and EN2, are deployed to serve two users. Figure 3b shows the information-theoretically optimal trade-off curve between the NDT $\delta^*$ and the fractional cache size $\mu$ as obtained in [5] under the constraint of uncoded inter-file caching. Note that this performance trade-off always results in a convex curve [5]. To take some exemplifying operating points on the curve, for $\mu = 1$, both ENs can store all files, and hence full cooperative transmission can take place (i.e., via zero-forcing beamforming for any set of users’ requests), yielding $\delta^* = 1$. This implies that the latency performance is the same as that of the mentioned baseline ideal system. On the other hand, at $\mu = 1/2$, which is the smallest cache size to enable delivery of any vector of requests, the NDT increases to $\delta^* = 3/2$ and is achieved via interference alignment [5], revealing the performance loss due to partial caching.

**CASE STUDY 2: FRONTHAUL PROCESSING AND EDGE CACHING FOR F-RANS**

We now elaborate on a full-fledged F-RAN scenario with cloud processing and edge caching for F-RANS as illustrated in Fig. 2. As a case study, we consider the F-RAN topology shown in Fig. 4a, in which the edge nodes EN1 and EN2 are endowed with caches, as discussed in the previous example, but are also connected to the cloud by means of fronthaul links with given capacities. From a signal processing viewpoint, the joint design of beamforming and fronthaul processing in hard-transfer mode, where the latter determines which ENs receive each non-cached file on the fronthaul links, is studied in [13, 14] for a fixed pre-defined cache allocation.

Figures 4b and 4c show the optimal NDT trade-off derived in [6], again under the assumption of uncoded caching. We first note that NDT trade-off identifies two distinct regimes in terms of the fronthaul capacity, a low-fronthaul capacity regime with $r \leq 1$ and a high-fronthaul capacity regime with $r > 1$. In the latter case, the use of both fronthaul and caching resources is necessary in order to obtain the optimal NDT performance, while in the former, if the cache capacity is sufficiently large (i.e., if $\mu \geq 1/2$), it is sufficient to leverage the cache storage resources to achieve the optimal performance.

To provide additional insights on the calculation and significance of the NDT metric, we now briefly discuss the scheme that achieves the NDT $\delta^*(\mu = 0, r) = 1 + 1/r$. The case $\mu = 0$ corresponds to the setting in which the ENs have no cache storage capability. A finite NDT can hence only be achieved by using the fronthaul resources. As discussed, the fronthaul links can be utilized in either hard- or soft-transfer mode. With hard transfer, the cloud can transmit both requested files to each EN, and then the ENs can use the same fully cooperative zero-forcing approach adopted for $\mu = 1$, as discussed above. Since the fronthaul links have capacities $C_F = r \log(P)$ each and $2L$ bits need to be sent to both ENs, the achievable NDT can be computed as $\delta = 1 + 2/r$. However, hard transfer turns out to be suboptimal in this scenario. The optimal NDT is in fact achieved through a soft-transfer scheme, whereby the cloud implements zero-forcing beamforming and quantizes the resulting baseband signals. It can be shown [6] that this scheme entails a fronthaul latency that equals the edge latency multiplied by $1/r$, since the scheme uses a resolution of around $\log(P)$ bits per downlink sample. As a result, it yields the optimal NDT $\delta^* = 1 + 1/r$. 

![Diagram](image-url)
In this section, we discuss two generalizations of the information-theoretic framework studied so far with the aim of accounting for imperfect CSI and for the impact of topology. Other generalizations of interest, not further discussed here, include the investigation of the impact of pipelining of fronthaul and wireless transmissions; the consideration of online caching strategies in which the caches can be updated based on the signals received from the cloud on the fronthaul [15]; and the study of the impact of limited reliability transmission in the finite blocklength regime.

**THE IMPACT OF IMPERFECT CSI**

In an F-RAN, for both time-division duplex (TDD) and frequency-division duplex (FDD) operations of the wireless channel, CSI is first estimated at the ENs, either directly through uplink training for TDD or indirectly via feedback for FDD, and then conveyed to the cloud through the fronthaul links. Furthermore, the CSI acquired at the EN is typically local in the sense that it only pertains to the channels describing propagation from the given EN, and not from the other ENs, to the users. As a result, in an F-RAN, the CSI model has the following two unique features:

- **Heterogeneous CSI timeliness**: CSI acquired at the ENs is typically local in the sense that it only pertains to the channels describing propagation from the given EN, and not from the other ENs, to the users. As a result, in an F-RAN, the CSI pertains the channels describing propagation from the given EN, and not from the other ENs, to the users. A result, in an F-RAN, the CSI model has the following two unique features:

- **Global vs. local CSI**: The ENs have local, more timely, CSI, and the cloud has global, but more delayed, CSI.

We next describe the proposed system model that aims at capturing these aspects, as well as the significant novel challenges that arise from the study of F-RANs with imperfect CSI in terms of both policy design and converse arguments.

As an exemplifying illustration of the main new challenges that arise in the design of caching, fronthaul, and transmission policies due to the heterogeneity of the CSI timeliness at ENs and cloud, we now consider the $M = 2$-EN and $K = 2$-user example studied above in which possibly delayed CSI is available at the ENs. We focus here for simplicity on the setup with $r = 0$ so that only caching, and no cloud transmission, is considered. The resulting NDT trade-off is shown in Fig. 6. The figure illustrates the impact of increasing delays at the ENs on the NDT, starting from no delay, to delay as large as coherence time or “stale” CSI, ending with no CSI at the ENs. Focusing on the operating point with $\mu = 1/2$, we observe that when CSI is timely, the minimum NDT of $3/2$ is achieved via a scheme based on interference alignment as discussed above and in [12]. Moreover, when CSI is outdated, the effect of “stale” CSI is reflected in an increase of the NDT to $5/3$, which can be achieved via a transmission scheme that uses “stale” CSI [5]. Finally, when there is no CSI, an NDT of $2$ is achieved by independent transmissions to each of the users (e.g., using time division), requiring twice the time compared to full CSI.

**IMPACT OF NETWORK TOPOLOGY**

Here, we focus on a general network topology, in which a user may be in the coverage of only a subset of ENs, hence receiving at negligible power for the rest of the ENs. This scenario captures the operation of larger-scale networks in which ENs cover different, but possibly overlapping, areas. Specifically, the system model discussed above is modified here by allowing the channel gains.
Assume that if requested. Instead, with coded caching, we split the file only to EN1 and EN2 cannot recover the file of a given file and EN3 the other, a user connects to the limited connectivity, there are users’ requests half of every file. Therefore, it can be seen that due to coded fragment A users. For instance, the choice and maintain tractability, we further assume that the minimal number of ENs that cover any user u in order to avoid uninteresting pathological cases and maintain tractability, we further assume that M = K and each EN covers the same number of users. For instance, the choice l = M corresponds to a fully connected network and l = 1 corresponds to non-interfering point-to-point channels.

We provide now an example that shows that coded caching can provide unbounded gains in general topologies. Consider a ring topology with l = 2, in which there are three equally spaced ENs, and three users placed between two successive ENs and connected only to the two nearby ENs. Assume that μ = 1/2 and that r = 0. With uncoded caching, each EN can hence at most store at most half of every file. Therefore, it can be seen that due to the limited connectivity, there are users’ requests that cannot be met, yielding an unbounded NDT. For instance, if EN1 and EN2 store the first half of a given file and EN3 the other, a user connected only to EN1 and EN2 cannot recover the file if requested. Instead, with coded caching, we split a file A into two equal size fragments A1 and A2. EN1 can cache the first half A1 of the file, EN2 can cache the second half A2, and EN3 the intra-file coded fragment A1 ⊕ A2 given by the XOR of the two fragments. With this coded caching scheme, the NDT is finite since a user attached to any two ENs can recover any file. Note that this amounts to the use of an (n, k) = (3, 2) MDS code.

CONCLUDING REMARKS AND OUTLOOK

The F-RAN architecture leverages the synergies between cloud processing and edge caching to offer performance advantages in terms of latency and spectral efficiency. In this article, we have introduced an information-theoretic framework that aims at capturing the key trade-off between delivery latency and main system resources, namely fronthaul capacity and caching storage capacity. We have provided a number of use cases, exemplifying examples, and open problems. In presenting the framework at a high level, the authors hope to stimulate research on the topic.

REFERENCES


BIOGRAPHIES

RAHUL TANDON (tandoren@email.arizona.edu) received his B.Tech. degree in electrical engineering from the Indian Institute of Technology, Kanpur in 2004 and his Ph.D. degree in electrical and computer engineering from the University of Maryland, College Park (UMCP) in 2010. From 2010 to 2012, he was a postdoctoral research associate in the Department of Electrical Engineering at Princeton University. He is currently an assistant professor in the Department of Electrical and Computer Engineering at the University of Arizona. Prior to joining the University of Arizona in fall 2015, he was a research assistant professor at Virginia Tech with positions in the Bradley Department of ECE, the Hume Center for National Security and Technology, and the Discovery Analytics Center in the Department of Computer Science. He was a co-recipient of the Best Paper Award at IEEE GLOBECOM 2011. He was nominated for the Graduate School Best Dissertation Award, and also for the IEEE Distinguished Dissertation Fellowship Award at UMCP. His current research interests include information theory and its applications to wireless networks, communications, security and privacy, distributed storage systems, machine learning, and data mining.

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Figure 6. Impact of CSI on latency for F-RANs.
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BACKGROUND
Communications standards enable the global marketplace to offer interoperable products and services at affordable cost. Standards Development Organizations (SDOs) bring together stakeholders to develop consensus standards for use by a global industry. The importance of standards to the work and careers of communications practitioners has motivated the creation of a new publication on standards that meets the needs of a broad range of individuals including: industrial researchers, industry practitioners, business entrepreneurs, marketing managers, compliance/interoperability specialists, social scientists, regulators, intellectual property managers, and end users. This new publication will be incubated as a Communications Standards Supplement in IEEE Communications Magazine, which, if successful, will transition into a full-fledged new magazine. It is a platform for presenting and discussing standards-related topics in the areas of communications, networking and related disciplines. Contributions are also encouraged from relevant disciplines of computer science, information systems, management, business studies, social sciences, economics, engineering, political science, public policy, sociology, and human factors/ usability.

SCOPE OF CONTRIBUTIONS
Submissions are solicited on topics related to the areas of communications and networking standards and standardization research, in at least the following topical areas:
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• Wireless LAN
• SDN
• Ethernet
• Media codecs
• Cloud computing
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• Patent policies, intellectual property rights, and antitrust law
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• The processes and organizations for creation and diffusion of standards, including the roles of organizations such as IEEE and IEEE-SA
• National and international policies and regulation for standards
• Standards and developing countries
The history of standardization, including, but not limited to:
• The cultures of different SDOs
• Standards education and its impact
• Corporate standards strategies
• The impact of Open Source on standards
• The impact of technology development and convergence on standards
Research-to-Standards, including standards-oriented research, standards-related research, research on standards Compatibility and interoperability, including testing methodologies and certification to standards
Tools and services related to any or all aspects of the standardization lifecycle
Proposals are also solicited for Feature Topic issues of the Communications Standards Supplement.
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COMMUNICATIONS, CACHING, AND COMPUTING FOR CONTENT-CENTRIC MOBILE NETWORKS

The Role of Cloud Computing in Content-Centric Mobile Networking

Jianhua Tang and Tony Q. S. Quek

ABSTRACT

Due to the ever growing popularity of smart handheld devices, the demand for multimedia services has had an upsurge in mobile networks over the past several years, and traditional mobile networking is turning into a content-centric mobile networking (CCMN). To maintain high quality of service for multimedia services, especially video services, caching is regarded as one of the most effective techniques. In this article, we first examine the limitations of caching techniques in conventional CCMN, including the core network and RAN. Then, by leveraging cloud computing into both core network and RAN caching, a cloud CCMN architecture is discussed, which overcomes the limitations of conventional CCMN architecture. Specifically, we elaborate on a cloud content delivery network in the core network and cloud RAN with caching as a service (CaaS) on the RAN side.

INTRODUCTION

The evolution of the Internet has recently been driven by fast data proliferation. Cisco predicts that global IP traffic will grow with a compound annual growth rate of 23 percent from 2014 to 2019, and this growth is mostly driven by video content; by 2019, video traffic, including TV, Internet, video on demand (VoD), and peer-to-peer (P2P), will constitute approximately 80 to 90 percent of global consumer traffic, and 70 percent of the IP VoD will be high-definition video. In terms of access modes, mobile data traffic will increase 10-fold from 2014 to 2019, and video traffic will account for 72 percent of global mobile data traffic by 2019.

With such fast data proliferation, network architecture is transferring from host-centric (or connection-centric) to content-centric [1] (or information-centric). In particular, in the current host-centric network, communication is based on named hosts (e.g., servers), while in the emerging content-centric network, communication is based on named data objects (e.g., video contents).

Since over 80 percent of global consumer traffic will be video traffic, and nearly three-fourths of mobile data traffic will be video traffic, there is an urgent need to develop and implement efficient techniques to distribute the high amount of video contents over networks, including both the core network and radio access network (RAN).

A well-known feature of video contents is that only a small portion of them will be requested most frequently. Based on this feature, lots of intelligent techniques have been proposed to improve the quality of service (QoS) for distributing video contents in both the core network and RAN. A common idea of these techniques is duplicating and caching the popular contents. By the places where the contents are cached, these techniques can be classified into two main categories: core network caching and RAN caching.1

Another feature of video traffic is the frequently occurring flash crowd phenomena (also known as the slashdot effect). A flash crowd occurs when there is an unexpectedly high amount of traffic during a short period of time. Unfortunately, with the super fast information spread in social networks, the occurrence of flash crowds, caused by some unexpected hot video contents, becomes more frequent nowadays. For example, the study of CoralCDN in [2] observes 2501 flash crowds appearing over a four-year span; that is, almost two flash crowds happen per day on average. Therefore, the ability to handle flash crowd traffic is also an essential point to be considered when developing caching techniques for video contents.

In the subsequent sections, we first overview traditional content caching networks, followed by cloud content caching networks. Next, we detail the investigated problems and open challenges, respectively. We then conclude the article.

OVERVIEW ON CONTENT CACHING

In this section, we overview the traditional techniques for both core network caching and RAN caching, and analyze their limitations.

CORE NETWORK CACHING

In core network, the deployment of content delivery networks (CDNs) is a commonly accepted solution to address the technical challenges introduced by the upsurge in video traffic. A CDN is an overlay network that duplicates pop-

1 Actually, there is one more caching category: users’ device level caching, such as P2P caching and device-to-device (D2D) caching. This category is out of the scope of this article, since here we only focus on caching techniques on the network side.

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ular contents from the origin content servers of content providers and caches these duplicated contents at the network edge. As a result, contents in a CDN are closer to the end consumers, and 72 percent of all Internet video traffic will be transmitted by CDNs by 2019 as predicted by Cisco.

There are many advantages of utilizing CDNs, such as decreasing origin content server load, reducing content retrieval latency, improving content availability, and increasing the number of concurrent users. Consequently, many commercial CDN providers have been founded since the end of the 1990s (e.g., Akamai).

However, there are some inherent limitations of utilizing CDN:
- It is not cost effective for most organizations or content providers to build up and maintain their own CDNs around the world to process global requests because of the high upfront and operation expenses.
- Although there are some third-party CDNs, the high setup fees and other hidden fees still keep small clients away. Moreover, support availability is another issue once third-party vendors are utilized to maintain the CDN.
- Due to the static resource allocation of traditional CDNs, they are inadequate to handle flash crowd traffic.

Later, we introduce the cloud CDN, which can resolve these inherent limitations of CDNs. However, core network caching still has its natural limitations, no matter what kind of technique is adopted. First, the contents stored in the core network are still not “near” enough to the end consumers. In addition, core network caching just reduces the amount of duplicate contents transmitted in the core network; however, the traffic amount to the RAN still remains challenging, which poses high pressure on RANs’ backhaul.

RAN Caching

Stemming from the idea of duplicating and caching popular contents, FemtoCaching [3] was recently proposed to cache popular contents in base stations (BSs) and access points (APs). It is becoming a promising solution to break through the natural limitations left by core network caching.

FemtoCaching has many advantages. First of all, the “distances” between contents and end consumers are further decreased compared to core network caching, and hence, the end-to-end delay can be further reduced potentially. Second, FemtoCaching mitigates the limited backhaul capacity problem by replacing backhaul capacity with storage capacity at the local BSs or APs. Furthermore, the request load to origin content servers can be reduced thereafter.

However, FemtoCaching also has its implicit problems:
- If every BS caches the same content set (i.e., the most popular ones), the total amount of content that can be cached will be limited since the caching size in each BS is very limited.
- If BSs can work cooperatively, the content set cached in each BS can be different.

However, the distributed content placement problem is nontrivial (“which file should be cached and where to cache it” is a complicated problem).
- To implement FemtoCaching, new devices (e.g., storage devices) have to be added to the current RAN, which is costly.

As can be seen above, although the recently proposed FemtoCaching can somewhat extend the capability of CDN caching, there are still many technical challenges in FemtoCaching. To solve the limitations and problems posed by both core network and RAN caching, in the next section, we briefly introduce cloud computing, present cloud CDN and cloud RAN, and discuss the caching issues in each of them sequentially.

Cloud Content-Centric Mobile Network

Along with content proliferation in the network, cloud computing has emerged as a widely adopted computing paradigm over the past several years. The increasing popularity of cloud computing is due to its attractive properties. The most typical property of cloud computing is the so-called five essential characteristics [4]: on-demand self-service, broad network access, resource pooling, rapid elasticity, and measured service.

As a result, many conventional communication systems are trying to migrate from hardware defined infrastructures to the software defined cloud environment since cloud computing offers the following benefits:
- Reducing capital expense. There is no need for individuals or organizations to spend huge resources to build up their own data centers, including hardware, software, or licensing fees. They only pay when they consume computing resources, and for how much they consume.
- Easy to maintain. Different from owning a data center, cloud service customers do not need to worry about maintaining the infrastructure. Or they can conduct maintenance work with just a few clicks.
- Elasticity. Individuals or organizations can access as much or little as they need, and scale up and down as required with only a few minutes’ notice.
- Improving accessibility and agility. Cloud data centers are deployed worldwide, so cloud service customers can enjoy lower latency and better experiences.

In the next two subsections, we leverage cloud computing for core network caching and RAN caching, respectively.

Cloud CDN

The emergence of cloud computing offers a natural way to extend the capabilities of CDNs and resolve the inherent limitations of CDNs mentioned earlier. To combine the merits of the CDN and cloud computing, cloud CDN was proposed as an architecture that migrates CDN into the cloud. We depict the evolution path from single server distribution to cloud CDN in Fig. 1. In a cloud CDN [5], virtual machines are carved out of an underlying hybrid cloud, forming a content distribution overlay. By migrating contents from CDN edge nodes to cloud data centers, now con-
tent consumers can obtain contents from cloud data centers instead of origin content servers or CDN servers. There are some existing cloud CDN providers, such as Rackspace Cloud Files and Amazon CloudFront.

We list the major advantages of cloud CDN, compared to traditional CDN, as follows:

• Inheriting the benefits of cloud computing. Since cloud CDN is an application of cloud computing, it carries on all the benefits, mentioned in the beginning of this section, of cloud computing.

• The ability to handle flash crowd traffic. With the rapid elasticity property of cloud computing, once a flash crowd is detected, a content provider can easily increase the computation and storage resources rent from a cloud service provider in real time.

Nevertheless, a disadvantage of cloud CDN is also introduced subsequently. To achieve better coverage and proximity, a content provider has to distribute its contents to multiple cloud service providers. This incurs additional content management and maintenance cost, since these multiple cloud service providers work independently.

CLOUD RAN WITH CACHING AS A SERVICE

By merging RAN and cloud computing together, cloud RANs (C-RANs) have been proposed as a prospective architecture for fifth generation (5G) wireless systems. A typical structure of C-RAN consists of three components: baseband unit (BBU) pool, fronthaul links, and remote radio heads (RRHs). The most significant innovation of C-RAN is utilizing a centralized cloud BBU pool, instead of the conventional distributed baseband processing devices co-located with the BSs, to conduct the computationally intensive baseband processing tasks. That means, in C-RAN, baseband signal processing functionalities are decoupled from RRHs, and RRHs just need to keep basic signal transmission and reception functionalities [6].

C-RAN possesses several advantages compared to conventional RAN. First, utilizing centralized signal processing in the BBU pool instead of the distributed BSs in the conventional RAN can significantly save capital and operating expenditure. Second, joint processing in the BBU pool and cooperative radio techniques over RRHs, which are interconnected via the BBU pool, improves the spectrum efficiency and link reliability, and the communication quality of mobile users. Third, applying cloud computing as the computing paradigm of the centralized BBU pool can reduce power consumption, and improve energy efficiency and hardware utilization through resource sharing.

Other than the advantages above, which are from the aspects of improving RAN performance, we are able to gain more benefits by further utilizing the BBU pool of C-RAN. The BBU pool, which consists of many general-purpose servers, redefines the RAN as a software defined environment instead of the conventional hardware defined infrastructure. The main functionality originally intended for the BBU pool is baseband processing. However, due to the software defined environment, it is easy to incorporate more functionalities into the BBU pool. In this article, we propose caching as a service (CaaS) as the functionality extension for C-RAN (Fig. 2), which means that some popular contents can be cached in the BBU pool, and content consumers can fetch these contents from an adjacent BBU pool [7].

As a new technique belonging to the category of RAN caching, C-RAN with CaaS can easily step over the natural limitations of core network caching mentioned later. Specifically, the BBU pool of C-RAN is located up to 20 to 40 km from the RRHs, and the RRHs are deployed close to the mobile users (typically within 100 m). That means, compared to core network caching, the content cached in the BBU pool is much “near-
er” to end consumers. Furthermore, the traffic amount in the backhaul can be reduced greatly, since most end consumers are able to obtain their requested contents in the BBU pool instead of in the core network.

When compared to the previous RAN caching technique (i.e., FemtoCaching), C-RAN with CaaS also has many strong points:

- The BBU pool can provide much higher caching size than those in the BSs or APs.
- The centralized BBU pool eliminates the distributed content placement problem over geographically dispersed BSs or APs, which is usually nontrivial.
- The RRHs still only need to keep basic signal transmission and reception functionalities, which saves the overhead of adding new hardware to BSs or APs (in FemtoCaching, additional hardware needs to be added to the current BSs or APs).

However, the prominent weak point of C-RAN with CaaS is also easily found. That is, it lacks the ability to reduce the fronthaul traffic amount. We discuss methods to overcome this weak point later.

With the assistance of cloud computing in both the core network and RAN, we depict an overall network structure of a cloud content-centric mobile network (CCMN) in Fig. 3. We conclude and compare different caching techniques in Table 1.

### Investigated Problems

In this section, we present a brief overview of the problems that have been investigated in CCMNs and cloud CCMNs over the decade and recently.

#### Two Classic Problems

Under a CDN structure, there are two main problems, which have been widely studied:

**Content Placement.** To duplicate the contents from a content provider to geographically distributed CDN edge nodes, a principal problem is “which content should be placed to which edge node.” A straightforward content placement scheme is the full-duplication approach, which means every CDN edge node caches all files from a content provider. However, this approach is usually impractical since it is too costly. An alternative scheme is the partial-duplication approach, under which every edge node only needs to cache a part of the whole content set. Sophisticated algorithms have been proposed to achieve the partial duplication content placement scheme, including popularity, object, empirical, or cluster-based methods.

**Request Routing.** Once contents have been distributed among CDN edge nodes, a main concern for content consumers is from where to fetch the desired content. A simple request routing method is directing the request to the “closest” server. However, this method may not always guarantee the best response time, especially from the aspect of load balancing. Lots of request routing algorithms have been proposed. We classify these algorithms into two categories: cost-blind and cost-aware. Cost-blind algorithms are normally heuristic and ignore the real-time system conditions, such as random balancing, round-robin, and weighted round-robin algorithms. Cost-aware algorithms are more accurate and complicated, which take currently available system resources into consideration, such as bandwidth, content, or energy-aware algorithms.

![Figure 3. The overall network structure of cloud CCMN.](image)

<table>
<thead>
<tr>
<th></th>
<th>Core network caching</th>
<th>RAN caching</th>
<th>C-RAN with CaaS</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CDN</td>
<td>Cloud CDN</td>
<td>FemtoCaching</td>
</tr>
<tr>
<td>Backhaul traffic reduction amount</td>
<td>None</td>
<td>None</td>
<td>High</td>
</tr>
<tr>
<td>Fronthaul traffic reduction amount</td>
<td>None</td>
<td>None</td>
<td>High</td>
</tr>
<tr>
<td>Caching size</td>
<td>High but fixed</td>
<td>High and elastic</td>
<td>Low and fixed</td>
</tr>
<tr>
<td>Handling flash crowd traffic</td>
<td>Incapable</td>
<td>Capable</td>
<td>Incapable</td>
</tr>
<tr>
<td>Content placement problem</td>
<td>Relatively simple</td>
<td>Relatively simple</td>
<td>Nontrivial</td>
</tr>
</tbody>
</table>

2 Actually, these two main problems also exist in FemtoCaching; here we take CDN as an example.
After the request route has been determined by those algorithms, the destinations of requests can be reached by varieties of mechanisms, such as HTTP redirection and Domain Name System (DNS)-based request routing.

Apart than these two classic problems, which exist in both CCMN and cloud CCMN, we now list two recently solved problems under cloud CCMN only.

**Handling Flash Crowd Traffic**

As mentioned earlier, handling flash crowd traffic is an unneglectable but thorny issue in CDN. With the network structure evolving from CDN to cloud CDN, we realize that this problem is now ready to be solved by utilizing the rapid elasticity characteristic of cloud computing.

In [5], we jointly study request routing and the flash crowd traffic detection problem. Specifically, we assume the request traffic can switch between two modes, normal mode and flash crowd mode. Our objective is to minimize Delay + $\beta$ Cost, where Delay stands for the average mean response time over the virtual machines (VMs), Cost includes the average computation and routing cost, and $\beta$ is a predefined weight between Delay and Cost. However, the objective Delay + $\beta$ Cost was originally formulated for single traffic mode only (i.e., either normal or flash crowd mode). Then we extend the objective formulation and ensure that the extended formulation is applicable for two switchable traffic modes. The extension is made by adopting the quickest detection [8] framework. In particular, a quickest detection problem includes a trade-off between detection delay and false alarm frequency. To strike this trade-off for our problem, we impose penalty functions, related to Delay + $\beta$ Cost, on detection delay and false alarm frequency, respectively. Finally, by solving the joint problem, we obtain the optimal request routing policy and change points (the points when traffic switches from normal to flash crowd mode and vice versa), respectively.

We test our proposed algorithm on a YouTube requests data set (Fig. 4a). In Fig. 4a, the dotted lines indicate two periods in which the average traffic arrival rate is significantly higher than other times. Those two periods can be treated as flash crowd modes. Then we compare the performance of our proposed algorithm (i.e., DRES) by solving the joint problem with the benchmark algorithms (N and F strategies), which are used in conventional CDNs as static resource allocation schemes in Fig. 4b. N strategy is based on the assumption that incoming requests always fall in the normal traffic mode, while F strategy inversely assumes requests are always in the flash crowd mode. Perfect strategy is the baseline of all algorithms, which assumes the system can detect the change points between traffic modes perfectly. The result shows that flash crowd traffic can be tackled effectively under the cloud CDN structure with our proposed algorithm.

**Resource Segmentation in the BBU Pool**

The BBU pool of C-RAN consists of many general-purpose servers, and each server has two main resources: computation resource (e.g., CPU and memory) and storage resource (e.g., hard disks). The two types of resources are limited within each server. In C-RAN with CaaS, servers can be segmented into two different types of VMs using virtualization [9]. The first type is the computation-intensive VM (CIVM), mainly for baseband processing. A CIVM needs a high amount of computation resource but only a small amount of storage resource. The second type is the storage-intensive VM (SIVM), mainly for content caching. An SIVM has the opposite resource requirements as a CIVM: low computation resource requirement but high storage resource requirement.

Therefore, under C-RAN with CaaS, there are two challenges:

- How to optimally segment each server’s computation and storage resources into two different VM types. In other words, how many CIVMs and SIVMs should be assigned in each server (Fig. 5a)?
- How does content caching affect baseband processing and wireless transmission, since

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3 In practice, there can be many types of VMs. But for simplicity, here we only consider two types.
these three functionalities are co-located in the same physical system (i.e., the C-RAN)?

In [10], we jointly consider the above two problems as a system cost (including server cost, VM cost, and wireless transmission cost) minimization problem. We formulate the problem as a mixed-integer nonlinear programming problem and propose efficient algorithms to solve it. To be more specific, we capture the relationship between content caching and wireless transmission by defining a concept of cache-based adaptive rate (CBAR), in which wireless transmission rate is proportional to the totally assigned cache size in the BBU pool, that is, the higher the cache size, the lower the wireless transmission rate. The intuition behind the CBAR is that if the cache size is smaller than the expected file length to be transmitted, the system should speed up the wireless transmission rate to compensate the potential file retrieval delay in the core network. We show the result of our proposed algorithm in Fig. 5b. It can be concluded from the result that first, to save system cost, not every server has to be turned on, and second, some of the servers may be segmented into just one type of VM.

OPEN CHALLENGES
The study of cloud CCMN is still in its infancy, especially the study of C-RAN with CaaS. We list some open challenges in this section.

REVISITING THE TWO CLASSIC PROBLEMS
Earlier, two classic problems, content placement and request routing, in CDN have been elaborated. However, in the cloud CCMN, these two classic problems have to be revisited since cloud computing brings new features into CCMN. For example, the computation and storage resources provisioning are now elastic, while previous studies on these two classic problems normally treat these resources as fixed.

Furthermore, the two classic problems also exist in C-RAN with CaaS. In particular, if we regard a BBU pool as a CDN edge node and assume that different BBU pools can work cooperatively, the content placement and request routing problems in multiple C-RANs with CaaS remain open.

HIERARCHICAL CONTENT PLACEMENT
Under cloud CCMN (Fig. 3), popular contents can be cached in either the core network, the RAN, or both of them. This forms a hierarchical caching structure. Under this structure, some potential questions have to be answered: for multiple mobile virtual network operators (MVNOs) and caching service for multiple content providers.

• For a content provider, what is the optimal leasing strategy to lease the caching size from clouds in both the core network and RAN to minimize its monetary cost? The leasing cost for a unit caching size may be different in cloud CDN and cloud RAN.

• With given caching size in both the core network and RAN, how is an optimal hierarchical caching strategy designed, that is, deciding a certain content should be cached whether in the core network only, the RAN only, both of them, or neither of them, to maximize the quality of experience of content consumers?

• Is it possible to jointly optimize leasing strategy and caching strategy in cloud CCMN?

PRICING SCHEMES FOR C-RAN
With the functionality extension (Fig. 2), a C-RAN now offers two services: RAN as a service (RANaaS) and CaaS. That means that the C-RAN owner can provide RAN service for multiple mobile virtual network operators (MVNOs) and caching service for multiple content providers.

All of these MVNOs and content providers lease resources from the C-RAN owner under a pay-as-you-go model. Under this scenario, a good pricing scheme is needed for the C-RAN owner. The current pricing scheme for cloud service providers, such as Amazon Elastic Compute Cloud (EC2), have three pricing schemes: pay-as-you-go (i.e., on-demand instances), subscription (i.e., reserved instances), and an auction-like spot scheme (i.e., spot instances). However, the pricing scheme for previous commercial clouds cannot be applied to C-RAN directly, especially for RANaaS, since there are some special properties included in wireless communications, such as spectrum and interference. Thus, a good pricing scheme for C-RAN is an open challenge.

Once the pricing scheme is fixed, there will be some follow-up problems:
• An MVNO or a content provider may aim to design an optimal leasing strategy for different numbers of instances under different pricing schemes to minimize its cost.

• The C-RAN owner may aim to maximize its profit by optimally segmenting its resources to different pricing schemes of different services.

Problems similar to the two problems above have been studied in previous commercial clouds, such as [11, 12]. However, the results in [11, 12] still cannot be utilized in C-RAN directly.

**Reducing the Fronthaul Traffic Amount**

As discussed earlier, leveraging C-RAN with CaaS can greatly reduce the traffic amount in backhaul. However, the amount of traffic in the fronthaul still remains unmitigated, which is the main drawback of C-RAN with CaaS compared to FemtoCaching.

To resolve this challenge, the following approaches can be resorted to:

**Cooperating with users’ device level caching.** To reduce the fronthaul traffic amount, a principal method is to push the cached contents further forward (i.e., caching contents in the “front” of the fronthaul). Due to the features of device-to-device (D2D) communication, D2D caching can be a decent complement for CRAN with CaaS, especially in the aspect of reducing the fronthaul traffic. But how to cooperate between CRAN with CaaS and D2D caching is still an open issue.

**Utilizing multicasting.** The current RAN is optimized from the aspect of unicasting to ensure that unique information can be delivered to specified individuals [1]. However, unicasting may become inefficient under some scenarios in CCMN (e.g., live video streaming), especially when the fronthaul capacity is scarce. In this article, we suggest that C-RAN with CaaS cache contents in the BBU pool, which offers another opportunity to implement multicasting services because of the centralized processing property of C-RAN. Therefore, utilizing multicasting in C-RAN with CaaS can reduce the fronthaul traffic amount as well. However, how to make full use of C-RAN for multicasting is still an open issue.

**Considering limited fronthaul capacity.** C-RAN facilitates coordinated multipoint (CoMP) transmission by its centralized BBU pool. In downlink CoMP, joint transmission is an effective technique to improve mobile users’ rate. However, joint transmission requires user data sharing among all the coordinated RRHs, which leads to demanding fronthaul capacity. Hence, to reduce the fronthaul traffic amount, people need to treat the fronthaul capacity as limited, although it is actually high, when designing the CoMP related algorithms in C-RAN.

**Improving Energy Efficiency and Reducing Latency**

As mentioned earlier, improving energy efficiency is one of the main merits of C-RAN. Many works have been studied the energy efficiency gain of C-RAN; for example, [13] identifies the energy efficiency in C-RAN under two different downlink transmission strategies, the data-sharing strategy and compression strategy. In the meanwhile, although FemtoCaching mainly aims to reduce backhaul traffic load and improve network throughput, its energy efficiency gain has been attracting extensive attention as well. Recently, the authors in [14] examined the condition when FemtoCaching can contribute to improving energy efficiency and analyzing maximal energy efficiency gain with FemtoCaching. However, the investigation of energy efficiency in C-RAN with CaaS is still open.

In the 5G era, a very stringent end-to-end latency requirement is expected. FemtoCaching is an effective way to help reduce end-to-end latency by moving contents “nearer” to mobile users. For instance, the problem of minimizing average content access delay under the in-network cache structure (which includes FemtoCaching) is explored in [15]. The results in [15] demonstrate that the average content access delay can be cut down by provisioning more cache size. Thus, a significant future study is how cloud CCMN helps reduce end-to-end latency, by considering hierarchical content placement and elastic caching size provisioning.

**Conclusion**

With the overwhelming video traffic in the upcoming future, content-centric mobile networks are likely to play a major role in future mobile networks. In this article, we start by analyzing the inherent limitations of previous core network and RAN caching techniques, respectively. We then bring cloud computing in both core network and RAN caching, which forms a cloud CCMN architecture. In particular, in a cloud CCMN, cloud CDN is used as the core network caching technique and C-RAN with CaaS is utilized as the RAN caching technique. In addition, we summarize some problems recently studied in cloud CCMN, and finally explore potential research directions in the future on cloud CCMN architecture.

**References**


BIographies

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Exploring Synergy between Communications, Caching, and Computing in 5G-Grade Deployments

The authors offer a first-hand tutorial on the most recent advances in content-centric networking, emerging user applications, as well as enabling system architectures. They bring into perspective additional important factors, such as user mobility patterns, aggressive application requirements, and associated operator deployment capabilities, to conduct comprehensive system-level analysis.

ABSTRACT

Decisive progress in 5G mobile technology, fueled by a rapid proliferation of computation-hungry and delay-sensitive services, puts economic pressure on the research community to rethink the fundamentals of underlying networking architectures. Along these lines, the first half of this article offers a first-hand tutorial on the most recent advances in content-centric networking, emerging user applications, as well as enabling system architectures. We establish that while significant progress has been made along the individual vectors of communications, caching, and computing, together with some promising steps in proposing hybrid functionalities, the ultimate synergy behind a fully integrated solution is not nearly well understood. Against this background, the second half of this work carefully brings into perspective additional important factors, such as user mobility patterns, aggressive application requirements, and associated operator deployment capabilities, to conduct comprehensive system-level analysis. Furthermore, supported by a full-fledged practical trial on a live cellular network, our systematic findings reveal the most dominant factors in converged 5G-grade communications, caching, and computing layouts, as well as indicate the natural optimization points for system operators to leverage the maximum available benefits.

Advances in Next-Generation Networking

With its regulatory timeline set by the International Telecommunication Union (ITU) and known as International Mobile Telecommunication (IMT), the standardization of novel fifth generation (5G) communications technology is finally at full speed in 2016. In line with that, the radio access network (RAN) 5G workshop held by the Third Generation Partnership Project (3GPP) in late 2015 defined the radio technology-related research roadmap to meet the proposed IMT-2020 milestones. Consequently, it is now the industry consensus that there will be a new, non-backward-compatible radio access technology as part of the 5G landscape. However, future 5G networks will be much more than yet another radio access standard, but rather an efficient integration of cross-domain networks to offer a sustainable solution for attracting new verticals beyond information and communications technology (ICT).

Going further, the emerging 5G interface will enable logical network slices within a unified communications ecosystem in stark contrast to a legacy collection of dedicated networks for different industries. This end-to-end network slicing should provide improved rate and latency performance, as well as cater for the more efficient use of wireless spectrum. Due to the dynamic and secure network slices, the integrated 5G system can deliver the needed flexibility to many diverse applications and services, thus radically transforming the existing business models. As the industry is currently answering the important questions of how to slice the network appropriately and at what granularity, it is becoming understood that network operators will take advantage of some of the already developed advanced technologies — including software-defined networking (SDN) and network functions virtualization (NFV) — to implement efficient network slicing.

Allowing the dynamic connection and configuration of various components, SDN is a relatively old technology (dating back to the 1990s), but only now do we have the computational power to finally put it to effective use. Building on top of modern high-volume servers, switches, storage, and cloud computing infrastructure, NFV is essentially the cloudification of the network itself, which has the power to virtualize entire classes of network node functions. With the inherent flexibility offered by SDN and NFV, prospective 5G operators may set up services quickly, and move them around as virtual machines in response to dynamic network demands.

Fueled by SDN, NFV, and network slicing, the communications and computing functionalities are beginning to converge within the 5G ecosystem, bringing up the notion of “computing for communications.” The latter concept leverages the synergy between the angles of communications and computing by addressing...
the challenge of efficient computation offloading over a wireless channel. With 5G-grade computation offloading, resource-constrained and energy-hungry user equipment will be able to migrate its heavy computation tasks to (nearby) resourceful servers. Hence, we are witnessing a dramatic paradigm shift from connection-oriented to content-oriented networking, which emphasizes data dissemination, storage, and retrieval capabilities, in contrast to past system architectures aimed solely at increased network capacity [5].

However, the aggressive bandwidth requirements of today’s and future user applications (which we discuss in the following section) keep pushing cellular network operators to respond promptly with decisive capacity scaling on their deployments. To this end, heterogeneous networks (HetNets) have recently matured as efficient system architectures, where tower-mounted macrocell base stations (BSs) for ubiquitous coverage and network management are complemented across the same geographical area with small cells of different sizes and by various radio access technologies to improve capacity [2]. Hence, contemporary HetNets allow innovations should allow the network to be brought closer to the actual content prosumers (producers-consumers) as well as enable network operators to further densify their deployments, especially in urban areas.

While ultra-dense small cell deployments with their improved area spectral efficiency do mitigate the backhaul capacity, at the same time they challenge the backhauling efficiency [3]. For many operators, deploying high-speed backhaul between increasingly large numbers of small cells becomes prohibitively expensive. In order to prevent the backhaul capacity from becoming the 5G system bottleneck (especially during peak traffic hours), caching at the BSs can be employed for providing content to users instead of straining the backhaul connections. As it is becoming recognized that caching is indeed an efficient solution to alleviate the backhaul capacity requirement — so that relevant data are deployed during off-peak hours in the caches and then accessed during peak traffic hours by the users — there is strong uncertainty around which solutions are most suitable.

Generally, there is a range of alternative architecture choices (some outlined in the course of this article) taking advantage of content reuse to replace backhaul connectivity with storage capabilities. In the end, a system may be desired where small cell BSs with low-rate backhaul but high storage capacity cache the most popular user content. Then backhaul connections can only be utilized to update the cached content at a rate proportional to how the overall demand distribution evolves over time [4]. Further powered by recent progress in affordable memory capacity, transparent caching in strategic locations should allow the speed-up of content distribution as well as improvement of network resource utilization, even when users do not request the same content simultaneously (i.e., leveraging the temporal variability of network traffic). We continue by understanding the origins of such variability.

**EMERGING USER APPLICATIONS AND SERVICES**

**TRANSFORMED CONTENT ACQUISITION HABITS**

As powerful smartphones and tablets increasingly permeate the fabric of our lives, humans are also taking more time utilizing them in their daily routines. Today, time spent using smartphones already exceeds web usage on computers. Indeed, a typical image of the last century was to see everyone reading a newspaper while commuting. Presently, this is forgotten, and people have reverted irrevocably to reading news online on their capable handheld devices. With news going mobile and more real time, the underlying business models begin to evolve as well, resulting in shorter publishing cycles and heavier multimedia streaming content. This, in turn, creates repetitive downloads of popular content, such as breaking news and online blockbusters, thus leading to excessively redundant data streaming.

Mobile reading as an emerging method of news discovery is but one example of how people share similarity in terms of content semantics and geography. Another example is represented by massive downloads of a new iOS release, which produces the biggest data spikes seen on the Internet so far. Beyond adopting mobile devices for news and collectively acquiring popular files, today’s Internet traffic shows a dramatic influx of on-demand video streaming. Global services, such as YouTube and Netflix, are already watched by millions and have in fact spawned a new generation of video consumption habits. For instance, Netflix — the world’s market leader for subscription video on demand — had over 60 million paying subscribers in the middle of 2015, and this number is predicted to double by the end of 2020.

However, very different from live streaming, with on-demand streaming people do not request the same content simultaneously. This important property, known as asynchronous content reuse, means that a few popular files account for the lion’s share of the overall traffic. Indeed, there is evidence that typical user demands concentrate on a relatively small library of files [5]. Furthermore, streaming video on demand requests are highly redundant over time and space, which accentuates the need to deepen explore the current statistical traffic properties and changed user content acquisition patterns. Many recent sources reveal that contemporary cellular technology and service providers are not yet capable of delivering seamless, cost-effective, and scalable on-demand video streaming as the underlying Internet architecture is still based on the historic end-to-end model, and we continue by introducing the associated challenges in the following subsection.

**CONTEMPORARY TECHNOLOGY CHALLENGES REVEALED**

In current cellular networks, mobile users located at the cell edges already suffer from high energy consumption due to the aggressive transmit powers, and are further disadvantaged by excessive latency in acquiring their desired content over a wireless access network. To make matters worse, humans are particularly sensitive to delay and jitter. The large data providers in the market, including Google and Akamai, which own

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1 Refer to “The Business Case for Caching in 4G LTE Networks,” white paper prepared for LSI by Hag Safari
a multi-billion dollar business built on content delivery networks, recognize that low latency is a key ingredient for the satisfaction of their users. Real-life examples from the Internet confirm that a modest increase in latency can decrease revenues for an Internet service significantly.\(^2\)

To enhance user delay experience, caching popular content at the wireless edge (i.e., small cell BSs) could be employed, thus mitigating the disproportion between the available and demanded wireless capacity [6]. However, when deciding exactly what and where to cache, unpredictable human mobility may complicate the process profoundly, especially since people increasingly rely on their portable and handheld equipment. Together with understanding mobility, the knowledge of user location is equally important to efficiently serve scenarios with co-located devices: passengers with mobile gadgets using public transit services; groups of people in a shopping mall, stadium, and airport; and so on. In the end, it is crucial to consider multiple real-world factors for effective content caching (e.g., popularity distributions, location, velocity, and mobility patterns) in order to accommodate challenging use cases with stringent quality of service (QoS) and computational requirements.

Today, there is already extreme diversity of user applications (with many more to come) that demand extensive computation and continuous processing of the collected data. These include car navigation systems, image processing for electronic games, video processing on smartphones, object recognition on mobile robots, speech synthesis, natural language processing, and wearable computing. However, further development of these novel 5G-grade applications and services is inherently constrained by the computing efficiency of current user equipment, which is not expected to scale indefinitely due to the fundamental limitations in form factor and battery life. Hence, these novel computation-hungry services will inevitably have to rely on advanced computation offloading capabilities and need to be carefully provisioned in emerging network architectures. We discuss their two characteristic classes in the next subsection.

**Characteristic Classes of 5G Applications and Services**

A particularly challenging use case in the above context, enabled by the latest advances in wearable display and computing technology, is augmented reality (AR), which opens the door to truly interactive user experience (Fig. 1). In contrast to virtual reality, AR aims at supplementing the real world, rather than creating an entirely artificial environment. To this end, physical objects in the individual’s surroundings become the backdrop and target items for computer-generated annotations, which requires complex real-time calculations. In light of the ongoing content delivery transformation spawning a myriad of computation-heavy and delay-sensitive applications [7], we propose to differentiate between two large classes of use cases based on whether the data flow is triggered by a user or by its surrounding network infrastructure.

**Pull-Based Use Cases:** This category includes user-initiated services (Fig. 1), such as multimedia processing for work and entertainment. Example applications range from editing and creating multimedia content in social networks by amateurs to serving the needs of roaming “deskless” workers up to potentially allowing for hands-free operation (e.g., in medicine, manufacturing, service). Common to all these use cases, the required functionality is “pulled” from its surrounding network by an individual user or a group of users on demand (i.e., similar to the PlanGrid solution for construction engineers).

**Push-Based Use Cases:** In this category, we collect network-initiated services (Fig. 1), including location-based viral advertising, hazardous environment monitoring, context-aware computing, and mobile AR scenarios. The corresponding applications vary from offering best-effort news and information services to providing real-time capability of object recognition and visualized digital information (spanning the areas of gaming and infotainment, utilities, service and education, guidance, etc.). These use cases commonly assume that the network proactively “pushes” certain services onto a user in a serendipitous fashion.

We proceed further with reviewing the recent progress in enabling network architectures to facilitate these use cases.

![Figure 1. Overview and synergy of emerging 5G-grade solutions.](image-url)
ly boosted computation offloading capabilities. Building on the virtualization of computationally intensive processing, CC leverages the possibility to run multiple operating systems and applications on the same machine(s), while guaranteeing isolation and protection of the programs and their data. This has been instrumental in migrating computation-heavy user applications into the more resourceful cloud and thus has led to enormous economic success. Today, modern cloud service providers enable their users to elastically utilize resources in an on-demand fashion, including infrastructures, platforms, and software [8].

Further integration of CC into the mobile environment has given rise to mobile CC, which allows many practical challenges related to performance, environment, and security to be overcome [9]. However, the major limitation of contemporary mobile CC solutions is high end-to-end latency experienced during data delivery (including access, transport, and server delay components). To this end, current wireless access networks may introduce extra unwanted latency due to the regulatory constraints on the available spectrum resources. As a result, cloud service providers are lacking cost-effective means to scale the bandwidth offered to their users, thus making mobile CC services cumbersome to deploy and maintain for handheld and wearable user equipment.

To improve performance on their access networks, mobile operators are taking many decisive steps outlined earlier. However, the ongoing race for a more efficient radio access technology leads to a situation where last-mile wireless connections are sometimes upgraded considerably faster than the corresponding backbone infrastructure. Therefore, some network operators do not have sufficient backhaul capacity (deployment type I in Fig. 2), and communicating data to the Internet and back across their deployments takes significant time. A more coherent (but costly) upgrade strategy involves enhancing the backhaul capacity together with the last-mile links so as to allow for sustainable growth and support higher traffic loads. This typically requires that the operators deploy more abundant backhaul capacity (deployment type II in Fig. 2). We review the attractive operator choices in more detail further on.

**Figure 2.** Available system architecture choices and related network “tree” levels.

**CONTENT DISTRIBUTION ARCHITECTURES OF TODAY**

An important line of development in networking architecture is related to content distribution [10]. Starting from the era of early peer-to-peer (P2P) overlays, contemporary Internet communications pays more attention to the content itself rather than where it is located physically. Powered by Akamai, content delivery networks (CDNs) support anycast methodology by choosing the most appropriate (i.e., topologically close) content replica to achieve self-organized, adaptive, and fault-tolerant content distribution. Furthermore, the concept of information-centric networking (ICN) has developed as a general infrastructure that provides in-network caching so that content is distributed in a scalable, cost-efficient, and secure manner. The essence of these advancements lies in decoupling content from its hosts (or their locations) not at the application layer, but rather at the network layer.

The above is a distinct departure from the conventional client-server architecture, where a client always moves its computational tasks to a more powerful server. However, if the CDN nodes are placed in the core network, there may be insufficient throughput at the wireless edge, and thus a lack of reliable, low-latency service on wireless links. This still remains reality as mobile (cell edge) users often have to run all the processing locally in their devices and then save the execution result on external memory sticks and flash drives for further sharing. Therefore, it has been quickly recognized that the network may need additional architecture options for deploying caches as well as performing data processing.

In light of the above, embedding caching and computing capabilities into heterogeneous wireless networks may achieve significant reduction in response times by mirroring data/service in various locations and in effect bringing the resources (radio access, storage, and computation cycles) closer to where they are actually used [11]. In addition, in-network caching allows shifting traffic from peak to off-peak hours, thereby naturally mitigating load variability and reducing congestion. Such distributed local caches typically operate in two phases, the content placement (storage) and delivery, but may also require extra system-wide information (e.g., hop count and content popularity distribution), which substantiates the need for intelligent caching strategies. As a result, predictive in-network caches support location transparency, facilitate efficient content...
Deploying cloudlets incurs extra costs for their installation and maintenance, and this does not offer any means to handle user mobility. An alternative approach is merging the cloud computing frameworks and the small cell networks as part of another concept named “femto-cloud” computing, where home eNodeBs would support the cloudlet functionality.

**STATE OF THE ART IN 5G-GRADE “COMPUTING CACHES”**

Facilitated by the all-IP nature of contemporary 3GPP Long Term Evolution (LTE) cellular networks, two types of locations appear attractive for deploying 5G-grade caches [12]:

- The Evolved Packet Core (EPC), which consists of the serving gateway (S-GW), the packet data network gateway (P-GW), and the mobility management entity (MME)
- The RAN, which features evolved NodeB (eNodeB) BSs

In some cases, it may also be beneficial to combine the caching functionality with the matching processing power, especially when an application requires repeated bursty access to a remote server or other complex interactions. We name the corresponding architecture node a computing cache, which is essentially a virtualized resource available in the 5G network and targeted specifically at remote execution of end-user applications.

Recent literature has been rich in proposing other hybrid deployments of communications, caching, and computing functionalities. Ever since the pioneering work in [13], various options for a mobile device to cyberforage by finding surrogate (i.e., helper) servers in the environment have been considered. Proposing to move computation resources closer to the user devices, the concept of a cloudlet has emerged offering mobile handsets the possibility to access nearby static resourceful computers, linked to a remote cloud with high-speed wired connections. Within the cloudlet vision, such helper servers would be located in public and commercial spaces (airports, train stations, cafes, etc.) where people congregate casually. Hence, user devices can offload their computations to a nearby server, at low latency and high bandwidth, rather than pushing them to the cloud.

However, deploying cloudlets incurs extra costs for their installation and maintenance, and this does not offer any means to handle user mobility. An alternative approach is merging the CC frameworks and the small cell networks as part of another concept named femto-cloud computing, where home eNodeBs would support the cloudlet functionality. Femto-clouds enable a capillary distribution of the CC capabilities, closer to the actual mobile clients. In complement, caching the content library at femtocell stations (so-called femtocaching [4]), and even in the mobile devices themselves, has demonstrated particular benefits by alleviating the backhaul requirement in HetNets [14]. Femtocaching has the potential to solve the network scalability challenge by providing user rates with better scaling behavior. In summary, Fig. 1 supports our above discussion on the relationship and synergy between communications, caching, and computing with an overview of the latest research progress.

In addition, a vast body of works has concentrated on development of advanced mobile content caching and delivery techniques, as well as focused on improving network resource utilization. However, all the relevant practical factors need to be taken into account comprehensively to leverage the full synergy of the converged communications, caching, and computing architecture, including the structure of content requests, cost per backhaul connection and operating costs, user mobility control, requirements of running applications, and so on. Inspired by this, in the rest of this article our aim is to offer a unique system-level analysis of such integrated architecture, supported by a live measurement campaign.

**REPRESENTATIVE SCENARIOS AND THEIR EVALUATION**

**EVALUATION METHODOLOGY AND ASSUMPTIONS**

Characterizing the converged communications, caching, and computing functionalities, we investigate two representative use cases belonging to the two classes introduced earlier:

- Streaming context-aware AR data (scenario 1, push-based)
- Using a web-based application, such as Adobe Photoshop (PS) cloud (scenario 2, pull-based)

Both example applications require intensive computations, which are cumbersome to run on small-scale user equipment and thus have to be offloaded. To this end, we assume that both storage and computation resources may in principle be located in the LTE network (RAN, EPC, etc.). An end-user session spawns small-size packets: files containing the extracted features for image classification and environment recognition, or PS brush track reports translated into formal commands.

With our detailed system-level simulations, we recreate an urban area of interest (or tracking area) where active users are moving according to a certain random walk model (calibrated with practical measurements in the following subsection). As a reference, we employ fractional Brownian motion with positively-correlated increments (Hurst parameter $H = 0.9$). Hence, our users tend to preserve their movement directions as they keep interacting with the network continuously. In scenario 1, the appropriate content holder is determined by the current geographical position of the user, while for scenario 2 a session with a particular content holder has a geometrically distributed duration with the average of 30 min.

In particular, for scenario 1 we consider continuous computing and data acquisition (i.e., a user’s wearable camera captures the context and annotations are “pushed” by the network) as the user moves across the area of interest. By contrast, for scenario 2, the “pull” requests are sent in ON/OFF fashion, such as when the user is drawing with the PS brush so that remote service is demanded. The period between the requests is taken as 33 ms and 50 ms for scenarios 1 and 2, respectively. For AR, the video frame size that has to be downloaded is 67 kb (i.e., video rate of 2 Mb/s at 30 fps rate), whereas for PS we assume a series of requests during the exponential ON periods with the average of 3 s, and the “silent” exponential OFF periods with the average of 6 s.
From the connectivity perspective, mobile users can communicate with a microcell BS if they are located within its coverage area. Alternatively, users may also connect to one of the femtocell BSs deployed across the tracking area according to a certain stationary repulsion point process. As a characteristic example of femto BS distribution, we consider a Strauss process with the inhibition coefficient 0.9 and the inhibition distance of 90 m. Given that cellular network topology is hierarchial, we further adopt the following abstraction of its structure. We represent the entire operator’s network as a forest of trees, where a certain tree (Fig. 2) corresponds to a particular access network “branch,” while “leaves” denote the end-user devices. Enumerating the network “tree” levels, we call the user level “level 0,” the RAN levels (femto and micro BSs) “level 1” and “level 2,” and further on through the aggregation nodes in the backbone network to the EPC (the root level is “level K”).

The network structure in our evaluation is instantiated with the typical numbers of descendants expected of a real operator network (10,10,30,8,variable) starting with EPC descendants and all the way down. The resources of any node are shared fairly between all the active descendants at a lower level (including the user level). In case of a backhaul bottleneck at some level, the maximum possible throughput of every user is decreased proportionally. Furthermore, both data storage and remote processing nodes may in principle be located at any given level of our network topology. Here, we assume that they are always deployed together (co-located) at every node of a certain level, thus mimicking the “computing cache” functionality discussed earlier.

Generally, the end-to-end latency comprises the time to:

- Upload the request \( t_{UL} \)
- Perform the calculations \( t_{compute} \) (either remotely or at the user device)
- Download the final result \( t_{DL} \)

The request timings that form \( t_{UL} \) at all levels are given in Table 1; \( t_{DL} \) and \( t_{compute} \) depend on the system load and are explained below. To estimate latency, we introduce the rates \( R_i \) to traverse the network tree from level \( i-1 \) to level \( i \) (shared between all the active descendants), which are directly related to the system load. Hence, \( R_i \) denotes the individual maximum data rate of a user at the femto BS, \( R_i \) is the rate on a backhaul connection between the femto and the micro BSs, while the rest of the network “tree” edges are wired.

The processing capacity of the computing nodes can be provisioned by the service provider appropriately based on the available funds per user. Therefore, our abstracted model postulates that a certain server at level 1 may process a computational task during 5 ms if there are no other requests. For level \( i \), \( i > 1 \), since the available computation resource is assumed to scale linearly, the user requests can be served proportionally faster due to additional parallel servers. However, a constant overhead of 10 ms is added to this variable delay regardless.

### Table 1. Key system parameters.

<table>
<thead>
<tr>
<th>Description</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of levels</td>
<td>5</td>
</tr>
<tr>
<td>Femtocell density</td>
<td>8/microcell</td>
</tr>
<tr>
<td>Effective user density</td>
<td>560 people/km²</td>
</tr>
<tr>
<td>Femtocell radius</td>
<td>50 m</td>
</tr>
<tr>
<td>Microcell radius</td>
<td>200 m</td>
</tr>
<tr>
<td>User speed</td>
<td>3 km/h</td>
</tr>
<tr>
<td>LTE femtocell capacity</td>
<td>10/10/30 Mb/s</td>
</tr>
<tr>
<td>LTE microcell backhaul</td>
<td>5/0/800 Mb/s</td>
</tr>
<tr>
<td>Aggregate 1 backhaul</td>
<td>30/30/300 Gb/s</td>
</tr>
<tr>
<td>Aggregate 2 backhaul</td>
<td>300/300/30 Gb/s</td>
</tr>
<tr>
<td>EPC capacity</td>
<td></td>
</tr>
<tr>
<td>LTE RAN latency</td>
<td>7 ms</td>
</tr>
<tr>
<td>Femto-micro BS latency</td>
<td>3 ms</td>
</tr>
<tr>
<td>Small-scale aggregator latency</td>
<td>0.5 ms</td>
</tr>
<tr>
<td>Large-scale aggregator latency</td>
<td>2 ms</td>
</tr>
</tbody>
</table>

In order to understand real-life user movement behavior and its impact on the performance of our envisioned system, we additionally implemented a full-scale user mobility study. The motivation behind this trial has been to collect live data from end-user devices connected to our open cellular network so as to reveal the effective frequency of serving cell changes by our test population of users. Furthermore, we employed thus collected information as a calibration dataset for a more detailed system-wide evaluation reported at length in this section. The present investigation of practical user mobility builds on our rich hands-on experience acquired during our recent implementation work in [15], which utilizes all of the key functionality expected of 3GPP LTE Release 10.

Our employed LTE testbed (Fig. 3b) is composed of:
- The RAN part, including several small cells
- The EPC part
- The IP multimedia subsystem (IMS) part

To provide a complete and unbiased picture, we also performed supporting experiments in other public mobile LTE networks served by telecommunication operators of the Czech Republic, including Telefónica O2, Vodafone, and T-Mobile. As our test user equipment, we utilized Samsung Galaxy S3 and S4, Jiayu S3 Advanced, as well as Samsung Galaxy Note 4 devices. In addition, we created an assessment tool in Java to collect live information on the cell ID, location of eNodeBs as the location area code (LAC) parameter, received signal strength, and connection latency between a user and the server.

Based on the results of the trial, we are convinced that user mobility is one of the most crucial factors in the present system performance evaluation. Hence, the obtained live measurements were processed to extract the values of the cell residence time (i.e., how long a user spends in one cell before changing it). Then we employed these data in our simulation study discussed in the previous subsection to yield substantiated conclusions on the practical system behavior. We report on our assessment by visualizing the collected results in the form of a day-time scattergram (Fig. 3a) for the cell residence...
Fig. 3. Trial implementation and interpretation of measurements.

times during business hours. Some of the test users demonstrate interesting variations in their mobility patterns, but on the whole the assembled data follow the trend of the unified sample on which we focus in the rest of this discussion.

Another curious finding is that the behavior of the random residence time is not stationary and might alter throughout the day (see box-and-whisker diagram in Fig. 3a for morning and afternoon hours). This is due to the characteristic habits of our test group: the participants tend to move more actively in the morning hours. Further, we note that the empirical probability density function as illustrated in Fig. 3c is very different from the standard exponential distribution, since the coefficient of variation for our sample is much higher than 1. This leads to the need of using more complex fitting options coming from the class of phase-type (PH) distributions, that is, mixtures of distributions. After calibrating with the experimental data, we continue by reporting our most important simulation-based findings.

SELECTED NUMERICAL RESULTS

To quantify the scaling laws behind the discussed use cases, we consider the system, where the relevant storage and processing functionalities are assumed to be available for a user at a particular “computing cache” node in the network (named “content holder”) for both Scenarios 1 and 2, as well as the three different network operator profiles (Fig. 4):

- Sufficient RAN and backhaul capacity (over-provisioned network, where the capacity of a higher-level node equals the total capacity of its subordinate nodes)
- Moderate RAN capacity and insufficient backhaul capacity (capacity of a higher-level node is decreased with respect to the total capacity of its subordinate nodes)
- Insufficient RAN and backhaul capacity (capacity scales down even more severely)

The corresponding deployment parameters are summarized in Table 1.

In real-world networks, computation delay decreases as the associated processing node is placed higher in the network “tree” (due to aggregating multiple computational tasks and allocating more resources). Hence, we expect that offloading to the higher “tree” levels may be more beneficial for the user in that respect. However, the data communications delay is always a non-decreasing function of the “tree” level index, and strongly depends on the current network load as well. Within the two considered scenarios, as user interactions with the network are rather intense, smaller network capacities may have difficulty in supporting the offered traffic load, thus creating an incentive to move the resources closer to the edge. These two conflicting objectives lead to nontrivial results, which also depend on other important factors.

In particular, the computing delay — which is higher at RAN nodes — impacts the total latency in an underloaded (well provisioned) network (Figs. 4a, 4d), thus moving the delay-optimal “computing cache” placement point toward the EPC. However, with degraded network capacity (Figs. 4b, 4c and then Figs. 4c, 4f), the computing delay loses its importance to the communications delay, which becomes the dominant factor in determining the user QoS. We confirm that the optimal “computing cache” level in a highly loaded network is at the edge (level 1 or 2, depending on the available computation resources), while in a more lightly loaded system the optimality point shifts to “higher” aggregation nodes. Furthermore, the more intensive data communications is, the sooner the optimal point slides toward the edge.

For our practical setup, we conclude that a typical network hardly copes with the latency requirements on the order of tens of milliseconds. Similarly, a legacy network with insufficient backhaul capacity cannot support the real-time restrictions of our Scenario 1 (AR). However, Scenario 2 (PS) may operate satisfactorily in all the considered deployments, since it is not as delay-critical and throughput-hungry. As seen in Fig. 5d, for the more user-mobility-sensitive Scenario 2, the transport delay at the edge of the network increases due to the fact that the user has to communicate with its original content holder farther away across the network. This certainly has a negative impact on the network
load as further demonstrated in Fig. 5b for the moderate-capacity operator deployment. The latter figure highlights the relative load on certain network levels (i.e., the “loaded level”) when the “computing cache” is deployed at the “storage level.”

In Fig. 5a, Scenario 2 creates more than twice as much extra load at “higher” levels when the “computing cache” is placed at the femtocell level. In contrast, Fig. 5a corresponds to Scenario 1 with its shorter intervals between the changes of the content holder (related to the cell residence times). This results in a stronger correlation of the user’s location with that of the helping network node. Note that an additional small portion of load (highlighted separately in the top left corner) is produced by serving users outside of the femtocell coverage. Importantly, due to the higher load of AR and insufficient backhaul capacity, the system bottlenecks (i.e., 99 percent loading) impact the service rate and the network load compared to when the computing cache resides at the edge, which leads to increasing delays (as we have seen in Fig. 4). For a resourceful and well provisioned operator infrastructure, one should expect a “flat” surface of backhaul load.

In summary, we learn that in case of sufficient operator network capacity, the deployment of computing cache nodes at “higher” levels would provide better end-user performance, likely at reduced equipment costs (computing cache nodes may be hosted on already existing server hardware), but then causing a significant load on the RAN (i.e., distribution network). For networks with insufficient backhaul capacity, attractive performance gains are only seen when the computing cache is brought closer to the user, as this mitigates congestion in the distribution network. However, there may be additional deployment costs, which have to be considered when the system is provisioned.

**Main Outcomes and Conclusions**

As our results conclusively indicate, appropriate deployment of the computing cache functionality in next-generation cellular networks does not have a single universal answer. To adequately quantify the cornerstone questions of what and where to cache, as well as how many computation nodes should be made available and at what level, we considered the realistic provisioning of the emerging 5G-grade applications and services, such as AR and offloaded computation. Not limited to a simple illustration of the attainable gains, the considered scenarios represent the two distinct classes of push-based and pull-based use cases introduced early on in this article.

Our subsequent numerical findings suggest that computing cache nodes have to be deployed by 5G system operators in a manner consistent with a broad range of practical aspects, many of which have been considered in synergy by this work for the first time. The most important of such factors are outlined below.

**Network Capabilities:** This includes the serving operator infrastructure, from the core down to femtocells, the capacity of all the transit nodes and RAN, the effective coverage ranges and cell density, as well as the computation and caching capabilities together with the cost of their deployment.
**Application Requirements:** Further attention should be paid to the actual service needs, including rate and delay, as well as the structure of computation demand in active periods.

**User Dynamics:** Other important factors are related to the number of active prosumers, their mobility (including speed and preferred movement patterns), as well as characteristic roaming behavior and cell residence times.

In summary, the analysis conducted in this article reveals that the optimal position of the computing cache nodes is determined by the application to be supported, the user mobility pattern, as well as the backhaul and RAN dimensioning.

More specifically, for the low-bandwidth, high-persistence use cases oriented toward computing (running longer than a user’s residence time in a particular area of interest), such as those illustrated by our Photoshop scenario, 5G operators should avoid the use of femtocells as computing cache nodes, contrary to popular belief. This is particularly true in a high-capacity network, where “vertical” backhaul transport delays are minimal, and the impact of handovers as well as low computational power of the femtocells become dominant. The general guideline for scenarios of this type is to deploy the computing cache functionality higher up in the distribution network, such that the more powerful computing nodes could be used, and the handover overheads would remain minimal.

However, for the emerging high-bandwidth location-bound services oriented at storage, such as our AR scenario, a different deployment strategy is preferred. Due to the properties of the AR use case, it becomes significantly more efficient to place the computing cache as close to the end user as possible. Surprisingly, even in this scenario femtocell-level caching is not always the best option, since considerable handover overheads may still exist in reality. In general, however, for download-oriented cases there is a reasonable motivation for the deployment of cell-level computing caches.

Overall, based on our results, it could be recommended that a 5G operator deploy different kinds of computing and caching solutions for various scenarios. In particular, cell-level computing caches should be deployed for bandwidth-hungry applications, whereas the higher-level computing cache positions should be considered for high-persistence, computation-oriented services. Furthermore, if the network is underprovisioned (i.e., its backhaul capacity is reduced considerably), the computing cache deployment choice has little effect on the overall service quality delivered to customers. It is thus imperative that the backhaul capacity between the cells housing the computing caches is sufficient to support handover-related traffic, and the QoS levels drop significantly.

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Enhancement for Content Delivery with Proximity Communications in Caching Enabled Wireless Networks: Architecture and Challenges

Min Sheng, Chao Xu, Junyu Liu, Jiongjiong Song, Xiao Ma, and Jiandong Li

ABSTRACT

To cater for the exploding growth of video traffic, small cell base stations (SBSs) and device-to-device-enabled caching and delivery have been regarded as promising techniques for future wireless networks. In this article, we design a proximity communications enhanced multilayer caching and delivery architecture. Then merits proposed by the building blocks are highlighted, and challenges and open issues are comprehensively presented. Specifically, we shed light on the trade-offs between key performance indicators (e.g., hit ratio, latency, and coverage) and operation costs (e.g., device storage space, wireless bandwidth, and device battery life), and then clarify fundamental coupling between content caching and delivery. To further verify the effectiveness of the cooperation among SBSs and user equipments, we propose a distributed content caching and delivery strategy, jointly considering popularity distribution, diverse storage capability, and user mobility. Simulation results demonstrate that the proposed strategy can significantly lower the content retrieval latency and reduce the traffic flowing to core networks.

INTRODUCTION

The recent proliferation of smart mobile devices (smartphones, large-screen pads, tablets, etc.) and multimedia services (video on demand, live broadcast, social networks, etc.) are accelerating the growth of video traffic on the Internet and users' expectation of retrieving contents with freedom of movement. Meanwhile, according to a recent Ericsson mobility report [1], video traffic has been the largest and fastest growing segment of all mobile data traffic; by 2019, it will increase about 13 times more than that in 2013, and make up around 50 percent of global mobile data traffic. Therefore, reliable delivery of video contents over wireless networks will be an essential issue in the near future.

While the content delivery network (CDN) has been proposed as the most widely adopted approach to avoid congestion near the server for traditional wired video transmissions, it would be inappropriate to directly transplant this technique into wireless networks. This is mainly due to the fact that not only wired links accessing servers, but also backhaul links between the radio access network (RAN) and core network (CN) as well as wireless links for user equipments (UEs) in RANs may be the bottleneck, which limits the achievable date rate. To alleviate this, efficient content caching and delivery techniques should be tailored for wireless networks. According to state-of-the-art research, several promising methods have been proposed by exploiting the redundancy in the users' demands to reduce the duplication transmission through backhaul links, and additionally improve the spectrum efficiency (SE) and energy efficiency (EE) performance. According to the position where video contents are cached, available strategies can generally be classified into two categories: small cell base station (SBS) enabled caching and delivery [2] and device-to-device (D2D)-enabled caching and delivery [3].

SBS-ENABLED CACHING AND DELIVERY

Heterogeneity would be one key feature demonstrated by future wireless networks augmenting an existing macrocell by deploying SBSs with small coverage areas [4]. In this context, each individual SBS is equipped with limited storage for caching, and UEs can retrieve contents directly from the SBS instead of from the remote server. Such a method is called SBS-enabled caching and delivery. Obviously, SBS-enabled caching has advantages of offloading traffic load from a carrier CN and alleviating backhaul congestion by pre-caching contents closer to the UEs during low traffic hours. However, utilizing SBS caching can hardly lower the traffic load carried via RANs (i.e., the problem of scarce wireless bandwidth is still pressing). Additionally, content sharing among neighboring SBSs is not permitted according to current Third Generation...
Partnership Project (3GPP) standards since X2 interface is specified only for signaling interactions between some SBSs (e.g., Home eNodeBs [HeNBs]).

**D2D-Enabled Caching and Delivery**

D2D communication is a novel technique enabling two proximal UEs to communicate directly, which can increase network SE, reduce transmission delay, offload traffic from the BSs, and alleviate congestion in cellular networks [5, 6]. In the past few years, D2D communication has emerged as an important research topic, and nowadays, this technique for Long Term Evolution (LTE) technology is sufficiently mature and has been standardized in 3GPP Release 12 [7]. Inspired by the standardization of D2D communication and the trend that UEs will possess larger and larger storage spaces, D2D-enabled caching and delivery has also been considered as one promising technique for wireless networks. Applying this approach, UEs can cache contents locally in their own devices and then directly share contents with each other via D2D links. Obviously, this technique can inherit all the benefits promised by D2D communication. Meanwhile, UEs can cache contents according to their own preference or the group’s demands, which provide higher caching flexibility. Nevertheless, compared to SBSs, UEs generally have relatively small storage capability and a battery with limited energy, which may result in a high cost for content caching and delivery in practice.

**New Caching and Delivery Architecture**

By combining the advantages of both SBS and D2D communication techniques, we design a multilayer caching and delivery architecture that is compatible with the standards currently specified. Contents can be cached in both SBSs and UEs, and moreover, caching resources between neighbor cells can be pooled together with the aid of multihop D2D communications, which can break through the limitation inserted by current specified standards. With this architecture, efficient multilayer content caching and delivery methods can be implemented, and the users’ experience can be further improved.

In the rest of this article, we first present our newly proposed architecture, the distinctive content delivery modes, and the related signaling interaction processes. Next, the open issues in this architecture are highlighted, and the state-of-the-art video caching and delivery strategies for wireless communications are summarized. Following that, a tailored cooperative caching and delivery strategy is devised to reduce the latency perceived by users; meanwhile, its effectiveness is evaluated through simulation results as well as testbed experiments. Finally, conclusions are drawn and possible extensions are briefly discussed.

**Multilayer Content Caching and Delivery Architecture**

We consider a multilayer architecture as shown in Fig. 1, which consists of devices in accordance with those specified by current 3GPP standards, including macrocell BS (MBS), SBS, UE, and so on. Here, SBSs are connected to the MBS via backhaul links, and the MBS connects to the gateway of a CN through a high-speed interface. Meanwhile, individual SBSs and UEs are equipped with storage spaces for content caching. Here, the caching resources between neighboring cells could be well joined with the aid of D2D communications. Based on this architecture, we next provide the corresponding typical distinctive delivery modes and present the enabling signaling procedures in detail.

**Distinctive Content Delivery Modes**

When requesting a content, the UE first searches for the content in its local storage. If the required content cannot be found, one of the following five distinctive delivery modes would be triggered. It should be noted that our proposed architecture also has full backward compatibility with the traditional MBS assisted delivery mode. To be specific, if a user cannot find the desired content in other UEs or SBSs, it will send a request to an MBS to fetch this content from a remote server.

**Neighbor D2D delivery (Fig. 1a).** One UE can retrieve contents via D2D links from nearby UEs. Higher UE density can lead to a higher probability that the required contents can be provided, thereby achieving the multi-UE caching diversity gain. In addition, allowing direct transmission between UEs in different cells can further improve the user experience, as some user requirements can be satisfied by adjacent cells instead of the associated SBS.

**Multihop delivery through D2D relay (Fig. 1b).** If neighbor UEs fail to find the required contents in their storage, they can serve as a relay for content delivery. This relay-based mechanism allows a broader range of cooperative caching and delivery, which improves the utilization of caching resources in each cell.

**Cooperative D2D delivery (Fig. 1c).** When one content’s duplications are cached in multiple UEs, these UEs can cooperatively deliver content by a D2D multiple-input multiple-output (MIMO) technique to provide a higher transmission rate. Furthermore, if the features of some video coding techniques (e.g., multiple description coding [MDC] and scalable video coding [SVC]) are considered, more benefits can be reaped [4].
We design three function modules in application layer protocol to enable the five previously depicted delivery modes, which are called REQProxy, ContentServer, and GateServer, respectively. REQProxy works as a proxy server that recognizes UEs’ demands and then requests the ContentServer for desired contents. ContentServer is used for content searching and forwarding. To be more specific, ContentServer first checks whether the required contents are in the local storage and transmits the contents to UEs if they are hit. Otherwise, the requirements will be sent to other UEs or SBSs. For GateServer embedded in SBSs, it is used to send requests that cannot be locally handled to other SBSs or remote servers.

In the following, we take three typical delivery modes as examples in Fig. 2 to illustrate the related signaling interaction procedures.

**Cooperative D2D delivery.** When one UE asks for some content, it first sends the request (Content REQ) to nearby UEs. Then an individual receiver’s REQProxy exchanges information with its ContentServer. If the content is hit, the REQProxy will respond to the requester’s REQProxy with an acknowledgment (RESP ACK). Next, the content requester and provider establish a connection with another two packets, Establish Connection and Connection ACK. This process is applicable in the neighbor D2D delivery process when the number of UEs that respond with RESP ACK equals 1, and also in the direct SBS delivery process when the associated SBS responds with ACK.

**Cooperative SBS delivery.** To enable cooperative SBS delivery, the content requester would send its request to the associated SBS. If the content is not available there, this requirement would be sent to an MBS with the aid of GateServer. Then the virtual link between SBSs would be established. After that, the required content is cached by the associated SBS and finally delivered to the original requester.

**Multihop D2D delivery.** To enable this mode, the relay UE may establish a connection with its associated SBS. Once the SBS confirms that the required content is available, a connection ACK would be fed back to this UE to establish the connection. Finally, the multihop transmission is realized.

Note that the above procedures can be completed in a distributed fashion without introducing a centralized controller tracking the content delivery of all UEs and SBSs. In particular, only the pre-caching-related information of neighboring devices needs to be locally exchanged, and hence, the scalability can be guaranteed.

**Open issues in proposed architecture**

Before efficiently reaping the merits possessed by our proposed architecture, some issues need to be thoroughly considered and addressed. Next, we shed light on some of them.

**Coupling in content caching and delivery:**

**Coupling between SBS and D2D caching:** The storage spaces of SBSs and UEs are extremely limited when meeting the entire file library.
Therefore, deciding which contents should be cached in which places (SBSs or UEs) is non-trivial. Moreover, implementing our proposed architecture, the caching resources in neighbor cells can be pooled together to improve users’ experiences, which makes the caching decisions of adjacent SBSs and UEs complicated to couple. Hence, the caching problem becomes more challenging. To address this, both the popularity distribution of video contents and potential information “hiding” in social networks should be learned and exploited [9, 10].

Coupling between caching and delivery: In practice, content caching distribution is the foundation for delivery strategy design, while content delivery results in turn affect content caching and replacement. To be specific, the distribution of cached contents in networks will affect the delivery path selection of content requesters. On the other hand, the statistical effects of content delivery after a long time can be used to explore the popularity distribution, with which the caches can be periodically updated. In our proposed architecture, this coupling will further spread from one single cell to a larger region because of the enabled D2D communications between neighbor cells. Therefore, study of this coupling is extremely important and remains a challenging issue.

Diversity-Aware Content Caching and Delivery:

Diverse mobility feature: Considering the movement of UEs, the D2D link utilized for content delivery is more likely to be interrupted, since the period of transmission may be longer than the time that the content requester stays within the coverage of the provider. To handle this problem, it is more suitable for SBSs to cache contents with larger size.

Diverse storage capability: The memory size gap between UEs and SBSs will affect the number of devices involved for cooperative content caching and delivery. For instance, given the target hit ratio, a small number of UEs would be involved if their memory sizes are large (i.e., the memory size gap between UEs and SBSs is small). Otherwise, cooperation among more devices is required to achieve the same target, which may increase the expected latency and energy consumption since more hops are used for content delivery on average.

Trade-off between Performance and Cost:

Latency vs. storage space: Caching contents in UEs and delivering them with D2D links can greatly reduce retrieval latency. Nevertheless, it may lead to much occupation of device storage spaces for pre-caching. Therefore, it is nontrivial to reduce content retrieval latency while minimizing the occupied storage.

Hit ratio vs. wireless bandwidth: The more devices are involved for cooperation, the more diverse contents can be retrieved without accessing remote servers (i.e., improving hit ratio). However, more wireless bandwidth is required to support content delivery among the cooperating devices.

Coverage vs. battery life: The content can be retrieved only when the requester is in the transmission coverage of the UEs pre-caching the required content. On one hand, higher power is required to improve transmission coverage, which quickly drains the device battery. On the other hand, to improve the “content coverage,” UEs are likely to cache popular contents and serve more nearby UEs, which results in the excessive energy consumption and battery life reduction.

Content Caching and Delivery Strategies:
The state-of-the-art content caching and delivery strategies can be mainly classified as three categories according to their design motivations, that is, caching and delivery with higher hit ratio, lower latency, and better EE performance.

Caching and Delivery with Higher Hit Ratio: Intuitively, the contents with the highest popularity should be cached in SBSs so that most contents duplication can be retrieved from the local SBS. However, the popularity profiles cannot be known in advance because of the variable demands of users and published contents. Without accurate knowledge of this information, the content placement problem was formulated as a multi-armed bandit (MAB) problem [11]. Authors in [12] proposed a collaborative framework, where SBSs can form a collation and share contents with each other so that the system hit ratio is improved. Further considering the scarcity of bandwidth resource, the D2D communication technique was introduced in caching strategy design [13], with which the local hit ratio can be further improved.

Caching and Delivery with Lower Latency: Latency can be reduced by shortening the path between the content provider and requester (e.g., caching diverse contents in neighbor UEs as much as possible). The authors of [14] considered the caching decision combined with the wireless bandwidth allocation, which was aimed at matching the available bandwidth resources and caching distributions in the BS and UEs. The latency minimization problem was formulated as the problem of minimizing the traffic transmitted through BS downlink and backhaul, and the Branch-and-Bound algorithm was developed to find the optimum caching distribution. To simplify the implementation, a greedy method was also devised.

Energy-Efficient Caching and Delivery: Improving EE is one of the major concerns for D2D caching and delivery. The energy saving problem was considered in [15], where two kinds of video stream requirements are investigated: synchronous (e.g., watching a live show) and asynchronous (e.g., requiring video-on-demand services) transmission. On one hand, in the synchronous scenario, both the centralized and distributed clustering algorithms were proposed for EE concerns. With the former, the BS controls the clusters’ conformation and decides the optimal cluster head for multicasting to achieve the optimal EE performance. In contrast, the latter one is more proper for practical implementation, although some performance degradation may occur. On the other hand, in each asynchronous D2D cluster, there is no cluster head, and the UEs independently cache video files to maximize the hit ratio.

Considering the movement of UEs, the D2D link utilized for content delivery is more likely to be interrupted, since the period of transmission may be longer than the time that the content requester stays within the coverage of the provider. To handle this problem, it is more suitable for SBSs to cache contents with larger size.
One UE is not in the coverage of any SBS after moving, it will be cancelled, and a new UE will be re-generated.

Figure 3. Average contents retrieving latency of different strategies.

Figure 4. Average region hit ratio of different strategies.

**PROPOSED STRATEGY AND CONSTRUCTED TESTBED**

Although there have been some progress and initial achievements in content caching and delivery in wireless networks, there are still many challenges, specified earlier, that need to be addressed. Hence, novel strategies aiming to improve the system performance in terms of latency, hit ratio, energy efficiency, and so on should be further devised. In this section, we are mainly concerned with latency and have proposed a neighbor nodes and neighbor cells cooperative delivery (N3D) strategy. When applying N3D, each device (UE or SBS) records its own content-related information in a local table, called LocalTable. Information of each cached content, such as Node ID, content ID, content size, and location, is recorded. When one UE requires a content, it broadcasts the request packet to nearby UEs and its associated SBS. The SBS and UEs, which have the required content and are willing to implement the delivery, respond with ACK packets (RESP ACK) containing their LocalTables. The content requester selects the UE or SBS that first replies with RESP ACK as its provider.

We note that our proposed N3D is just a content delivery method, and can be paired with any caching strategy. In fact, considering the effects of storage space difference between SBSs and UEs as well as their individual mobility features, different caching and replacement strategies should be devised. In addition, although the content popularity profile can be characterized by, say, a Zipf-like distribution, it cannot be known in advance by each SBS in practice. To validate the effectiveness of N3D with simulation, we have also proposed a caching method considering all the above important factors, which is presented in detail later.

**SIMULATION RESULTS**

To verify the effectiveness of the N3D strategy, we compare its performance with those of the gateway search (GS) strategy and cell-based caching (CBC) strategy in this subsection. When GS strategy is implemented, UEs directly ask the gateway for required contents if the contents are not available in their own individual storage spaces. Applying CBC strategy, each UE can only retrieve the requested content from the associated SBS or remote servers if the UE has not pre-cached it. Note that both GS strategy and CBC strategy can be applied into our proposed multilayer architecture.

Here, we consider a square with a side $L = 300$ m under the coverage of one MBS, where $N = 36$ SBSs are uniformly deployed. Each SBS has a circular coverage of radius $r = 30$ m and serves on average $K = 30$ users. The size of content library is $M = 1000$, and the storage capacity of UE and SBS is $M_e = 20$ and $M_S = 100$, respectively. In each normalized time unit, there are $P_r \cdot K$ UEs making content requests, and the content popularity follows a Zipf distribution with parameter 0.8, where $P_r$ denotes the request probability and is set to 0.1. Considering the fact that SBSs cannot actually know the content popularity in advance, each SBS would randomly pre-cache $M_e$ contents from the library and then periodically replace the contents (e.g., in the period of low traffic) according to the content request frequency in history. In this simulation the replacement decision is made every $T = 100$ normalized time units. In addition, each UE would cache the previous requested content with probability $P_c$, and replace the content least requested by itself and neighbors with the new one when the storage is full. Here, the SBS is static, but UE will move at a fixed speed (no more than 1.5 m/s) and random direction with the angle from 0 to $\pi$.2

Figure 3 shows the average content retrieving latency when N3D, CBC, and GS are applied, respectively. The latency is normalized by the largest content delivery latency, achieved by the GS strategy. We notice that the latency decreases with increasing caching probability. This is because more content duplications in local storage will lead to a higher probability for local con-
tent sharing. Meanwhile, both N3D and CBC outperform the GS strategy. The reason is that when implementing N3D and CBS, content can be retrieved from either the associated SBS (CBC) or nearby devices (N3D), which leads to significant latency reduction.

Besides latency, we further evaluate the performance of N3D in terms of region hit ratio (RHR) in Fig. 4, where RHR indicates the ratio of locally retrieved contents without visiting remote content servers to the total requests. It is shown that our proposed N3D also outperforms the other benchmark strategies in terms of RHR; moreover, the gap between the RHR achieved by N3D and that achieved by GS increases with respect to the UE caching probability. This improvement is mainly due to the fact that more contents would be attained via D2D links when P_c increases but the reduction of latency is evident (Fig. 3). This is because the storage space of SBS is much larger than that of UE, and hence only a few new contents can be found in the UE, and RHR varies little when P_c increases. However, when P_c is larger, more contents can be locally obtained instead of being retrieved from the SBS, which reduces the delivery latency.

**Testbed Establishment and Strategy Evaluation**

To verify the validity of our proposed architecture and designed delivery strategy, we have set up a testbed consisting of five UEs and two SBSs (i.e., two small cells), as shown in Fig. 5. This testbed is placed in a 6 m × 10 m laboratory where WLAN links are used to emulate the links from SBS to UE and from UE to UE. Each SBS is equipped with a TP-LINK TL-WN322G+ plug-and-play WLAN card working on 2.4 GHz frequency and can access the Internet via the wired link connected to our campus network. Meanwhile, each UE is a PC only equipped with the TP-LINK WLAN card and hence has to first establish the connection with an SBS before accessing the Internet. The operating system of each device is Ubuntu 14.04, and all protocol modules are developed with C++ language. In order to emulate the practical backhaul of small cells, we have made a restriction on the transmission rate of the link from SBS to the campus network, which is set to be 200 kbps. We note that the scale of our testbed is relatively small due to the restrictions of the area of our test site and number of available physical devices, but all important functions of our architecture can be demonstrated via it (e.g., neighbor D2D delivery, multihop D2D delivery, and direct SBS delivery). Its scale could easily be enlarged when more UEs and SBSs are incorporated.

Here, we consider that UEs in cell 2 would request the video content only pre-cached in SBS. According to an earlier section, if the CBC and N3D are implemented, the available video delivery paths for one user may vary according to the order of other users’ requests. For instance, if the request made by UE4 is later than that by UE3 when CBC strategy is implemented, RHR varies little as P_c increases but the reduction of latency is evident (Fig. 3). This is because the storage space of SBS is much larger than that of UE, and hence only a few new contents can be found in the UE, and RHR varies little when P_c increases. However, when P_c is larger, more contents can be locally obtained instead of being retrieved from the SBS, which reduces the delivery latency.

<table>
<thead>
<tr>
<th>Test case index</th>
<th>Strategy</th>
<th>Normalized latency for UE3</th>
<th>Normalized latency for UE4</th>
<th>Normalized latency for UE5</th>
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</thead>
<tbody>
<tr>
<td>Case-A</td>
<td>GS</td>
<td>0.9718</td>
<td>0.9947</td>
<td>0.9816</td>
</tr>
<tr>
<td></td>
<td>CBC</td>
<td>1.0000</td>
<td>0.1100</td>
<td>0.0901</td>
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<tr>
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<td></td>
<td>CBC</td>
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<td>0.1089</td>
<td>0.3381</td>
</tr>
</tbody>
</table>

Table 1. The normalized latency for UE3, UE4, and UE5.
can be delivered via the direct D2D link (one hop). Therefore, for UE3, the underlying latency can be further reduced with N3D when the case changes from case A to case C.

**CONCLUDING REMARKS**

In this article, we have proposed a multilayer content caching and delivery architecture for future heterogeneous wireless networks, which reaps the benefits of both SBS and D2D techniques. This a fundamental framework enabling reliable content delivery by pooling together caching resources in neighbor small cells. To fully explore and exploit its merits, more concern should be focused on caching and delivery strategy design. For instance, compared to uncoded caching, coded caching is more flexible and storage-efficient but may require more computation resources [2]. Hence, different caching strategies would be suitable for devices with differentiated capabilities (e.g., computation ability and battery capacity). Meanwhile, delivery latency is the main performance metric studied in this article. However, when more types of content requests are considered, diverse metrics (e.g., SE and EE) should be defined, and proper trade-offs should be made on demand. To realize this, more information needs to be obtained and utilized, such as the dynamic channel conditions of wireless links and power consumption models of devices.

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ABSTRACT
As mobile services are shifting from connection-centric communications to content-centric communications, content-centric wireless networks emerging as a promising paradigm to evolve the current network architecture. Caching popular content at the wireless edge, including base stations and user terminals, provides an effective approach to alleviate the heavy burden on backhaul links, as well as lower delays and deployment costs. In contrast to wired networks, a unique characteristic of content-centric wireless networks (CCWNs) is the mobility of mobile users. While it has rarely been considered by existing works on caching design, user mobility contains various helpful side information that can be exploited to improve caching efficiency at both BSs and user terminals. In this article, we present a general framework for mobility-aware caching in CCWNs. Key properties of user mobility patterns that are useful for content caching are first identified, and then different design methodologies for mobility-aware caching are proposed. Moreover, two design examples are provided to illustrate the proposed framework in detail, and interesting future research directions are identified.

INTRODUCTION
Mobile data traffic is undergoing unprecedented growth, further propelled by the proliferation of smart mobile devices (e.g., smart phones and tablets). In particular, the data services subscribed to by mobile users have gradually shifted from connection-centric communications (e.g., phone calls and text messages) to content-centric communications (e.g., multimedia file sharing and video streaming). One main effort to meet such a strong demand is to boost the network capacity via network densification (i.e., to deploy more access points). While this approach is expected to significantly increase the capacity in future fifth generation (5G) networks, it incurs a tremendous demand for backhaul links that connect the access points to the backbone network. Thus, it will cause a heavy financial burden for mobile operators who are required to upgrade the backhaul network, and such a comprehensive approach will not be cost effective to handle content-centric mobile traffic, which may be bursty and regional. Consequently, a holistic approach is needed, and cache-enabled content-centric wireless networking emerges as an ideal solution.

Nowadays, abundant caching storage is available at the wireless edge, including both base stations (BSs) and user terminals (UTs), which can be used to store popular contents that will be repeatedly requested by users. Since the prices of caching devices, such as solid state drives (SSDs), have been coming down year after year, it has become more and more cost effective to deploy caches instead of laying high-capacity backhaul links [1]. Moreover, the ample storage at mobile UTs, currently as large as hundreds of gigabytes, is also a potential resource to be utilized for caching. Besides reducing the demand and deployment costs of backhaul links, caching popular content is also an effective technique to lower delays and reduce network congestion [2], since mobile users may acquire the required files from the serving BSs or the proximal UTs directly without connecting to the backbone network.

The idea of content-centric networking has already been explored in wired networks, where named pieces of content are directly routed and delivered at the packet level, and content packets are automatically cached at routers along the delivery path. Accordingly, caching design at the routers, including content placement and update, is crucial to system performance. Caching at the wireless edge can draw lessons from its wired counterpart, but it also enjoys new features. The broadcast nature of radio propagation will fundamentally affect the content caching and file delivery, which has recently attracted significant attention. Another important feature of content-centric wireless networks (CCWNs) is user mobility, which has been less well studied. While mobility imposes additional difficulties on caching design in CCWNs, it also brings about new opportunities. User mobility has been proven to be a useful feature for wireless network design; for example, it has been utilized to improve the routing protocol in wireless ad hoc networks [3]. Unfortunately, most previous studies on caching design in CCWNs ignored user mobility and assumed fixed network topologies.
Figure 1. A sample cache-enabled CCWN. A mobile user may download the requested file from the BSs or UTs along its movement path that have this file in cache. Once the requested files match the cached data, transmissions over the backhaul network will be avoided. Otherwise, mobile users have to request from the central controller via backhaul links.

which cannot capture the actual scenario. There have been initial efforts on caching designs by incorporating user mobility [4]. However, only some special properties of user mobility patterns were addressed, and there is a lack of systematic investigation.

The main objective of this article is to provide a systematic framework that can take advantage of user mobility to improve the caching efficiency in CCWNs. Specifically, a comprehensive discussion of spatial and temporal properties of user mobility patterns is first provided, each of which is linked to specific caching design problems. We then propose mobility-aware caching strategies, with two typical design cases as examples. Finally, we identify some future research directions.

**Exploiting User Mobility in Cache-Enabled CCWNs**

In this section, we illustrate the importance of considering user mobility when designing caching strategies in CCWNs. A sample cache-enabled CCWN is shown in Fig. 1, where both BSs and UTs have cache storage and are able to cache some pieces of content from the file library. In the following, we first introduce the main caching design problems in CCWNs and then identify important properties of the user mobility patterns, and associate them with different caching problems.

**Key Design Problems of Caching in CCWNs**

The fundamental problem in caching design for CCWNs is to determine where and what to cache. The design principles may depend on different types of side information, including long-term information obtained from observations over a long period of time, such as the statistics of users’ requests and average communication times with BSs and other UTs, and short-term information generated by instant changes (e.g., instantaneous channel state information and real-time location information). The collection of long-term information incurs low overhead, while the usage of short-term information can provide better performance but requires frequent update. In the following, we categorize different caching design problems in CCWNs according to the timeliness of the available information.

**Caching Content Placement**: Caching content placement typically relies on long-term system information and is used to determine how to effectively pre-cache content in the available storage. To reduce overhead, the update of side information and caching content will not be very frequent. It is normally assumed that the long-term file popularity distribution is known a priori, and the network topology can be either fixed or subject to some assumptions in order to simplify the design.

Previous works have provided some insights into caching content placement at BSs. In particular, without cooperation among BSs, the optimal caching strategy is to store the most popular files [5]. However, if users are able to access several BSs, each user will see a different but correlated aggregate cache, and in this scenario, allocating files to different BSs becomes nontrivial. Moreover, the coded caching scheme, where segments of fountain-encoded versions of the original file are cached [5], outperforms the uncoded caching scheme where only complete files are cached. By carefully designing the caching content placement via combining multiple files with a given logic operator, different requests can be served by a single multicast transmission [6], which results in a significant performance improvement compared to the uncoded scheme.

Meanwhile, caching content placement at UTs is also attracting noticeable attention. Caching at UTs may allow users to download requested content in a more efficient way with device-to-device (D2D) communications, where proximal users communicate with each other directly. Compared to caching at BSs, the advantages of caching at UTs come from the lower deployment costs and automatic promotion of the storage capacity when the UT density increases, as the ensemble of UTs forms an aggregate cache; while the drawbacks include the difficulty of motivating UTs to join the aggregate cache, and the more complicated randomness in the D2D scenario. Pioneering works have shed light on caching content placement at UTs [7].

However, it is noted that previous studies rarely considered user mobility, which can be tracked without much difficulty with today’s technologies. If we could make use of long-term statistics of user mobility, such as the average steady-state probability distribution over BSs, the efficiency of content caching will be significantly improved.

**Caching Content Update**: Although long-term information incurs low overhead to obtain, it contains less fine-grained information, which may also expire after a period of time and thus cannot assure accuracy. For example, the BS-UT or UT-UT connectivity topology may change quickly due to the movement of UTs. Consequently, it may cause significant errors by using the expired long-term information to design caching strategies. If short-term information is available, such as the real-time information of the file requests and transmission links, caching content can
be updated to provide a better experience for mobile users. In the following, we introduce two caching content update problems.

**Adaptive Caching:** Since caching storage is limited, it is critical to replace stale caching content to improve caching efficiency. Common adaptive caching schemes to increase the cache hit ratio include replacing the least recently used content and the least likely requested content [8]. Another typical application of adaptive caching is to serve users that follow regular mobility patterns and have highly predictable requirements. When the mobility regularity and request preference of mobile users are known, BSs can update the caching content according to the estimation of future requests. The main challenges come from the accurate prediction of users’ future positions and requirements, the frequency of conducting the adaptive caching strategy, as well as the replacement priorities for the caching content.

**Proactive Caching:** In practice, a user can only download a portion of its requested file rather than the entire file from a BS, as a moving user may not have enough communication time with the BS. Proactive caching aims to provide seamless handover and downloading for users by pre-fetching the requested content at the BSs that will be along the users’ future paths with a high probability. Nevertheless, user requests and locations are usually unknown in realistic environments, and thus the accuracy of location prediction is critical to performance.

**Modeling User Mobility Patterns**

As can be inferred from the above discussions, taking user mobility into consideration is critical for caching design in CCWNs. In this subsection, we provide detailed descriptions of different user mobility properties, which can be classified into two categories, the spatial and temporal properties. The spatial properties contain the information of user mobility patterns related to physical locations, while the temporal properties characterize the time-related features.

**Spatial Properties:** The mobility pattern of a mobile user can be visualized by the *user trajectory* (i.e., the user’s moving path). Crucial information for caching design in CCWNs, such as serving BSs, and distances between BSs and mobile users, can be obtained from the trajectories of the mobile users. It is an ongoing research topic to investigate realistic models for user trajectory (e.g., the random waypoint model in [9]). As an example, the trajectories of two mobile users are shown in Fig. 2a, which are based on data collected on a university campus.

The *cell transition*, which denotes the transition pattern of a user moving from one cell to another, implies the information of serving BSs for each mobile user, which is one of the most critical pieces of information in caching design at BSs. Compared to user trajectory, the cell transition contains less fine-grained information as the moving path inside each cell cannot be specified. It is appropriate to capture the transition property using a Markov chain model [10], where the number of states equals the number of BSs. In the Markov chain, one state denotes a specific user being served by a given BS, and the transition probabilities represent the probabilities of a specific user moving from the serving area of one BS to that of another BS.

Recently, it has been found that user mobility patterns also largely depend on the social relations among mobile users. For example, it was claimed in [11] that mobile users having relatively strong social ties are more likely to have similar trajectories. In [12], Musolesi et al. proposed a two-level mobility model, which first establishes a social graph, where the nodes represent mobile users and the weighted edges represent the strength of the social connection between mobile users. Then social groups are built, and mobile users in each group move

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Exploiting mobility for caching in CCWNs.

Table 1. Exploiting mobility for caching in CCWNs.

<table>
<thead>
<tr>
<th>Spatial properties</th>
<th>Temporal properties</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>User trajectory</td>
</tr>
<tr>
<td>Caching content</td>
<td>✓</td>
</tr>
<tr>
<td>placement at BSs</td>
<td></td>
</tr>
<tr>
<td>Caching content</td>
<td>–</td>
</tr>
<tr>
<td>placement at UTs</td>
<td></td>
</tr>
<tr>
<td>Adaptive caching</td>
<td>–</td>
</tr>
<tr>
<td>Proactive caching</td>
<td>✓</td>
</tr>
</tbody>
</table>

General: ✓ means that the mobility property can be utilized in the corresponding caching design problem, and – means that the mobility property may not be utilized.

Temporal Properties: To capture the information of the frequency and duration that two mobile users are connected with each other, the timeline of a pair of mobile users is represented by contact times and inter-contact times, where the contact times are defined as the time intervals during which mobile users are within the transmission range, and the inter-contact times are defined as the time intervals between two consecutive contact times. The timeline of two users shown in Fig. 2a is illustrated in Fig. 2c. Such a mobility model has been applied to routing problems in ad hoc networks. For instance, in [3], Conan et al. modeled locations of contact times in the timeline of each pair of mobile users as a Poisson process so as to capture the average pairwise inter-contact times in an ad hoc network.

The cell sojourn time denotes the time duration of a specific user served by a given BS, which may affect the amount of data that this user can receive from the BS. Figure 2b shows the cell sojourn times of the two users whose trajectories are shown in Fig. 2a. Specifically, in [10], Lee et al. provided an approach to obtain the sojourn time distributions according to the associated moving history of mobile users.

The user mobility pattern always possesses a periodic property, which can be exploited to tackle the adaptive caching problem. The return time, which is defined as the time for an arbitrary mobile user to return to a previous visited location, is considered as a measure to reflect the periodic property and the frequency of mobile users revisiting a given area. In [13], Gonzales et al. measured the distribution of the return time and figured out that the peaks of the return time probability are at 24 h, 48 h, and 72 h.

Exploiting Mobility for Caching in CCWNs

Built on the information given in the above two subsections, potential approaches are now proposed to take advantage of user mobility patterns to resolve different caching design problems in CCWNs, as summarized in Table 1.

Caching Content Placement at BSs: In CCWNs, as a user moves along a particular path, the user may download the requested file from all the BSs along this path, and different BSs may cooperatively cache this file to improve efficiency. For this purpose, the statistic and predictive information of the BSs along the user trajectory, which can be obtained based on user trajectory or cell transition probabilities, will be needed. Compared to cell transition probabilities, the user trajectory provides additional information (i.e., different transmission distances from BSs in different cells), which can help better design the BS cooperative caching in CCWNs. For example, different transmission distances may result in different transmission rates, which will affect the amount of data that can be downloaded from different BSs. Furthermore, the cell sojourn time is also a critical factor to determine the amount of data that can be delivered, and thus will also affect the caching content placement at BSs.

Caching Content Placement at UTs: By enabling caching at UTs, mobile users may get the requested files via D2D links. For caching design in such a setting, the information related to inter-user contacts is essential. In particular, the information about inter-contact times and contact times is valuable, and are further illustrated in this section. As shown in the next section, in addition, social relations may help to decrease a large network into several small social groups, and thus reduce the complexity of caching design. Meanwhile, social groups also imply some contact information, that is, mobile users in the same social group are more likely to have more contacts [14]. Thus, social group information can also be utilized to design caching content placement at UTs.

Adaptive Caching: The caching content can be adjusted adaptively based on the periodic mobility pattern, for which the knowledge of return times will be very useful. Moreover, mobile users in different social groups may have different content preferences. Thus, the mobility pattern of each social group can be utilized to improve the adaptive caching design. For example, in a restaurant, there may be several customer groups with different content preferences during different time periods; for example, elders may enjoy morning tea, students have lunch with friends, and office workers may have dinner together. The BSs around the restaurant may perform adaptive caching accordingly.

Proactive Caching: If the user trajectory or cell transition property can be estimated based on past data, the future serving BSs for mobile users can be predicted. In this way, if a mobile user requests a certain file, the BSs that are predicted to be on its future path may proactively cache the requested file, each with a certain segment, and then the user can download the file when passing by. While it may slightly increase the backhaul traffic, such proactive caching can significantly improve the caching efficiency and reduce download latency.

The above proposals are by no means complete. Nevertheless, they clearly indicate the great potential and importance of mobility-aware caching in CCWNs. We hope this discussion will inspire more follow-up investigations.
In this section, we present two specific design examples for mobility-aware caching content placement, including caching at BSs and caching at UTs. Sample numerical results will be provided to validate the effectiveness of utilizing user mobility patterns in wireless caching design problems.

**Mobility-Aware Caching at BSs**

We first consider utilizing the cell sojourn time information to design caching content placement at BSs, which may be macro BSs or femtocell BSs. A sample network is shown in Fig. 3a. For simplicity, we assume the transmission rate for each user is the same while passing by each BS, and cell sojourn times are estimated based on available data. Mobile users will request files in the file library based on their demands, which is assumed to follow a Zipf distribution. Both uncoded and coded caching schemes are considered. In the uncoded case, we assume that each file is either fully stored or not stored at each BS. In the coded case, rateless fountain codes are applied, where each BS may store part of a coded file, and the whole file can be recovered by collecting enough coded message of that file [5]. When a mobile user requests a file, the user will try to collect the requested file while passing by each BS. The proportion of the requested file that can be downloaded from a BS is limited by the transmission rate and the sojourn time in this cell, as well as the proportion of the requested file stored at this BS. We aim to minimize the cache failure probability, which is the probability that the mobile users cannot get the requested files from cached contents at BSs. The coded caching placement problem can be formulated as a convex optimization problem, while uncoded caching placement can be obtained by solving a mixed integer programming (MIP) problem.

We evaluate the performance of the proposed mobility-aware caching strategies based on a real-life data set of user mobility, which was obtained from the wireless network at Dartmouth College [2]. The following caching placement strategies are compared:

- Mobility-aware coded caching strategy, which is the proposed coded caching strategy obtained by solving a convex optimization problem.
- Mobility-aware uncoded caching strategy, which is the proposed uncoded caching strategy obtained by solving an MIP problem.
- MPC strategy, which is a heuristic caching strategy, for which each BS stores the most popular contents [8].

The comparison is shown in Fig. 4, where a larger value of the Zipf parameter $\gamma_p$ implies the requests from mobile users are more concentrated on the popular files. We see that the mobility-aware caching strategies outperform the heuristic caching strategy, and the performance gap expands with $\gamma_p$, which demonstrates the value of mobility information. Moreover, the coded caching strategy performs better than the uncoded caching strategy, which validates the advantage of coded caching.

There are many interesting problems for further investigation. For example, the user trajectory can be utilized to consider variant download rates, which will affect the amount of data obtained in different cells. In addition, based on the user trajectory, it is possible to jointly deal with the caching problem and interference management.
In this subsection, we focus on caching at UTs. We consider taking advantage of average inter-contact times among mobile users to improve the caching efficiency at UTs. An illustrative example is shown in Fig. 3b. The locations of contact times in the timeline for any two mobile users are modeled as a Poisson process, as in [3], where the intensity is estimated from the history data. For simplicity, the timelines for different pairs of mobile users are assumed to be independent, and each file is assumed to be either completely stored or not stored at each UT. Mobile users will request files in the file library based on their demands, which are assumed to follow a Zipf distribution. When a mobile user generates a request, it will first try to find the requested file in its own cache, and will then wait to encounter users storing the requested file. The delay time is defined as the time between when a user requests a file and when it encounters the first user storing the requested file. We assume that if the mobile user stores the requested file or its delay time is within a predetermined delay threshold, it will be served via D2D links; otherwise, it will get the file from the BS. To offload the traffic from BSs and encourage proximal D2D transmissions, we set the objective as to maximize the data offloading ratio, which is the fraction of users that can get requested files via D2D links. This turns out to be a challenging problem and falls in the category of monotone submodular maximization over a matroid constraint, which can be solved by a greedy algorithm with an approximation ratio of 1/2.

The performance of mobility-aware caching at UTs is evaluated based on a real-life data set, which was collected at the INFOCOM conference [15]. Considering that most requests occur in the daytime, we generate average inter-contact times according to the daytime data during the first day of the conference. The following caching placement strategies are compared:

- Mobility-aware greedy caching strategy, which is the proposed caching strategy using a greedy algorithm.
- Mobility-aware random caching strategy, which is similar to the random caching strategy proposed in [7]. In this strategy, each UT caches files according to a Zipf distribution with parameter \( \gamma_c \). The optimal value of \( \gamma_c \) is estimated from the fraction of users that can get requested files via D2D links, by solving a linear search.
- MPC strategy, which is the same as the one used in Fig. 4.

Based on the data during the daytime on the second day of the conference, the performance of three caching strategies is compared in Fig. 5 by varying the file request parameter. It shows that both mobility-aware caching strategies significantly outperform the MPC strategy, and the performance gain increases as \( \gamma_c \) increases. Furthermore, the mobility-aware greedy caching strategy has better performance than the mobility-aware random caching strategy, since the former strategy incorporates average pairwise inter-contact times more explicitly and allows more optimization variables. Through extensive simulations, we also observe that as the number of users increases, the data offloading ratio using mobility-aware caching strategies increases, while the MPC strategy remains the same. Meanwhile, the greedy caching strategy always outperforms the random one. This implies that better utilization of user mobility patterns can further improve the caching efficiency.

While this initial study provides promising results, lots of challenges remain. For example, since the number of mobile users in a CCWN is usually very large, collecting the pairwise inter-contact times will cause a high overhead. One potential solution is to decompose the large number of mobile users into several social groups, and then design caching content placement at UTs based on the inter-contact times of mobile users within the same social group. Moreover, coded caching strategies can also be applied, which is a promising approach to further optimize the caching efficiency.

**Conclusions and Future Directions**

In this article, we conduct a systematic study that investigates the exploitation of user mobility information in cache-enabled CCWNs. Useful spatial and temporal mobility properties are identified and linked to key caching design problems. Through two design examples, the advantages and effectiveness of mobility-aware caching are demonstrated. To fully exploit mobility information in CCWNs, more work will be needed, and the following are some potential future research directions.

**Joint caching content placement at the wireless edge**: In practice, many caching systems consist of more than one layer of caches, which leads to a more complicated hierarchical caching architecture. In CCWNs, while most existing works, as
well as our discussion in this article, treat caching at BSs and UTs as separate problems, a joint design of caching at both BSs and UTs will be essential to further improve the system performance.

**Dynamic user caching capacities:** Unlike BSs, the caching capacities at UTs may not be fixed, since they are related to the storage usage of mobile users, which are different from user to user and change over time. It is thus important to investigate how to adaptively cache according to the dynamic user caching capacities, while also taking user mobility into consideration.

**Big data analytics for mobility information extraction:** With the explosive growth of mobile devices, collecting user mobility information will generate huge amounts of data. Thus, big data analytics to extract the required mobility information is another challenge in mobility-aware caching. Meanwhile, accurate prediction is also critical. Although some existing user mobility models can predict the future mobility behavior via historical data (e.g., the Markov chain model in [10] can jointly predict the cell transition and cell sojourn time), more work will be needed, for example, on how to predict the user trajectory. It is also important to investigate how different mobility models will affect the performance of caching strategies.

**Privacy issues:** In order to take advantage of the user mobility pattern, some personal information (e.g., home locations and work place locations) may be divulged in the collected mobility information. This will certainly cause some privacy concerns. Thus, how to extract useful user mobility information without touching individual privacy is important. Location obfuscation and fake location injection mechanisms may serve as potential approaches for anonymous traces.

**REFERENCES**


**Figure 5.** Comparison of different caching content placement strategies at UTs with 78 mobile users and a file library consisting of 1000 files, while each UT can cache at most one file.

**BIographies**

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The authors demonstrate that the integrated network-based content dissemination scheme outperforms its CI-based counterpart in terms of both content delivery ratio and its various energy and delay metrics. Furthermore, the opportunistic network is capable of offloading a large fraction of tele-traffic from the overloaded CI-based network.

**ABSTRACT**

The classic centralized infrastructure (CI) exhibits low efficiency in disseminating the content of common interest across its requesters. In order to overcome the limitations of CI-based content dissemination, smart mobile devices are capable of activating direct opportunistic communications among mobile users, which returns in integrated cellular and opportunistic networks. During the content dissemination process, the social characteristics of multiple users, including their common interest in the content, their mobility patterns, their social ties, and their altruistic forwarding behaviors, should be carefully considered in order to design an efficient content dissemination scheme. We demonstrate that the integrated network-based content dissemination scheme outperforms its CI-based counterpart in terms of both content delivery ratio and its various energy and delay metrics. Furthermore, the opportunistic network is capable of offloading a large fraction of tele-traffic from the overloaded CI-based network.

**INTRODUCTION**

In typical scenarios multiple mobile users (MUs) are capable of forming a community of interest [1] for jointly requesting the content of common interest (CoCI). For instance, the crowd participating in the inauguration of the new Pope may form a community of interest in order to share close-up video clips of the Pope on the podium. Similarly, supporters in a football stadium may also form a community of interest so as to share video clips of a spectacular goal from different angles or the score updates from another stadium. At the time of writing, numerous mobile applications are capable of monitoring MUs' interests for the sake of disseminating their requested content and for enabling their interactions. However both the content dissemination and interactions among MUs have to be completed by the communication centralized infrastructure (CI).

CI-based content dissemination has the following three major limitations:

- Due to the sparse deployment of the CI in rural/disaster areas, MUs often suffer from intermittent connections to the CI. The CoCI can only be delivered to MUs when they enter the transmission range of the CI.
- Numerous dedicated channels have to be established in order to deliver the CoCI to multiple requesters. However, disseminating the same copies of the CoCI via dedicated channels results in a waste of precious resources and potential network congestion.
- Since the performance of the multicast/broadcast technique is dominated by that of the “worst link” [2], as the requesters of the CoCI become denser, the “worst link” becomes even worse. As a result, CI-based multicast/broadcast often suffers from a long dissemination delay in densely populated scenarios.

During the CI-based content dissemination process, some MUs might benefit from more opportunities to contact the CI or from having better connections to the CI and hence succeeding in receiving the CoCI earlier than their less privileged peers. As a result, activating direct communications among the MUs is capable of providing better opportunities for the unserved MUs to contact the CoCI owners and may also provide diversity gains in order to counteract the “worst link” phenomenon. Hence, based on the common interest of MUs, the opportunistic communication among the MUs themselves — either opportunistic contact or opportunistic multicast depending on the application scenarios — can be integrated into conventional CI-based content dissemination. To elaborate a little further:

- Common interests motivate MUs to share the CoCI with their hitherto unserved peers. MUs sharing common interest in a certain type of content might decide to form a community.
- Opportunistic contacts can be relied on for delivering the CoCI in large and sparsely populated areas. In this scenario, an unserved MU may only fetch the desired CoCI from a CoCI owner when he/she moves into the CoCI owner’s transmission range. Opportunistic contacts can be activated without any central control by the dedicated short-range communication stan-
In its own storage, moves within the area studied desired CoCI, where the owner saves the CoCI increase the chances of an MU acquiring the carry-and-forward protocol of [3] is adopted to connect adequately by the CI, which results in intermittent connectivity.

Due to the sparse deployment of CI in rural/disaster areas, MUs are not always covered adequately by the CI, which results in intermittent connectivity.

Apart from CI-based transmission, the store-carry-and-forward protocol of [3] is adopted to increase the chances of an MU acquiring the desired CoCI, where the owner saves the CoCI in its own storage, moves within the area studied by carrying the CoCI, and then forwards it to the unserved MU within its transmission range. Both the WiFi-Direct and Bluetooth techniques support the function of discovering peer devices. Following the device discovery stage, either the CoCI owner “pushes” the CoCI to the unserved MU, or the unserved MU “pulls” the CoCI from its owner. As a result, the unserved MU does not have to continuously request the CoCI. Since the mobility-dependent contacts among MUs are not predictable, they are referred to as opportunistic contacts. Figure 1a exemplifies the content dissemination process in integrated CI and opportunistic networks. After two observations of the mobile scene, as illustrated by Fig. 1a, all five MUs have successfully received the CoCI. Among these five MUs, MU1, MU3, and MU5 receive the CoCI from the CI deployed in this area. If the CI is the sole transmitter disseminating the CoCI, only these three MUs would successfully receive the CoCI, while the other two would fail to receive it. Hence, it is plausible that the integrated CI and opportunistic network becomes capable of outperforming its CI-based counterpart in terms of its successful CoCI delivery ratio.

It is also readily recognized that opportunistic contact aided content dissemination is affected by the following factors:

**Transmission range of MUs:** The transmission range of MUs is jointly determined by numerous factors, including the transmit power, the statistical channel attenuation, and the capability of the error correction decoder, just to name a few. The range is often assumed to be a constant specified by the different short-range transmission techniques.

**Transmission rate of opportunistic links:** Once a pair of MUs enter each other’s transmission range, an opportunistic link can be established for delivering information. Correspondingly, the transmission rate of this opportunistic link is determined by the specific set of parameters and techniques invoked in the physical layer as well as by the channel quality. However, if accurate power control is used, this parameter may be assumed to be near-constant.

**Maximum number of opportunistic links:**

![Diagram](image-url)
Multiple opportunistic links can be established for delivering the CoCI from the CoCI owner to the unserved MUs within its coverage range. The maximum number of opportunistic links supported by the CoCI owner’s device may affect the pace of the content dissemination process. However, in a large sparsely distributed area, it rarely happens that multiple unserved MUs enter the transmission range of the CoCI owner. Hence, the number of opportunistic links has a limited impact on the content dissemination in this specific scenario.

**File size of the CoCI:** The file size and transmission rate jointly determine the **downloading time** of the CoCI.

**Lifetime of the CoCI:** MUs are only interested in up-to-date content. Hence, after the informative lifetime of the specific content has expired, the MUs lose interest in it.

**Inter-contact duration of a pair of MUs:** Inter-contact duration specifies the length of a period between the instants when an MU pair leaving each other’s transmission range and when they re-enter the transmission range. A shorter inter-contact duration indicates that the MU pair is capable of making opportunistic contacts more frequently.

**Contact duration of a pair of MUs:** Contact duration specifies the period in which an MU pair is within each other’s transmission range. A longer contact duration may increase the chance of an MU pair completing a file transfer between each other.

By jointly considering all the aforementioned factors, an effective contact between an unserved MU and a CoCI owner is defined by simultaneously satisfying the following two conditions:

- The unserved MU enters the transmission range of the CoCI owner.
- The unserved MU stays within the transmission range of the CoCI owner for a period longer than the sufficient downloading time of the CoCI.

The content dissemination process in the integrated network can be modeled by a continuous-time pure-birth Markov chain (CT-PBMC) [3]. Relying on the CT-PBMC, we are capable of analyzing the various delay metrics as well as average successful delivery ratio, also bearing in mind when the CoCI expires.

Let us now present a crisp performance characterization. Observe from Fig. 1b that if more MUs participate in the integrated network-based content dissemination, the average delivery ratio is substantially improved. Furthermore, if the informative lifetime of the CoCI is prolonged, more MUs may successfully receive the desired CoCI before it expires.

### Offloading Tele-traffic from the CI

In contrast to the rural/disaster scenario studied in the previous section, we assume adequate coverage provided by the CI deployed in the large area studied, where the MUs are always capable of maintaining reliable connections to the CI. In this scenario, we can reasonably assume that the CoCI can be nearly instantly disseminated by the CI across the target community of interest.\(^2\) This CI-based approach requires having dedicated channels established between the MUs and their associated CI. However, disseminating the same copies of the CoCI via the limited number of dedicated channels imposes an increased burden on the CI. Fortunately, as discussed earlier, a CoCI may attract the interest of the MUs for the duration of its lifetime. MUs do not have to receive the CoCI at the same time. In contrast, disseminating the CoCI before it expires is expected to satisfy the requesters. As a result, exploiting the delay-tolerant nature of the CoCI, the opportunistic contact-based communication among the MUs can be relied on for delivering the CoCI to a large fraction of its requesters, which hence offloads tele-traffic from the overloaded CI. Furthermore, offloading tele-traffic from a high-power CI to a low-power opportunistic contact-based transmission is capable of significantly reducing energy consumption.

As detailed in [4], the optimal content dissemination approach may be described by the following procedure:

1. **Step 1:** The CI initially transmits the CoCI to some of the MUs in the opportunistic network.
2. **Step 2:** The CoCI is spontaneously disseminated via opportunistic contact-based communications among the MUs before it expires.
3. **Step 3:** At the very end of the CoCI’s lifetime, the CI is invoked again for delivering the CoCI to the hitherto unserved MUs.

In order to offload as much tele-traffic from the CI to the opportunistic networks as possible in step 2, our main task is to find an optimal MU receiver set with members initially receiving the CoCI from the CI in step 1.

**Social network analysis** assists us in identifying the most significant MUs in an opportunistic network. However, we have to first model the social ties among the MUs according to their contact patterns. As shown in Fig. 2a, the shaded boxes represent the contact duration between a pair of MUs. Several conventional metrics, that is, the contact frequency, average inter-contact duration, and average contact duration, have been recorded for characterizing how socially close a pair of MUs may be deemed to be according to their contact pattern. However, these conventional metrics cannot always accurately specify the social closeness of a pair of MUs. For example, observe from Fig. 2a that the pair of MUs associated with the contact pattern of (a-2) has a weaker social connection than those associated with (a-3) and (a-4) in terms of the contact frequency, average inter-contact duration, and average contact duration. However, these metrics all fail to characterize the difference between cases (a-3) and (a-4). In order to jointly consider the regularity of contact patterns with the above-mentioned conventional metrics, the so-called social pressure metric (SPM) was proposed in [5] for accurately modeling the social strength of a pair of MUs. Explicitly, given a specific period \(T\), the SPM between \(MU_i\) and \(MU_j\) is defined as \(SPM_{ij} = \frac{1}{T} \frac{1}{E_{ij}} \sum_{\tau=1}^{T} \frac{E_{ij}(\tau)}{E_{ij}(\tau)}\), where \(E_{ij}(\tau)\) is the \(\tau\)th inter-contact duration. The reciprocal of the SPM represents the weight of the social tie. With the aid of this SPM, we can finally decide that the pair of MUs associated with (a-4) has stronger social strength than that associated with (a-3).
Real-life social ties do correspond to the contact pattern of MUs. For example, people sharing strong social ties are more likely to be found in each other’s proximity. The contact pattern of MUs may change their real-life social ties. For example, people frequently being in each other’s proximity may indeed strengthen their social ties, while people rarely visiting each other may typically weaken their social ties. As a result, our SPM modeling closely reflects the relationship between the MUs’ contact pattern and their real-life social ties.

After evaluating the weight of the social ties between all pairs of MUs, we only retain the specific social ties having a higher weight than a predefined threshold. The resultant social network is uniquely and unambiguously characterised by a directed and weighted graph. Relying on the toolbox of social network analysis, we can finally rank all the MUs according to their social significance. An example of a social network is portrayed in Figs. 2b–2d, obtained by analyzing the contact patterns of MUs during the hour spanning from 8:00 to 9:00 a.m., on the opening day of INFOCOM 2006. Then the following three schemes may be conceived for selecting the initial MU receiver set by considering the MUs’ social significance:

**Out-degree-based selection:** The out-degree of an MU is defined as the number of social links that emerges from this MU and terminates at its neighbors in the social network. More explic-
The size of the initial receiver set vs. the number of MUs served by the opportunistic network. Contact patterns of the 78 mobile users are extracted from the mobility traces during 8:01–9:00 a.m. on the opening day of INFOCOM 2006. The opportunistic contacts between an arbitrary pair of MUs are activated by their Bluetooth devices, and the transmission rate of the opportunistic link is 1 Mb/s. Bluetooth is a half-duplex peer-to-peer communication technique. Hence, while a source is transmitting data to a target, no extra links can be established, even if another unserved MU enters the transmission range of the source. The file size of the CoCI is 40 Mbits (5 MB).

Figure 3 exemplifies the capability of the opportunistic contact-based communication of the MUs offloading the tele-traffic from the CI-based communication. Please refer to [3] for the detailed simulation settings of the scenario we would like to characterize next. Observe from Fig. 3 that the component-based selection outperforms all its counterparts in terms of its ability to offload the tele-traffic. According to both the out-degree-aided and the betweenness-based selections, the MUs of the initial receiver set all belong to the largest component. Hence, these two strategies have similar performance, as shown in Fig. 3. Furthermore, when the size of the initial receiver set is small, random selection performs worst, because it is likely to select isolated MUs. However, when the size of the initial receiver set is large, random selection becomes the second best scheme, because its initial receiver set is capable of covering many isolated components.

**EFFICIENT CONTENT DISSEMINATION IN DENSELY POPULATED AREAS**

In the scenarios shown earlier, where the MUs are sparsely distributed in a large area, the delay-tolerant nature of the CoCI-based communications and the opportunistic contact among the MUs are relied on for disseminating the CoCI across the set of requesters. In this section, a different scenario is considered, with the main characteristics summarized as follows:

- The MUs are densely distributed in a small area. The size of this small area is comparable to the transmission range of the MUs.
- An MU is connected to any of its peers by a single-hop/multihop link.
- The CoCI in this scenario is delay-sensitive.

CI-based multicast constitutes an effective technique for disseminating the delay-sensitive CoCI across its coverage area. When the population of MUs requesting the CoCI becomes dense, even CI-based multicast becomes inefficient. Due to its quality of service (QoS)-guaranteed nature, the performance of the CI-based multicast is dominated by the worst link in the set of links connecting this CI to all the requesters [2]. Since the MUs are more densely populated, the probability of the worst link becoming much worse is increased, as shown in Fig. 4a. As a result, CI-based multicast suffers from a long delay during disseminating the CoCI to all its requesters. Furthermore, during the CoCI dissemination process, the CI has to repeatedly multicast the CoCI until all the requesters successfully receive it; hence, the CI fails to satisfy other communication demands, which results in inefficient exploitation of the CI.

We can exploit the following facts in order to improve the CoCI delivery. During CI-based multicast, some of the MUs benefiting from statistically better channels may successfully receive the CoCI much earlier than their less privileged peers. If we could exploit the redundant copies of the CoCI in the storage buffer of these successful MUs and hence activate direct peer-to-peer (P2P) communications between them and their hitherto unserved peers, we may increase the diversity gain of CoCI delivery and thus overcome the “worst link” phenomenon.
often encountered in densely populated areas, as shown in Fig. 4a. Furthermore, we may also relieve the CI from the cumbersome multicasting task. Since we cannot predict which specific MUs may successfully receive the CoCI first, opportunistic multicasting is invoked for activating the P2P communications among the MUs by exploiting the broadcast nature of wireless channels. However, as also portrayed in Fig. 4a, we should note that if the number of requesters is not very high, activating P2P communications may not necessarily produce a better link than the worst link of CI-based multicast, so it may not improve the speed of content delivery.

In order to cooperatively disseminate the CoCI in densely populated areas, MUs form a community of interest by obeying the following conditions:
- MUs share a common interest in the same content.
- The CoCI is delay-sensitive.
- MUs roam within a bounded area having a relatively small size.

Let us elaborate a little further on modeling the MUs’ interests. The statistics of users’ YouTube viewing behaviors suggest that a small fraction of popular contents attract the interest of a large fraction of users [6]. We model an MU’s interest in a specific piece of content $i$ by the probability $Pr(C_i)$ of this MU requesting $C_i$ from the CI. If we have a range of contents $C_i | i = 1, ..., M$, their corresponding requesting probabilities $Pr(C_i) | i = 1, ..., M$ obey the Zipf distribution [7].

As shown in Fig. 4b, we characterize our integrated-network-aided hybrid content dissemination scheme in the scenario of densely populated areas. In the first stage of our regime, CI-based multicast is invoked for initially disseminating the CoCI to all the requesters. CI-based multicasting will be curtailed once at least a single MU requester successfully received it. Then, the CoCI owners cooperate with each other in order to disseminate the CoCI to the hitherto unserved MUs with the aid of opportunistic multicasting. Once an unserved MU successfully received the CoCI, it will join the CoCI-owner set and it will further multicast the CoCI during the next stage.

In order to avoid any unexpected collision and interference, a time-division multiple access (TDMA) scheme is implemented in the medium access control (MAC) layer. The related transmission frame structure is also portrayed in Fig. 4b, where a single time slot of a transmission frame only allows a single transmitter to forward the CoCI. Classic round-robin scheduling can be applied in the MAC layer due to its low-complexity nature. However, for the sake of further improving the attainable performance of the CoCI dissemination, we could carefully select the most suitable CoCI owners for forwarding the CoCI to the hitherto unserved MUs during the next stage of CoCI dissemination. As a result, we should evaluate the potential impact of a CoCI owner on all the unserved MUs with the aid of social network analysis tools [8]. During the second stage of the cooperative multicast aided CoCI dissemination portrayed in Fig. 4, the CI plays the role of a central controller in order to facilitate control signaling exchange and to efficiently schedule the transmissions of multiple CoCI multicasters.

In the physical layer, the channel effects are modeled by both the uncorrelated Rayleigh fading and the path loss. Since the MUs roam across the area studied, the path loss between the CI and an MU as well as that between a pair of MUs varies. The movement of MUs in this scenario can be modeled by the ubiquitous uniform mobility model, which has been widely adopted for the performance analysis of mobile ad hoc networks (MANETs) [9].

Furthermore, we shall incorporate MUs’ altruistic behavior into the modeling of the opportunistic multicast aided content dissemination process [10, 11]. We assume that at the beginning of each sub-stage portrayed in Fig. 4, a) Activating P2P link among MUs overcomes the “worst link” phenomenon; b) the hybrid content dissemination scheme and the TDMA-based frame structure.
4h, the CoCI owners may independently make a decision as to whether they are willing to forward the CoCI received with a probability of $q$, which is defined as the factor of altruism (FA). The MUs’ altruistic behavior is affected by numerous factors, such as the users’ interest in the content, their energy concerns, and a range of other random factors. Apart from the MUs’ altruistic behavior, the social ties among the MUs may also constrain the multicasting of the CoCI. Due to security and privacy concerns, an MU is only willing to share the CoCI with its social contacts. As demonstrated by [12], people’s social ties are largely determined by their geographic distances. The probability $p$ of a pair of persons sharing a social tie is proportional to $d^{-\alpha}$, where $d$ is the distance between this pair of persons, and $\alpha$ is the associated social exponent. By further considering the small world phenomenon exhibited by social networks, the social contacts of an MU are divided into short-range contacts and long-range contacts [13, 14]. Short-range contacts of an MU are formed by all the MUs within the neighborhood range of the target MU, while long-range contacts of an MU are formed by the MUs outside the neighborhood range of the target MU with a probability proportional to $d^{-\alpha}$.

In Fig. 5, we study the impact of content popularity on the average content dissemination delay when all the requesters successfully receive the CoCI. The CI in this example is a typical BS. We set the FA to 0.5 and the total number of MUs roaming within the area studied to $N = 100$. Furthermore, 10 different pieces of content are considered in this scenario, with rank one being the most popular piece and rank 10 being the least popular piece. Observe from Fig. 5 that our hybrid content dissemination scheme outperforms BS aided multicast when disseminating the most popular content. However, BS aided multicast still constitutes a more efficient option of disseminating the less popular pieces of content.

**Open Research Issues and Other Applications**

Several open problems still remain to be solved in the design of our integrated network-based content dissemination regime. In order to accurately estimate the associated social properties, MUs have to have an awareness of the global network and have to intelligently process this knowledge on their own devices. However, in a purely distributed network, this awareness can only be acquired by information exchange among MUs by relying on their opportunistic contacts, which is a slow process consuming a large amount of energy. Furthermore, the computing capability of a mobile device may turn out to be too limited to process the required knowledge right across the global network. In order to overcome these inefficiencies, MUs may rely on their specific “virtual machines” in the mobile cloud for exchanging information, sharing storage space, and carrying out computing tasks.

The MUs’ altruistic behaviors largely depend on the intensity of their interest. However, the correlation between the MUs’ interests and their resultant altruistic behaviors remains a mystery. Their mutual influence has to be translated into a mathematical model. Furthermore, the altruistic behavior can be viewed as a strategy adopted by an individual MU for striking a balance between the associated delay reduction and the energy dissipation. Game theory may come to serve for finding the optimal solution.

In order to compensate for the energy loss of MUs, we can enable them to harvest additional energy during the content dissemination process. For example, the device of an MU may harvest energy from the environment as a solar cell does. Alternatively, simultaneous information and energy transfer can be used during the content dissemination process, where the intermediate relay may harvest additional energy from the RF signal, as a reward for forwarding the content.

Disseminating the delay-sensitive content across large areas may impose other challenges. Due to the slow movement of the CoCI owners, their local multicasting cannot forward the CoCI to the set of distant and hitherto unserved MUs until they enter the proximity of their unserved peers. Therefore, the CoCI owners may become clustered, so none of them can efficiently forward the CoCI to the distant unserved MUs. In order to guard against this clustering effect, novel resource scheduling techniques may be designed by jointly considering both the CI’s resources and the MUs’ resources.

Despite the above mentioned challenges, our integrated CI and opportunistic networks may find numerous applications in practical scenarios. Apart from the content dissemination process designed for distributing the CoCI, our integrated networks may also be applied in disaster areas for uploading important messages. Furthermore, our integrated networks may be applied in vehicular ad hoc networks in order to disseminate important traffic information and facilitate the design of smart vehicles. Furthermore, our integrated networks may be introduced into aeronautical ad hoc networks in order to implement low-complexity information exchange among aircraft. Moreover, they may also be implemented in some communities, such as a university campus, railway stations, and airports, in order to disseminate important information. Last but not least, our integrated networks may improve the
charity project One Child One Laptop in order to further reduce the expense of delivering educational programs.

CONCLUSIONS
In this treatise, we study content dissemination in integrated CI and opportunistic networks by exploiting MU’s common interest and activating their opportunistic communications. With the aid of integrated networks, we may significantly increase the average successful delivery ratio of the CoCI and offload tele-traffic from the CI to the opportunistic network. We can also substantially reduce the content dissemination delay, and may reduce the total energy dissipated by exhaustively disseminating the CoCI to all its requesters, since the CI-based high-power transmission is offloaded to the low-power direct transmission among MUs. However, as the price of implementing the above mentioned benefits, the MUs have to rely on their own energy in order to assist in the content dissemination process, while the control overhead imposed by scheduling the direct communications among the MUs has to be increased. Interested readers may refer to the seminal references listed in Table 1 for more technical details.

REFERENCES

Table 1. Seminal references for further reading.

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The authors introduce a novel transport protocol for Content-Centric Networking (CCN), named Content Name-Based Mobile Real-Time Streaming (NMRTS). NMRTS utilizes "symbolic interest" for mobile users to receive real-time streaming without sending bursty requests, and the regular interests (CCN generally uses) to recover lost packets during handover.

NMRTS utilizes "symbolic interest" for mobile users to receive real-time streaming without sending bursty requests, and the regular interests (CCN generally uses) to recover lost packets during handover.

**Abstract**

In light of the rapid growth in mobile video traffic and demand for stable video viewing with higher video resolution, we introduce a novel transport protocol for content-centric networking (CCN), named Content Name-Based Mobile Real-Time Streaming (NMRTS). NMRTS utilizes "symbolic interest" for mobile users to receive real-time streaming without sending bursty requests, and the regular interests (CCN generally uses) to recover lost packets during handover. NMRTS can support seamless handover by adjusting the symbolic interest lifetime and its transmission frequency according to network conditions. Through simulations, we demonstrate that NMRTS reduces the amount of both control message transmission and redundant data traffic, and minimizes data reception delay, compared with the general CCN.

**Introduction**

Video streaming has been gaining popularity among mobile users. Because of the recent improvement in wireless network connection speeds, users expect the demand for more stable high quality streaming video services to be met more easily. This trend has accelerated the growth in video traffic transmitted over mobile networks. In fact, according to Cisco’s prediction [1], mobile video traffic will increase 13-fold between 2014 and 2019, accounting for 72 percent of total mobile data traffic, and furthermore, data traffic from wireless and mobile devices will exceed traffic from wired devices by 2019.

The rapid growth in mobile data traffic poses challenges for both mobile network operators and users on the current connection/host-oriented communication architecture. To guarantee high quality connectivity and better user quality of experience (QoE), mobile and fixed-line operators suffer from the increase of investments in the infrastructure and redundant traffic. In fact, it is reported that 15 to 60 percent of redundant traffic was observed in enterprise and university access links [2], which is likely to occur in mobile networks due to the significant increase in mobile data traffic. It is hard to promptly invest in improving connection capacity including wireless and wired back-haul links in terms of capital and running cost. It is thus indispensable to enable efficient video delivery by saving bandwidth consumption and eliminating redundant traffic. From mobile users’ point of view, QoE degradation is likely to occur by video service disruption or disconnection during handover. In addition, mobile users tend to experience longer delays since connection-oriented communication itself does not try to obtain desired data or content from nearby caching points.

Content-centric networking (CCN) [3] or named-data networking (NDN) [4], which has been introduced as an alternative to the existing connection-oriented communication architecture, is a promising approach to deal with the aforementioned issues. By shifting the host IP to named data as the "thin waist" of the network protocol stack, name-based communication in CCN enables consumers (i.e., data receivers) to obtain desired data or content without establishing and maintaining continuous end-to-end communication channels between hosts. The data flow in CCN is organized from named data requests sent by consumers and the corresponding data forwarding from in-network cache. In-network cache reduces redundant data traffic and server resources. It also lowers data transmission delay for consumers, especially when nearby nodes (e.g., routers, mobile base stations) become caching points and provide the requested data to the consumers.

Conceptually, CCN intrinsically supports mobility and multicast communication. Unlike connection-oriented communication, mobile nodes in CCN can easily switch between multiple available links owing to the link independence of data acquisition. Furthermore, name-based communication does not require mobile nodes to know about the location of the data in advance. These features simplify mobility management for mobile nodes. In the case of live streaming or broadcast multimedia services, multicast communication is indispensable for the conservation of network and CPU resources. Multicast communication is also possible in CCN, because intermediate nodes can aggregate identical data requests issued by consumers, receive a single copy of the data from the upstream, and forward the multiple copies of the data in the downstream to all the interested consumers.

In the past several years, many researchers have reported that CCN provides advantages for realizing efficient video delivery services in mobile networks. However, there is still a gap...
between the benefits reported in the literature and what should be implemented in practice.

This article describes the concept of CCN-based mobility and the way to improve consumer mobility for real-time video streaming services. We then propose a novel transport protocol for CCN, named Content Name-Based Mobile Real-Time Streaming (NMRTS). We conducted a simulation to verify the effectiveness of NMRTS on the basis of the derived design concept. The results show that NMRTS reduces the amount of both control message transmission and redundant data traffic, and minimizes data reception delay, compared with the general CCN.

BACKGROUND AND MOTIVATION

CCN OVERVIEW

CCN advocates receiver-driven communication, in which a consumer initiates communication by issuing an interest packet for a specific content object packet. CCN routers forward the interest to the publisher responsible for the requested content, using name prefixes for routing (known as name-based routing). A forwarding information base (FIB) is a lookup table used to determine incoming interfaces (or incoming faces) for receiving the content. Its entries consist of a name prefix and incoming face pairs. If there are multiple incoming faces for a certain name prefix in the FIB, the CCN "forwarding strategy" plays a role in finally determining which and how many of the available incoming faces to use. Each CCN router also maintains a pending interest table (PIT), which is a lookup table containing the name prefix and outgoing faces used to forward received content.

When a CCN router receives an interest, it examines the content name to determine whether it is cached in its local content store (CS). If the named content is in the CS of the router, the router forwards the cached content to the face on which it arrived. If the named content is not present in the CS of the router, the router searches its PIT to determine whether another interest for the same content has already been processed. If the relevant content name is in the PIT of the router but the arrival face is not set in its PIT, the router updates the PIT entry by adding the arrival face as the outgoing face and discards the interest (interest aggregation). If the relevant content name is not in the PIT of the router, the router creates a new PIT entry and forwards the interest through the incoming faces determined by the forwarding strategy.

If no router along the path of the content to the publisher has the requested content in its cache, the interest packet will finally reach the publisher. When an interest is received, the publisher forwards the named content to its downstream router. Each CCN router along the path that receives the interest forwards the content through the outgoing faces, stores the forwarded content in its CS if required (in-network cache), and deletes the corresponding PIT entry.

IP-BASED MOBILITY VERSUS CCN-BASED MOBILITY

In IP-based connection-oriented mobility, it is assumed that hosts basically keep the same location identifications, which are the IP addresses, in order to enable continuous communication. This assumption requires running a mobile routing protocol such as Mobile IP [5]. On the other hand, CCN communication that adopts name-based routing does not require any host location information or identifier. In CCN, a mobile consumer can receive data only by sending the corresponding interest without depending on the location. Hence, unlike IP-based mobility, it is not necessary for a consumer to update location information/identifier during handover, thus a dedicated mobile routing protocol is not necessary in CCN.

Although CCN-based mobility overcomes the limitations of traditional connection-oriented mobility, it is not trivial in practice. We describe the challenges in the next section.

PRACTICAL CHALLENGES

CCN offers benefits to real-time video delivery services in mobile networks. However, there are the following three practical challenges to be addressed (also shown in Fig. 1): bursty interests; duplicate data traffic due to failure of multicast communication; data loss during handover.

Bursty Interests: When a consumer sends an interest, they specify the complete name prefix of the requested content, and the corresponding data (i.e., the content object packet) is received from the network. CCN enables packet-level content delivery, in which the content name prefix specified in an interest indicates a transmission unit of the data. However, real-time streaming content changes over time, and streaming content in a given time slot is not identical to other content in a different time slot. Therefore, consumers must specify each fragment of content on receiving it, and send interests per segment number in order to obtain data. The segment number increases rapidly in proportion to the streaming bit rate, and hence consumers must send bursty interests continuously and consecutively (e.g., 1250 interests per second for a 10 Mb/s stream when the content object size is 1000 bytes) in order to continue receiving streaming data.

![Figure 1. Practical challenges in CCN-based mobile communication.](image-url)
Bursty interests could cause congestion, especially in wireless networks, and degrade throughput. The possible solution is to increase the content object size as in MPEG-Dash over CCN [12], where the content object size is about 25K bytes. However, a content object of large size can cause packet fragmentation (e.g., IPv4 fragmentation) that causes unnecessary packet processing and degrades throughput. Moreover, it introduces redundant data traffic since retransmissions for lost data due to a single fragment loss cause the whole data to traverse in the network again. The effective solution to suppress interest traffic and bandwidth cost by retransmission is absolutely imperative.

**Failure of Multicast Communication:** Multicast communication in CCN potentially fails in the case of real-time video broadcast services, because of the gap in the arrival timing of interests at a base station or router that aggregates identical interests issued by each consumer who receives the same video content. For example, as shown in Fig. 1, the base station receives an interest through an arbitrary face and forwards the corresponding data after receiving the data from the upstream router. Then, if another identical interest arrives at the base station through the same face and the requested data is not present in its CS, both the base station and the upstream router need to forward the data again. This situation causes duplicate data traffic in both wired and wireless links and throughput degradation [8]. If the requested data is in the CS of the base station, the gap in the arrival timing of identical interests at the base station makes duplicate data transmission inevitable in the wireless link. It is quite difficult for each consumer to send interests in a perfectly synchronized manner. Thus, in order to consistently avoid duplicate data traffic, universal multicast communication should be achieved without depending strictly on the timing for transmission of interests at the consumer.

**Data Loss during Handover:** Consumer mobility during handover is not effective in terms of the throughput and the interest traffic overhead, when using simple default forwarding strategies such as flooding and smart-flooding adopted by NDN [9]. The forwarding strategy proposed in [7] can improve handover performance, but it requires accurate network estimation of each face, which is quite difficult in practice. The inaccurate estimation leads to frequent interest retransmissions for recovering lost data, especially when consumers send bursty interests, which results in generating more interest and data traffic. In this context, seamless and tolerant handover management is indispensable in order to enable a simple but efficient way of estimating face conditions and recovering lost data.

**RELATED WORK**

In [6] consumer handover is supported by recovering lost data using a simple interest retransmission mechanism at the consumer side. The performance of this scheme largely depends on the parameter of the loss detection timer that is relevant to the parameter of the interest lifetime of the PIT entry. A shorter interest lifetime enables a fast recovery, but makes it difficult to enable interest aggregation at routers (to support multicast communication). Moreover, it has not been investigated how consumers should send interests for saving consumption bandwidth while receiving data generated in real-time. Schneider et al. [7] proposed a forwarding strategy suitable for mobile nodes with multiple faces, and argued that QoE can be improved by adaptively selecting faces to use according to the network condition estimations. However, in practice, accurate network estimations of each face are quite difficult during frequent handovers, since handover latency varies depending on wireless technology and the device type. Inaccurate estimation increases bandwidth waste by redundant traffic. In addition, it has not been investigated how a large number of consumers effectively receive real-time video content by saving interest and data traffic.

**IMPLEMENTATION DESIGN**

In this section we describe the protocol implementation of Content Name-Based Mobile Real-Time Streaming (NMRTS). NMRTS is the extension of the Content Name-Based Real-time Streaming (NRTS) [8] for mobile communications. NRTS avoids bursty interests and enables universal multicast communication by using symbolic interest, while NMRTS implements the additional functions to enable seamless handover using interest control.

Figure 2 illustrates the implemented NMRTS components and message flows. NMRTS utilizes two types of interest packets: symbolic interest (SMI) without segment identification, and regular interest (RGI) with segment identification. Typically, each face corresponds to an available device interface, through which interest packets are sent. Meanwhile, NMRTS creates and utilizes logical faces, each of which is linked to an available base station for the mobile node. NMRTS appropriately sends SMI and RGI through the logical faces according to link conditions, in order to suppress both bursty interests and redundant data traffic, and enable seamless handover.
SMI transmitted through logical faces toward non-attached base stations results in redundant interest and data traffic. It is thus necessary to understand the current valid logical face toward the attached base station. NMRTS realizes it without any information from a lower layer in a simple way. NMRTS measures the RTTs to logical faces (i.e., neighborhood base stations) using ccnping [10], and selects one logical face (i.e., a valid logical face) giving the smallest RTT, because the RTTs to other logical faces become longer due to the larger number of hops. ccnping is also invoked whenever additional data loss is detected after 50 ms has passed from the last ccnping. The RTT measurement of each logical face may waste the bandwidth of the network. A longer time interval of ccnping, however, delays the valid logical face selection. Owing to the SMI benefit, NMRTS can suppress bursty interests and enable universal multicast data delivery. Therefore, this mechanism using ccnping does not adversely affect NMRTS performance, as shown later.

**FORWARDING STRATEGY**

Although SMI enables universal multicast data delivery, there are two drawbacks to be considered. First, after handover, the previous base station that the consumer associated with continues receiving data traffic until the PIT entry expires. It could result in a large amount of unnecessary data traffic if there is no consumer receiving the same content from the base station. Second, since the sending interval of SMI is a few seconds, the consumer may fail to promptly obtain data from the new base station. NMRTS forwarding strategy overcomes these drawbacks by adjusting SMI_LT and the frequency of sending SMIs (SMI_Freq), while keeping the benefit of SMI. In order to stop forwarding redundant data traffic at the previous base station, NMRTS attempts to remove the PIT entry within a shorter period by sending SMI with the reduced SMI_LT in response to the occurrence of data loss. In addition, NMRTS increases SMI_Freq in order to seamlessly obtain data from the new base station. To keep the benefit of SMI, NMRTS increases SMI_LT and reduces SMI_Freq during a period of no data loss. If data loss occurs, NMRTS sends RGI to recover the lost data.

**Adaptive Interest Lifetime and Frequency Setting:** NMRTS advocates the consumer-driven

![Figure 3. Simulation topology.](Image 298x638 to 570x783)
mechanism without making intermediate nodes operate any procedures.

NMRTS adjusts the SMI_LT based on data packet loss conditions using the Additive Increase Multiplicative Decrease (AIMD) algorithm. NMRTS receives data packets sequentially by SMI, which enables packet loss detections by using the segment number of real-time receiving data. NMRTS sets the additive increase parameter “a” and the multiplicative decrease factor “b” \((0 < b < 1)\) to 0.5 and 10 ms, respectively, based on our comprehensive simulation results. If some lost packets are detected, SMI with the reduced SMI_LT is transmitted immediately. Currently, the initial/maximum and minimum values of SMI_LT are 5.0 and 0.5 sec, respectively. According to the determined SMI_LT, SMI_Freq is also adjusted. In order to avoid the timer expiry of the PIT entry at the base station and routers, NMRTS sends two SMIs at regular intervals during the SMI_LT minus 100 ms \((\text{SMI}_\text{Freq})\).

There is a case that a consumer attaches to a new base station without data loss. In this case, the SMI_LT remains longer, and the new base station does not receive the SMI from the consumer and hence does not forward the streaming data to the consumer. In addition, if we assume the streaming bit rate is more than 1 Mb/s and the content object size is 1,000 bytes, then the average data reception interval becomes less than or equal to 8 ms. It means that if a consumer does not receive the streaming data for 50 ms, there is a high possibility that data loss occurred. According to these analyses, when there is no subsequent streaming data reception during 50 ms after the last data reception, NMRTS recognizes that data loss occurred and adjusts SMI_LT and SMI_Freq in order to promptly make SMI arrive at the new base station.

**Data Loss Recovery by RGI:** NMRTS sends RGI through an arbitrary logical face in order to utilize in-network cache at the corresponding base station to recover lost data. NMRTS periodically checks whether data loss occurred per 10 ms, and sends RGIs to recover lost data through the current logical face. If a consumer cannot get the corresponding data by the RGI within 50 ms, the RGI is transmitted again.

Since bursty data loss widely occurs in wireless networks, the frequency of sending SMI and RGI can be increased. In fact, CCN generally increases the risk of the bursty interest problem by retransmitting many interests, since it sends bursty interests to timely receive real-time video data. On the other hand, in NMRTS, SMI can be suppressed even when bursty loss happens, because the highest SMI_Freq is set to 200 ms. Owing to this feature, NMRTS mitigates the bursty interest problem caused by sending RGIs during a bursty loss period.

![Figure 4. Simulation results: a) number of sent interest packets; b) number of data packets each BS received; c) CDF of data reception delay at consumer 1; d) SMI lifetime transition at consumer 1.](image-url)
EVALUATION

SIMULATION SETUP
Using an NS-3 based NDN simulator (ndnSIM) [11] extended with NRTS, NMRTS, and CCN with the consumer handover scheme proposed in [6], we evaluated the performance of NMRTS to confirm the validity of the proposed mechanism of the adaptive SMI lifetime and frequency setting. The SMI lifetime of NRTS was a constant value of 5.0 sec, and its transmission frequency was set to 50 packets/sec in common with NMRTS. The max/min value of the SMI lifetime of NMRTS was set to 5.0/0.5 sec. The SMI and CCN have the same parameters of the loss detection timer and RGI lifetime. The CCN sends RGIs at an adequate frequency according to the video data rate to be received.

Figure 3 shows the simulation topology. Each NMRTS/NRTS consumer-1 and consumer-2 creates five logical faces. We set the cache capacity of each BS and router to 1 GB, with consideration for the fact that cache size is limited by memory access speed. The propagation delay in each link was set to 5 ms, and the bandwidth was set to 50 Mb/s. We used the following settings for the BSs. The CCN features: 802.16 operating at 12 Mb/s (OFDM). Consumer-1 and consumer-2 move at the speed of 5 m/s, while receiving the real-time video content of 3 Mb/s. The packet size of interest and data is set to 120 and 1200 bytes, respectively. To generate background traffic, we set the three CCN consumers that send interests and receive the content object packets at 3Mb/s. The simulation duration was set to 30 sec.

SIMULATION RESULTS
Figure 4 shows the following performance results:
1) The total number of interest packets sent by consumer-1 and consumer-2.
2) The total number of data packets the BSs received (i.e., data traffic traversing the backhaul links, excluding background traffic).
3) The cumulative distribution function (CDF) of data reception delay that means the time gap between the data transmission time at the real-time video publisher and the reception time at consumer-1.
4) The SMI lifetime transition at consumer-1. Because there is no large performance difference between consumer-1 and consumer-2 in this scenario, we show the results of consumer-1.

As shown in Fig. 4a, thanks to the adaptive SMI lifetime and transmission frequency mechanism, NMRTS drastically suppressed interest traffic compared to NRTS and CCN. The numbers of the ccnping packets sent by NMRTS consumer-1 and consumer-2 were 1,250 and 1,240, respectively. Because CCN frequently sent RGIs to receive real-time video data, the number of interests sent by CCN became larger than NMRTS. The CCN feature becomes more noticeable when the streaming bit rate is much higher, but the number of interest packets sent by an NMRTS consumer does not depend on the streaming bit rate. The reason why the number of interest packets of NRTS became larger than NMRTS is that NRTS consumers cannot quickly start to receive real-time video data from a new BS after handover due to the low frequency of SMI transmission, which results in more data packet losses that necessitates many RGIs for recovering the lost data. The numbers of ccnping packets sent by NRTS consumer-1 and consumer-2 were 1,840 and 1,835, respectively.

The low frequency of SMI transmission of NRTS caused much data traffic over backhaul links due to the longer lifetime of the PIT entry at the previous BS, as shown in Fig. 4b. On the other hand, frequent RGI transmission of CCN for receiving real-time video data enables quick data receptions from a new BS, which decreased the data traffic. In the case of NMRTS, the SMI lifetime was adjusted according to the observed data loss conditions, as shown in Fig. 4d. Since the SMI with shorter lifetime was transmitted to the BS before handover, data traffic also can be suppressed. Note that owing to universal multicast communication, NMRTS can reduce more redundant data traffic than CCN in a situation in which we cannot expect that the BSs always cache the real-time video data.

Concerning the performance of packet reception delay, Fig. 4c shows that NMRTS outperforms the others. The reason why the delays of the performance of CCN decreased is that bursty RGIs sent by CCN caused both RGI and data losses, which increased the data reception delays. The total number of data losses first detected by the two CCN consumers was 7,175 packets, while those of NRTS and NMRTS were 6,204 and 3,757 packets, respectively. NRTS cannot promptly retrieve real-time video data from a new BS, whereas NMRTS adjusts the SMI transmission frequency so as to promptly send SMI to the new BS, which can suppress data loss and improve the data reception delay. In this way, NMRTS can suppress interest traffic and reduce the number of data losses as well as the amount of data traffic over backhaul links by the adaptive SMI lifetime and transmission frequency mechanism according to data loss conditions.

CONCLUSION
As we envision a scenario in which the rapid growth in video data traffic over mobile networks will strongly necessitate both saving consumption bandwidth and maintaining user QoE, in this article we introduced a new CCN transport protocol, NMRTS. NMRTS is useful for real-time video streaming services, because universal multicast communication is enabled without requiring consumers to send interests in a perfectly synchronized manner, and routers to cache real-time video data in a consistent way. Furthermore, the adaptive symbolic interest lifetime and loss recovery mechanism of NMRTS suppress bursty interests and support seamless handover while avoiding redundant data traffic. Our simulation results confirmed NMRTS overcomes the drawbacks of the existing method in terms of saving both interest and data traffic and minimizing data reception delay.

In future work we will thoroughly analyze the tradeoff between symbolic interest lifetime and the performance of data reception delay in more complex and congested networks. This analysis will improve the algorithm of the adaptive sym-
bolic interest lifetime setting. We will evaluate the performance of the actual NMRTS implementation we have been developing in real wireless networks.

REFERENCES


BIOGRAPHIES

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BACKGROUND
Pushing computing, control, data storage and processing into the cloud has been a key trend in the past decade. However, cloud alone is encountering growing limitations in meeting the computing and intelligent networking demands of many new systems and applications. Local computing at the network edge is often necessary to, for example, meet stringent latency requirements, integrate local multimedia contextual information in real time, reduce processing load and conserve battery power on the endpoints, improve network reliability and resiliency, and overcome the bandwidth and cost constraints for long-haul communications.

To meet the growing local and distributed computing needs, the cloud is now “descending” to the network edge and sometimes diffused onto end user devices, which forms the “fog.” Fog computing distributes computing, data processing, and networking services closer to the end users. Instead of concentrating data and computation in a small number of large clouds, fog computing envisions many fog systems deployed close to the end users or where computing and intelligent networking can best meet user needs. Fog computing and networking present a new architecture vision where distributed edge and user devices collaborate with each other and with the clouds to carry out computing, control, networking, and data management tasks.

Fog computing and networking see rapidly increasing applications in, and demands from, many industries such as manufacturing, smart cities, connected transportation, smart grids, e-health, and oil and gas. Fog computing will also be a key enabler for the Internet of Things (IoT) and 5G mobile networks. For example, fog-based services can prove effective ways to address a wide range of unique IoT challenges such as help securing resource-constrained endpoints or supporting local analytics. Fog-enabled 5G radio access networks can improve network performance, enable direct device-to-device wireless communications, and support the growing trend of network function virtualization and separation of network control intelligence from radio network hardware.

Realizing fog computing and networking imposes many new challenges. For example, how to compose, deploy, and manage distributed fog services, how to enable highly scalable and manageable fog networking and computing, how to secure fog computing systems, how should the fog interact with the cloud, and how to enable users to control their fog services provided by fog operators. Addressing these challenges necessitates rethinking of the end-to-end network and computing architecture.

This Feature Topic (FT) is designed to attract papers that will address key challenges such as those mentioned above. Authors are invited to submit complete unpublished papers that are not under review in any other conference or journal in any of, but not limited to, the following or related topic areas:

• Fog computing and networking architectures, including fog-based radio access networks
• Fog system and service management
• Fog-cloud interactions and enabling protocols
• Fog-based data services, including distributed data centers, edge data analytics, edge caching
• Edge resource pooling
• Security and privacy in fog computing environment
• Fog-enabled applications
• Trials and experimentation on fog computing and networking

SUBMISSIONS
Papers should be tutorial in nature, and authors must follow the IEEE Communications Magazine guidelines for preparation of their manuscripts. For further details, please refer to Information for Authors’ on the IEEE Communications Magazine web site at http://www.comsoc.org/pubs/commag/sub_guidelines.html. Manuscripts should be submitted through Manuscript Central at http://commag-ieee.manuscriptcentral.com/. Please select “April 2017 / Fog Computing and Networking” in the drop-down menu.

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Abstract
Mobile devices today are constantly generating and consuming a tremendous amount of content on the Internet. Caching of such “massive” data is beyond the capacity of existing cellular networks in terms of both cost and bandwidth due to its connection-centric nature. The increasing demand for content poses fundamental questions where, what, and how to cache and retrieve cached content. Leveraging the shift toward a content-centric networking paradigm, we propose to cache content close to the mobile user to avoid wasting resources and decrease access delays. Therefore, we present SAVING, a socially aware vehicular information-centric networking system for content storage and sharing over vehicles using their computing, caching, and communication (3Cs) capabilities. The encapsulated 3Cs are exploited first to identify the potential candidates, socially important, to cache in the fleet of vehicles. To achieve this, we propose a novel vehicle ranking system allowing a smart vehicle to autonomously “compute” its eligibility to address the question, where to cache. The identified vehicles then collaborate to efficiently “cache” content between them based on the content popularity and availability to decide what and how to cache. Finally, to facilitate efficient content distribution, we present a socially aware content distribution protocol allowing vehicles to “communicate” to address the question how to retrieve cached content. Implementation results for SAVING on 2986 vehicles with realistic mobility traces suggests it as an efficient and scalable computing, caching, and communication system.

Introduction
Recent advances in communication technologies along with the soaring number of smart mobile devices result in growth of content demand by lots of consumers in closer urban proximity, each with multiple portable devices. For example, a large number of users on the move in an urban environment, such as passengers in buses, taxis, and vehicles, are interested in watching video of the latest episode of a hot TV show/drama or sports highlights. Provisioning of such popular content to each user requires lots of redundant connections between users and the service provider, given that the content is requested by lots of spatio-temporally co-located users with similar social interests. It is now challenging for the current connection-centric network infrastructure to facilitate content availability for such large numbers of mobile users in close proximity in an urban environment while offering attractive tariff plans supporting unlimited bandwidth. We advocate use of the recently proposed information-centric networking (ICN) [1–3] decoupling the content provider–consumer and support in-network caching at intermediate nodes. Content caching at intermediate nodes is studied for using different caching policies based on typical content replacement strategies such as first-in first-out (FIFO), least recently used (LRU), and least frequently used (LFU). This article highlights, for the research community, an advanced dimension of the underlying content caching challenge by posing the following fundamental questions. First, there is a need to identify eligible candidates to cache content by answering the question where to cache, thus finding the criteria for a node to be an important information hub in the network. Once such nodes are identified, another question follows: what and how to cache, regarding decisions based on content popularity and availability in the network. There is also a need to decide among them which nodes should keep which content to avoid redundant caching as well as different cache replacement policies. Once the content is cached in the network, the question of how to retrieve cached content also needs to be addressed.
To address the aforementioned questions, we present a socially aware vehicular information-centric networking model (SAVING) encapsulating them into three classes, computing, caching, and communications (3Cs), where mobile nodes such as vehicles with their intrinsic processing, storage, and communicating capability can “compute” their eligibility to “cache” and “communicate” with each other to facilitate efficient content delivery in a content-centric mobile network. We define a new notion of the computing class where a mobile node computes its eligibility to be selected as an important information hub in order to cache content. Similarly, the caching class incorporates all the questions regarding the content popularity and availability in the network, including different cooperative caching schemes, once the nodes compute their social importance in the network. The communication class involves different content distri-
SAVING presents a new concept of finding important vehicles as a ranking system comprising three novel centrality schemes, InfoRank [4], CarRank [5], and GRank [6]. Each vehicle first classifies different cached information using InfoRank based on its popularity, availability, and timeliness with respect to user interest. The vehicle then autonomously computes its relative importance in the network using CarRank and GRank. CarRank allows a smart vehicle to rank itself based on its popularity with respect to user interests, its spatio-temporal availability, and its neighborhood connectivity as local vehicle eligibility metric. GRank considers the information reachability in an urban environment beyond local importance by allowing a vehicle to consider itself as a global “city-wide” information hub to cache content in the network. Finally, we propose a social content distribution protocol where the novel vehicle centrality schemes are deployed to relay and retrieve cached content in the network.

The remainder of the article is organized as follows. The next section provides an overview of SAVING followed by a description of the computing class, later discussing the ranking system to identify important information hubs as new trends of autonomous computing by smart vehicles. We describe the caching class, explaining different criteria for classifying content. We discuss the communication class, describing the social content protocol as an example. We discuss performance evaluation, describing the results from each class. We conclude the article with some future insights.

SAVING SYSTEM OVERVIEW

SAVING aims to provide a novel concept of distributed content caching and distribution framework to complement an infrastructure network for urban mobile users in order to maximize content availability with minimum delays.

The named data networking concept introduced by the information-centric networking paradigm is able to coexist with the mobility and intermittent connectivity challenge in mobile networks. ICN’s inherent in-network caching and provider-consumer decoupling maximize content availability by allowing users to retrieve content cached at “any” nearby source independent of the underlying network connectivity. We propose below an ICN-enabled SAVING system by describing a use case for the location-aware content caching and distribution in an urban environment.

USE CASE: LOCATION-AWARE INFORMATION SHARING

We consider the case of location-aware content where interests in information regarding available parking lots, traffic/weather conditions, fuel prices, virtual tours to local attractions, or snapshots/videos of nearby resort areas are generated by applications targeting vehicles in a given area, regardless of their IP address. To address this, SAVING comprises a publish-subscribe ICN model allowing a mobile node, such as a vehicle in our case, to subscribe for the following three roles.

Information Provider: An information provider vehicle acts as the content source to publish content. For example, it can subscribe itself to publish sensory information collected from urban streets using vehicle-embedded cameras and sensors.

Information Facilitator: This is a vehicle responsible for collect, caching, and relaying data generated by information provider vehicles as well as forwarding the user interest in content to facilitate efficient content caching and distribution.

Information Consumer: VehicleS subscribed to request different content from the information facilitators/providers within the vehicular network are considered as information consumers to pull content in an information-centric vehicular network.

The three distinct roles are defined since certain vehicles can be subscribed only as consumers or providers, therefore not participating to facilitate other subscribers in the network. Each ICN-enabled vehicle maintains three routing parameters.

Forwarding Information Base: It resembles a routing table that maps content name components to interfaces. Each vehicle forwarding information base (FIB) is populated by the routes discovered using our proposed centrality-based interest/data forwarding protocol.

Pending Interest Table: It keeps track of all the incoming interests that the vehicle has forwarded but not satisfied yet. Each pending interest table (PIT) entry records the content name carried in the internet, together with its incoming and outgoing interface(s).

Content Store: It is a temporary cache to store content each intermediate vehicle has received while forwarding content. Since a named-data packet is meaningful independent of where it comes from or where it is forwarded, it can be cached in a content store (CS) to satisfy future interests.

An overview of the proposed SAVING system is illustrated in Fig. 1. The user vehicle plays the role of a consumer vehicle interested in information regarding a location in a particular zone, assuming the city is divided into different urban zones following, say, an ICN hierarchical naming convention. It forwards the interest to an information facilitator vehicle in range, which subsequently facilitates by caching and providing the desired content. The source vehicle acts as an information provider by providing the content to the information facilitators responsible for the content delivery in the network.

COMPUTING: WHERE TO CACHE?

The identification and selection of suitable vehicles to cache content among the fleet of thousands of vehicles poses an economic and bandwidth challenge along the inherent issue of mobility and intermittent connectivity. The challenge exists in finding the right set of vehicles available at the right time and place for efficient data collection, storage, and distribution through low-cost inter-vehicle communications. We believe that of all the vehicles, only a set of appropriate vehicles can be considered important based on their daily commute while considering
In the temporally evolving vehicular network topology, it is nontrivial to use the vehicle contact frequency and duration to decide its importance due to rapid changes. To overcome this, we propose CarRank, which simultaneously considers three novel albeit essential parameters: information importance, vehicle spatio-temporal availability, and network connectivity.

We define a novel concept of computing by allowing mobile nodes with sufficient processing, storage, and communication capabilities to perform autonomous computing. A ranking system is presented as an example of such autonomous computing where mobile nodes can select information or a location as popular if it observes an increase in the amount and frequency of interest in the associated content.

A vehicle can consider information or a location as popular if it observes an increase in the amount and frequency of user interest in the associated content.

We address the question of where to cache by using multi-hop interest forwarding. We consider the following local parameters known to the vehicle for analytically finding its importance:

- Information importance: Information importance measures vehicle relevance to users for a particular content; that is, the interest-response frequency is a vital factor to classify a content’s importance. A vehicle associated to contents related to popular locations is considered as an important information hub in the network.
- Spatio-temporal availability: It reflects social behavior based on a vehicle’s habitual routes as a factor of its daily commute. Spatial availability reflects the vehicle’s recursive presence in an area, while temporal availability refers to its relevance in time for a location.
- Neighborhood importance: Neighborhood importance shows vehicle topological connectivity in order to be able to distribute information. An easily reachable and well-connected vehicle in a network topology can act as an efficient facilitator.

The vehicle ranking algorithm CarRank (Algorithm 1) is used for the identification of information facilitator vehicles to find the vehicles responsible as information hubs in the network. The vehicle first classifies the information associated with it, taking into consideration the relevance to the users’ interest. It then considers the associated information popularity to find its relative importance in the network using the CarRank algorithm as its vehicle centrality:

$$LC_v(t_{k+1}) = \theta \times LC_v(t_k) + (1 - \theta) \times \left[ \alpha f_v^I(t_{k+1}) + \beta f_v^X(t_{k+1}) + \gamma f_v^T(t_{k+1}) \right]$$

where $f_v^I$, $f_v^X$, and $f_v^T$ are the importance func-
tions for the information, vehicle spatio-temporal availability, and vehicle neighborhood, respectively.

Each function’s contribution is normalized by the terms $\alpha$, $\beta$, and $\gamma$, where $\alpha + \beta + \gamma = 1$, and $\theta \in [0, 1]$ allows the vehicle to increase its importance with respect to the previous time slot. The impact of each parameter differs with respect to different applications. For example, if the vehicle is located in a better connected neighborhood, it can easily spread information. Therefore, the corresponding vehicle weights the information importance along the neighborhood more than the spatio-temporal availability.

Global Information Hubs — GRank: Inspired by the concept of communicability in complex networks [7], GRank (Algorithm 2), a global vehicle centrality scheme, allows a vehicle to use a new stable metric named information communicability to rank different locations in the city and rank itself accordingly. Using GRank, the vehicle finds each location’s reachability and popularity taking into consideration the user interest satisfaction related to the location. It also considers its mobility pattern between different locations in the city through its availability at each location. Vehicles available in popular locations in the city qualify as important information facilitator vehicles with higher vehicle centrality scores in the network. We can identify popular locations in the city with the maximum global centrality with respect to all information facilitators. However, popularity of locations depends on several factors such as the information type, depending on the application requirements as well as time of day. Similarly, we can use the maximum location importance to identify popular neighborhoods over a longer time span.

The vehicle centrality function at the time instant is given as the average information global centrality for all associated locations. For a vehicle, the vehicle global centrality $GC(t_{k+1})$ for the next time instant $t_{k+1}$ is updated as the exponential weighted moving average (EWMA) function of the current and previous global centrality as shown in the relation below:

$$GC(t_{k+1}) = \theta \times GC(t_k) + (1 - \theta) \times f_C(t_{k+1}), \quad (1)$$

where $\theta \in [0, 1]$ is the tuning parameter, which allows the vehicle to adjust its importance with respect to the previous time slot, $GC(t_k)$ is the vehicle global centrality at the beginning of the current time slot, and $f_C(t_{k+1})$ is the vehicle global centrality computed at the end of the current time slot.

The difference between CarRank and GRank can be explained by the fact that each interest specifies two satisfaction deadlines, $I_{\text{max}}$ and $I_{\text{min}}$, where $I_{\text{max}} \geq I_{\text{min}}$ indicates the maximum and minimum threshold time to provide the corresponding content. Thus, in case the interest cannot be satisfied by a local facilitator vehicle (CarRank-based) in the vicinity by an initial threshold $I_{\text{max}}$, the interests can be forwarded to more globally central vehicles (GRank-based) to $I_{\text{max}}$: the maximum interest deadline to avoid bandwidth and time utilization.

## CACHING: WHAT AND HOW TO CACHE?

In this section we discuss a novel approach for content cache management by classifying the cached content importance with respect to the intended user. Let us assume vehicles encountering each other in a vehicular network constantly receive interests for content from neighboring vehicles regarding different information. Some of such information can be of more importance to the intended users in the network, which the vehicle can easily recognize from the amount of user interest received for it. Therefore, a vehicle can consider an information popular if it observes an increase in the amount of user interest in the associated content. We assume that it is capable of recording the time and position each time it responds with the desired content to an interested user. Thus, SAVING incorporates a novel distributed algorithm InfoRank with the concept of enabling a mobile node to rank important information associated with it based on the satisfied user interests and the information validity scope.

### Interest Satisfaction Frequency

We define interest satisfaction frequency as the frequency of user interests satisfied in the previous time slot and the total successful responses for the content associated with the vehicle. Thus, the vehicle regularly updates each content importance value depending on the interest satisfaction frequency. We assume that each vehicle is able to record the time and position each time it responds as the content provider to a user interest. Interest in each content specifies the temporal scope of information validity. For instance, road congestion information is only valid during congestion. Therefore, it should be ensured that the information importance is not substantially augmented after the desired deadline.

### Information Timeliness

The information timeliness $t$ is the measure of the temporal information validity scope, which can be adjusted by a tuning parameter depending on the application needs (e.g., 1 h for accident information validity). If there are no active inter-
Facilitator discovery process.

Figure 2. Facilitator discovery process.

 estimates and the average interest validity time has passed, the information importance adapts an exponential delay since the information is of less importance in the network. However, \(\tau\) is set to unity for content to be always available in the network.

The content importance depends on its importance at the beginning of the time slot. If it has not responded in the previous slot, the content importance is not increased unnecessarily.

We also consider the percentage of time the vehicle itself acted as the original source for any content. InfoRank is updated regularly to ensure the content relevant to the vehicle maintains its value in case the vehicle does not respond in the previous slot. The interest later in time could finally route to the vehicle, which maintains its value as the original source for particular content. A tuning parameter decides the importance value with respect to the associated content in cache. For all contents associated with a vehicle, we also consider the ratio of missed interest to the total interest received by the vehicle. Missed interest provides the vehicle reliability regarding successful response to incoming interests.

To summarize with an example, assume a vehicle visiting an area at some future time slot places an interest for the content regarding that area. This interest is propagated to a potential facilitator vehicle. Each vehicle, upon receiving the interest message, checks its cache to find a match regarding the desired content. If the interest cannot be satisfied, it is forwarded to neighboring vehicles. If a match is found, it responds to the interest message by providing the corresponding content where each vehicle computes its cached information importance by finding its respective InfoRank score. Once the information importance is agreed between different facilitators, collaborative caching between nodes (i.e., peer-to-peer networking) can be ensured by mutually respecting a social norm to avoid redundant content caching in the network.

**Communication: How to Retrieve?**

In this section we discuss efficient retrieval of the cached content in the network by focusing on the communication aspect. For this reason, we present an idea of a socially aware content caching and distribution scheme where the consumer pulls content of interest cached at important information facilitator vehicles in the network. We use the above mentioned two novel vehicle centrality schemes to identify important information facilitator vehicles based on cache management for content suggested by the InfoRank scheme.

**Content Distribution Protocol**

The centrality-based content distribution protocol leverages the facilitator centrality to forward consumer interests in content as well as route the content from the corresponding information providers. The provider as well as the consumer search for a nearby information facilitator vehicle using its centrality score to forward interest/content. We propose a hybrid content distribution protocol with ICN inherent pull-based content retrieval for the consumer and a push-based approach for the provider to publish content.

**Information Facilitator Discovery:** The facilitator discovery process allows a vehicle to search in its vicinity for the highest centrality facilitator vehicle using the FACILITATOR() function. It compares the facilitator centrality score of all the neighboring vehicles and returns the best facilitator centrality vehicle among the vehicles that are in the vicinity of a vehicle. The PROVIDER() function assigns a vehicle to be the provider vehicle to publish the content for the consumer vehicle. The publishing of content by the provider can be either solicited or unsolicited. In the case of solicited interest, the provider can publish content destined for the vehicle to a nearby information facilitator using the PUBLISH() function. Similarly, unsolicited publishing with a nearby facilitator can be performed by an information facilitator discovery process initiated at any time by the information provider. The CONTENT() function is used for the content availability check at each intermediate vehicle CS.

Figure 2 depicts the social content distribution protocol. Consumer vehicle \(v\) generates interest in content as INTEREST() toward the best ranked facilitator in the vicinity. The facilitator discovery process continues to search for the content at each intermediate relay vehicle by constantly discovering the next best ranked vehicle in the vicinity of each intermediate relay vehicle. Thus, each relay vehicle becomes the responsible vehicle to facilitate the content. If it is unable to find the content in its CS, it performs facilitator discovery to find a vehicle with a higher facilitator centrality score, and a PIT entry is created. The process is repeated at each intermediate facilitator until either the desired content is found or there are no more facilitators to discover.

The convergence of the facilitator discovery process is two-fold. The first obvious convergence occurs when the desired content is available at the corresponding facilitator. In this case, the content is published at the consumer vehicle following a reverse path to the initial requester using breadcrumbs left in the PIT at each intermediate node. The intermediate vehicles subsequently populates the corresponding FIB entry for the content. If the content is not available and there are no further facilitators to discover, the responsible vehicle declares itself as the content...
tent provider to publish content at the consumer vehicle.

**Performance Evaluation**

The performance of SAVING is validated by a set of simulations under a realistic mobility scenario using traces from Cologne, Germany as an accurate mobility trace available for a vehicular environment [8]. The number of vehicles in each region vary at different times of day. We analyze up to 2986 vehicles over the entire simulation duration with 1 s time granularity. The Cologne city center is simulated for 1 h by clustering $6 \times 6$ km$^2$. The number of regions can vary between different cities depending on the size, although we divide Cologne into 36 neighborhoods. Urban roads with vehicle communication range around 300 m is considered. The Nakagami path loss model is combined with a log-distance propagation model to cater for the impact of buildings and other obstacles.

**Simulation Scenario**

We simulate an urban vehicular network using ndnSIM (http://named-data.net/techreports.html) to integrate the named data networking (NDN) communication model. The simulation scenario implements the following applications:

1. **Consumer**: Consumer vehicles are the potential users planning to visit an area. Each consumer vehicle generates an interest in a content associated with a location in the city, which is routed to provider vehicles.

2. **Provider**: We define a vehicle to be the content provider in the network for the areas visited in a time slot before the consumer interest generation time. The areas visited are considered as locations associated with the provider.

3. **Facilitator**: Vehicles satisfying incoming requests generated from consumers regularly compute their centrality score to consider themselves as information facilitators. Similarly, constant content forwarding and cache hits also count toward the facilitator centrality score.

We associate each vehicle with a different set of location-dependent content as its cached content. Each vehicle is enabled to randomly generate interest with varying frequency at different time intervals for different (predefined) content as a consumer. The interest profile characteristics are two-fold. First, we evaluate information using InfoRank considering its popularity based on the number, frequency, and spatio-temporal validity of generated interest in the content. Then, considering the cached content importance, we imply our ranking schemes CarRank and GRank to evaluate the interest profile for the associated vehicle.

We assume the interests follow a Zipf distribution, where we observe frequent interests in content regarding popular locations.

**Simulation Results**

For better performance analysis of the proposed SAVING system in different simulation scenarios, we compare it with the state-of-the-art socially aware routing schemes. Such schemes typically rely on centrality schemes as degree, betweeness, and eigenvector centrality. Therefore, we perform a comparative analysis of the proposed vehicle centrality-based routing with the benchmark centrality schemes regarding the following performance metrics:

- Success rate for satisfying consumer interests in the network
- Aggregated content store (cache) hit rate at the information facilitators

**Success Rate**: Success rate refers to the percentage of the generated consumer interests successfully satisfied over the entire simulation duration.

**Benchmark Centrality Schemes Comparison**: The proposed vehicle centrality-based content distribution is compared to the state-of-the-art centrality schemes as benchmark. Figure 3 shows the percentage of consumer interests at different locations successfully satisfied by the corresponding information facilitators/providers. We...
observe that forwarding the interest toward a socially important vehicle using CarRank and GRank as a metric results in more successful interest satisfaction. The vehicles identified by the proposed vehicle centrality metric satisfied around 40 percent of interest compared to other centrality metrics despite high mobility and intermittent connectivity. This is because typical centrality schemes only take into account physical topology when computing a node’s importance in the network, ignoring satisfied user interest.

**Socially Aware DTNs and Socially Unaware Schemes Comparison:** We also compare the success rate of SAVING with two relevant socially aware DTN routing schemes, MS-LOR [9] and Bubble-Rap [10], as well a variant without considering social awareness. BubbleRap uses a hybrid metric based on community and betweenness centrality, while MS-LOR uses a three-layer social metric based on degree centrality. The socially unaware approach implements interest flooding in which each consumer rebroadcasts interest to all of its neighbors except the one from which it received information. Figure 4 depicts the results from the comparative analysis. We observe that using CarRank and GRank-based routing yields a success rate around 40–50 percent, where socially aware DTN schemes achieve a maximum of 38 percent at 50 min from ML-SOR. An interesting observation made is the stability of SAVING unseen in other schemes. It is because they fundamentally rely on centrality measures, which assume a static graph topology with respect to time. Moreover, the host-centric nature instead of the content-centric approach with no in-network caching support limits DTN capability to maximize content availability. Thus, content distribution based on adapted metrics (e.g., CarRank and GRank) better copes with the dynamic nature of the vehicular network.

**Cache Hits:** We evaluate the ICN built-in feature of in-network caching at intermediate nodes at the selected facilitator vehicles. For this purpose, we compute the cache hit rate at the facilitator vehicles. A second successful response by a vehicle for the same content is considered a cache hit. The cumulative cache hit rate is computed for the entire simulation duration. Figure 5 shows the cache hit rate for the facilitator vehicles identified by each centrality scheme. The vehicles identified by our proposed vehicle centrality scheme yield a higher hit rate than all the other schemes. This is because we consider content popularity as a key factor; thus, a vehicle containing important information responds and subsequently caches more frequently compared to other vehicles.

We also observe that vehicles identified using betweenness centrality follow our proposed scheme yielding better cache hit rate due their frequent availability as intermediate bridges at most of the shortest paths, thus allowing them to cache more content. Moreover, the intermediate facilitators identified by our vehicle centrality scheme cached more important content due to their better neighborhood connectivity and spatio-temporal availability in the network. This proves that in-network caching offered by ICN along the proposed vehicle centrality scheme overcomes the mobility and intermittent connectivity constraints in vehicular networks for efficient content distribution.

**Conclusions and Open Research Issues**

We propose SAVING as an alternate solution to leverage smart vehicles with their caching, computing, and communicating capabilities. Socially aware network computing, caching, and communication capabilities to facilitate content availability for an urban mobile user with minimum content access delay. SAVING is a socially aware vehicular information-centric system focusing the research community interest on the application of combining a socially aware content distribution scheme with the information-centric networking paradigm. We explored possible answers to the fundamental questions of how and where to cache what content in mobile networks under the increasing growth of mobile traffic. Moreover, this article highlights another perspective by equally considering efficient content retrieval by caching at vehicles. The suggestion for vehicles can also be generalized for all sorts of mobile nodes depending on nodes’ computing, caching, and communication capabilities.

Open research issues include efficient socially aware routing strategies, a flexible and scalable naming scheme for novel applications, and the possibility to support high-bandwidth-consuming content video streaming in the content-centric networking paradigm. However, each of the 3Cs still lacks exploration by current research, requiring intelligent algorithms for nodes to make real-time decisions regarding the cached content. Similarly, the need for distributed cache management schemes with collaborative content replacement strategies with redundancy avoidance for the daily massive content generated needs to be considered. Thus, we invite the research community to explore the new trends exploiting socially aware network computing, caching, and communication in a content-centric approach to overcome the limitations of the existing connection-centric approach.
**References**


**Biographies**

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YACINE GHAMI-Doudane [M] is currently a full professor at the University of La Rochelle and director of its Laboratory of Informatics, Image and Interaction. Before that, he held an assistant/associate professor position at ENISIE, a major French post-graduate school located in Evry, France, and was a member of UGM-UMR 8049. From February 2011 to July 2012, he regularly visited the Performance Engineering Laboratory of University College Dublin, Ireland. He received an engineering degree in computer science from the National Institute of Computer Science, Algiers, Algeria, in 1998, an M.S. degree in signal, image, and speech processing from the National Institute of Applied Sciences, Lyon, France, in 1999, a Ph.D. degree in computer networks from University Pierre & Marie Curie, Paris 6, France, in 2003, and a Habilitation to Direct Research in computer science from Université Paris-Est in 2010. His current research interests lie in the area of wireless networking and mobile computing, with a current emphasis on topics related to IoT, wireless sensor networks, and vehicular networks. He holds three international patents, and has authored or co-authored seven book chapters, about 25 peer-reviewed international journal articles, and more than 80 peer-reviewed conference papers. Since 1999, he has participated in several national and European-wide research projects in his areas of interest, among them two regional research projects, two national research projects, nine European-wide research projects (one ongoing one in 2015–2016), as well as three EU COST Actions. As part of his professional activities linked to the computer networking research community, he also acted as Chair the IEEE ComSoc Technical Committee on Information Infrastructure & Networking from January 2010 to December 2013, and is currently chairing the IEEE ComSoc Humanitarian Communications Technologies Ad Hoc Committee (HCT). He is an Editorial Board member of the IEEE IoT Journal, Springer AoT, Elsevier IJNCA, and Wiley WCMC, is co-Editor-in-Chief of Springer/KICS CT Express, as well as the founding Editor-in-Chief of the IEEE ComSoc Ad Hoc and Sensor Network Technical Committee Newsletter. Among other conference involvements, he has acted as the TPC Chair of IEEE CCNC 2015, and Symposium Co-Chair of IEEE ICC 2009, 2010, and 2012 as well as IEEE GLOBECOM 2012 and 2015.
The field of optical communications continues to evolve to meet the ever increasing demands for capacity, flexibility, scalability, and security. In this second installment of the Optical Communications Networks Series in 2016, we have selected six contributions that address physical layer security in optical networks, resource optimization in cloud-based radio over fiber networks, transceiver configuration in flex-grid optical networks, control plane architecture in multi-domain elastic optical networks, cost assessment of FTTdp networks with G.Fast, and hybrid ROADM architectures for scalable C/DWDM metro networks. Diverse topics addressed in this issue are indicative of the pervasive role of optical communications technologies in modern communications networks.

In the first contribution, “Physical Layer Security in Evolving Optical Networks,” N. Skorin-Kapov, M. Furdek, Z. Szilard, and L. Wosinska present an overview of potential physical-layer vulnerability scenarios in current and future optical networks. They propose a generalized security framework and outline strategies for dealing with such vulnerabilities to assist in the development of effective service provisioning, network monitoring, protection, and restoration schemes within the context of optical-layer security. With increasing capacity and performance requirements, security issues in optical networks are expected to continue to attract a great deal of attention.

In the second contribution, “C-RoFN: Multi-Stratum Resources Optimization for Cloud-Based Radio over Optical Fiber Networks,” H. Yang, J. Zhang, Y. Ji, and Y. Lee address multi-layer resource optimization by proposing a novel cloud-based radio over optical fiber network architecture and using software-defined networking. In the continuing evolution of 5G wireless networks, the cloud radio access network paradigm has been introduced by operators for aggregating all base station computational resources into a cloud baseband unit pool while the radio frequency signals from geographically distributed antennas are collected by remote radio heads. These RF signals are transmitted to the cloud platform through optical transmission systems. In such networks, the multiple-stratum resources of radio, optical, and BBU processing units have been interwoven with each other so that the legacy architectures cannot efficiently implement resource optimization and scheduling for high quality of service guarantee.

In the third contribution, “Control Plane Solutions for Sliceable Bandwidth Transceiver Configuration in Flexi-Grid Optical Networks,” R. Martínez, F. Cugini, R. Casellas, F. Paolucci, R. Vilalta, P. Castodi, and R. Muñoz present a discussion of control plane enhancements for elastic optical networks (EONs) with their goal of leveraging flexibility and improved spectral efficiency. In EONs, bandwidth-variable transceivers and bandwidth-variable cross-connects need to be configured to establish optical connections. This complex configuration operation is performed via a control plane architecture and extensions to account for the client signal data rate, the required modulation format, the adopted forward error correction mechanisms, the number of carriers, and the required spectrum resources. Thus, this contribution provides an analysis of selected candidate control plane architectures.

In the fourth contribution, “A Control Plane Architecture for Multi-Domain Elastic Optical Networks: The View of the IDEALIST Project,” R. Casellas, O. González, F. Paolucci, R. Morro, V. López, D. King, R. Martínez, F. Cugini, R. Muñoz, A. Farrel, R. Vilalta, and J-P Fernández-Palacios present an overview of the design and implementation of a generalized multi-protocol label switching and path computation element based control plane for multi-vendor and multi-domain EONs. A control plane enables the automation of the provisioning, recovery, and monitoring of end-to-end optical connections. This article covers the macroscopic system along with the core functional blocks, control procedures, message flows, and protocol extensions, and the results shown target follow-up standards development in international standards organizations.

In the fifth contribution, “Cost Assessment of FTTdp Networks with G.fast,” J. R. Schneir and Y. Xiong address cost considerations in the construction, installation, and operation of fiber to the distribution point (FTTdp) networks that use G.fast, a recently ratified international standard for high-speed (up to 1Gb/s) transmission over twisted pair copper cables. These FTTdp networks are expected...
to complement the rollout of fiber to the home services, especially in areas where the deployment of the fiber near and inside a building is difficult and expensive. Thus, in this article, the primary cost components of FTTdp networks with G.fast are analyzed, and cost advantages that can be achieved in comparison with FTTH networks are identified.

In the sixth contribution, "Low-Cost Hybrid ROADM Architectures for Scalable C/DWDM Metro Networks," M. Nooruzzaman and E. Halima present reconfigurable optical add/drop multiplexer (ROADM) architectures and explore a new optical node structure of hybrid coarse wavelength-division (CWDM) and dense WDM (DWDM) (C/DWDM) networks. CWDM networks have been considered as promising first-step metro and access network architectures, offering a significant cost advantage over DWDM due to the lower cost of lasers and the filters used in CWDM modules. In the proposed approach, existing CWDM channels are merged with new DWDM channels on existing fibers. They demonstrate that the hybrid ROADM architecture offers significant advantages in scalability and the initial cost of deployed nodes and networks.

This installment of the Optical Communications Networks Series marks the beginning of our term as Series Editors. It is our great pleasure and honor to take over this responsibility from the former Series Editor, Dr. Osman Gebizlioglu, who handled this installment as his last in this role. With help from our authors and reviewers, and with feedback from readers, we expect the Optical Communications Networks Series in IEEE Communications Magazine to continue contributing to the field of optical communications, networks, and applications.

Biographies

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Xiang Liu (Xiang.Liu@huawei.com) received his Ph.D. degree in applied physics from Cornell University in 2000. He is currently the director of Optical Access Networks Research at the U.S. R&D Center of Huawei Technologies, focusing on next-generation optical access technologies. He spent the early part of his career at Bell Laboratories in New Jersey, working on high-speed optical fiber transport technologies. He is a Fellow of the OSA and a Deputy Editor of Optics Express, and was the TPC Co-Chair of OFC ’16.
We are witnessing the evolution of optical networks toward highly heterogeneous, flexible networks with a widening area of application. As the bandwidth and reliability performance requirements of mission-critical applications tighten, and the amount of carried data grows, issues related to optical network security are becoming increasingly important. Optical networks are vulnerable to several types of attacks at the physical layer, typically aimed at disrupting the service or gaining unauthorized access to carried data. Such security breaches can induce financial losses to clients or loss of privacy, or cause network-wide service disruption, possibly leading to huge data and revenue losses. Awareness of system weaknesses and possible attack methods is a prerequisite for designing effective network security solutions. As optical networks evolve, new and changing vulnerabilities must be identified and dealt with efficiently. To this end, this article provides a comprehensive overview of potential physical-layer attack scenarios in current and future optical networks. It then proposes a general security framework, outlining possible strategies for dealing with such attacks, meant to aid in the development of efficient provisioning, monitoring, protection, and restoration schemes in the context of optical-layer security.

INTRODUCTION: OPTICAL NETWORK EVOLUTION

Optical networks have evolved from simple point-to-point systems operating on a single wavelength at megabit per second rates over a few kilometers to ultra-long-haul multirabit systems supporting over a 100 wavelengths per fiber and all-optical transmission schemes [1]. Due to ever increasing bandwidth demands, and new traffic requirements and models stemming primarily from the proliferation of cloud services, further developments in optical networking are pushing toward increased dynamism and flexibility.

To support the increasing complexity associated with future dynamic optical networks, software defined networking (SDN) has been proposed as a promising solution for simplifying control and management. The SDN paradigm focuses on decoupling the control and data planes, and shifting the control logic from routers and switches to a logically centralized controller, essentially acting as a network operating system. The network can then be made programmable through software applications running on top of this operating system and interacting with underlying physical network devices [2].

To enhance flexibility for more efficient resource usage, mixed line rate (MLR) networks allow the coexistence of different channel sizes over the existing infrastructure based on the legacy fixed 50 GHz grid pattern set by the International Telecommunication Union (ITU). A flexible grid option based on 12.5 GHz frequency slots was adopted by the ITU in 2012 to support finer grid granularity. Further flexibility is envisioned with the elastic optical networking (EON) paradigm where use of a flexible grid and adaptive transceivers will allow connections to be established with variable bit rates and the spectrum allocated to demands to grow/shrink as needed [3].

The foreseen evolution of optical networks to dynamic elastic SDN-based networks will lead to new security vulnerabilities that must be addressed in order to provide a secure networking environment. In fact, there is growing interest of major telecom operators and vendors in optical-layer security, driving the development of solutions for optical fiber intrusion detection and encryption-enabled transponders. However, many aspects of determining and countering security breaches are yet to be addressed. Identifying vulnerabilities and possible threats in both current and future networks is a crucial prerequisite for developing secure planning approaches and effective protection and restoration strategies to support this migration. In this article, we provide an overview of physical-layer vulnerabilities and potential attack scenarios in evolving optical networks, aimed at highlighting important security issues and challenges. We then present a general security framework outlining possible methods of dealing with some of them and provide directions for future work.

PHYSICAL-LAYER ATTACKS IN EVOLVING OPTICAL NETWORKS

Physical-layer attacks have traditionally been categorized according to the type of damage they cause: service disruption or eavesdropping. Some
of these attacks have previously been identified and described in [4, 5, references therein], primarily exploiting the transparency associated with optical-bypass-based networks. Here we extend these works with several new attack scenarios exploiting the vulnerabilities of evolving optical networks and classify them according to the attack method used, which may call for distinct protection schemes. We distinguish between three types of attacks: signal insertion attacks, signal splitting attacks, and physical infrastructure attacks, as shown in Table 1.

### SIGNAL INSERTION ATTACKS

Signal insertion attacks reduce the quality of transmission of legitimate connections by injecting harmful signals into the network, causing service degradation (or possibly service denial). Depending on the network architecture and optical components used, signal insertion attacks may propagate through the network, causing system-wide damage, and can appear sporadically, potentially incurring multiple restorations. Examples of such attacks are described below.

#### HIGH-POWERED JAMMING ATTACKS

High-powered jamming attacks can be realized by inserting an optical signal of excessive power (e.g., 5–10 dB above normal signals) on a legitimate channel used in the network (in-band jamming) or on a wavelength outside the signal window (out-of-band jamming) [4]. Such high-power signals, which exceed the component specifications, can degrade co-propagating user channels due to increased nonlinear effects and crosstalk in fibers and switches. Furthermore, a jamming signal out of the working range of the amplifier can cause so-called gain competition in optical amplifiers where the high-powered signal robs weaker legitimate signals of gain.

Typical point-to-point data center applications or filterless network architectures for terrestrial or submarine applications route traffic using only passive couplers with amplifiers, and cannot thwart or readjust such jamming signals. For actively switched networks, this type of attack can be especially harmful in older networks deploying fixed optical add/drop multiplexers (FOADMs) without power equalization capabilities, as a jamming signal inserted on a legitimate connection could also propagate through the network unthwarted, causing excessive damage. Since approximately 40 percent of current networks still employ this type of technology, this scenario still poses a significant threat. FOADMs are particularly dominant in Asian markets, as well as having significant presence in U.S. metro applications.

In more modern networks, which deploy reconfigurable OADMs (ROADMs) equipped with variable optical attenuators (VOAs) to regulate the output power of transiting signals, such a jamming signal would be attenuated at the first downstream node, limiting its propagation. However, it would still cause increased crosstalk to co-propagating signals on the link where it was inserted, in addition to disrupting the jammed signal itself. Additionally, even if electronic equalization in amplifiers counteracts gain competition in the steady-state, initial brief oscillations in gain (called transients) can occur. Some example effects of a high-power jamming attack are illustrated in Fig. 1a.

#### AMPLIFIER TRANSIENT ATTACKS

Amplifier transients, arising in both erbium doped fiber amplifiers (EDFAs) and Raman amplifiers [1], occur when there is a sharp change in the input power level, causing the amplifier gain to change briefly before returning to its nominal value (steady-state). The power of the remaining channels increases or decreases as the optical amplifier attempts to maintain a constant total power level on the fiber. Besides high-power jamming, establishing and tearing down normal connections (e.g., for restoration purposes) in optical bypass networks can cause undesirable transients. Although these oscillations are short-lived, they can still cause error bursts and may even propagate, where transients on one link cause transients on successive links. These effects could be exploited by a malicious attacker by repeatedly or sporadically inserting a high- or even normal-powered jamming signal (e.g., inserting fast pulsing signals in less than a 1 ms timeframe), which would not only cause transients itself, but could incur multiple restorations of the affected legitimate signals, subsequently causing additional transients on other links.

Systems that better manage transients are evolving as a prerequisite for future dynamic networking. Consequently, the amplifiers used in the newest systems work in constant gain mode with transient suppression control. However, there are still significant deployments with constant-power-based amplifiers where transient attacks could cause significant signal deterioration. This is especially true in submarine cables where loading lines are used to maintain a constant input power in undersea amplifiers. Based on the implementation, the reaction time of such loading lines may be much slower than the transient controls.
implemented in terrestrial amplifiers, making them more vulnerable to transient-based attacks.

**Mixed Modulation Attacks**

In mixed line-rate (MLR) and future elastic optical networks (EONs), a key security vulnerability stems from the nonlinear effects arising between line rates using different modulation formats. Typically, on-off keying (OOK) is used for 10 Gb/s rates, binary phase shift keying (BPSK) or quadrature phase shift keying (QPSK) for 40 Gb/s rates, and dual-polarization QPSK (DP-QPSK) for 100 Gb/s rates. OOK 10G channels strongly deteriorate the quality of the higher-bit-rate phase modulated channels due to cross phase modulation (XPM). Although it is technically possible to have 10G and 40/100/200G channels at 50 GHz spacing, larger guardbands should be employed to reduce the nonlinear penalty imposed on the higher line rate channels to an acceptable level. This is especially critical in more nonlinear G655 fiber types.

A possible attack causing signal degradation could be realized by inserting an OOK channel near a 40/100/200G channel without allowing for enough guardband, as shown in Fig. 1b, significantly deteriorating the optical signal-to-noise ratio (OSNR) of the higher-line-rate signal. If such a signal was inserted as a legitimate connection by an insider attacker, it could propagate through the network, degrading multiple co-propagating neighboring channels. As opposed to high-power jamming, such a signal would not be thwarted by power equalizing components or even be detected as malicious by power monitoring equipment.

To perform any of the aforementioned signal insertion attacks, an attacker can access the network via various entry points. An insider with node access can directly tamper with the associated lasers and patch panels to increase power or insert harmful signals. Note that depending on the nodal architecture (e.g., a ROADM in a broadcast-and-select configuration with a wideband coupler on the add side), an attacking signal may be launched into the network unfiltered. Alternatively, a harmful signal can be inserted by an external attacker with physical access to part of the fiber by removing the cladding, slightly bending the fiber and radiating light into it.

Another possible way to access the channel is via monitoring ports typically present at network components, such as amplifiers, wavelength selective switches (WSSs), or (de)multiplexers, which mirror the signal with an optical splitter to allow for monitoring devices to be connected without traffic interruption. In addition to providing a means for potential eavesdropping, these ports could also be used to insert malicious signals into the network and damage live traffic.

Another entry point to the network can be realized through alien wavelengths. Namely, in order to allow for network upgrades and efficient transmission of high-capacity connections over the existing infrastructure, operators are forced to implement alien wavelengths in their networks. Such connections can traverse multiple domains without optical-electronic-optical (OEO) conversions at domain boundaries in multi-vendor environments. In simple FOADM-based networks, the network management system (NMS) has no information on the performance of the alien channels or control over their power and frequency. In more intelligent networks, the alien channels are configured as friendly wavelengths, allowing the management system to have information on signal parameters, but still no control over their values. If the nodal architecture is such that there is no WSS on the add side, such as a colorless FOADM with no WSS on the add/drop ports or a 1 × 9 broadcast-and-select ROADM with a WSS only on the drop side and a coupler on the add side, these alien wavelengths can enter the network unfiltered, providing a point of entry for any of the aforementioned signal insertion attacks. In newer generation networks, a dedicated interface will be defined to host alien wavelengths in order to tune their power levels, but there will still be no control over frequency and modulation format.

**Signal Splitting Attacks**

We refer to signal splitting attacks as attacks that split and remove part of a legitimate signal carried in the network, for either eavesdropping or signal degradation purposes. Such attacks may be difficult to detect and locate due to low losses.
incurred at the insertion point, going undetected by the network management system and/or raising alarms only downstream of the attacking point. Two examples of such attacks are described below.

**Attacks**

Tapping attacks physically tap into the optical signals traversing the network in order to gain unauthorized access to confidential privileged information. In today’s digital era, eavesdropping occurs on all network layers from the application to the physical layer, primarily targeting governments and the financial, energy, transport or pharmaceutical sectors. This becomes even more important with the development of cloud services and tremendous amounts of data, including mission-critical data, being stored in data centers. One way of performing such an attack at the physical layer is by directly accessing the optical channel via fiber tapping, that is, removing the fiber cladding and bending the fiber to cause part of the signal to leak out onto a photo detector [6]. Tapping devices, which can be clipped onto the fiber and cause micro-bends to leak signals, are easily accessible on the market. Furthermore, existing tapping devices can cause losses below 1 dB, which would not be detected by most commonly used NMSs unless sensitive power monitors and intrusion detection systems (IDSs), typically based on bulk power measurements, are employed. Another possibility of accessing user channels could be via component monitoring ports, as mentioned in the previous subsection.

**Low-Power QoS Attacks**

A low-power quality of service (QoS) attack, identified in [7], deliberately attenuates the power of a legitimate channel by inserting a splitter along the link for signal degradation purposes. Since optical amplifiers are placed such that they compensate only for the losses on the previous fiber span, the induced attenuation could significantly degrade the performance metrics of the attacked connection. Even if power monitoring is employed, the power degradation may not be significant enough to cross the alarm threshold at the attack location, but may do so on downstream links far from the attacking point, making source identification and localization more difficult. Furthermore, if nodes are equipped with fixed attenuation-based power equalization, the attack could propagate since legitimate co-propagating channels would be attenuated to ensure a flat power spectrum.

**Physical Infrastructure Attacks**

Physical infrastructure attacks include all attacks that physically damage or tamper with the optical network infrastructure, such as cutting a fiber, unplugging connections, or damaging optical components. These attacks typically persist until repaired and do not propagate through the network. They can often be modeled as single or multiple component faults and require efficient protection and/or restoration mechanisms. We distinguish between three types of physical infrastructure attacks as follows.

**Single Component Attacks**

Single component attacks refer to attacks that mimic single link or node component failures, such as deliberately cutting a fiber or damaging a switch or amplifier. Individual connections could also be unplugged at the patch panel. Such attacks are typically detected by a loss of light and rely on standard survivability techniques.

**Disaster-Like Attacks**

Disaster-like attacks are human-made attacks that have the effect of multiple failures in a specific geographical area, such as weapons of mass destruction (WMD) and electromagnetic pulse (EMP) attacks [8]. EMP attacks are realized by radiating a short burst of electromagnetic energy that can disrupt the electronic components needed to operate the fiber plant in a large geographical area. From a network perspective, these attacks have similar effects as natural disasters and can affect all generations of optical networks.

**Critical Location Attacks**

Critical location attacks are aimed at specifically attacking weak points or critical hub nodes in the network to cause system-wide damage. Although these attacks can fall under single component failures or disaster-like failures, we consider them separately since they also include coordinated multiple failures that are not geographically localized.

For example, a critical location attack could target undersea landing points. Generally, undersea landing points are highly localized due to regulatory limitations, and as a consequence are at a high risk of location-based attacks. Given the remote location and fairly sparse undersea connectivity of some islands or even continents, targeting specific undersea landing points could impact the overall reachability of such areas.

As demands for cloud services proliferate, content service providers (Amazon, Google, Facebook, Yahoo, etc.) increasingly store and replicate massive amounts of content in multiple data centers. Although storing data in multiple geographically distributed locations generally leads to increased reliability in case of localized failures, if an attacker were to acquire information regarding where the content is replicated, a coordinated attack targeting all mirrored sites could cause tremendous damage. A more covert attack could target replica synchronization by exploiting network latency. Namely, synchronous replication used between data centers is very sensitive to signal latency, limiting the maximum distance suitable for synchronous replications to 100–200 km. If the topology of the network is known, a set of systematic fiber cuts or component damaging attacks could be performed to take advantage of this vulnerability by forcing the path between two data centers to be restored to a path that does not meet these requirements, as illustrated in Fig. 2. Although latency measurements are defined in optical transport network (OTN) frames, real-time latency monitoring is not implemented in most networks, which complicates the detection of such attacks.

In future SDN-based networks, network control logic will be moved to an external entity, referred to as the SDN controller, which will...
Figure 2. Example of a critical location attack targeting latency sensitivity of synchronous replication between data centers. Paths P1 and P2 are assumed to satisfy the latency requirements, while P3 does not.

serve as a network-wide operating system. This approach offers many benefits, such as simplification of network management, easier modification of control software, and increased control to carriers and enterprises. It can also provide enhancements in security by providing a centralized point of control to collect and distribute security information. However, the SDN controller will also be prone to a whole set of new security threats since a compromised controller can compromise the entire network [2]. At the physical layer, a centralized architecture for the SDN controller would make it very vulnerable to physical infrastructure attacks, presenting a single point of attack. Due to reliability and scalability issues, the SDN controller will most likely be implemented in a distributed computing environment. However, a coordinated attack could still target the multiple locations of the SDN controller, causing massive system-wide damage.

**A Physical-Layer Security Framework**

Operators are often faced with making complex routing and fiber infrastructure deployment decisions, which are heavily influenced by security issues and involve significant trade-offs with route length, latency, and cost. For example, networks across the Middle East could be serving traffic between Asia and Europe (approximately 7 TB exchanged every second) with the lowest latency due to the shortest physical distance. However, due to security risks and geo-political factors, most commonly alternative, longer options (e.g., SEA-ME-WE undersea cables or routes via Russia or the Pacific and Atlantic Oceans) are used.

To aid operators in making the network provisioning decisions considering physical-layer security, we present a general security framework for evolving optical networks, outlining potential approaches to mitigate, avoid, and/or reduce the damage caused by the aforementioned physical-layer attacks, as summarized in Table 2. Depending on the quality of protection and security required, we propose to differentiate between best effort, standard protected, and gold user connections, and define security-aware provisioning methods corresponding to each class.

**BEST EFFORT**

Best effort connections would rely on efficient network planning approaches, but no specific protection resources would be allocated. Attack-aware optical network planning, original-
As a second line of defense, standard defragmentation would be used to protect data from eavesdropping. In cases where physical infrastructure attacks are a concern, attacker can be dropped to allow for spectral reallocation over “safer” paths, even if extra bandwidth is required.

### Table 2. The proposed security framework indicating possible approaches to deal with physical-layer attacks for best effort, standard protected, and gold user connections.

<table>
<thead>
<tr>
<th>Approach</th>
<th>Best effort connections</th>
<th>Standard protected connections</th>
<th>Gold user connections</th>
</tr>
</thead>
<tbody>
<tr>
<td>IDS</td>
<td>Soft AA-RWA/RSA schemes</td>
<td>Spectrum reallocation with priority over best effort channels</td>
<td>Hard Survivable AA-RWA/RSA schemes</td>
</tr>
<tr>
<td>Encryption</td>
<td>Optical encryption</td>
<td>IDS with a less sensitive reaction threshold</td>
<td>Optical encryption – Hard AA routing schemes</td>
</tr>
<tr>
<td>Restoration</td>
<td>-IDS for detection</td>
<td>-IDS with a more sensitive reaction threshold</td>
<td>-Proactive recovery using fiber tapping detection methods</td>
</tr>
</tbody>
</table>

### Figure 3. An example of: (a) a non-attack-aware; (b) a soft attack-aware RWA scheme for in-band jamming attacks in FOADM-based networks. While both schemes use equal resources in terms of wavelength links, the soft attack-aware solution faces less potential damage in the presence of in-band jamming.

As an example, a survivable AA-RWA scheme that ensures that the primary path and backup path of protected connections cannot be affected by the same jamming signal was proposed in [12].

Where spectrum is scarce and larger spacing cannot be allocated to reduce the effects of mixed modulation attacks in MLR and elastic networks, interleaving higher-priority (protected) and lower-priority (best effort) connections can ensure that at most one higher-priority connection is affected by an attack on a link. Furthermore, if higher-priority connections are alternated with sets of best effort channels in elastic networks, lower-priority channels could potentially be dropped to allow for spectral reallocation of higher-priority connections in case of an attack, as illustrated in Fig. 4. Besides attack scenarios, interleaving low- and high-priority connections could also leave more space for real-locating higher-priority connections in case of standard defragmentation.

As with best effort traffic, encryption methods would be used to protect carried data from eavesdropping. As a second line of defense for tapping and other component insertion attacks, power drops detected by the IDS would incur restoration. Two alarm thresholds could be incorporated into the IDS, where only alarms corresponding to the higher threshold would trigger reaction mechanisms for standard protected users, to avoid excessive false alarms and frequent restorations.

Regarding physical infrastructure attacks, regular protected users would rely on standard shared or dedicated path protection for single link and node failures. Such protection schemes typically do not consider multiple failures and disasters, and thus would not specifically deal with disaster-like and critical location attacks.

### Gold Users

The user class that we refer to as gold users would comprise a smaller portion of customers who are willing to pay more for gold services in terms of security. Such connections would be provisioned with special dedicated 1+2 attack-aware path protection schemes. Security measures for gold users would potentially incur higher costs, but would provide enhanced protection.

As opposed to soft attack-aware planning schemes, which are not aimed at protecting individual connections, gold users would be specifically routed over “safer” paths, even if extra bandwidth is required.
resources are required. We refer to this as hard attack-aware (AA) planning. Hard (survivable) AA-RWA schemes for gold users could be applied in future SDN networks by incorporating data and statistics collected by the SDN controller to avoid using untrustworthy links, nodes, domains, or channels. Since attacks can occur sporadically, nodes and links that have previously raised alarms or have shown suspicious behavior, such as power fluctuations indicating jamming, transients, or tapping, could be avoided. Furthermore, gold users could be routed over paths or domains that do not support alien wavelengths. They could also be routed to avoid higher-risk areas, such as locations of major military facilities or political unrest, which are more probable targets of WMD and EMP attacks. As an example, in 2012 Gulf Bridge International (GBI) integrated a terrestrial link in addition to its submarine network for Asia-Europe route diversity that avoids Egypt [13]. Routing schemes could also favor links and nodes employing better optical monitoring equipment or more advanced technology, such as amplifiers with more effective transient control or programmable transponders that can change the modulation format if the quality of transmission falls below a certain threshold.

Gold users would also take advantage of spectral reallocation in elastic networks in case of a mixed modulation or jamming attack. Larger guardbands could be employed to ensure sufficient space for reallocation, as well as enhanced versions that allow other standard protected connections to be re-allocated and best effort connections to be torn down to make room for the necessary shift of gold users.

In addition to optical encryption methods to protect against eavesdropping, another layer of privacy could be provided for gold users transmitting highly sensitive information by applying optical steganography where secret so-called stealth channels are hidden among regular public channels [6, 14]. In future dynamic networks, with fast setup times, enhanced privacy for mission-critical demands could also be realized using route hopping [1]. In this approach, the route of a connection is changed rapidly to avoid eavesdropping, analogous to frequency hopping in military radio applications.

Figure 4. An example of the effects of distinct spectrum allocation schemes in the presence of a mixed modulation attack.

Assuming a double threshold IDS, the more sensitive reaction threshold would be applied to gold users to cease transmission on the intruded path. The proactive recovery scheme described in [15] could also be applied, where connections are preemptively restored before conventional performance monitoring detects power variations or before a failure (e.g., a fiber cut) actually occurs by detecting vibrations and other environmental variations around a fiber.

For physical infrastructure attacks, the dedicated path protection schemes designed for gold users not only could tolerate single component failures, but would also be designed to deal with multiple failures or attacks. A disaster-resilient protection and content placement scheme for data center networks was proposed in [8] where anycast routing is exploited to protect mission-critical demands accessing data center content by routing their protection paths to data centers in different geographical regions. Similar approaches should be investigated to deal with critical location attacks.

**Main Research Challenges and Opportunities**

Developing new attack-aware planning methods for both current and future networks brings forth many new research challenges and offers opportunities for novel research ideas on topics such as latency-aware and attack-aware anycast protection schemes for inter-data center networks, and enhanced survivable RSA schemes and defragmentation algorithms for elastic and flexible-grid networks. At the component level, improvements in amplifier transient control will play an important role, not only in the context of attacks, but also as a prerequisite for future dynamic networks. Furthermore, perhaps one of the biggest challenges in the context of future optical networks security will lie in the secure design of the SDN controller in future SDN-based networks. Methods to exploit the benefits of this centralized approach for secure network planning should be developed in hand with effective methods to protect the SDN controller, not only from physical infrastructure attacks, but also from unauthorized access. That is, breaching the
SDN controller and gathering intelligence on the network topology, routing, availability, and so on could be used to realize any of the aforementioned attacks. As such, secure authentication and access will pose big challenges, particularly as network management systems become accessible via mobile phone applications (and not only from secured offices). Advances in optical encryption methods can aid in the development of new authentication schemes and help protect user privacy, while effective alarm correlation and attack localization algorithms for improved IDS software can further facilitate attack detection and isolation.

**CONCLUSION**

Due to increasing bandwidth and performance requirements, primarily stemming from the growth of cloud services, security issues in optical networks are of ever increasing importance. Although the dynamicity and flexibility associated with future optical networks offers many advantages, it also incurs additional security vulnerabilities, which must be identified in order to develop efficient and cost-effective countermeasures. In this article, an overview of physical-layer attacks in evolving optical networks is given, identifying various attack scenarios, their potential consequences, and possible entry points for attackers. A general security framework is then proposed, describing potential methods for dealing with such attacks for different classes of users aimed at reducing the overall network vulnerability. This proposed framework serves as a starting point for developing new and enhanced security solutions in current and future optical networks.

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**References**

C-RoFN: Multi-Stratum Resources Optimization for Cloud-Based Radio over Optical Fiber Networks

Hui Yang, Jie Zhang, Yuefeng Ji, and Young Lee

ABSTRACT

C-RAN has become a promising scenario to accommodate high-performance services, which provides ubiquitous user coverage and supports real-time cloud computing using cloud BBU’s. The interaction between RRH and BBU or resource schedule among BBU’s in cloud have become more frequent and complex due to the development of system-scale and user requirements. The heavy-duty interaction can promote the networking demand among RRHs and BBUs, and forces to form elastic optical fiber switching and optical networking according to the characteristics of high bandwidth, low cost, and transparent multi-rate traffic transmission. In such a network, the multiple stratum resources of radio, optical, and BBU processing unit have interwoven with each other, so the traditional architecture cannot efficiently implement the resource optimization and scheduling for the high-level QoS guarantee. In this article, we present a novel C-RoFN architecture for MSRO using software defined networking. The proposed architecture can globally optimize radio frequency, optical spectrum, and BBU processing resources effectively to maximize radio coverage and meet the QoS requirement. The authors present a novel C-RoFN architecture for MSRO using software defined networking. The proposed architecture can globally optimize radio frequency, optical spectrum, and BBU processing resources effectively to maximize radio coverage and meet the QoS requirement.

INTRODUCTION

Due to the rapid evolution of the fifth generation (5G) mobile network, network operators are rethinking the way their networks are controlled to provide efficient access between resources and users [1]. A large number of users or providers are deploying their services in radio access networks (RANs), which are characterized by supporting higher data rates, excellent end-to-end performance, and ubiquitous user coverage with lower latency, power consumption, and cost [2]. To adapt to 5G requirements, the cloud RAN (C-RAN) is a paradigm introduced by operators that aggregates all base stations’ computational resources into a cloud baseband unit (BBU) pool [3], while the radio frequency signals from geographically distributed antennas are collected by remote radio heads (RRHs) and transmitted to the cloud platform through an optical transmission system [4]. C-RAN can reduce the number of cell sites by maintaining similar coverage and enhance real-time cloud computing by offering better services [5], which cuts down capital and operating expenditures (CAPEX and OPEX) of a network.

Recently, the interaction between RRH and BBU or resource schedule among BBUs in cloud (e.g., virtual migration inter-BBU) have become more frequent and complex due to the development of system-scale and user requirements (e.g., ubiquitous access in the Internet of Things). Due to the diversity and hugeness of user demands, a large number of high-performance services have presented high burstiness and high heterogeneity, especially for multi-rate traffic in fronthaul. The heavy-duty interaction caused by services can promote the networking demand among RRHs and BBUs, and force flexible optical fiber switching and optical networking to be formed [6] according to the characteristics of high bandwidth, low cost, and transparent multi-rate traffic transmission. The traditional technology of the optical network (e.g., wavelength-division multiplexing) is inefficient and unsuited for carrying these services flexibly due to the strictly fixed International Telecommunication Union Telecommunication Standardization Sector (ITU-T) wavelength grids and spacing. The elastic optical network (EON) is achieved by taking advantage of orthogonal frequency-division multiplexing (OFDM) technology [7], which can allocate necessary spectrum resources at fine granularity for various user connection demands, and offer cost-effective and highly available connectivity channels [8]. In such a network, the multiple stratum resources of radio, optical, and...
BBU processing unit have interwoven with each other, so the traditional architecture cannot efficiently implement the resource optimization and scheduling for high-level quality of service (QoS) guarantee. In addition, as a centralized software control architecture, the software defined networking (SDN) enabled by OpenFlow has gained popularity by supporting programmability of network protocols and functionalities [9, 10], which can provide unified control over various resources for the joint optimization of functions and services with a global view [11–13]. Therefore, it is very important to apply SDN technique to control and optimize the resource assignment in such an environment.

Cross-stratum optimization (CSO) with the benefits of SDN is proposed as an architecture that allows global optimization and control across an elastic optical network (EON) and data center application stratum heterogeneous resources to meet the QoS in our previous work [14]. On the basis of it, this article extends to propose a cloud-based radio over fiber optic network (C-RoFN) architecture with multi-stratum resources optimization (MSRO) using SDN. The C-RoFN architecture can provide the basis of resource scheduling and management. The additional processing time caused by digitization can be avoided in C-RoFN to enhance the responsiveness to end-to-end user demands. Based on the proposed architecture, the global evaluation strategy is introduced to give a global evaluation factor, which can consider all multi-stratum resources using a unified measure. Under the centralized scheduling, radio frequency, optical spectrum, and BBU processing resources should be globally optimized effectively to maximize radio coverage and meet the QoS requirement. The overall feasibility and efficiency of the proposed architecture are experimentally verified on an SDN-enabled testbed in terms of resource occupation and path provisioning latency.

The rest of the article is organized as follows. We describe the advancement of C-RoFN architecture and functional modules in detail. A cooperation procedure for C-RoFN is proposed and discussed. A global evaluation strategy is proposed based on C-RoFN. The testbed deploying CSO with OpenFlow-enabled devices is presented and discussed with numerical results. Finally, we conclude the article and discuss future research issues.

**Technology Advancement Statement**

In the current C-RAN, the digitized signal is transmitted in the fronthaul network between the RRH and BBU, such as in the common public land mobile network radio interface (CPRI). The digitization can abstract and sample the radio frequency and change it into a discretized digital signal, which promotes the radio’s capacity of resisting disturbance through transmission. However, the disadvantage of digitization has become more and more clear, especially with regard to the development and number of user requirements. First, much traffic has been aggregated into fronthaul, while digitization may occupy larger bandwidth and bring enormous overhead to an optical network. Second, the digitized signal transmission needs the deployment of analog-digital conversion in antennas. Since different data compres-

Since different data compression technologies require various bandwidth, operators expect a just-right-size spectrum for each data path to improve spectrum utilization. This means that multi-type signals should adopt different encoding and decoding modes, so as to increase the complexity of the system design. Third, the radio and optical resources have been separated due to radio signal digitization, and are controlled in various parts without unified control. To address these issues, this article proposes a C-RoFN architecture with elastic optical networking under unified SDN orchestration. The advancement of the proposed solution is threefold. First, the analog signal in the architecture can reduce the fronthaul bandwidth caused by digitization. Second, since a flexi-grid optical network is used to connect the RRH and BBU, multi-rate or multi-type traffic can be carried with elastic spectrum switching capability through transparent transmission. Also, the additional processing time caused by digitization will be avoided to meet the delay requirement of under a millisecond. Third, due to the same physical essence in electromagnetic wave theory, the radio and optical spectrum can be centrally controlled and scheduled with SDN orchestration in a unified manner to enhance the responsiveness to end-to-end user demands. C-RoFN globally optimizes radio frequency, optical spectrum, and BBU processing resources effectively to maximize radio coverage and meet the QoS requirement with vertical integration and horizontal merging.

**C-RoFN Architecture for Multi-Stratum Resources Optimization**

We present the C-RoFN architecture to implement MSRO. The networking modes for C-RoFN have extended in two orthogonal directions, as shown in Fig. 1. One mode is from the perspective of resource form. In this mode, optical and computing resources are interconnected across the optical network, and BBU strata along the east-west direction. This leads to the interconnection and networking of heterogeneous resources cross-stratum in a latitudinal direction, which is established as cross-stratum. The other mode is from the perspective of carrying capacity. The related entity with small granularity of switching can be abstracted as the high-layer network (e.g., radio network), while the related entity with large granularity of switching should be abstracted into the low-layer network (e.g., EON). The networking of multiple layers is established along a longitudinal direction, which is called multi-layer. Based on the networking modes, three C-RoFN applications in this architecture are formed: the interaction between RRHs (e.g., collaborative radio), the service from RRH to BBU, and resource scheduling among BBU (e.g., virtual resource migration inter-BBU). The logical relationship among the networking modes and the application scenarios for C-RoFN is shown in Fig. 1.

**C-RoFN Architecture**

The software defined C-RoFN architecture for MSRO is illustrated in Fig. 1. The EON is used to interconnect the cloud BBU’s, which deploy
network and processing (e.g., computing and storage) stratum resources, respectively. The distributed RRHs are interconnected and converged into an EON, which allocates the customized spectrum with finer granularity for RF signals. Note that the C-RoFN consists of three strata: radio resource, optical spectrum resource, and BBU processing resource. Each resource stratum can be software defined with OpenFlow protocol (OFP) and controlled by a radio controller (RC), an optical controller (OC), and a BBU controller (BC), respectively, in a unified manner. To control heterogeneous networks for MSRO with OFP, OFP-enabled RRH and a bandwidth-variable optical switch with OFP agent software are required, which are referred to as OF-RRH and OF-BVOS, respectively, as proposed in [14]. The motivations for MSRO in software defined C-RoFN architecture are twofold. First, the MSRO can emphasize the cooperation between the RC and OC to overcome the interworking obstacles derived from multi-layer overlaid networks, and it effectively realizes vertical integration. Second, in order to provide the end-to-end QoS, multiple stratum resources can be merged through controllers’ interaction with horizontal merging, while achieving global cross-stratum optimization of EON and BBU resources.

The proposed architecture with EON leverages flexible transponders and elastic optical switches to interconnect the RRH and BBU. In the current environment, optical devices (e.g., flexible transponders) are much more expensive than electric equipment generally due to the costly optical module. With the evolution of optics component technology, the cost of flexible optical modules has declined continuously, especially employing photonic integration. It can contribute to lessening the cost difference between digital and analog solutions. Additionally, due to the continuous extension of the optical communication ecosystem, the market size may react to cost reduction of device manufacture. Note that we need time and have to make efforts on the development of this market. The innovation of flexible optical devices may constitute a high-efficiency and low-cost solution so that optical communication ecosystem will present a virtuous circle.
Radio Controller
The RC is responsible for analyzing RF status with RF resources maintained in the RF domain, and performs multi-layer resource vertical integration with other controllers. The corresponding functional modules are described as follows.

Radio Frequency Monitoring: is responsible for compiling and managing the status of OF-RRH via OF control, interworks with processing resources through the radio-BBU interface (RBI), and provides the RF information to the control allocation module.

Radio Frequency Allocation: is the core module of RC, which can allocate the RF resource for the services and provide the resource integration request to the OC via radio-optical interface (ROI) including QoS parameters.

OpenFlow Control: is used to send flow modification messages to assign the RF by updating the control entries of RRH while receiving their utilized status.

Optical Controller
The OC sustains optical fiber network information abstracted from a physical network, and accordingly performs lightpath provisioning in an EON to achieve CSO of networks and application resources and multi-layer resource integration. A typical OC consists of six essential modules, which are described as follows.

MSRO Control: is the core module for messages scheduling to other modules. It can perform the integration request through ROI and forward it to a path computation element (PCE), returning a success reply containing the lightpath information.

Path Computation Element: A PCE is capable of computing an end-to-end network path or route based on a network graph, and applying computational constraints.

CSO Control: is an engine on the location selection of a BBU server using CSO according to the status of EON and BBU processing stratum resources. The latter is perceived by the CSO agent in the BC via the optical-BBU interface (OBI).

Network Virtualization: can abstract and manage the network topology from a physical network through PCE calculation, and provide abstracted resource information to a CSO control module.

Spectrum Control and Monitor: can monitor physical optical network elements and flexibly control the spectrum bandwidth and modulation format in the underlying network. The end-to-end elastic lightpaths are provisioned by controlling all corresponding OF-BVOSs along the computed path by using OFP.

Database Management: conserves the elastic lightpath information after the connection setup from PCE and the abstract topology information from the network virtualization module.

BBU Controller
The centralized BC is responsible for monitoring and allocating processing resources (e.g., computing and storage) in BBUUs and arranging the services. The corresponding functional modules are described as follows.

CSO Agent: is the communication module to interact with OC and RC through OBI and RBI, and provides computing and storage resource utilization periodically or based on event-based trigger.

BBU Monitoring: can monitor and maintain virtual processing resources obtained from BBUUs periodically.

Database: stores the abstract computing and storage information from BBUUs.

Cooperation Procedure for MSRO in C-RoFN with Different Service Modes
Cooperation among the RC, OC, and BC in various service modes is one of the key issues for MSRO in C-RoFN. It helps to achieve the scenarios of multi-layer resource vertical integration and cross-stratum resource horizontal merging for MSRO. This section summarizes different cooperation procedures among three types of controllers. In the control plane, there are two kinds of scheduling ways and procedures among the controllers. One control manner is the load distribution, while the other is the distribution determined by control functionality. Note that the heterogeneous resources have intertwined with each other in C-RoFN, where each resource stratum has its own functionality achieved by a corresponding controller. Therefore, we adopt the functionality distribution for the control manner and procedure, while each controller (i.e., RC, OC, and BC) controls the corresponding resources and completes the functionalities. If network architecture just contains one kind of resource (e.g., RF), the generic load distribution should be the optimal choice in the control plane.

Multi-layer Vertical Integration Model
The multi-layer resource vertical integration mode in C-RoFN can utilize radio and EON resources effectively. It is necessary for the RC to collaborate with the OC and BC when a service request needs to cross RF and optical spectrum layers in such a model. Thus, the carrying and integration of multi-layer resources is best done in the vertical direction. The OC and RC can fulfill the spectrum and RF resource assignment in their own layers, respectively.

Multi-layer resource vertical integration mode in C-RoFN could ensure the QoS of a user who requires the services, which is shown in Fig. 2. For the status detecting RoFN in real time, the RC sends a flow monitor request to each OF-RRH periodically via OFP, while obtaining the RF status information from them and interweaving the processing resources with the BC. If a new request arrives at the OF-RRH for the service, the device forwards this request to the RC. The multiple layer resource vertical integration control can be triggered in the RC, which then sends the request to the OC. After session establishment, the OC receives the resource vertical integration request, estimates the request status with GES, and computes a path considering the CSO of the EON and BBU processing resources cooperating with the BC. Then the OC proceeds to set up an end-to-end elastic lightpath by controlling all corresponding OF-BVOSs along the computed path by using OFP. When
the OC obtains setup success reply from the last OF-BVOS, it responds the integration reply to RC with provisioning lightpath and abstracted optical spectrum resources information. After that, the RC sends an setup message to the RRH such that the radio frequency is modulated to the optical spectrum for utilizing the multiple layer resources effectively, while the computing and storage resource usage in BC is updated to keep the synchronization by receiving the update message from RC. Note that, these existing OpenFlow messages are reused to simplify our implementation, which are interworking and illustrated in the higher-right corner of Fig. 2.

CROSS-STRATUM HORIZONTAL Merging MODEL

Cross-stratum resource horizontal merging model in C-RoFN architecture could ensure the high-performance QoS of the demands among BBUs, for example, content distribution, traffic load balancing, and BBU migration. Figure 3 shows the cooperation procedure of cross-stratum horizontal merging model for BBU services in the proposed architecture. The OC monitors the spectrum information of optical nodes by the interaction with OF-BVOSs, while the BBU stratum resources are monitored and maintained in the BC. Through interworking the BBU processing resources, the BC sends the service request to OC to ask for the optical network resources information. After the session establishment, the global evaluation strategy with cross stratum horizontal merging in OC can be completed to choose the optimal destination node based on various service types and parameters, and the utilization of optical network resources over a period of time, and then responds to the setup request. The end-to-end elastic lightpath is set up by controlling the corresponding OF-BVOS along the computed path by using OFP. When the OC obtains a setup success reply from the last OF-BVOS, it responds with a setup reply to the BC provisioning a lightpath. Then the BBU service can be transmitted through the elastic lightpath. After that, the processing usage in the BC is updated to keep synchronization by receiving the update message from the OC.

GLOBAL EVALUATION STRATEGY IN C-RoFN

In the C-RoFN, the multiple stratum resources of radio network, optical network, and BBU processing resources are deployed in this scenario used for optimization. The C-RoFN architecture can provide the basis of resource scheduling and MSRO. The traditional resource evaluation strategies assess the resource utilization only partly considering one kind of resource. Based on functional architecture, we propose a global evaluation strategy (GES) to give a global evaluation factor [15], which can consider all multi-stratum resources using a unified measure. Under the centralized scheduling, RF, optical spectrum, and BBU processing resources should be globally optimized by GES effectively to meet the QoS requirement.

The GES strategy first selects a new BBU based on the processing status collected from a BBU, and the radio and optical condition provided by RC and OC dynamically. To measure the choice rationality of service provisioning, we define the global evaluation factor, which considers all multi-stratum parameters. The CPU usage and storage utilization describe the current usage of BBU resource, while optical network parameters comprise the traffic engineering weight of the current link, and the latency and hop of the candidate path. Radio parameters contain the symbol rate and RF of the current radio signal. We assume that the BBU node includes computing and storage resources, while the BBU pool can be seen as a data center. According to BBU resource utilization, the GES strategy first selects the best K candidate BBU nodes with minimum BBU usage in the BBU stratum preparing for radio signals and a continuous spectrum path. In radio and optical strata, the node with minimum value based on the global evaluation factor will be selected from the K candidates. Note that the GES strategy uses the global evaluation factor to choose the optimal destination BBU node, and allocate the optimal radio and optical spectrum resources to globally optimize the RF, optical spectrum, and BBU processing resources. Then the GES strategy can complete the path computation in the connection and service parameters constraints, and decide to allocate and adjust the radio frequency and optical spectrum resources for the request according to QoS priority, request status, and resource utilization selected from controllers. The elastic lightpath can be established with spectrum and modulated RF allocation through OFP between the source and destination nodes after the choice of BBU.
EXPERIMENTAL SETUP AND RESULTS

To evaluate the feasibility and efficiency of the proposed architecture, we set up an EON with software defined C-RoFN based on our testbed including both control and data planes, as shown in Fig. 4. In the data plane, two analog RoF intensity modulators and detect modules are utilized, driven by a microwave source working at 40 GHz frequency to generate double sideband. Four OF-enabled elastic reconfigurable optical add/drop multiplex (ROADM) nodes are equipped with Finisar bandwidth variable wavelength selection switches (BV-WSSs) in EON. We use Open vSwitch (OVS) as the software OFP agent according to the application programming interface (API) to control the hardware and interact between the controller and radio and optical nodes. In addition, OFP agents are used to emulate other nodes in the data plane to support the C-RoFN with OFP. The BBU cloud and OFP agents are realized on an array of virtual machines created by Redhat VMware ESXi V5.1 running on IBM X3650 servers. Since each virtual machine has an operating system, its own CPU, and storage resource, it can be considered as a real node. The virtual operation system technology makes it easy to set up experiment topology for large-scale extension. For the OF-based C-RoFN control plane, the OC server is assigned to support the proposed architecture and deployed by means of three virtual machines for MSRO control, network virtualization, and PCE strategy as a plug-in, while the RC server is used for RF resource monitoring and assignment. The BC server is deployed as a CSO agent to monitor the computing resources from BBUs. Each controller server controls the correspond-
Based on the testbed, we have designed and verified experimentally the MSRO for service in software defined C-RoFN. Figures 5a and 5b present the whole signaling procedure for MSRO using OFP through a Wireshark capture deployed in the OC and RC, respectively. The features request message is responsible for monitoring by regularly querying OF-BVOSs about the current status. The OC obtains the information from OF-BVOSs via a features reply message. The RC obtains the service usage of BBU resource through the interworking between the RC and BC via a User Datagram Protocol (UDP) message, where we use UDP to simplify the procedure and reduce the performance pressure of controllers. When a new request arrives through a packet in a message, the RC sends the request to the OC for MSRO via a UDP message. After completing the GES strategy, the OC obtains computing and storage usage from the BC through a UDP message, and then computes a path considering CSO with multiple stratum resources to realize cross-stratum resource horizontal merging. Then the OC and RC provision spectral path and assign the RF to implement multi-layer resource vertical integration, and control the corresponding nodes via a flow mod message. Then the OC updates the resources usage to the RC with a UDP message to keep the synchronization. The experimental results correspond to the procedures depicted above.

The spectrum of lightpath for analog C-RoFN is reflected on the filter profile, as shown in Fig. 5c. The radio signals can be modulated on the spectral channel with MSRO in C-RoFN. Two 40 GHz band signals are multiplexed onto two elastic optical spectral channels, and these optical channels can be controlled to carry the radio signals by the SDN orchestration.

We also evaluate the performance of MSRO with GES strategy under a heavy traffic load scenario of C-RoFN and compare it to the traditional CSO strategy [14] using virtual machines. The requests are set up with bandwidth randomly distributed from 500 MHz to 40 GHz, where the spectrum slots in the EON are 12.5 GHz. The service processing usage in a BBU is selected randomly from 1 to 0.1 percent for each demand. They arrive at the network following a Poisson process, and results have been extracted through the generation of 100,000 demands per execution. After deciding the destination BBU node, we accommodate the service considering RF assignment with available optical spectrum first, and then use another suitable optical spectrum resource. The first-fit principle can be performed as the RF and spectrum assignment for the provisioning path between the source and destination. Then the path can be established with spectrum and modulated RF allocation through OFP after the choice of the BBU. Figures 6a and 6b compare the performance of two strategies in terms of resource occupation rate and path provisioning latency. Resource occupation rate reflects the percentage of occupied resources to the entire radio, optical network, and BBU resources. As shown in Fig. 6a, the GES strategy can enhance the resource occupation rate more effectively than the other strategy, especially when the network is heavily loaded. The reason is that the GES strategy can globally optimize the radio, optical, and BBU stratum resources to maximize radio coverage, and realize cross-stratum resource horizontal merging and multi-layer resource vertical integration in two directions considering three dimensionalities of resources. Figure 6b shows that the GES strategy can reduce the path provisioning latency compared to the other. The latency reflects the average setup delay including computing and procedure time. That is because the GES strategy chooses the destination BBU considering RF and spectrum resources allocation before the service arrives, which leads to low computation and provisioning time.

**Conclusion**

In order to meet the QoS requirement of 5G network services, this article presents a novel C-RoFN architecture for multi-stratum resources optimization using software defined networking, which can allow the multi-layer vertical integra-
Numerical results show that the MSRO in C-RoFN with GES strategy can utilize across radio frequency, elastic optical network and BBU processing stratum resources effectively to maximize radio coverage with higher responsiveness to end-to-end service demands.

Figure 6. Comparison on a) resource occupation rate and b) path provisioning latency in heavy traffic load scenario between various strategies.

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Optical Communications

Control Plane Solutions for Sliceable Bandwidth Transceiver Configuration in Flexi-Grid Optical Networks

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Abstract

In EONs, the establishment of flexi-grid optical connections requires configuring both BVTs and bandwidth variable cross-connects. Such configuration has to be performed through proper control plane architecture and extensions, accounting for the client signal data rate, the required modulation format, the adopted forward error correction mechanisms, the number of carriers, and the required spectrum resources. Nowadays, two candidate control plane architectures with similar functions (like endpoint and node addressing, automatic topology discovery, network abstraction, path computation, and connection provisioning) can be adopted: distributed GMPLS — with optional PCE and instantiation/ modification — and a control plane based on SDN with a logically centralized controller and an open protocol such as OpenFlow. In this work, by relying on the data modeling of 1 Tbit/s SBVTs, we aim to design and provide feasible control plane enhancements for both GMPLS/PCE and SDN/OpenFlow architectures to automatically configure endpoint SBVTs while dynamically establishing end-to-end connections in EONs. Such extensions, successfully validated in experimental scenarios, are qualitatively compared to ease eventual EON operator selection.

Introduction

The recent deployment of the flexi-grid technology together with the introduction of coherent detection strategies for high rate optical transmissions have enabled the successful implementation of a single-carrier 100 Gb/s transponder that is, a bandwidth variable transponder (BVT). Such a transmission signal requires frequency slots (FSs) occupying 37.5 GHz of the flexi-grid spectral resources [1]. The advantages of BVTs are to generate optical signals supporting a number of modulation formats, forwarding error correction (FEC) codes, and bit rates that are dynamically selected according to the transported client signals and network conditions (transmission distance, physical impairments, etc.) [2]. The optical signal being created uses the same FS (defined by the pair of central frequency parameter n and slot width parameter m [3]) on each link between the ingress and egress nodes. This is referred to as the spectrum continuity constraint. The bandwidth-variable optical cross-connect (BV-OXC) for each traversed network node is configured based on the selected FS (i.e., n and m).

To further improve the overall spectral resource efficiency, flexibility, and reconﬁgurability of elastic optical networks (EONs), a new generation of transponders, called sliceable BVTs (SBVTs), is expected to be deployed in the near future [4]. SBVTs support generating multiple flows (subcarriers) each operated at, for example, 100 or 200 Gb/s bit rates, enabling, for instance, the creation of super-channel optical connections achieving transmission rates of 1 Tbit/s.

High flexibility in SBVTs will be enabled by the capability to configure each subcarrier with different transmission parameters ( modulation format, FEC, etc.), to be adapted according to the requested optical reach. Moreover, the subcarriers will have the possibility to be either independently routed in the network (applying the sliceable functionality) or efﬁciently co-routed along a common path in the EON. These co-routed subcarriers compose the above mentioned super-channel connections.

Within super-channels, no guard bands among subcarriers are required. Instead, guard bands are usually required between independent channels to account for the non-zero roll-off function of the traversed filters, applied by the bandwidth-variable wavelength selective switches (BV-WSS) implemented in the optical BV-OXC. The appealing flexibility benefits brought by SBVTs in EONs also impose important challenges to automatically setting up end-to-end flexi-grid connections. The aim of this work is to detail feasible EON control plane enhancements enabling the explicit control of both transit BV-OXCs and reconfigurable SBVT parameters (e.g., modulation format, bit rate, FEC) at the endpoints. By doing so, the flexibility and beneﬁts in terms of spectrum efﬁciency provided by EONs are fully leveraged. Hence, the protocol enhancements, evaluation, and standardization...
of a control plane for the new generation 1 Tb/s SBVT are addressed. To this end, any control protocol could be used to steer the configuration of the SBVT parameters. However, herein we consider the two most widely adopted network control plane solutions: the traditional general multiprotocol label switching (GMPLS)/path computation element (PCE) control plane technology and the software defined networking (SDN) approach using the OpenFlow protocol. For both solutions, the proposed SBVT-oriented control extensions are specifically designed to support multi-vendor SBVT interoperability, thanks to the definition of a common data information model.

A qualitative comparison between both distributed and centralized solutions for effectively controlling endpoint SBVTs is provided to enable the identification of pros and cons associated with each control plane option.

**Sliceable Bandwidth Variable Transponder**

This section describes the considered SBVT architecture along with the definition of all the key parameters to be configured when creating either a single optical flow or a super-channel connection (two or more subcarriers). These key parameters represent a generic SBVT data modeling that is then used for the control protocol implementation in either the GMPLS/PCE or SDN context.

**Architecture**

The SBVT, also called multi-flow optical transponder, enables the client tributary traffic to be transported in the network over a number of $C$ optical subcarriers, with transmission parameters configured according to the required optical reach. For example, 1 Tb/s SBVT can be implemented with $C = 10$ subcarriers operating at 100 Gb/s polarization multiplexed quadrature phase-shift keying (PM-QPSK) or with $C = 5$ subcarriers, each transmitted at 200 Gb/s PM 16-quadrature amplitude modulation (PM-16-QAM).

The architecture of a reference SBVT is reported in Fig. 1 [4, 5]. It includes the modules below.

**Tributary Interface**: The interface electronically processes the client tributaries. This way, the content provided by the IP layer can be suitably transmitted at the photonic layer. The client traffic is flexibly adapted according to a predefined granularity (e.g., 100 Gb/s) to optical flows. The adaptation is typically performed in the framework of the optical transport hierarchy (OTN) within the optical transport network (OTN), as detailed in [6]. The interface also implements key functionalities for FEC mechanisms, and operation, administration, and maintenance (OAM).

**Flow Distributor**: It is an electronic switching matrix enabling each OTN flow provided by the tributary interface to be routed toward any subcarrier. The flow distributor is an optional component providing additional flexibility in provisioning and restoration operations.

**Subcarrier Generation**: It consists of either an array of $C$ independent laser sources or a single multi-wavelength source [5]. At the transmitter side, it is used for carrier generation. At the receiver side, it is used as a local oscillator.

**Multi-Flow Module**: It consists of $N$ subcarrier modules where each flow is modulated/detected on the related optical subcarrier. At the transmitter side, digital-to-analog conversion (DAC) is typically exploited to configure multi-level signals for high order modulation formats (e.g., 16-QAM). Shaping filters are also included to improve transmission performance. At the receiver side, coherent detection is exploited. In particular, high-speed analog-to-digital converters (ADCs) are exploited.

**Multiplexer/Coupler**: All subcarriers are optically coupled/multiplexed into a single add/drop port of the BV-OXC. The BV-OXC typically consists of a broadcast or switch and select architecture implemented with BV-WSS [7].

**Super-Channel and SBVT Data Model**

For the above considered SBVT architecture, a number of key parameters need to be configured for generating a super-channel connection and for each individual subcarrier. In this sense, Fig. 2a depicts a (class) diagram via Unified Modeling Language (UML) determining the data model to be used when setting up a super-channel (formed by one or more subcarriers; Fig. 2b) and configuring each involved subcarrier.

The parameters and attributes related to the super-channel creation include the following.

**Number of Active Subcarriers**: (a non-zero positive integer). In the case of fully loaded SBVTs, all $C$ subcarriers are activated to carry the tributary traffic at full rate. Optionaly, under networking conditions where the full rate is not required, only a subset $c \leq C$ of the available subcarriers may be activated.

**Super-Channel Frequency Slot**: (freq_slot). The frequency/optical spectrum range allocated to the super-channel connection. As stated above, an FS is completely defined by its nominal central frequency (NCF) using the parameter $n$ and its slot width (parameter $m$). In the data model, we use the object label to specifically determine the FS. Label format (Internet Engineering Task Force [IETF] RFC 7699) is defined by the grid parameter (i.e., International Telecommunication Union Telecommunication Stan-
Figure 2. a) UML for the super-channel and SBVT creation and configuration; b) examples illustrating a super-channel formed by two subcarriers and candidate subcarrier configuration according to CFG and SWG parameters.

dardization Sector [ITU-T Flex [3] in EONs), the channel spacing (CS) specifying the spacing between adjacent channels (e.g., 6.25 GHz), the identifier used to distinguish different lasers within the same node when they can transmit the same wavelength, the n parameter representing an offset with respect to an anchor frequency (193.1 THz), and the parameter m defined as a multiple of a given slot width granularity (i.e., 12.5 GHz). The super-channel FS information is used to configure the BV-WSS within all traversed BV-OXCs along the entire connection.

**Spatial Path:** (path) It specifies the route/path in terms of the sequence of nodes and links from the ingress to the egress of the connection. This determines the BV-OXCs that need to be configured to accommodate the super-channel with its specified FS.

The capabilities and features related to each subcarrier forming the super-channel (described in Fig. 2a) include the following.

**Subcarrier Frequency Slot:** The spectrum utilization of each subcarrier is identified by the label object (i.e., grid, CS, identifier, and the NCF defined by the integers n and m). To achieve ultra-high spectral efficiency, a debate is ongoing to relax the 6.25 GHz granularity to CS of 1 GHz or even to any frequency value of the spectrum. The subcarrier NCF (n) is used to configure the laser source at the transmitter and the local oscillator at the receiver. The subcarrier slot width (m) can be used to determine the proper filter shaping configuration within each subcarrier module.

**Baud Rate:** (baud_rate) The baud rate (in symbols per second) of each subcarrier is configured according to the available electronic processing capabilities of the transponder in such a way that, considering the applied modulation format and FEC/coding, the generated gross rate is able to guarantee the requested net bit rate. Typically, a restricted set of candidate values is considered. The baud rate configuration is applied to the subcarrier module.

**Modulation Format:** (mod_format) A pool of supported modulation formats. Examples include PM-OPSK and PM-16-QAM. To be applied within the subcarrier module.

**FEC/Coding:** (fec_coding) At the transmitter side, to guarantee adequate robustness, specifically designed redundancy (i.e., overhead) is applied to the net bit rate. FEC is typically applied within the OTN operations at the tributary interface. Examples of FEC are Reed-Solomon, Hamming code, BCH, and so on.

**Central Frequency Granularity:** (CFG) It indicates the multiple of CS between supported adjacent NCFs. Figure 2b depicts some examples about the feasible FS width depending on the subcarrier SWG.

**Slot Width Granularity:** (SWG) It determines the slot width granularity (in multiples of 2 × CS) between two consecutive FSs generated by the same subcarrier. Figure 2b depicts some examples about the feasible FS width depending on the subcarrier SWG.

**Minimum Slot Width:** (min_width) It indicates for a subcarrier the minimal slot width that can be generated.

**Maximum Slot Width:** (max_width) It indicates for each subcarrier the maximum slot width that can be generated. Observe that specifying both min_width and max_width strictly defines the range of feasible FSs to be generated by the subcarrier.

To successfully set up super-channel connections, the aforementioned parameters need to be adequately considered and incorporated in the adopted control plane approach as detailed in the next section.

**CONTROL PLANE SOLUTIONS FOR FLEXI-GRID OPTICAL NETWORKS**

A control plane aims to fulfill the requirements of fast and automatic end-to-end provisioning and re-routing of flexi-grid connections, covering common functions like addressing, automatic topology discovery, network abstraction, path computation, connection provisioning, and recovery.

The control plane functions can, on a first
approach, be distributed or centralized. Either way, they need to be extended to address the new requirements associated with the aforementioned optical technologies: adequate robustness and efficient flexible spectrum allocation of co-routed connections (i.e., super-channel establishment and configuration). The selection of a centralized or distributed control plane is conditioned by aspects such as flexibility and extensibility, availability, already installed deployments, actual network size, and scalability. Later, we qualitatively provide some discussions addressing those aspects with respect to the selected control plane approach.

**Gmpls/PCE Distributed Control Plane**

A set of cooperating entities (controllers) execute the control plane functions in a distributed manner. Each controller governing a network node (e.g., BV-OXC) disseminates the topological elements that are directly under its control via a routing protocol (e.g., Open Shortest Path First, OSPF). The routing dissemination enables each control plane entity to have a unified view of the network topology (i.e., node/link connectivity and network resource availability). Such information allows computing quality of service (QoS)-enabled services at either the connection ingress node or via a centralized dedicated element, the PCE. Afterward, the connection is set up by relying on a signaling protocol (Resource Reservation Protocol, RSVP) along the nodes involved in the computed path. The reference architecture is defined in the ITU-T automatic switched optical network (ASON) and relies on the Gmpls set of protocols defined by the IETF. A data communication network, based on IP control channels allows the exchange of control messages between Gmpls controllers. Each Gmpls controller manages the state of all the connections (i.e., label switched paths, LSPs) originating at, terminating at, or passing through a node, and maintains its own network state information (topology and resources), collected in a local traffic engineering database (TED). Figure 3a shows the distributed Gmpls-based control plane architecture along with the exchanged control protocol messages used to discover the network topology (OSPF-TE), computing end-to-end paths via PCE protocol (PCEP) and establishing the flexi-grid LSPs (RSVP with traffic engineering, RSVP-TE).

**SDN/OpenFlow Control Plane**

A single entity (controller) is responsible for the control plane functions, commonly using open and standard protocols, such as the OpenFlow protocol (OF/OFF). The SDN controller performs path computation and service provisioning, configuring the forwarding and switching behavior of the nodes. A centralized control plane provides a method for programmatic control of network resources and simplification of the control plane process. By deploying the control plane intelligence in the controller, resources allocated in hardware nodes for control plane functions can be reduced. Such solutions may involve deploying hardware (computational and storage) that is orders of magnitude more powerful than individual controllers. Figure 3b depicts the (logically) centralized SDN controller responsible for setting up the connections directly configuring (via OFF FLOW MOD messages) the involved BV-OXC nodes. That is, the switching (WSS) of BV-OXCs is modified according to the FS of the flexi-grid connection being established.

**CONTROL PLANE ENHANCEMENTS IN SUPPORT FOR SBVT**

Regardless of the adopted control plane solution, the automatic and dynamic configuration of flexi-grid networks must take into account the functionalities, capabilities, and restrictions imposed by devices such as (S)BVTs. This is done to attain the most efficient use of spectrum resources when serving end-to-end connections. In this section, we detail the required extensions in terms of protocols and processes considering the data modeling described above for the effective control of SBVTs in the context of both Gmpls and SDN/OFF protocol extensions.

**Gmpls Protocol Extensions**

The Gmpls OSPF-TE protocol provides to each controller the network topology (i.e., graph) and TE link and node attributes such as the link metric, link NCF availability, and switching node restrictions. This information is gathered into repositories (TEDs) at each Gmpls controller and at the centralized PCE. By doing so, the PCE is able to compute end-to-end paths taking into account the specifics of the optical layer. The characteristics and capabilities of SBVTs are thus required when computing end-to-end flexi-grid optical paths, especially to compose super-channel connections. With this aim, both Gmpls OSPF-TE and RSVP-TE protocols are extended [8]. In general, whenever a link or node attribute is modified (e.g., resources are dedicated for connection setup/tear down) an OSPF-TE Link State (LS) Update message is flooded within the network domain updating controllers’ TEDs. The link and node attributes carried in the LS Update message are formatted via the type length value (TLVs) structure. In particular, to flood SBVT TE attributes the existing Gmpls port label restriction (PLR) sub-TLV is extended as depicted in Fig. 4a. Such SBVT information along with the topology graph and NCF link availability constitute the necessary input to devise routing and spectrum assignment (RSA) algorithms at the PCE and enable online computation of feasible path connections, thus improving the overall spectrum efficiency [8].

The carried SBVT attributes in the Gmpls PLR sub-TLV are:

- \text{TxSubTrnsp} and \text{RxSubTrnsp}, specifying the number of total/equipped subcarriers for Tx and Rx directions on the interface, respectively
- \text{AvailTxSubTrnsp} and \text{AvailRxSubTrnsp}, specifying the number of available subcarriers (i.e., not used) for Tx and Rx directions, respectively
- Aggregated NCF status for both Tx and Rx directions on the interface between the SBVT device and the BV-OXC (Fig. 1)

To this end, a standard OSPF-TE label TLV is used where bitmap coding allows identifying
Each bit the status of a supported NCF: 1 means available and 0 occupied.

– **Num labels** specifies the supported number of NCFs (which determines the size of the bitmap coding).

– **Grid, CS, Identifier, n, and m** form the standard lambda label for EONs (IETF RFC 7699) as aforementioned. Using the bitmap coding, the parameter n specifies the lowest NCF (e.g., anchor frequency of 193.1 THz) which is used as the reference wavelength to determine the frequency of each supported NCF.

With the above information at the time of computing and selecting the resources (i.e., optical spectrum and SBVT subcarriers) for a new connection, the RSA algorithm is aware whether sufficient unused subcarriers are available at the endpoints’ SBVT, and whether the optical spectrum continuity and contiguity can be ensured [8]. For the sake of completeness, in EONs without wavelength conversion, the RSMAs need to deal with two constraints: the spectrum continuity constraint, that is, the same unused optical spectrum portion should be allocated on each link of the path; and the spectrum contiguity constraint, that is, the NCFs occupied by the pool of subcarriers forming the super-channel need to be spectrally contiguous. Finally, observe that other relevant SBVT information (described in the data model of Fig. 2a), such as the list of supported modulation formats and FEC coding, could easily be added into the extended PLR sub-TLV contents.

Once the route is computed by the RSA algorithm at the PCE, RSVP-TE conveys the necessary information to allow resource reservation and configuration of the optical sub-systems (i.e., SBVTs and BV-OXCs). To do that, the path is passed as an explicit route object (ERO) to the RSVP-TE protocol.

For the sake of completeness, in EONs without wavelength conversion, the RSMAs need to deal with the spatial_path parameter in the data model of Fig. 2a, followed by the computed FS (i.e., n and m parameters) carried within the ERO Label subobject. This label information corresponds to the freq_slot parameter of the super-channel shown in Fig. 2a.

To handle the SBVT configuration at the connection endpoints, RSVP-TE is extended, carrying the so-called explicit transponder control (ETC) subobject into the Path message [8]. The ETC subobject depicted in Fig. 4b is formed by a variable list of transponder (subcarrier) TLVs. Each transponder TLV contains four sub-TLVs enabling the configuration per subcarrier basis:

1. Sub-transponder (subcarrier) Id locally identifies a subcarrier within the SBVT.
2. Subcarrier-FS specifies the FS (n and m) to be allocated by the respective subcarrier.
Modulation format configures the selected subcarrier modulation format (e.g., QPSK, 8-QAM).

FEC allows its configuration.

Observe that the above subcarrier configuration attributes are directly bound to the subcarrier-specific parameters described in the data model depicted in Fig. 2.

**OpenFlow Protocol Extensions**

OFP version 1.3 supports optical flows provisioning and control. However, flexi-grid is not supported yet, and a number of significant parameters and attributes are missing to properly configure SBVTs. Initial proposals for extensions specifically suitable for flexi-grid optical networks and SBVT have been presented [9, 10].

The OFP FLOW MOD message sent by the SDN controller to the OF agent is responsible for the configuration of the flow entries in each node. The selection of the SBVTs and the BV-OXC to accommodate the connection is performed by the path computation process used by the SDN controller. Once the end-to-end path is computed, to support configuring SBVTs the FLOW MOD has to be extended.
• Single/multi-carrier optical channel
• Number of subcarriers forming the super-channel
• Specification of the subcarriers including baud rate (and thus supported bit rate)
• NCF (i.e., parameter $n$)
• Subcarrier width (parameter $m$)
• Modulation format
• FEC, code type, and rate

For the sake of completeness, it is worth mentioning that such attributes follow the data model presented in Fig. 2a at the time of setting up super-channel connections and configuring their respective set of subcarriers.

When a single co-routed media channel is computed, a single FLOW MOD per node is forwarded. When sliceability is considered, the configuration of different media channels originating from the same SBVT is provided with different FLOW MOD messages (possibly bundled, if originating from combined computation), each one with different configurations of the BV-OXC output ports and BV-WSS filters. Therefore, such extensions fully enable the sliceable functionality.

Specific extensions enable the implementations of sliceability comprising programmable and asymmetric multichannel (PAMW) signal generation [11]. In particular, the laser type, laser id, and tone id attributes identify the laser source type, the physical laser device, and the laser instance utilized for that flow. BV-OXC and SBVT configurations are acknowledged by means of asynchronous or synchronous mechanisms (e.g., using extended PORT STATUS messages or novel FLOW ACK messages).

The extensions imply that detailed SBVT information (e.g., node architecture, type and number of installed transponders, node/card capabilities) are available at the controller performing advanced path computation including spectrum and transmission parameters assignment. Node/SBVT information can be provided by means of either discovery procedures or enhanced controller-agent OpenFlow session handshake describing the OpenFlow switch architecture and available capabilities.

Table 1a summarizes the key features for each of the above control plane approaches to support automatic SBVT configuration in EONs. For the sake of clarification, these features include the key control plane functions such as the service provisioning, the required routing information, and the path computation.

**Conducted Experimental Validations of the SBVT-Controlled Approaches and Qualitative Comparison**

In [9], the authors conducted an experimental validation and performance evaluation of the GMPLS SBVT-oriented protocol extensions to dynamically serve flexi-grid LSPs. The tests were performed at the control plane level within the CTTC ADRENALINE testbed. A 14-node Spanish EON topology was used where each link supports 128 NCFs (with a channel spacing of 6.25 GHz). Every connection/LSP request to be served, besides specifying the source and the destination endpoints (equipped with SBVTs), demands a specific bit rate (ranging from 100 to 500 Gb/s). The collected TED information (i.e., topology, link spectrum resource, and SBVT attributes) flooded by the OSPF-TE protocol is used as input for an RSMA algorithm executed at the PCE. If a feasible connection is computed, the output is formed by the spatial path (ERO), that is, nodes and links to be traversed; the control...
connection FS (n and m); and the number and ids of the subcarriers (constituting the super-channel LSP) to be activated along with their modulation format and FEC configuration.

The attained results in [8] reflect the feasibility of the proposed GMPLS routing and signaling extensions in support of SBVT configuration on dynamic flexi-grid traffic along with supporting the automatic establishment of super-channel connections. The RSMA algorithm (on average) takes 10 ms for path computation, while the signaling process requires (on average) 40 ms to set up the end-to-end connection, not including hardware latencies.

The OpenFlow extensions reported above have been employed in a real SDN data/control plane experimental testbed at CNIT labs. The EON data plane includes SBVT prototypes with advanced functionalities generating configurable PM-QPSK-modulated super-channels at up to 1 Tb/s exploiting the time frequency packing (TFP) technique and coherent detection; commercially available BV-WSSs with configurable filter of 1 GHz granularity; and fixed-length optical links (optical amplifiers, fiber spans) and a configurable optical recirculating loop to set up different optical reach distances. Such devices are controlled and dynamically configured by OpenFlow-enabled agents called FlexSwitch, collocated at each BV-OXC. OpenFlow sessions are established among each FlexSwitch, and the southbound interface of the central OpenFlow controller called FlexController [9], capable of lightpath setup, tear down, and dynamic adaptation. In particular, adaptation options include path rerouting, elastic operations (e.g., additional subcarrier activation due to bandwidth increase request), hitless spectrum shifting, and code rate adaptations. The TED is enriched with information related to node architecture, functionalities, and available modules. The controller implements impairment-aware path computation, also accounting for the SBVT capabilities at source and destination (i.e., available/configurable physical parameters). Different path computation and adaptation algorithms are utilized, mainly based on least congested spectrum K-shortest path subject to spectrum continuity constraint and QoT validation, and may include multiple (either parallel or sequential) actions output [13].

Results on the OpenFlow-based control plane testbed show that path computation at the controller (including spectrum and TFP-based transmitter transmission parameter assignment) is performed within 5 ms for provisioning [12] and within 100 ms for multipath restoration [9]. Node configuration, exploiting OpenFlow messages,

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<td></td>
<td>- BV-OXC and SBVT sequentially configured</td>
<td>- SDN controller's topology manager keeps track of the (abstracted) network topology and resources (i.e., TED).</td>
<td>- Distributed path computation done at each ingress controller is viable.</td>
<td></td>
</tr>
<tr>
<td>Centralized</td>
<td>- Parallelized configuration of BV-OXCs and SBVTs</td>
<td>- SDN controller either has an internal PCE for path computation or relies on an external PCE service invoked by the controller.</td>
<td>- OFP FLOW MOD message between the SDN controller and OF agent at BV-OXC and SBVTs</td>
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**Table 1.** a) Main features of distributed and centralized control plane solutions to support automatic SBVT configuration in EONs; b) qualitative (pros and cons) comparison of both control plane approaches.
is performed in a parallel fashion, and therefore is practically independent of the number of involved nodes. The configuration requires a time contribution within 5 ms for the OpenFlow control plane (i.e., including message exchange, flow entries processing, and TED update) and a variable time ranging from 40 ms up to 1.6 s for hardware configuration (e.g., actual filter re-shaping enforcement of BV-OXC) [12].

Concerning computation related to advanced SBVT features, a significant example refers to PAMW sources generating multiple sliceable subcarriers. In this case, the OCH SPEC information carried by each FLOW MOD to source and destination SBVTs specifies the selected laser source identifier and multiple (incremental) laser tone identifiers. The extended FLOW MOD message (Fig. 5) was captured at the Flex-Controller southbound interface during PAMW experimental validation in which three sliceable lightpath setup requests and configurations have been performed. In this specific case, three lightpaths are activated, generated by the same physical source and routed along two different paths (port 1 for flows 1 and 2, and port 2 for flow 3) through proper BV-OXC configuration. The capture shows the complete flow configuration of three lightpaths, including request, configuration messages, acknowledgments, and final reply, enabling flow traffic utilization.

The above summarized experiments and experiences conducted by the authors allow validating the proposed protocol extensions in each control plane approach to support the objective of automatically configuring SBVTs when setting up flexi-grid connections. In addition, the experimental tests also provide detailed insights when comparing both control plane approaches. In Table 1b, both control plane approaches are qualitatively compared in terms of pros and cons using a number of performance parameters including technology maturity, scalability, extensibility, complexity, multi-vendor interoperability issues, multi-domain support, recovery, and so on. In general, each approach presents its own advantages and disadvantages, outlined here, aiming to support network operators at the time of selecting one option or the other.

**CONCLUSIONS**

This work focuses on the recent advances attained in the context of the EU IDEALIST project [14] in the effective control of EON infrastructures taking into account the configuration of SBVTs at the endpoints. In this regard, first a generic data model addressing most of the capabilities and features of SBVTs is provided (number of subcarriers, modulation formats, FEC, etc.). This is made with special interest in supporting the appealing flexi-grid SBVT advantage of creating super-channel connections (e.g., 1 Tb/s) as a result of merging a number of subcarriers. The adopted data model represents a baseline to define two control plane approaches to enable the automatic control and configuration of SBVTs in EON. The control plane architectures are traditional distributed GMPLS and centralized SDN/OpenFlow. In both cases, we have presented design considerations, and the proposed protocol extensions and procedures to illustrate and validate each of the control plane approaches, two scenarios have been detailed and executed in two experimental testbeds at CTTC and CNT labs. Furthermore, a qualitative comparison in terms of different technical aspects (e.g., scalability, maturity, multi-vendor interoperability support) is provided. In light of the obtained results in both experimental activities, we can confirm the feasibility of the adopted control plane approaches, leaving the network operator with the option to select any of them for controlling their EON network.

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**BIOGRAPHIES**

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A Control Plane Architecture for Multi-Domain Elastic Optical Networks: The View of the IDEALIST Project

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ABSTRACT

A key objective of the IDEALIST project included the design and implementation of a GMPLS and PCE-based control plane for multi-vendor and multi-domain flexi-grid EON, leveraging the project advances in the optical switching and transmission technology, an enabling interoperable deployment. A control plane, relying on a set of entities, interfaces, and protocols, provides the automation of the provisioning, recovery and monitoring of end-to-end optical connections. This article provides an overview of the implemented architecture. We present the macroscopic system along with the core functional blocks, control procedures, message flows, and protocol extensions. The implemented end-to-end architecture adopted active stateful hierarchical PCE, under the control and orchestration of an adaptive network manager, interacting with a parent PCE, which first coordinates the selection of domains and the end-to-end provisioning using an abstracted view of the topology, and second, delegates the actual computation and intra-domain provisioning to the corresponding children PCEs. End-to-end connectivity is obtained by either a single LSP, or by the concatenation of multiple LSP segments, which are set up independently by the underlying GMPLS control plane at each domain. The architecture and protocol extensions have been implemented by several partners, assessing interoperability in a multi-partner testbed and adoption by the relevant Internet SDO (standards development organization).

INTRODUCTION

FLEXI-GRID NETWORKS

Optical transport networks [1] (OTN) are composed of network elements connected by optical fibers allowing the transport, multiplexing, routing, management, supervision, and survivability of optical channels carrying client signals. Such channels were constrained by a DWDM fixed frequency grid, inefficient for low rate signals and not adequate for high rate signals. The term “flexible grid or flexi-grid” [2] relates to the updated set of nominal central frequencies (NCF), defined within an abstract grid anchored at 193.1 THz, a new channel spacing (6.25 GHz), and other optical spectrum management considerations covering the efficient and flexible allocation of optical spectral bandwidth. A frequency slot (i.e., a variable-sized optical frequency range) is thus characterized by its nominal central frequency and its width, expressed in multiples of a given width granularity (12.5 GHz), and can be allocated to a connection, based on the signal modulation format and data rate.

The functional architecture of an OTN is decomposed into independent layers [1] and, in our context, the media layer is the server layer of the optical signal layer, and the optical signal is guided to its destination by means of a network media channel where the switching is based on a frequency slot.

HARDWARE MODELS

An information model is an abstract description used to represent and manage objects (such as a network device) on a conceptual level, independent of any specific protocols used to transport data. A data model is protocol specific and includes many technology specific details. Using well-defined standards-based common information and data models, provides interoperable data exchange between different implementations.

Standardization, notably at the Internet Engineering Task Force (IETF), is often influenced by early implementations and cooperative development by vendors and open source projects. Particularly pertinent to this article is the fact that the data models that were used to represent and configure optical interfaces with flexi-grid capabilities, or to describe a network topology (nodes, links, and connectivity) enhanced with details of optical capabilities and available resources, enabling network optimization and dynamic and online path computation, were developed by the project members themselves and contributed to the IETF.
Drivers and Motivations for an “Idealist” Control Plane

Backbone networks are intended to transport the aggregated traffic from several metropolitan networks. However, existing transport networks are based on the assumption that the traffic demands are predictable, and are not adapted to varying traffic requirements. Therefore, current networks require multiple manual configurations in the metro and core network nodes.

Dynamic optical networks are possible thanks to a distributed generalized multi-protocol label switching (GMPLS) control plane. There is a need for an end-to-end architecture to reduce the provisioning process of legacy network management systems (NMS), using standard network configuration interfaces, which will trigger automated standard control plane for multi-domain/vendor/layer operation. The control plane allows the reconfiguration of the optical service, its protection and restoration capabilities, not only for a single domain, but also for multi-domain scenarios. The benefits of a standardized control plane extend beyond the absolute functions enabled by the control plane itself, because such a common approach also facilitates interoperability between equipment supplied by different vendors, and so enables a network operator to construct a heterogeneous network yet operate it in a homogeneous way.

The implemented control plane architecture covers the automated provisioning and recovery of network connectivity services in a multi-domain setting. Such developments are increasingly driven by use cases such as interconnecting distributed data-centers, associated traffic patterns, and dynamicity.

Existing Control Plane Framework

There is extensive experience in the use of a dynamic distributed control plane. Standardization of this work has been conducted principally within the IETF, with some architectural and use-case documents developed within the ITU-T. The GMPLS architecture [3] comprises the following elements:

- A link/neighbor discovery/verification protocol, such as the Link Management Protocol (LMP), that allows neighboring nodes part of the control plane adjacency to unambiguously associate data plane adjacencies (e.g., fiber links), correlate identifiers, and assure compatible capabilities.
- A routing protocol. The Open Shortest Path First (OSPF) protocol describes the characteristics of nodes and links, so the state and capabilities of the resources are distributed and updated to all of the nodes, knowing which resources are in use, faulted/out of service, or available.
- A signaling protocol. The ReSerVation Protocol with Traffic Engineering extensions (RSVP-TE) is used to set up label switched paths (LSPs). RSVP-TE messages specify the path of the LSP, request specific capacity on the path, and report back the exact allocated network resources to support the LSP.
- A path computation service. A key aspect is determining which path an LSP should follow. This function can be performed externally (the path is supplied to the control plane), or delegated to the control plane. In either case, the computation can be complex. The path computation element (PCE) is a functional component that can be queried using the Path Computation Element communication Protocol (PCEP), recently extended to allow the network to delegate control of an LSP to a PCE, and to allow a PCE to direct the establishment of new LSPs (becoming an active PCE) [4].
- A network state reporting mechanism. The Link State Border Gateway Protocol (BGP-LS) allows an entity to collect, synthesize, and report the full set of state and capability information from the network to an external consumer such as a management system [5]. A coherent view of these protocols in a managed or software defined networking (SDN) context is provided by the IETF through their application based network operation (ABNO) architecture [6].

Control Plane Architecture

Our GMPLS/PCE control plane for multi-domain, flexi-grid networks addresses the provisioning of either a network media channel or a constant bit rate service between optical transceivers, which can support multiple bit rates. A media channel is a media association representing the topology path and the allocated resource (i.e., the frequency slot). It is similar to the GMPLS concept of LSP where, from a data plane perspective, it is the path in the network resulting from reserving and configuring transmission and switching resources across TE links and nodes in a way that can transport client signals and data from its entry point or interface to the exit point or interface. It represents a (effective) frequency slot supported by a concatenation of media elements. GMPLS labels locally represent the media channel and its associated frequency slot, which is the switched resource. Network media channels are considered a particular case of media channels when the end points are transceivers, and transport a single optical tributary signal (OTS), as shown in Fig. 1. The control plane deals with the resource reservation and configuration of media layer matrices that switch frequency slots and the configuration of the transceivers at the endpoints, with an agreed hardware model that, as of today, is not standard. No signal layer (e.g., OTS) switching is considered. Switching at the media layer is configured by configuring optical filters and configuring cross-connections.

From a bottom-up approach, each domain deploys its own GMPLS control plane instance. On top of it, each domain deploys an active stateful PCE (AS-PCE) for the purposes of both optimal path computation and service provisioning within its domain. Multi-domain path computation and provisioning is carried out by means of a hierarchical path computation element (H-PCE) [7], with the parent PCE (pPCE) coordinating the procedures between children PCEs (cPCEs) and under the control and orchestration of an adaptive network manager (ANM). The macroscopic architecture is shown in Fig. 2.
Figure 1. Relationship between optical tributary signal, network media channel, and media layer elements, and its view as a GMPLS LSP construct.

ADAPTIVE NETWORK MANAGER AND IN-OPERATION NETWORK PLANNING

The control plane has relied only on distributed functionalities, but the advent of PCE demonstrates that having a central entity can provide multiple benefits. The ANM was conceived with the idea of orchestrating network processes beyond the PCE capabilities. Its functionalities are to monitor network resources, and to decide the optimal network configuration based on the status, bandwidth availability, and user service. It does not replace the control plane, but extends and complements it (e.g., interacting with the client layer) and delegating specific functions (e.g., path computation) to it.

The ANM was implemented, utilizing the ABNO architecture, and relies on standards-based and open interfaces, providing the capability for application interaction via a north bound interface (NBI) and south bound interface to the data plane, either directly to each network element or via the control plane. The link between the ANM and the control plane is the parent PCE, which receives queries to carry out path computation and provision end-to-end connections.

The ANM platform allows automatic IP link provisioning, multi-layer restoration, dynamic bandwidth allocation based on traffic changes, periodic defragmentation, and network reoptimization after network failure recovery [8], so an operator planning tool has updated network information and maintains a provisioning interface with the network. This architecture benefits from the GMPLS/PCE control plane, reducing network CAPEX by minimizing the over-provisioning required in today’s static environments.

HIERARCHICAL PATH COMPUTATION ELEMENT

A parent PCE (pPCE) is responsible for inter-domain path computation, while in each domain a local child PCE (cPCE) performs intra-domain computation. The pPCE resorts to the hierarchical traffic engineering database (H-TED) storing the list of the domains and inter-domain connectivity information, to determine the sequence of domains. Moreover, to perform effective inter-domain computation, the pPCE is allowed to ask cPCEs for the path computation of the several border-to-border LSP segments.

A number of innovative extensions have been implemented by IDEALIST. First, besides reachability information, abstract intra-domain TE information is announced to the pPCE (e.g., in the form of mesh of abstracted TE links between border nodes) with the aim of improving the effectiveness of the domain sequence computation. In particular, the north-bound distribution of link state and TE information using BGP-LS is utilized by domains’ BGP speakers to populate the H-TED. Second, in order to enable advanced TE functionalities, e.g., elastic operations and re-optimizations [9, 10], the H-PCE architecture has been extended to support the active stateful PCE with instantiation capabilities.

In summary, the H-PCE achieves end-to-end path computation by performing domain sequence selection and segment expansion, based on spectrum availability information provided by BGP-LS and PCEP requests submitted to cPCEs. The same H-PCE deployment is used in some use cases to perform the provisioning, where the end-to-end path is split in segments, sent to the cPCE by means of instantiation messages, and each cPCE performs segment instantiation. The end-to-end LSP is set up in the form of a “stitching on the wire” of several segments.

GMPLS DISTRIBUTED CONTROL PLANE

Within each domain, there is an instance of a GMPLS control plane. GMPLS controllers execute several collaborative processes, and a data communication network based on IP control channels allows the exchange of control messages between controllers. Noteworthy processes are the connection controller, the routing controller, or the link resource manager. We assume that a GMPLS controller is associated with a single flexi-grid optical node.

Under distributed control, each GMPLS controller manages the state of the connections (i.e., LSPs) originating, terminating, or passing-through a node and maintains its own network state information (topology and resources), collected in a local TED and synchronized thanks to the routing and topology dissemination protocol. Controllers then appropriately configure the underlying hardware (filter, transceiver, or switch configuration) during the establishment of an LSP, as per the basic operation of a GMPLS control plane [3]. In the next section, we overview the main involved procedures focusing on the specific aspects of the optical technology (see [11] for a detailed view).

CONTROL PLANE PROCEDURES

INTRA-DOMAIN AND INTER-DOMAIN TOPOLOGY DISSEMINATION

Within a domain, each node routing controller is responsible for disseminating changes in the network state regarding the resourc-
es under its control (e.g., originating links) through OSPF-TE link state advertisements (LSA). Each LSA is sent to the neighboring nodes, which update their TED repositories and forward the LSA in turn. This mechanism allows synchronizing all the nodes’ repositories within a given time, referred to as the routing convergence time. The basic procedures remain mostly unchanged, relying on extending the actual information objects within the LSAs.

OSPF-TE has been extended to support the dissemination of per-node and per-link TE attributes, reflecting device restrictions and overall optical spectrum availability. In particular, nodes may have asymmetric switching capabilities or different minimum slot size restrictions; optical transmitters/ receivers may have different tunability constraints. Other extensions have been implemented for disseminating the capabilities of sliceable bandwidth variable transceivers (S-BVTs), including, for example, the number of available sub-transponders and their parameters. Let us note that in this approach, OSPF-TE is one of the methods by which a cPCE obtains the TED to perform constrained routing and spectrum assignment (RSA) and is the source of the (abstracted) information conveyed toward the pPCE.

BGP-LS has also been suitably extended to support specific information exchange, such as spectrum availability, transponders’ physical parameters, and interoperability capabilities. BGP-LS is also used to report the relevant attributes of inter-domain links. Without disclosing the internal domain topology, this allows a pPCE to have, at least, a graph that represents inter-domain connectivity and to perform basic multi-domain path computation.

**MULTI-DOMAIN PATH COMPUTATION**

Following Fig. 3, when a service request, driven by an operator, is received by the ANM, the controller asks the pPCE for an inter-domain path (step 1). The pPCE, based on the (possibly abstracted and aggregated) information obtained from the cPCEs, computes the domain sequence (including each domain entry and exit nodes) and subsequently requests from the cPCEs the corresponding border-to-border expansion (also by means of PCEP PCReq messages, step 2). Once the pPCE receives the responses (PCRep, step 3), which include, among other objects, the segment spectrum availability, the pPCE performs a detailed end-to-end path computation including the routing, spectrum assignment, and transponder selection. Optical constraints are considered based, for instance, on node switching capabilities, optical reach, and transponder capabilities. For example, in the case of an end-to-end spectrum continuity constraint, the pPCE has to assign a frequency slot such that it is able to convey the requested bandwidth, it is available across all the end-to-end path, including inter-domain links, and it allows the selection of available end-point transponders. Then, the pPCE answers the ABNO controller via a PCEP Reply (PCRep) message.

**INTER-DOMAIN SERVICE PROVISIONING VIA ANM WITH ACTIVE STATEFUL CAPABILITIES**

Once the path is computed, the ABNO controller asks the pPCE to establish the path with a PCEP Initiate message. There are several provisioning models, with varying requirements of control plane interoperability. Here, we focus on the contiguous LSP with a single end-to-end RSVP-TE session, and the model relying on the stateful capabilities of the H-PCE structure with multiple (one per domain) RSVP-TE sessions.

In the single session case, the provisioning interface is a dedicated PCEP session with either the cPCE of the ingress domain or directly the ingress node, and there is a single RSVP-TE session from the source node within the source domain to the destination node. The multiple session case requires that all PCEs are stateful with instantiation capabilities. The connectivity at the data plane level is insured by concatenating compatible media channels at every domain, each set up by the local RSVP-TE session. Note that the first case implies interoperability at the control plane signaling level between different optical vendors’ respective RSVP-TE implementations at the inter-domain boundaries, since there is a single end-to-end session that crosses the external network-to-network interfaces. On the contrary, for the second case, interoperability requirements are limited to PCEP, vertically, from the cPCEs to the pPCE, between each vendor and the provider of the pPCE. Both approaches can be seen in Figs. 3 and 4.
Specific extensions were defined for the RSA procedures in a hierarchical framework. Upon request from the pPCE, all cPCEs compute the path segment (sequence of nodes and links) inside their respective domain and reply this information to the pPCE, along with spectrum availability.

In either case, once the end-to-end path or the specific segment is computed, the assigned slot is included in the explicit route objects (EROs) after each hop. In the first case (Fig. 3) the end-to-end ERO is sent to the ingress node in a PCInitiate message (step 5), triggering the signaling process (6, 7) and final report to the ANM (8). In the second case (Fig. 4), the obtained ERO per segment are enclosed in PCInitiate messages sent by pPCE to each involved cPCE (step 4). Once intra-domain provisioning is performed (step 5-8), PCE Report (PCRpt) messages are sent to pPCE to acknowledge the segments’ status (step 9). Finally, the multi-domain LSP is stored in the H-TED and provisioning response is provided to the ANM (step 10). Similar procedures for inter-domain LSP update and LSP deletion are envisioned.

**CONTROL PLANE PROTOCOL EXTENSIONS**

Control plane extensions affect all the protocols of the GMPLS suite together with the those adopted as northbound interfaces (i.e., PCEP and BGP-LS).

**PROVISIONING AND LSPDB SYNCHRONIZATION INTERFACE**

The provisioning of LSPs relies on the use of the PCEP protocol, enhanced with stateful and instantiation extensions. Specific extensions to PCEP to cope with flexi-grid involve the BANDWIDTH object to convey the traffic descriptor that specifies the requested or allocated frequency slot width, and the ERO object with the resources to use along the path, which has been extended to carry the information describing the configuration of the optical transponders, such as the selected modulation format, baud rate, FEC, and so on. To this end, a new sub-object, called explicit transponder control (ETC), has been defined. It is formed by a variable list of sub-transponder TLVs, each of them describing one of the specific sub-carriers forming the super-channel LSP. To overcome scalability limitations, we enable the summarization of a set of parameters in a single parameter, the transceiver class, which considers the main parameters such as trunk mode and type, framing, channel band and grid, minimum and maximum chromatic dispersion, maximum polarization mode dispersion, differential group delay, and so on. A transceiver vendor is thus responsible for specifying the class contents and values. The vendor can publish the parameters of its classes or declare them to be compatible with already published classes.

**INTRA-DOMAIN TOPOLOGY DISSEMINATION**

The OSPF-TE protocol has been extended to convey, on a per link basis, the status of each possible central frequency or NCF (referred to as NCF availability) and the presence and attributes of transceivers. The former is done by means of a new object within the switching capability-specific information (SCSI) field. NCF availability is advertised using a bitmap format with bit position zero representing the lowest central frequency, each succeeding bit position representing the next central frequency; a bit set to 1 means the NCF is not in use.

**MULTI-DOMAIN TOPOLOGY ABSTRACTION**

BGP-LS extensions addressed both the propagation of the NCFs’ availability and the announcement of an S-BVT transceiver’s capabilities to the pPCE, in order to perform routing and spectrum assignment (RSA) for the multi-domain path. The first extension involves adding a new LINK_STATE attributes object TLV into the BGP-LS Update message, further characterizing a given optical link. The latter extension involves announcing the capabilities of an S-BVT attached to a given link using two new BGP-LS TLVs called “MF-OTP encoding” (for multi-flow optical transponder) and “transceiver class and application code”, respectively. Both BGP-LS extensions reuse the same encoding as those proposed in OSPF-TE.

**PATH COMPUTATION**

Specific extensions were defined for the RSA procedures in a hierarchical framework. Upon request from the pPCE, all cPCEs compute the
path segment (sequence of nodes and links) inside their respective domain and reply this information to the pPCE, along with spectrum availability. This is accomplished by sending a PCEP Reply (PCRep) message containing the ERO object and two new objects: a LABEL_SET object that encodes the free NCFs along the computed path, and a SUGGESTED_LABEL object, suggesting (but not mandating) the label (i.e., the specific frequency slot) to be used in that domain. The pPCE performs an end-to-end allocation with this information.

**Signaling Protocol**

The extensions to the signaling protocol include:

- A new 64-bit label format, used in all the objects carrying a label (GENERALIZED_LABEL, SUGGESTED_LABEL, LABEL_SET, ERO, etc.) specifying frequency slot center and width in terms of two integer values, \( n \) and \( m \), according to the following formulas:
  
  \[ \text{Center Frequency (THz)} = 193.1 + n \cdot 0.00625, \text{slot width (GHz)} = 12.5 \cdot m. \]

- A new traffic descriptor type for the SENDERSPEC and FLOWSPEC objects to specify traffic parameters, carrying the slot width and label value.

Note that the label value, used in GMPLS to define what is switched, indicates, in this case, the slot features and, in particular, its width, therefore also affecting the LSP bandwidth. The same ERO extensions already described apply to the ERO object contained in the signaling messages.

**Experimental Validation**

The architecture and its integration with the underlying data plane has been demonstrated in several stages, starting from control plane testbeds and ultimately integrating both control and data planes. In [12] the optical channel provisioning was evaluated in a distributed multi-partner control plane testbed with locations in Madrid (Telefónica I+D), Barcelona (CTTC), Torino (Telecom Italia), and Pisa (CNIT). The testbed was connected at the control plane level by means of dedicated IPsec tunnel, emulating a multi-domain network (Fig. 5). On top of this connectivity, logical relationships between PCEs were established. We reported the details of the interoperability of routing (BGP-LS), path computation and instantiation (PCEP), and signaling (RSVP-TE) implementations [12]. In [13], a higher degree of interoperability was achieved, demonstrating the aforementioned different provisioning models. Experimental results showed all protocol interactions and LSP setup times. The adoption of BGP-LS extensions fully enabled multi-domain TE and was demonstrated in a limited number of domains. The system was integrated and demonstrated at both the control and data plane levels [14], where domains can have real hardware optical nodes that switch frequency slots, although by necessity inter-domain links between remote locations are emulated. The data plane included both real and emulated flexi-grid nodes and SBVTS. Two real S-BVT prototypes were provided by different IDEALIST vendors (e.g., CNIT/Ericsson and Coriant). These S-BVTs performed super-channel transmission with a configurable number of PM-16QAM Nyquist-shaped carriers, overall providing up to 1Tb/s. At the receiver, coherent strategy with off-line post-processing was adopted. The S-BVTs supported the configuration of the number of active carriers, their central frequencies, modulation format, symbol rate and FEC.

**Future Considerations**

The evolution of transmission and data plane technologies, supporting rates at 1Tb/s and beyond, will reach its maximum potential when supported by automatic configuration procedures enabling the deployment of spectrally-efficient plug-and-play transponders. Control plane solutions will have to be improved to provide procedures for the commissioning and self-tuning of the transmission parameters (e.g., upon failure recovery) while aiming to optimize the use of network resources. Plug-and-play 1Tb/s transponders will also have to operate in interoperable multi-vendor environments.

While the control plane supports the dynamic configuration of transceivers, the full automation and self-tuning of parameters will rely on the integration with functional components related to cognitive and self-adaptive networks. The solutions require, for example, the deployment of passive and active monitoring and measurement systems beyond what currently exists, along with the adoption of formal languages and frameworks for the specification of rules and policies typical of expert systems.

Multi-vendor interoperability still remains a major issue to solve. While there are incentives (e.g., from operators or service providers trying to drive down costs), there is huge pressure for vendors to increase margins and differentiate from competing offers.

The decoupling of the data plane and the control plane is expected to also be applied in the context of optical core networks through the concept of transport SDN. A unified control plane architecture is expected to successfully orchestrate the core with metro and data center premises, enabling the challenging support for future front/back-haul networks and 5G applications. Once flexible and open frameworks and interfaces have been adopted for the control and orchestration of network connectivity services across, for example, multiple heterogeneous domains, extending the know-how and conceiving new architectures for the joint allocation of heterogeneous resources is the next logical step, and addresses uses cases that require the allocation of both computing and storage resources.

To achieve the goal of effective interoperability, two factors are also expected to play key roles in addition to standardization, i.e., the definition of common, standard data models, and the use of open source software, offering common core components and allowing “plug-ins” for different applications and vendor devices. Although some vendors may still include proprietary optimizations, a common basis is expected to improve interoperability performance.

Ongoing efforts at the SDOs regarding the definition of common information models (e.g., related to network topologies) are a step in this
direction. Nevertheless, the goal of achieving total interoperability still remains a hard issue, even more difficult in the domain of optical transport networks.

**Conclusions**

A GMPLS/PCE control plane for flexi-grid networks orchestrated by the ANM requires an architecture and protocols fulfilling the initial requirements while ensuring robustness, security, and scalability. Although the framework is considered to be stable and quite mature, addressing the constraints associated with flexi-grid DWDM networks, variable bandwidth transceivers, and programmable devices is a complex problem. We have detailed the components of such a multi-domain control plane. The summarization of TE capabilities per domain, underlay network abstraction, and applicability of stateful PCE capabilities to end-to-end path computation across multi-domain networks, are part of the IDEALIST solutions based on a hierarchy of the PCEs, which have been implemented, demonstrated in a multi-vendor testbed, and reported for standardization.

While other standardization bodies are working on the specification of the architecture of an SDN-based solution for multi-domain transport networks, our original goal was to extend the GMPLS protocol suite. The final architecture shares several aspects with SDN, since each domain is scoped and encapsulated by an active stateful PCE, and architectural elements still apply even if the network is composed of heterogeneous control technologies, including, for example, SDN and Openflow [15].

The architecture is hybrid, combining distributed and centralized elements. An additional role of the ANM and PCE is to enable a progressive migration to a transport SDN, since the architecture fits in a wider SDN applicability context in which driving a GMPLS domain is one south-bound interface of an orchestrator.

**Acknowledgments**

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REFERENCES

BIOGRAPHIES
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ROBERTO MORENO received a Dr. Ing. degree in electronic engineering from the University of Genoa, Italy, in 1988. After six years with Marconi, he joined TIM (at that time CSELT) in 1995, where he is currently in the IP & transport innovation unit. He has been involved in European projects (LION, NOBEL, NOBEL-2, MURED, STRONGEST, IDEALIST) and technology transfer activities. His research interests include network control with focus on multi-layer, multi-domain and traffic engineering aspects.

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The authors analyze the main cost components of FTTdp networks with G.fast and the cost differences that can be achieved in comparison to FTTH networks. The two metrics employed in the article, the cost per home passed and the cost per home connected, were applied to six different geotypes that range from dense urban to rural scenarios. The results show that the cost per home passed of an FTTdp network with the distribution point located in the street is lower than the cost of FTTH. The cost per home connected for FTTdp is lower than that of FTTH for all of the scenarios studied. The cost reduction rates achieved by FTTdp networks are higher when employing a brownfield deployment scheme than when employing a greenfield deployment scheme.

**INTRODUCTION**

Different telecommunications operators are examining the features of various access networks to provide a high-speed fixed broadband service. In terms of transmission capacity over long distances, fiber to the home (FTTH) networks outperform cable- and copper-based networks. However, one of the drawbacks of FTTH networks is the cost of deploying fiber to each home, which can be quite high for some network scenarios. Moreover, from the viewpoint of implementation, in some cases it can be problematic to roll out fiber in the last segments of the network because of the necessary approvals that are required, especially inside buildings and in customers’ apartments. One possibility to overcome this limitation consists of reuse of the copper cable in the last segments of the end-to-end network. This can be achieved by employing a fiber to the distribution point (FTTdp) network, which is a hybrid fiber- and copper-based network that requires a distribution point located close to the end user’s premises. The distribution point is connected to the end user’s premises through the existing copper cable and to the central office by means of a fiber cable. The distribution point can be located in different places; for example, it can be placed by the door of the apartment of the end user, on the floor, in the cellar of the building, or on the street.

The following network technologies can be employed for transmission over unshielded twisted pair (UTP) cabling: extended bandwidth asymmetric digital subscriber line 2 (ADSL2+), very-high-speed digital subscriber line 2 (VDSL2), VDSL2-vectoring variants, or G.fast — fast stands for fast access to subscriber terminals. In this article we use the terms copper cable and copper line to refer to UTP cabling. In almost all cases, there is existing ADSL2+ or VDSL2 equipment already deployed to provide up to several dozen megabits per second. In theory, with G.fast it is possible to achieve a combined upstream and downstream bandwidth of up to 1 Gb/s over a category 3 (CAT3) cable shorter than 100 m [1]. Based on this high-speed transmission capacity, a few operators are pondering whether G.fast could be employed as an alternative to FTTH in regions where the rollout of the fiber infrastructure to the in-building segment is complicated. Fiber to the basement or building (FTTB) networks have a network architecture similar to an FTTdp network that has the distribution point in the cellar of the building.

Some studies have addressed technical and strategic aspects of FTTdp and G.fast. The concept of hybrid fiber-copper network architectures is described in [2]. The way in which hybrid FTTH-copper networks can be employed to provide a transmission capacity of several hundred megabits per second is explained in [3]. Technical aspects of G.fast, such as medium access, coding, channel characterization, reverse power, and crosstalk cancellation, are addressed in [4]. Different studies analyze several aspects of the cost of FTTH networks [5–8]. However, according to the authors’ knowledge, little research has been published so far on the cost of hybrid networks with G.fast. The penetration rate that can be achieved by hybrid fiber- and copper-based networks that work with G.fast for some network deployments is explained in [9]. The cost of deploying an FTTdp network with G.fast for a few geotypes is explained in [10], and it is shown that for certain network deployment scenarios FTTdp networks with G.fast can have lower costs than FTTH networks.

Telecommunications operators are keen to know the cost implications of FTTdp networks.
with G.fast and how they might fit into their broadband strategy. The goal of this article is to examine the cost implications of the rollout of FTTdp networks with G.fast. The research questions addressed are: What are the main cost drivers of rolling out an FTTdp network with G.fast? What are the cost differences in comparison with FTTH access networks?

These questions were addressed by means of a cost model that was employed to obtain the cost of FTTdp and FTTH networks. The cost model was based on the fact that a cost model is needed because it is expected that in several regions FTTdp and FTTH networks will have a broadband transmission capacity on the same order of magnitude. The following six geotypes, which are based on fixed access networks in various regions in Europe, were employed: dense urban, urban, dense suburban, suburban, dense rural, and rural. There are several differences between the research done for this article and the research presented in [10]. For instance, in this article we consider six different geotypes and different building types.

The rest of the article is organized as follows. In the next section, the concept of FTTdp networks and copper-based aspects of G.fast is described. We then explain the network architectures used for cost analysis and the features of the cost model. We present the results of the cost analysis by analyzing the following two types of results: the cost of a home passed and the cost of a home connected. Finally, some conclusions of the study are made.

### FTTdp Networks and G.fast

Various hybrid fiber-based and copper-based networks can be built up according to the location of the distribution point. For instance, the FTTdp networks that can be constructed include fiber to the street (FTTS), FTTB, fiber to the floor (FTTF), and fiber to the door (FTTD). Operators can face some difficulties when trying to deploy fiber inside buildings or apartments. For example, in some cases the process of obtaining permission to deploy fiber in the building can be long and burdensome. In some European countries, for instance, a decision about the possible deployment of fiber inside the building must be approved first by the community of neighbors, who in some cases hold a meeting only once a year. Furthermore, it is not always easy to obtain the approval of the owner or occupant of the apartment to roll out fiber in the household. These inconveniences can result in a long time to market (TTM) for the provisioning of a high-speed fixed broadband service.

VDSL2, VDSL2-vectoring, and G.fast are the possibilities being considered by operators for the copper network section of FTTdp networks. The International Telecommunication Union (ITU) Study Group 15 (SG15) has been working on the standardization of G.fast, which is a high-speed fixed broadband copper-based access technology. The specifications of the power spectral density appear in ITU Telecommunication Standardization Sector (ITU-T) Recommendation G.9700 [11], whereas the physical layer specifications of subscriber terminals are described in ITU-T Recommendation G.9701 [1]. It is possible that enhancements will be standardized in the future in amendments to Recommendation G.9701. The Broadband Forum (BBF) defined the specifications of the distribution point unit (DPU), which is an essential component of the FTTdp network architecture [12]. Theoretically, with G.fast it is feasible to achieve a transmission rate of up to 1 Gb/s for combined uplink and downlink transmission over a length shorter than 100 m [1]. The transmission capacity decreases for distances higher than 100 m.

### COST MODELING

#### NETWORK ARCHITECTURES

Tables 1a and 1b show a few features of the six geotypes employed. The geotypes differ in various aspects. Table 1a shows the values of the following parameters: the feeder, distribution, and drop segment lengths; the number of households per central office and per street cabinet; and the cost of trenching and duct deployment in the feeder and distribution segments. Table 1b shows the building features and the gigabit-capable passive optical network (GPON) splitting ratios. Several of the input parameters employed in the cost model were derived by contacting companies that deploy and maintain fixed access infrastructure in France, Germany, and the United Kingdom. Cost studies of fixed access networks in different regions of Europe were also employed as information sources [13, 14]. Based on the information collected, average values were derived.

We have assumed that for the dense urban, urban, and dense suburban geotypes multiple dwelling units (MDUs) with the number of households per building that appear in Table 1b are employed. Table 1b also shows that single dwelling units (SDUs) are employed for the suburban, dense rural, and rural geotypes. Moreover, we have considered one operator deploying G.fast in the building or house, and that there is no mix of technologies.

The network architectures of the following three types of networks employed for the six geotypes are shown in Fig. 1: FTTH, FTTdp with the distribution point located in the cellar of the building (FTTdp-Building), and FTTdp with the distribution point located in the street (FTTdp-Street). An FTTdp-Building network is in fact an FTTB network, but as this article focuses on FTTdp networks, we employ the term FTTdp-Building. Figures 1a, 1b, and 1c show the network architectures employed for the dense urban, urban, and dense suburban geotypes. These geotypes work with MDUs. Figures 1d, 1e, and 1f show the network architectures employed for the suburban, dense rural, and rural geotypes, which work with SDUs. The cost of the active and passive network components of the following sections of the network, which are depicted in Fig. 1, were considered in the analysis: the central office, the feeder segment, the street cabinet, the distribution segment, the drop segment, the DPU elements, the in-building segment, and the equipment in the customer’s premises.

The three networks studied in this article employ GPON. The downstream capacity provided by GPON is 2.5 Gb/s, and the splitting ratio...
employed is 1:32, which is typical for certain GPON deployments in Europe. This splitting ratio provides every household with an average transmission capacity of 78 Mb/s. Usually, operators assume that all users who share the same GPON port are not transmitting simultaneously, and, as a consequence, the broadband commercial offer of the operators, based in many cases on the peak bandwidth that can be provided, can be of several hundred megabits per second.

From the central office up to and including the distribution segment, the network elements shown in Fig. 1 are the same for the three network architectures. The central office contains the optical line terminal (OLT) and the optical distribution frame (ODF). The OLT contains upstream Ethernet and downstream GPON ports. The number of GPON ports depends on the geotype employed. The feeder and distribution segments contain ducts, fiber, and manholes. A small street cabinet was employed, because it will need to contain only the splitters. The distribution segment contains at least two fibers per building. The drop segment contains the ducts and fiber for the FTTH and FTThdp-Building network as well as the copper cable for the FTThdp-Street network.

As shown in Fig. 1b, for FTThdp-Building a DPU cabinet is located in the basement of the building. As shown in Fig. 1c, for FTThdp-Street the DPU cabinet is located on the street between the distribution segment and the drop segment. In our study, the energy of the DPU cabinet is provided from the end user’s premises by using reverse powering. The reverse power feeding (RPF) located in the end user’s premises sends energy to the DPU. The DPU cabinet includes the following network components: a DPU per FTThdp subscriber, an internal main distribution frame (MDF), 1:8 splitters, and the cabinet. The DPU cabinet considered in the analysis can contain up to eight DPUs, which enables the provisioning of a broadband service for up to eight FTThdp users.

The FTThdp networks work with an existing external MDF, which is placed in the basement of the building and appears in Figs. 1b and 1c. This MDF is not the same as the internal MDF of the DPU cabinet. Figure 1a shows the fiber cable located in the in-building segment of the FTThdp-Building and FTThdp-Street network architectures, as shown in Figs. 1b and 1c, respectively. A copper cable is also employed in the drop segment of the FTThdp-Street network architecture. It is assumed that the existing CAT3 cable, which has been used for telephony service for decades, is employed for the transmission with G.fast.

Two splitting levels are employed for the three network architectures shown in Figs. 1a, 1b, and 1c: a first splitting level of 1:4 in the street cabinet, and a second splitting level of 1:8, which is located in the basement of the building for FTTH and in the DPU cabinet for the case of FTThdp-Building and FTThdp-Street. In total,

<table>
<thead>
<tr>
<th>Geotype</th>
<th>Length of the feeder segment</th>
<th>Length of the distribution segment</th>
<th>Length of the drop segment</th>
<th>Number of households per central office</th>
<th>Number of households per street cabinet</th>
<th>Cost of trenching and duct deployment in the feeder and distribution segments</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dense urban</td>
<td>950 m</td>
<td>200 m</td>
<td>4 m</td>
<td>17,000</td>
<td>500</td>
<td>US$110/m</td>
</tr>
<tr>
<td>Urban</td>
<td>1100 m</td>
<td>230 m</td>
<td>8 m</td>
<td>15,000</td>
<td>480</td>
<td>US$110/m</td>
</tr>
<tr>
<td>Dense suburban</td>
<td>1040 m</td>
<td>300 m</td>
<td>10 m</td>
<td>10,000</td>
<td>420</td>
<td>US$992/m</td>
</tr>
<tr>
<td>Suburban</td>
<td>1800 m</td>
<td>980 m</td>
<td>30 m</td>
<td>4000</td>
<td>350</td>
<td>US$992/m</td>
</tr>
<tr>
<td>Dense rural</td>
<td>500 m</td>
<td>650 m</td>
<td>18 m</td>
<td>1000</td>
<td>180</td>
<td>US$880/m</td>
</tr>
<tr>
<td>Rural</td>
<td>800 m</td>
<td>1300 m</td>
<td>75 m</td>
<td>800</td>
<td>115</td>
<td>US$880/m</td>
</tr>
</tbody>
</table>

Table 1. Input parameters according to the different geotypes: a) access network features; b) building features and GPON splitting ratios.
Figure 1. Network architectures for the geotypes dense urban, urban, and dense suburban: a) FTTH; b) FTTdp-Building; c) FTTdp-Street. Network architectures for the geotypes suburban, dense rural, and rural: d) FTTH, e) FTTdp-Building, f) FTTdp-Street.

Usually operators assume that all the users that share the same GPON port are not transmitting simultaneously and, as a consequence, the broadband commercial offer of the operators, based in many cases on the peak bandwidth that can be provided, can be of several hundred Mb/s.
both splitting levels yield a total splitting factor of 1:32. The optical network terminal (ONT), the customer premises equipment (CPE), and RPF active equipment can be located in the subscriber’s premises.

For the network architectures that work with SDUs, which are shown in Figs. 1d, 1e, and 1f, there is only one splitting level in the street cabinet, which is 1:32. Furthermore, for the FTTdp network architectures, the DPU cabinet does not contain a splitter and supports a single-port DPU.

**COST METHODOLOGY**

For the cost analysis it was assumed that almost all network components that appear in Fig. 1 should be deployed. The external MDF and the copper cables employed in FTTdp architectures were not considered as capital expenditures (CAPEX) in the cost model because it was assumed that they exist before the FTTdp network deployment takes place.

Two metrics were employed for the total cost of ownership (TCO) analysis: the cost per home passed and the cost per home connected. The main difference between both values is that the cost per home passed does not include all the network components or the effect of the market share value. For example, frequently the cost of a home passed in a VDSL2 or G.fast deployment does not include the cost of the CPE, any active network equipment located in the subscriber’s premises or the cost of the cable in the in-building segment.

For FTTH, the cost of a home passed encompasses the network components located from the central office up to the drop segment. For FTTdp-Building and FTTdp-Street the cost per home passed includes all the network elements from the central office up to the drop segment and distribution segment, respectively. The cost of a home connected includes all the network elements considered in the cost per home passed, but also the additional elements that will permit an end-to-end connection. For FTTH, the cost of a home connected also includes the cost of the splitters in the basement of the building, the internal network components, and the cost of the RPF and CPE. It was assumed that every DPU unit would be installed in the DPU cabinet on an on-demand basis. The cost of the DPU and RPF of the FTTdp networks were obtained by taking into account the present cost of the internal network components and assuming that over the next years several network components will have a price reduction due to a rise in sales volume.

The cost per home passed includes the CAPEX values. It was assumed that it takes four years to pass all the households in the deployment area. The rate of households passed was 25, 50, 75, and 100 percent over the first four years. The cost per home connected includes CAPEX and operational expenditures (OPEX) over a timeframe of 15 years. The cumulative present value (CPV) formula with a discount rate of 10 percent was employed to calculate the present value of the CAPEX and OPEX.

The CAPEX involves all the investment needed to roll out the active and passive infrastructure. The cost of digging, the deployment of ducts, and the rollout of fiber and manholes belong to CAPEX for the feeder, distribution, and drop segments. All the fiber installation considered in the study is underground, and there are no aerial components. The OPEX includes the annual cost of maintaining the infrastructure, and the repair and replacement of damaged network components. It was derived by using mark-up values of 1 percent for the passive infrastructure and 7.5 percent for the active infrastructure applied to the CAPEX. In the central office, the OPEX also includes the cost of energy consumption and the rental of floor space. For the calculation of the OPEX associated with the maintenance of the copper cable in the in-building segment and in the drop segment, monthly values of US$1.1 and US$0.4 were employed, respectively.

The values of the asset lifetimes are six years for the active network equipment placed in the subscriber’s premises (ONT, RPF, and CPE); eight years for the rest of the active equipment (DPU and OLT); 25 years for the fiber cables; and 45 years for the active network components from the central office up to the drop segment. For FTTH, the cost of a home passed encompasses the network components located from the central office up to the drop segment and distribution segment, respectively. The cost of a home connected includes all the network elements considered in the cost per home passed, but also the additional elements that will permit an end-to-end connection. For FTTH, the cost of a home connected also includes the cost of the splitters in the basement of the building, the internal network components, and the cost of the RPF and CPE. It was assumed that every DPU unit would be installed in the DPU cabinet on an on-demand basis. The cost of the DPU and RPF of the FTTdp networks were obtained by taking into account the present cost of the internal network components and assuming that over the next years several network components will have a price reduction due to a rise in sales volume.

The results of the cost per home passed for all the geotypes are shown in Fig. 2a. The results were derived by calculating the CAPEX with a coverage of 100 percent. There are substantial differences between the costs for the six geotypes because of the different feeder and distribution segment lengths and the different household densities. The cost of a home passed for FTTH and FTTdp-Building is similar. This is because all the network components from the central office up to the drop segment are the same. Given that there is no initial investment in the drop segment for FTTdp-Street, the cost of FTTdp-Street is lower than the cost of FTTH and FTTdp-Building. The cost reduction achieved by FTTdp-Street in
comparison with FTTH and FTTdp-Building for dense urban, urban, dense suburban, suburban, dense rural, and rural areas is 2.4, 6.4, 6.8, 55.4, 41.2, and 55.5 percent, respectively. This gives an average total cost reduction of 28.0 percent.

The cost differences and similarities among the three network architectures are explained by Fig. 2b, which shows the cost composition of the cost per home passed for the dense suburban area. The cost of the central office, feeder segment, street cabinet, and distribution segment is the same for FTTH, FTTdp-Building, and FTTdp-Street, whereas the cost of deploying the fiber in the drop segment is the same for the FTTH and FTTdp-Building architectures. Due to the fact that the DPU cabinet is placed in the street between the distribution and drop segments, there is no CAPEX-related cost of the drop segment for FTTdp-Street. As depicted in Fig. 2b, the largest part of the cost for the three network architectures depends on the feeder and distribution segments. For FTTH and FTTdp-Building networks, the cost percentage of both segments is 89 percent of the cost per home passed, whereas it is 96 percent for the FTTdp-Street network.

**Cost per Home Connected**

For the analysis of the cost per home connected, we have used the following two approaches for the network deployment, which are explained below: greenfield and brownfield. For both cases, it is assumed that the copper cables and MDF already exist.

**Greenfield Deployment:** It is assumed that there is no fiber-based infrastructure previously deployed, and almost all network elements located in the access network will be deployed. As mentioned above, some sections of the existing copper-based infrastructure will be reused.

**Brownfield Deployment:** It is considered that the following components of the fiber-based infrastructure already exist: the network elements located between the central office and the distribution segment. The operator will only need to deploy the following network elements: drop segment, in-building segment, DPU elements for FTTdp networks, and the ONT/RPF and CPE equipment.

Tables 2a and 2b show the cost per home connected and the cost reductions achieved.
For the greenfield and brownfield deployment schemes, the cost delta or cost difference between the FTTH and FTTdp network architectures for each geotype is the same. However, the cost reduction rates that can be achieved by FTTdp in comparison to FTTH when employing the greenfield and brownfield rollout schemes change.

As shown in Table 2a, in all cases the cost of FTTH is higher than the cost of FTTdp networks. The average cost reduction rate achieved by FTTdp-Building in comparison to FTTH is 1.8 percent. The average cost reduction rate of FTTdp-Street in comparison to FTTH is 22.3 percent. For the dense urban geotype, the cost of FTTdp-Building is 2.3 percent lower than the cost of FTTdp-Street. This is because of the low cost of the drop segment for this geotype. Moreover, in all cases the total cost of the DPU elements for FTTdp-Building is lower than the total cost of the DPU elements for FTTdp-Street. The DPU equipment deployed outdoors for FTTdp-Street is more expensive than the DPU equipment deployed indoors for FTTdp-Building. For the rest of the geotypes, the cost of FTTdp-Street is lower than the cost of FTTdp-Building. The average cost reduction rate achieved by FTTdp-Street in comparison with FTTdp-Building for the urban, dense suburban, suburban, dense rural, and rural geotypes is 25.2 percent.

Table 2. Cost per home connected and cost reduction vs. FTTH, CAPEX, and OPEX, 50 percent market share: a) greenfield deployment; b) brownfield deployment.

<table>
<thead>
<tr>
<th>Geotype</th>
<th>Dense urban (US$)</th>
<th>Urban (US$)</th>
<th>Dense suburban (US$)</th>
<th>Suburban (US$)</th>
<th>Dense rural (US$)</th>
<th>Rural (US$)</th>
</tr>
</thead>
<tbody>
<tr>
<td>FTTH</td>
<td>1613</td>
<td>3027</td>
<td>3808</td>
<td>5317</td>
<td>4031</td>
<td>10480</td>
</tr>
<tr>
<td>FTTdp-Building</td>
<td>1570 (2.7%)</td>
<td>2923 (3.4%)</td>
<td>3707 (2.6%)</td>
<td>5275 (0.8%)</td>
<td>3990 (1.0%)</td>
<td>10439 (0.4%)</td>
</tr>
<tr>
<td>FTTdp-Street</td>
<td>1607 (0.4%)</td>
<td>2855 (5.7%)</td>
<td>3596 (5.6%)</td>
<td>2974 (44.1%)</td>
<td>2874 (28.7%)</td>
<td>5322 (49.2%)</td>
</tr>
</tbody>
</table>

For FTTdp-Building, the cost reductions achieved in the geotypes that work with MDUs are higher than the cost reduction achieved in the geotypes that work with SDUs. For FTTdp-Street, the cost reduction rates obtained in the geotypes that work with SDUs are higher than the cost reductions obtained in the geotypes that work with MDUs. This is because of the relatively high cost of the drop segment for FTTH networks in SDU-based geotypes. The average cost proportion of the drop segment in SDU-based geotypes for FTTH and FTTdp-Street networks is 42.5 and 0.7 percent, respectively.

The cost reduction rates achieved by FTTdp networks when using a brownfield deployment scheme, which are shown in Table 2b, are in all cases higher than the cost reduction rates achieved when using a greenfield deployment scheme, which are shown in Table 2a. Where- as the average cost reduction rate achieved by FTTdp-Building when employing the greenfield deployment approach was 1.8 percent, the average cost reduction rate achieved when employing the brownfield rollout scheme was 4.9 percent. Regarding FTTdp-Street, the cost reduction rates achieved when employing the greenfield and brownfield deployment schemes were 22.3 and 41.8 percent, respectively.

Figure 3a shows the cost structure of the cost per home connected in the dense suburban geotype. The values of CAPEX and OPEX for the three networks are the same for the central office, the feeder segment, the street cabinet, and the distribution segment. The cost percentage of the feeder and distribution segments for the FTTH, FTTdp-Building, and FTTdp-Street networks is 67, 69, and 71 percent, respectively. The FTTH and FTTdp-Building networks have the same cost of the drop segment: US$195. The US$25 of the drop segment in the FTTdp-Street network corresponds to the cost of the maintenance of the copper network. The cost percentage of the in-building segment for FTTH is 15 percent of the cost per home connected, whereas the cost percentage of the DPU elements for FTTdp-Building and FTTdp-Street is 13 and 15 percent, respectively. The cost of the active equipment in the end user’s premises is higher for FTTH than for the FTTdp network architectures. Even though the cost of the CPE is the same for the three networks, the difference lies in the fact that the cost of the ONT for the FTTH network is higher than the cost of the RPF for the FTTdp networks.

An essential cost difference between the
FTTH and FTTdp networks is motivated by the costs of the in-building segment and the FTTdp DPU elements. As shown in Fig. 3a, the cost of the in-building segment for the FTTH and FTTdp networks is US$576 and US$63, respectively. For FTTdp-Building and FTTdp-Street, the cost of the DPU elements is US$496 and US$555, respectively. Based on the values of Fig. 3a, Fig. 3b shows in detail the cost components of the CAPEX and OPEX for the in-building segment and the DPU elements.

As is depicted in Fig. 3b, the US$568 CAPEX for FTTH is higher than the US$407 CAPEX for FTTdp-Building and the US$463 CAPEX for FTTdp-Street. Nonetheless, the OPEX for FTTdp-Building, US$152, and the OPEX for FTTdp-Street, US$155, are higher than the US$8 OPEX for FTTH. The required CAPEX for connecting a new FTTH subscriber, which is US$354 and corresponds to the task of deploying the fiber cable in the building, is higher than the CAPEX of a new DPU for FTTdp networks, which is US$183. The OPEX allocated to the maintenance of the DPU elements in FTTdp networks is higher than the OPEX needed to maintain the FTTH network inside the building. Moreover, the cost of maintaining the copper line in the in-building segment for FTTdp networks, US$63, is higher than the cost of maintaining the fiber line for FTTH networks, which is US$3. The maintenance cost of the DPU, which is an active element, is higher than the maintenance cost of the DPU cabinet, which is a passive network element.

Figure 3. Cost per home connected, CAPEX and OPEX, 50 percent market share, greenfield deployment, dense suburban area: a) cost composition of all the network elements; b) cost composition of network elements in the in-building segment and in the DPU.
CONCLUSIONS

This article has shed light on the main cost components of FTTdp networks with G.fast. Moreover, the cost differences that are achieved in comparison with FTTH networks that provide similar broadband transmission rates have been analyzed. The cost per home passed of FTTH and FTTdp-Building networks is the same, but the cost of FTTdp-Street is lower than the cost of FTTH. The results show that for all the network scenarios studied when analyzing the cost per home connected, the usage of FTTdp-Building and FTTdp-Street leads to cost reductions in comparison to FTTH. With FTTdp-Street, it is possible to obtain for the majority of cases lower costs and therefore higher cost reductions than with FTTdp-Building.

The cost reduction rates achieved by FTTdp networks are higher when employing a brownfield deployment scheme than when employing a greenfield deployment scheme. The cost reduction achieved by FTTdp networks with G.fast in comparison with FTTH and the lower rollout time of FTTdp networks make FTTdp with G.fast a good candidate for the provision of high-speed fixed broadband services, especially in buildings where the rollout of fiber inside the building is challenging.

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BIOGRAPHIES

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Low-Cost Hybrid ROADM Architectures for Scalable C/DWDM Metro Networks

Md. Nooruzzaman and Elbiaze Halima

ABSTRACT

CWDM networks have proven to be a promising first-step metro and access network architecture, offering a significant cost advantage over DWDM due to the lower cost of lasers and the filters used in CWDM modules. If demand grows beyond the capacity covered by CWDM channels, DWDM network elements can be introduced to merge CWDM and DWDM traffic at the optical layer. This ensures two advantages: reduced initial investment and scalability for deploying DWDM channels in the future. This article presents various ROADM architectures, and explores the novel optical node architecture of hybrid C/DWDM networks, consisting of CWDM, hybrid C/DWDM, and junction nodes connecting two rings. Evaluation has shown that the hybrid ROADM architecture is superior to other conventional ROADM architectures in terms of scalability and the initial cost of optical nodes and networks.

INTRODUCTION

The emergence of various applications such as online video on PCs and smartphones, online gaming, and high-definition television over the Internet have engendered enormous pressure on network industries to develop optical network infrastructures for effective broadband services in the metro networks, large-scale LANs of commercial complexes, passive optical networks (PONs), and other access networks. Access data rates have increased from kilobits per second to megabits per second, and new emerging demands, such as ultra high definition and 3D videos, will require even higher data rates of more than 100 Mb/s per subscriber [1]. Driven by this unabated growth of traffic, the demand for effective broadband services based on wavelength-division multiplexing (WDM) technologies continues to grow. In metro networks, the traffic volume is relatively low but varies from network to network compared to that of core backbone networks. The flexibility to support different topologies, dynamic wavelength allocation, and automated network control and upgradability are key elements in the design of next-generation metro networks. Introducing these capabilities requires reconfigurable optical add/drop multiplexers (ROADMs) with dynamic add/drop structures, embedded control planes, and lightpath reconfigurations. Colorless, directionless, contentionless (CDC) ROADM has become one of the most attractive architectures, as it can be programmed to switch connections operating at any wavelength to any outgoing direction without contention [2].

However, traffic volume in access networks is low, and subscribers generally have much lower bandwidth requirements, concentrating more on service cost. This is where coarse WDM (CWDM) proves to be a promising candidate. Low-cost ROADMs for CWDM optical networks were proposed [3, 4], as they offer significant cost advantages over dense WDM (DWDM) systems. DWDM solutions are more expensive because they use cooled distributed feedback (DFB) lasers [5]. In contrast, systems based on CWDM technology deploy uncooled DFB lasers and wideband optical filters, which allow for more wavelength deviations. These present several advantages for CWDM systems, such as lower power dissipation, a smaller footprint, and lower cost [5]. Therefore, installing high-cost DWDM infrastructures for metro/access networks at the first step is undesirable, as doing so will not guarantee maximum return on investment (ROI). In this situation, implementation of CWDM networks is much more cost effective [5], and when and where necessary, an operator can grow the capacity by introducing DWDM wavelengths into CWDM systems. In this way, some of the nodes in a CWDM ring can be converted to hybrid nodes, which permits both CWDM and DWDM channels to be transmitted on the same fiber.

In this article, we present the novel structure of hybrid CWDM and DWDM (C/DWDM) ROADMs, which can be used for developing scalable metro/access networks that initially use CWDM wavelengths and allow carriers to add new DWDM channels as demand grows. This merges existing CWDM channels with new DWDM channels on existing fibers. A CWDM system is migrated to a hybrid C/DWDM system simply by connecting a DWDM multiplexer (Mux) to the given CWDM module as demand grows.

METRO NODE ARCHITECTURES WITH ROADMs

Metro networks require significant improvement in both capacity and functionality to cope with the foreseen bandwidth demand increase [6, 7].

The authors present various ROADM architectures and explore the novel optical node architecture of hybrid CWDM and DWDM C/DWDM networks, consisting of CWDM, hybrid C/DWDM, and junction nodes connecting two rings. Evaluation has shown that the hybrid ROADM architecture is superior to other conventional ROADM architectures in terms of scalability and the initial cost of optical nodes and networks.
The technological advancements in the realm of backbone and access portions of a network have so far not been matched with progress in the metro portion. The challenge for next generation metro networks is to flexibly aggregate, transmit, and switch high-volume traffic in a highly cost-efficient way to handle new dynamic services and applications.

A metro network structure comprises synchronous digital hierarchy (SDH)/synchronous optical network (SONET) rings that can be subdivided into metro/access rings and metro/core rings [6, 7], as illustrated in Fig. 1a. Metro/access rings (also referred to as collector, distribution, aggregation, or edge rings) collect and aggregate the data from customer sites. Metro/core rings (also referred to as feeder rings) do further data aggregation and feed data to long-haul/backbone networks. However, current metro/core rings do not feed all data that end users request. As demand for video content increases, video storage units are distributed in metro networks instead of backbone networks [8]. In this scenario, end users’ video on demand services are streamed from their respective metro networks instead of backbone networks. This approach ensures a superior quality of experience (QoE), as most popular content is now sourced from within the metro/core network and not from a single video cache located in the backbone network. Consequently, the traffic within metro networks is growing significantly, driven largely by IP video and growth in data center or cloud traffic. Distributing content sources within the metro will result in a change in traffic flows in the network. Service providers need to increase flexible metro/access solutions, which deliver agility, scalability, and efficiency.

Since CWDM nodes are cost effective, most access nodes are generally preferred to be CWDM components. However, some of the nodes and WDM PONs may require higher data rates, requiring DWDM channels to be merged with current CWDM channels [9]. While current optical nodes enable switching, they cannot merge CWDM and DWDM channels in a node. Therefore, there has been an increasing need for the development of nodes with functionalities providing a transparent mechanism for both CWDM and DWDM channels to establish an all-optical link between access and core networks. This actually means that there is a need for hybrid wavelength (circuit) routers that are able to aggregate and merge traffic from one network with lower-speed traffic to another network with higher-speed traffic. For example, the metro/access ring in Fig. 1b has three CWDM nodes. The four hybrid C/DWDM nodes were originally installed with CWDM filters as a first-step solution and later on migrated to hybrid nodes. Now all the nodes can add/drop or pass through CWDM channels. However, only hybrid nodes handle DWDM channels. Since video on demand services are sourced from caches locat-
ed in metro/core networks, some of the heavy access nodes are expected to be served over CWDM networks without any optical-electrical-optical (OEO) conversion using the premises to core route. For example, a DWDM channel can establish a heavy transparent link between nodes 19 and 8 over the CWDM access network without requiring OEO conversion at node 15. Transmitting DWDM channels over CWDM networks without interrupting CWDM channels is thus possible.

Figure 1c shows the required component list at each node. This was assumed for the purpose of evaluation, although the number of components depends greatly on the scalability and flexibility levels operators expect from the initial deployment. Figure 1c also includes unit cost values (in arbitrary units) of the components, which are normalized to the cost of a 120 arrayed waveguide grating (AWG) [10]. Figures 1d and 1e show conventional and three-port OADM-based access node architectures, respectively. The schematic diagram of a hybrid ROADM appears in Fig. 1f. The detailed architectures of various generic and hybrid ROADMs and functions are presented in the following sections.

**Generic ROADM Architectures**

A ROADM enables network operators to quickly and flexibly respond to traffic changes, such as establishing new or releasing existing lightpaths. Each node in the network behaves like a junction point for optical signals, where a particular optical signal may be dropped and the rest routed in different directions as defined by the software control system. A ROADM with maximum flexibility has three features: it is CDC. Notably, a ROADM with maximum flexibility (CDC features) requires a large number of high-cost wavelength selective switches (WSSs), multicast switches, and other components. Therefore, this type of ROADM is appropriate for core/backbone networks rather than for metro/access networks to achieve the highest wavelength agility.

ROADMs are able to transparently switch traffic within the same network. Different types of ROADM architectures have been implemented [2, 11] for switching DWDM streams in core or backbone networks. However, the switching of data between networks (e.g., core and access/premises rings) is performed using OEO conversion. This OEO conversion makes the nodes expensive and complex, with large footprints and power consumption [12].

Figure 2a shows the internal architectures of the ROADMs used in WDM metro networks. In most cases, current core infrastructures are based on DWDM rings, and the nodes are based on ROADMs. Typically, metro networks connect core networks via points of presence (PoPs), where there are also core routers. The
metro rings connect multiple PoPs, containing an aggregation layer that may consist of layer 2 switches, which terminate CWDM access networks. These PoPs can also be equipped with PONs and optical line terminals (OLTs). The access nodes may comprise OADMs, and toward the core, nodes are equipped with ROADMs for achieving higher flexibility. The CWDM access rings connect several central offices (COs) or OLTs where digital subscriber line access multiplexers (DSLAMs) aggregate access traffic at the first stage. Figure 2b shows the structure of hybrid ROADM structures comprising three-port OADMs based on different applications. The following sections present detailed implementation examples of various hybrid ROADM architectures, functions, and costs.

ADVANTAGES OF HYBRID C/DWDM NODES AND NETWORKS

DWDM offers greater channel capacity; however, it typically incurs more capital expenditures (CAPEX) and operational expenditures (OPEX). For applications beyond the metro/core, the need for such DWDM bandwidth expansion tends to be quite rare. Therefore, proven hybrid C/DWDM infrastructures are considered promising for core and access networks.

Reduction of OEO Conversions: The major advantages of hybrid C/DWDM systems that the carriers can enjoy are as below.

- Reduced Cost: CWDM offers a significant cost advantage over DWDM because the narrower channel spacing of DWDM systems (100 or 50 GHz) is not tolerant of channel deviations, requiring the thermally compensated transceiver lasers to confine the DWDM channels within the narrower optical pass bands. To maintain these narrower wavelength tolerances, DWDM transceiver lasers need proper control circuitry to stabilize the transmission wavelength, which is costly. This cost savings becomes especially significant for large-scale deployments of access networks.

- Reduced Power Consumption: DWDM does not need complex circuitry for channel stabilization, and hybrid ROADM reduces the instances of OEO conversions by setting all-optical links between core and access nodes, resulting in reduced power consumption.

- Reduced Express (Pass-Through) Losses: The express sections of DWDM ROADMs consist of splitters/couplers and Mux/DeMux or WSSs. In hybrid ROADMs, some of the express channels do not enter the DWDM section; they pass through the CWDM filters, which causes less insertion loss compared to that of conventional nodes.

- Pay-as-You-Grow: CWDM is an excellent cost-effective first-step solution for metro and access networks. If the demand increases, the number of channels can be increased, as one module handles one channel. Adding one new channel at a time allows on-demand service introduction with minimal initial investment.

- Investment Protection: If the required capacity grows beyond the capacity supported by CWDM channels, DWDM can be introduced in the node without replacing the existing CWDM filters; this plug-and-play feature ensures upgradability and protects the initial investment.

HYBRID C/DWDM ROADM STRUCTURES

Figure 3 shows the internal architectures of hybrid C/DWDM ROADMs. The ROADM schematic is divided into two major sections: the DWDM section for add/drop and pass-through of CWDM channels, and the DWDM section to support DWDM channels. Figure 3a shows the structure of hybrid ROADMs, which are used for interconnecting two hybrid C/DWDM rings (e.g., node 15 in Fig. 1b). Such a situation is expected to become important in the event of network alliances, acquisitions, or mergers between two networks or network operators. If the east and north directions, as labeled in Fig. 3a, are connected with the metro/access ring in Fig. 1b, the west and south ports will be connected to the metro/access ring. This node serves as a junction between the channels from two rings. This ROADM can drop or pass through a CWDM or DWDM channel in any direction.

As Fig. 3a shows, eight three-port CWDM OADMs with wavelength $\lambda_3$ are used for reconfiguring (add/drop or pass-through) one wavelength, hence $\lambda_3$. Each pair of OADMs handles the add/drop of one wavelength to/from one direction. When an optical signal comprising multiple channels enters the hybrid ROADM from the east direction, the OADM filter allows it to pass the $\lambda_3$ channel to the E2 port. The E2 port is again shown in Figs. 3b and 3c. Figures 3b and 3e show two types of connections of the switches (SWs) for switching the CWDM channels. In Fig. 3b, the 3 channel arriving at the N1 port using the switching function of the 2 × 2 add/drop SW 2. In this way, the directions of each wavelength are controlled by the optical SWs [3]. Figure 3c shows the structure of the CWDM switch connection for the directionless pass-through feature. In this scenario, a particular wavelength from one nodal degree can be passed through to any of the other three directions through a 1 × 4 optical SW. The $x$, $y$, and $x'$, $y'$ fiber joints indicated in Fig. 3a are used for inserting new CWDM OADMs when necessary.

The ROADM architecture presented in Fig. 3d is suitable for interconnecting DWDM networks with a hybrid C/DWDM ring. For example, in Fig. 1b, this type of ROADM is used at node 8. Figures 3e and 3f show the internal architectures of the DWDM switching sections comprising AWGs and WSSs, respectively. The optical signal allowed by the 20 nm pass band of the CWDM filters ($\lambda_4$) enters this section. In this section, DWDM channels from different nodal degrees are dropped or passed through, and this is done by the Mux (e.g., AWG or WSS). Some ports of the DWDM Mux and DeMux are connected to the local transceivers, and the other ports are connected to the other DeMux and Mux, respectively, for routing the particular express channels. Generally, AWGs or WSSs are used as Mux/DeMux devices.

In Figs. 3a and 3d, a single DeMux structure (e.g., AWG) per degree, able to demultiplex all possible channels, divides all dropped channels. Multiplexing of the pass-through and locally added channels is accomplished using generally the same elements, only in the reverse direction.
The flexibility as the WSS is capable of directing any wavelength to a particular port. Figure 3f shows the structure of a colorless DWDM section, where any wavelength can be switched to any transceiver using the switching function of a WSS, and the tunable transceivers can receive any wavelength. This DWDM switching section has higher flexibility compared to the one in Fig. 3c.

**COLORLESS, DIRECTIONLESS, CONTENTIONLESS ROADM IMPLEMENTATION**

In a colorless ROADM, any wavelength can be assigned to any add/drop port. In the CWDM part, the colorless feature is not available, as the OADM filters are fixed, and the filters allow a particular wavelength to pass through it. For example, in Fig. 3c, the OADM filters support only the \( \lambda_3 \) wavelength. This module is neither
directionless nor contentionless, as a specific transceiver is used to add/drop the $\lambda_3$ wavelength to/from a specific degree.

The DWDM section shown in Fig. 3f is colorless; however, a particular transceiver is connected to a given nodal degree. Thus, adding signals to any desired direction from a particular transmitter is not possible. This is a significant constraint for wavelength planning at the optical layer.

A directionless add/drop structure provides the freedom to direct a channel to any degree of the ROADM and is implemented by connecting an add/drop port to every degree. This can be realized by adding an array of $1 \times \delta$ optical switches with drop ports as shown in Fig. 4a [11], where $\delta$ is the nodal degree. In this scenario, the colorless channels to the tunable transceivers are switched by these switches and received from any nodal direction. This structure is also contentionless, as the same wavelength can be added/dropped simultaneously from all directions through different transceivers.

**Hybrid ROADM Architectures for Coherent Transmissions**

With efficient impairment compensation capability using digital signal processing (DSP) and the freedom of tuning the receiving wavelengths, coherent detection will be the main transmission technology choice for high-performance 100 and 400 Gb/s applications [13]. In coherent transmission, the local oscillator in the receiver can be tuned to select the desired channel from a group of DWDM channels, allowing simplification of the ROADM drop structure [2, 11]. In the drop section of Fig. 4b, splitters (Sp) replace the WSSs, and tunable filters are used in receiving ports 1, 2, and 3 immediately before the receivers. Splitters split the optical signal, and the tunable filter allows a particular wavelength to pass through from a plurality of wavelengths before the receiver. The tunable filter (TF) is not required after SW A, as a coherent receiver is used. In coherent detection, the splitters provide multiple wavelengths to the receivers, and the receivers select the desired wavelengths from the plurality of received wavelengths by tuning the local oscillators.

In Fig. 4c, the drop section consists of WSSs, whereas the WSSs in the add section are replaced by combiners. When combining channels with a combiner, crosstalk between overlapping channels can be an issue. Designs may need to specify the laser side mode suppression ratio or filter the signal to reduce its bandwidth (and filter out noise) before the colorless combining is performed. To eliminate this problem, tunable transceivers are equipped with DSP [14], which is able to compensate for more than 50,000 ps/nm of chromatic dispersion (CD) and more than 100 ps of peak differential group delay led by polarization mode dispersion (PMD) [13].

In Fig. 4d, the drop and add sections consist of splitters and combiners, respectively. This is the most cost-effective implementation of colorless and directionless hybrid ROADM based on filterless power splitters and coherent receiver technology [14]. By using this type of DWDM section in hybrid ROADMs, gridless coherent detection can be realized without requiring bandwidth variable filtering components, such as liquid crystal on silicon (LCOS)-based WSS.

**Impact of Hybrid ROADMs on Available DWDM ITU Grid**

In 2002, International Telecommunication Union Telecommunication Standardization Sector (ITU-T) G.694.2 standardized 18 CWDM
channels from 1270 nm through 1610 nm with a channel spacing of 20 nm. Out of the 18 channels, the eight most commonly used channels are from 1470 nm through 1610 nm. The DWDM wavelengths are positioned in a grid having exactly 100 GHz (about 0.8 nm), 50 GHz, or even 25 GHz spacing in optical frequency, with a reference frequency fixed at 193.10 THz (1552.52 nm) [15]. The standard ITU-T grid specification has three bands: the S-band (1492.25–1529.55 nm), C-band (1530.33–1569.59 nm), and L-band (1570.42–1611.79 nm). However, the wavelengths within C-band are the most commonly used for DWDM communication, as this region can be amplified using an erbium-doped fiber amplifier (EDFA). Therefore, in hybrid ROADMs, the wavelengths within 1530 nm, 1550 nm, and 1570 nm envelopes are used for DWDM transmissions and other windows for CWDM systems. When diameters of the metro rings exceed 50–60 km, longest channels may require amplification; therefore, wavelengths within C-band are generally assigned for the longest lightpaths [4].

Within the 20 nm pass band in each of these channels, 25 DWDM channels with 100 GHz space can be accommodated if the filters ideally produce perfect square add/drop or pass-through windows. Figure 5a shows the add, drop, and pass-through windows for the 1550 nm CWDM filter. Although the width of the CWDM window is 20 nm, the maximum available space for DWDM channels is 15.23 nm, as depicted in Fig. 5a. Therefore, in practice, adding 25 100 GHz channels within the pass-band of a 1550 nm CWDM channel is not possible. In 14.42 nm (the distance between nominal central wavelengths 1542.94 and 1557.36 nm) or (194.3–192.5) = 1.8 THz of space, 19 channels with 100 GHz spacing can be accommodated, as Fig. 5b shows. Therefore, 6 100 GHz channels adjacent to CWDM guard-bands cannot be used in hybrid ROADM-based networks.

However, if the ring transmits only DWDM channels (i.e., the west-south degrees of the hybrid ROADM shown in Fig. 3d), those six channels on the DWDM ring are not affected, as they do not pass through any CWDM filters inside the ROADM. Accordingly, the hybrid ROADM in Fig. 3d can be used to deploy DWDM core or metro rings.

**APPLICATION OF HYBRID ROADMS AND SCALABILITY FEATURES**

CWDM is a cost-effective first-step infrastructure solution for access networks in terms of scalability and ROI. When additional channels are essential, a DWDM section can be inserted. Thus, the ROADM infrastructure can be upgraded. Several types of networks can take advantage of these ROADM architectures. For example, DWDM traffic can be allowed to overlay an existing CWDM network at a predetermined crossover point. In this situation, the aggregated traffic from CWDM access networks is transparently switched to the core networks through the DWDM channels between premise/access and core networks.

Two networks are configured in such a way to allow DWDM traffic to travel across a CWDM ring. None of the CWDM nodes through which the DWDM channels pass require DWDM filters, as the CWDM filter itself passes through those DWDM signals, and the insertion loss of a CWDM OADM is approximately 0.4 dB [3]. Therefore, the pass-through loss of the DWDM channels incurred by the intermediate CWDM nodes will be 0.4 times the number of OADM filters cascaded along the route at the intermediate nodes.

The structure shown in Fig. 3 has four nodal degrees; however, the setup can be implemented for other nodal degrees. For example, nodes of three nodal degrees (e.g., node 13 in Fig. 1) can be implemented by simply eliminating one of the south or west degrees. Core nodes can directly establish a DWDM lightpath over a CWDM ring for transmitting the aggregated traffic to/from the PON through this type of node. While most of the backbone networks have mesh topology, many metro/core/regional networks may also require mesh topology for providing more
connectivity as time evolves. In this case, C/DWDM or DWDM ROADMs can be scaled to any nodal degree required.

**Comparative Cost Analysis of Conventional and Hybrid Metro Networks**

This section presents a comparative cost analysis between conventional and hybrid networks. The results offer insight into the cost benefits that can be obtained with the scalability feature of the hybrid network. The cost evaluation used the network shown in Fig. 1, where the metro/core network connects seven metro/access networks. In this analysis, all metro/access networks were CWDM initially. Then, with the traffic increase, some of the metro/access nodes were upgraded to hybrid nodes, as Fig. 1b shows. As the traffic continues to grow, all metro/access nodes will be converted to hybrid C/DWDM nodes. Figure 1 illustrates the total available wavelengths and capacity of each wavelength.

In this evaluation, the access data are aggregated at the metro nodes; the metro data are further aggregated at the core nodes. An increasing percentage of traffic does not leave the core network as end users sourcing their video on demand from local distributed data centers and storage attach directly to their respective core networks. The distribution of the video caching within the metro/core network results in bandwidth savings in the backbone network, since multiple subscribers requesting the same content are served from this cache instead of being served from central storage accessible via the backbone network [8]. Thus, not all of the traffic between the access and core nodes enters the backbone network. However, the traffic between core nodes highly depends on the number of content storage (CS) components located in the core.

As Fig. 6a shows, if the core network contains a single CS unit, traffic between that and other core nodes is the same as the aggregated traffic requested by the subscribers connected to all metro/access networks served from that CS. The traffic between core nodes and the LH network is less than the average traffic between core nodes because a high volume of demand for video contents is served from the core network by a local CS. While access-metro-core network topologies and traffic patterns differ widely from one operator to another and from one city to another, Fig. 6a plots the average inter-node traffic demands as a function of total traffic at the core node. The traffic growth rate ranged between 17 and 22 percent in different nodes,
but the rate of each node was assumed to be constant over all periods.

Figure 6b illustrates two patterns of costs for metro networks differentiated by HF and LF. The normalized costs of the components shown in Fig. 1c were used. In Fig. 6b, the costs of the metro networks were taken for LF nodes in both conventional and hybrid systems. The cost of the hybrid metro network increases as traffic grows, indicating that the network was initially installed with CWDM nodes. Later, nodes were migrated to hybrid nodes. When the total traffic in the core network reached 9.0 Tb/s, all the metro nodes were converted to hybrid nodes, and they support both CWDM and DWDM channels simultaneously. A cost difference between conventional and hybrid metro is apparent, as conventional metro nodes require WSSs for pass-through while hybrid metro requires three-port ROADMs.

Figure 6c shows the total cost of the metro networks in Figs. 1a and 1b, which comprises the cost of 7 core nodes, 7 metro/access ring networks, and 7 × 4 = 28 access rings. This result also includes the cost of 10G CWDM and DWDM transceivers used in the metro and access rings. The 10G CWDM transceivers cost was assumed to be one-third that of 10G DWDM transceivers. The normalized cost of 10G DWDM transceivers assumed to be 0.5 a.u. Note that this chart contains the cost of only the different components used in conventional and hybrid networks. Clearly, the cost of the hybrid network is significantly lower than that of its conventional counterparts. The relative cost savings in graphs Fig. 6d show a significant initial investment savings compared to that for conventional networks.

**SUMMARY**

Hybrid ROADMs, along with an advanced portfolio of optical components and devices, offer flexibility, cost efficiency, and efficient scalability. A tutorial background of different types of ROADMs used in metro networks is presented. The above presents the key structures of hybrid ROADMs, which can be used in large-scale LAN, metro, and access networks. Cost evaluations show that hybrid ROADM usage in metro and access networks reduces the initial installation investment required. As capacity demands continue to grow, CWDM networks can be migrated to C/DWDM hybrid networks thanks to the scalability of hybrid ROADMs.

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**BIographies**

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Physical Layer Security Issues in Interference-Alignment-Based Wireless Networks

Nan Zhao, F. Richard Yu, Ming Li, Qiao Yan, and Victor C. M. Leung

Abstract

IA is a promising solution for the interference management of future wireless networks. On the other hand, physical layer security is a critical issue of wireless communications in the presence of adversaries. Recently, these two important fields tend to be researched closely together. In this article, some of the key results are summarized, and two primary attacks at the physical layer of IA-based networks, adversarial jamming and eavesdropping, are further studied. We first propose an anti-jamming scheme by aligning the jamming signal together with interference among users cooperatively when an adversarial jammer exists. Then an AN scheme is proposed, in which the external eavesdropping is disrupted by AN without introducing any additional interference to the legitimate network. To further analyze the potential threat, a collusive eavesdropping scheme by some hostile IA users in the network is also proposed. Simulation results are presented to show the effectiveness of these schemes. Finally, some future challenges are also summarized.

Introduction

Interference is fundamental in wireless communications, and will hinder the tremendous expansion of connected devices and growth of mobile data traffic in next generation wireless networks [1]. Recently, interference alignment (IA) has been proposed as a promising solution to the interference management problem of modern wireless systems [2]. In IA-based wireless networks, the interference among users is cooperatively aligned into certain subspaces at receivers through precoding matrices, and thus the desired signal can be decoded free of interference [3, 4]. In [2], Cadambe and Jafar derived the degrees of freedom (DoFs) and sum rate of a K-user IA network. The closed-form solutions of an IA network with plenty of users are difficult to obtain [2, 3]. In [3], Gomadam et al. proposed two easily implemented algorithms to calculate the solutions iteratively. The feasibility conditions were analyzed in [5], based on which we can determine whether a certain IA-based network can be achieved according to the number of antennas, users, and data streams. When signal-to-noise ratio (SNR) becomes lower, the sum rate of an IA-based network may fall short of the theoretical maximum, and thus in [4], the idea of opportunistic communication was leveraged to improve its performance. Due to its prospective performance, IA has been successfully applied to many wireless networks.

On the other hand, due to the broadcast and superposition characteristics of wireless channels, the security of information transfer in wireless networks is still a critical and challenging issue [6]. Thus, physical layer security is becoming more and more important, which has attracted significant interest from both academia and industry. Eavesdropping and jamming are two main attacks at the physical layer of wireless networks. When eavesdropping is involved, it is critical to guarantee that the confidential information transmitted between legitimate transceivers cannot be retrieved by passive eavesdroppers [7]. With regard to jamming, legitimate transmission should not be attacked by the jamming signal from active jammers outside the network [6]. From another point of view, the jamming signal can also be exploited as artificial noise (AN) to improve the physical layer security of legitimate transmission by disturbing the eavesdropper’s reception [8].

Recently, due to its superposed nature, IA tends to be adopted in analyzing the precise secure DoFs of many kinds of wireless networks based on the information-theoretic concept, including X networks [9], interference networks [10], one-hop networks [11], multicast networks [12], broadcast networks [13], and so on. Also, some research works have also been conducted to make IA-based networks more secure by additional secrecy precoding [14] or minimization of total mean square error [15]. Nevertheless, the physical layer security issues of IA-based networks have not been analyzed systematically, and more research should be done to further improve the anti-jamming and anti-eavesdropping performance. Thus, in this article, the physical layer security issues of two primary attacks, jamming and eavesdropping, are analyzed in IA-based wireless networks. Specifically, when an adversarial jammer with multiple antennas exists, an anti-jamming IA scheme is proposed through designing the precoding matrices to align the jamming signal along with interference among users. Also, an AN scheme is proposed for the...
IA-based wireless networks, and AN is generated by each IA transmitter to disrupt the external eavesdropping without introducing any additional interference to the legitimate network. To further analyze the potential threat of IA-based wireless networks, a collusive eavesdropping scheme (CES) by some hostile IA users in the network is also proposed. Simulation results of these schemes are presented, and some research challenges are pointed out for the physical layer security issues of IA.

The rest of this article is organized as follows. In the next section, the physical layer security issues of IA are analyzed, and then we propose an anti-jamming IA scheme when an adversarial jammer exists. To disrupt external eavesdropping, an AN scheme for IA-based wireless networks is proposed. Then a collusive eavesdropping scheme by some hostile IA users is proposed to analyze the potential threat of IA-based wireless networks. Some challenges are then presented, and we conclude the article in the final section.

**Physical Layer Security Issues of IA-Based Wireless Networks**

With the interference aligned in certain subspaces, IA seems secure, as demonstrated in Fig. 1, and we analyze the physical layer security issues of IA as follows.

**External Jammer**
When there is an adversarial jammer that wants to disrupt the transmission of IA users, more antennas can be equipped, or some of the streams in the network can be reduced to align the interference with the jamming signal as in Fig. 1, which can be eliminated together at each receiver. This is demonstrated later.

**External Eavesdropper**
IA seems to be secure due to the superposition of signals at the eavesdropper. Nevertheless, it can still be eavesdropped when adequate antennas are equipped at the eavesdropper. An AN scheme is proposed later for IA-based wireless networks to battle external eavesdropping.

**Internal Eavesdropper**
When a hostile IA user that intends to eavesdrop a certain legitimate user exists, it is difficult to achieve this due to the alignment of signals from other users at the eavesdropper. However, when it is helped by some IA users, the information of a legitimate user can be eavesdropped, as introduced later.

**Anti-Jamming IA Scheme with Adversarial Jammers**
In this section, an anti-jamming IA scheme is proposed to eliminate the jamming signal along with interference from other users for IA-based wireless networks.

**Anti-Jamming IA Scheme**
In a symmetric IA-based wireless network of \((M \times N, d)^K\), it is feasible with \(M + N \geq d(K + 1)\) according to [5]. When IA is performed, interference among users can be constrained into certain subspaces at unintended receivers by cooperative precoding matrices, and then perfectly eliminated through decoding matrices. However, when some external jammers exist, the transmission of IA users will be influenced severely. We know that there may be several jammers with total \(N_j\) antennas or only one \(N_j\)-antenna jammer. When the \(N_j\) antennas on the jammers are independent, these cases are equivalent. Thus, only the case of one jammer with \(N_j\) antennas is discussed in this article for simplicity. To guarantee the transmission of IA users when adversarial jammers exist, the jamming signal should be eliminated together with the interference among users, and the precoding and decoding matrices should be re-designed accordingly. Thus, an anti-jamming IA scheme is proposed in which the jamming signal should also be zero-forced at each receiver besides the IA conditions (4) and (5) in [3].

In the anti-jamming scheme, the IA conditions should be satisfied and the jamming signal should be zero-forced. To achieve this, the iterative IA algorithm in [3] can be adopted with some necessary modifications. In the forward direction of iterations, the jamming signal is also considered to calculate the decoding matrices, while in the reverse direction, the jamming signal is not involved due to the fact that the jammer can only transmit signal instead of receiving. Therefore, when enough antennas are equipped at the transceivers of users, the jamming signal and interference can both be eliminated effectively through the anti-jamming IA scheme. Besides, to achieve the anti-jamming scheme, the channel state information (CSI) from the jammer should also be estimated along with that from the legitimate transmitters by each IA receiver.\(^2\)

**Feasibility Conditions**
The anti-jamming IA scheme will achieve reliable performance only when the feasibility conditions can be satisfied; otherwise, neither the jamming signal nor the interference can be eliminated further. However, when adequate antennas are equipped at the eavesdropper, AN is generated accordingly. \(V^K\) means that in the \(K\)-user IA network, \(M\) and \(N\) antennas are equipped at each transmitter and receiver, respectively, and \(d\) independent data streams are transmitted by each user.\(^2\)

\(^1\) \((M \times N, d)^K\) means that in the \(K\)-user IA network, \(M\) and \(N\) antennas are equipped at each transmitter and receiver, respectively, and \(d\) independent data streams are transmitted by each user.

\(^2\) Perfect global CSI is assumed to be available at each IA node. CSI acquisition is a common issue of IA and is not further discussed in this article.
eliminated perfectly. The feasibility conditions of IA-based wireless networks are well demonstrated in [5], and a generic polynomial system is solvable if and only if the number of variables \( N_v \) is no less than the number of equations \( N_e \) according to Bezout’s theorem. In the anti-jamming scheme, the total number of variables \( N_v \) can be expressed as \( dK(M + N - 2d) \). The number of equations when considering IA is equal to \( dK(dK - d) \), and the number of equations when zero-forcing the jamming signal is \( dKN_j \). Thus, the total number of equations of the anti-jamming IA scheme \( N_e \) can be denoted as \( dK(dK - d + N_j) \). When the anti-jamming IA scheme is feasible, it must follow \( N_v \geq N_e \), and we can obtain \( M + N \geq (K + 1)d + N_j \). Besides, we know that zero-forcing the jamming signal can be only solved by the variables in decoding matrices, in which the number of variables can be expressed as \( N_v = Kd(N - d) \). The total number of equations when zero-forcing the jamming signal is \( N_e = KdN_j \). Thus, the \( KdN_j \) equations can only be solved when \( N_v \geq N_e \), and we obtain \( N \geq N_j + d \). According to the DoF requirement of a point-to-point multiple-input multiple-output (MIMO) channel, the value of \( M \) should also satisfy \( M \geq d \).

Actually, the conditions derived above are proper conditions, that is, when the anti-jamming IA scheme is feasible, it must follow \( M + N \geq (K + 1)d + N_j \), \( N \geq N_j + d \), and \( M \geq d \). When only one stream is transmitted by each user in a symmetric network, the proper conditions are equal to feasibility conditions; however, when multiple streams are transmitted by each user, the proper conditions are not always equivalent to the feasibility conditions with few counter-examples. Fortunately, the iterative algorithm in [3] can be utilized as a theoretical tool to examine the feasibility of a proper anti-jamming IA scheme easily. In a feasible IA-based wireless network of \((M \times N, d)\) with \( M + N = d(K + 1) \), we can know that the corresponding anti-jamming IA scheme will surely be infeasible when some external jammers of total \( N_j \) antennas exist. What should we do to make it also feasible when the jamming signal is also involved? From the proper conditions obtained above, we can add \( N_j \) more antennas at each user to make the anti-jamming IA scheme feasible, as long as \( N \geq N_j + d \). We can also make it feasible for the anti-jamming IA scheme by reducing one stream of a certain user. When \( d = 1 \), reducing one stream of a certain user is equivalent to turn one user of the network into sleep mode.

**Simulation Results**

In the simulation, we consider a three-user IA-based wireless network with one data stream for each user. The transmit power of each user is set to \( P \), and the transmit power of each antenna of jammers is also equal to \( P \). The sum rate and the interference leakage per user in the anti-jamming IA scheme are compared in Fig. 2 with different numbers of \( M + N \) and \( N_j \).

In Fig. 2a, \( N_j \) is set to 3. From the results, we can see that when the feasibility conditions of the anti-jamming IA scheme can be satisfied, that is, \( M + N \geq (K + 1)d + N_j \), the jamming signal and interference can be perfectly eliminated, and the sum rate of the network is high. However, when \( M + N = 6 < (K + 1)d + N_j \), the performance of sum rate degrades severely, due to the fact that it becomes unfeasible. In Fig. 2b, the SNR is set to 10 dB, and the number of antennas at the jammers, \( N_j \), changes from 1 to 5. From the results, we can see that only when the feasibility condition of the scheme can be satisfied as \( M + N \geq (K + 1)d + N_j \), the interference leakage per user in the network can be mitigated.
to zero, which means that the network is feasible. Otherwise, the interference leakage will become higher with larger \( N_e \), which will significantly reduce the performance of the scheme.

**Artificial Noise Scheme for IA-Based Wireless Networks**

In this section, an AN scheme for IA-based wireless networks is proposed, in which AN is generated at each transmitter to disrupt external eavesdropping without introducing any additional interference to the legitimate network.

**Artificial Noise Scheme**

When an external eavesdropper of the IA-based wireless network exists, zero-forcing can be performed by multiple antennas to eavesdrop the information from a certain legitimate user with interference from other users eliminated. Nevertheless, as the CSI between the legitimate receivers are usually unknown at the eavesdropper, the number of data streams of each user is difficult to determine at the eavesdropper, and thus the interference among streams of the targeted user cannot be eliminated. To achieve the eavesdropping with interference perfectly eliminated, at least \( N_{eav} = dK - d + 1 \) antennas should be equipped at the eavesdropper outside an IA-based wireless network of \((M \times N, d)^K\). As the eavesdropper usually works passively without access to the legitimate network, the CSI of the eavesdropper is unknown to the IA users. Thus, the transmit power for each user is adopted to be in constant with the external eavesdropper for IA-based wireless networks, as shown in Fig. 3. In the proposed AN scheme, \( d \) data streams are transmitted by the \( k \)th user through precoding matrix \( \mathbf{W}_k \), \( k = 1, 2, \ldots, K \), through which the interference is constrained into a certain subspace at each receiver. One-stream AN is also generated at the \( k \)th transmitter through precoding vector \( \mathbf{w}_k \), by which the AN is aligned into the same subspaces as the interference as in Fig. 3. Thus, AN can be eliminated along with the interference at each receiver through decoding matrix \( \mathbf{U} \), which is orthogonal to the subspace of interference and AN. Based on the above analysis, the transmission power of AN can be increased to reduce the eavesdropping rate without introducing any additional interference to the legitimate network. In the scheme, the AN generated by each user should be aligned into the same subspaces as the interference at each receiver, and to realize this, the iterative IA algorithm \([3]\) can be adopted with some necessary modifications. In the forward direction of iterations, both the AN and interference should be involved to calculate the decoding matrices, while in the reverse direction, the interference is considered to update the precoding matrices for data streams, and the signals from all the users should be involved to calculate the precoding vectors for AN.

To achieve the above AN scheme for IA-based wireless networks, enough antennas should be equipped at each transmitter to make it feasible. In the AN scheme of \((M \times N, d)^K\), the number of variables in the precoding and decoding matrices for data streams is \( dK(M + N - 2d) \), the number of variables in the precoding vectors for AN is \( K(M - 1) \), and thus the total number of variables can be expressed as \( dK(M + N - 2d) + K(M - 1) \). On the other hand, the number of equations when considering IA is equal to \( d^2K(K - 1) \), the number of equations to eliminate the AN at all the users is equal to \( dK^2 \), and thus the total number of equations can be denoted as \( d^2K(K - 1) + dK^2 \). To make the proposed AN scheme feasible to solve, the number of variables should be no less than that of equations, and we can obtain its feasibility conditions as \( (d + 1)M + dN \geq d^2K + d^2 + dK + 1 \). Besides, we can know that the total number of antennas equipped at each user \( M + N \) to make the scheme feasible becomes smaller with larger \( M \). Thus we can also conclude that at least \( M + N = d(K + 1) + 1 \) antennas should be equipped at each user to make the scheme feasible, that is, only one more antenna is needed for each user than the conventional IA-based wireless network, when \( dK - k + 1 \leq M \leq dK + 1 \) can be satisfied.

Although the proposed AN scheme will not introduce additional interference to the legitimate transmission of the network, the sum rate will be decreased compared to the conventional IA-based wireless networks, especially when \( M \) is small. This is due to the fact that the decoding matrix \( \mathbf{U} \) will eliminate the AN from its corresponding transmitter, which will also affect the received power of the legitimate transmission. As \( M \) becomes larger in a feasible AN scheme, the transmission rate will become larger. Thus, it is better for us to set \( dK - k + 1 \leq M \leq dK + 1 \) to achieve relatively high performance with fewer antennas.

**Simulation Results**

In the simulation, we consider the AN scheme for an IA-based wireless network with \( K = 5 \) and \( d = 2 \), and SNR is set to 20 dB. The antennas at the eavesdropper \( N_{eav} \) is set to 13 to zero-force the inter-user interference. The transmit power of each user is set to \( P \). The transmission rate, eavesdropping rate, and secrecy rate of a certain legitimate user are compared in Fig. 4.

In Fig. 4a, the transmit power of AN is varying. From the results, we can see that the eavesdropping rate will be decreased when the
transmit power of AN becomes higher without affecting the transmission rate, and thus the secrecy rate becomes higher accordingly. In Fig. 4b, the transmission rate becomes higher when $M$ becomes larger, which is consistent with the analysis above. The eavesdropping rate is independent of the value of $M$, and thus the secrecy rate also becomes higher with larger $M$. Besides, we can find that the curves have a fluctuation at $M = 4$, this is because, to make $N$ be an integer, more antennas than needed are equipped at each receiver to make the scheme feasible.

**Collusive Eavesdropping Scheme in IA-Based Wireless Networks**

As analyzed above, when only one hostile eavesdropper exists in the IA-based wireless network, it is difficult to achieve eavesdropping due to the overlapping of signals from different users. Nevertheless, if some of them cooperate to eavesdrop a certain user, it can be achieved free of interference. In this section, we present a CES in IA-based wireless networks.

**CES in IA-Based Wireless Networks**

In a feasible IA-based wireless network of $(M \times N, d)^P$ with $M + N = d(K + 1)$, without loss of generality, we assume that the first user is eavesdropped by the $K$th user, with the help of the other $K − 2$ users. Thus, we can denote the $K$th user as the eavesdropper, and the second to the $(K − 1)$th users as cooperators. Because the precoding and decoding matrices of the first user cannot be changed, $G$ out of the $K − 2$ cooperators should sacrifice their own quality of service (QoS) to make a CES feasible. Thus, the requirements of CES should be satisfied as follows:

- The data transmission of the first user should be performed free of interference, and it cannot perceive that it is being eavesdropped by other users.
- The transmission QoS of the second to the $(G + 1)$th users is sacrificed to make the CES feasible; nevertheless, the received signal-to-interference-plus-noise ratio (SINR) of these $G$ users should be maximized.
- The data transmission of the $(G + 2)$th to the $(K − 1)$th users can still be carried out free of interference.
- The transmitted information of the first user is eavesdropped at the eavesdropper (the $K$th receiver) free of interference.
- The transmission of the cooperators and the eavesdropper should not be terminated, or the first user may suspect that it may be eavesdropped.

To achieve the requirements presented above, the precoding and decoding matrices should be re-designed as in Fig. 5. In the CES, the precoding and decoding matrices of all the users should be calculated, and then the precoding and decoding matrices of the eavesdropper and cooperators should be re-designed to eavesdrop the first user at the $K$th receiver. The signals from the second to the $K$th transmitters should be aligned at the eavesdropper, and thus the information transmitted by the first user can be eavesdropped. The data transmission of the $(G + 2)$ th to the $(K − 1)$th users can still be performed free of interference, while the alignment of interference is relaxed at the second to the $(G + 1)$
th receivers to make the CES feasible; thus, the QoS of these $G$ cooperators is sacrificed. Nevertheless, the received SINR of the second to the $(G + 1)$th users should be maximized. The interference from the second to the $K$th transmitters should be aligned in the same direction at the first receiver as in the original IA scheme, which is orthogonal to $U^{[1]}$; thus, it will not notice that it is being eavesdropped. To re-design the precoding and decoding matrices of the cooperators and eavesdropper in the CES according to Fig. 5, we can adopt the iterative IA algorithm in [3] with some necessary modifications, which will not be specified due to the simplicity of this article. Besides, according to the relations of the number of equations and variables, we can conclude that the proposed CES in a feasible IA-based wireless network with $M + N = d(K + 1)$ is also feasible when $(K + 1)M + N = 2d – dK$ can be satisfied. From this feasibility condition, we can also know that the value of $N$ should be less than or equal to $(GdK - dK + d)/G$ to make the CES feasible when $M + N = d(K + 1)$.

Although the eavesdropping can be performed free of interference in the CES, the eavesdropping rate will be decreased when $N$ becomes smaller. This is because the first user and the $K$th user utilize $V^{[1]}$ and $U^{[2]}$ to eliminate the signal from the first transmitter at the $K$th receiver in the original network. When $N$ is smaller, $V^{[1]}$ will play a more important role in the interference mitigation, and thus the eavesdropping rate will become lower although $U^{[2]}$ are re-designed. There are two special cases. When $N = d$, the eavesdropping rate will be zero even when the feasibility condition can be met; when $N = Kd$, the eavesdropping can easily be done with better performance without the help of any cooperators, even though the feasibility condition cannot be met. Thus, to guarantee the feasibility condition and eavesdropping rate, $N$ should be set as large as possible in $[d + 1, (GdK - dK + d)/G]$ or to be $N = dK$.

**Simulation Results**

In the simulation, we consider the CES in a five-user IA-based wireless network with $M + N = 12$, $d = 2$, and $G = 2$. The transmission rate of the first user, the average rate of the cooperators, and the eavesdropping rate are compared in Fig. 6 with SNR = 40 dB. From the results, we can know that the proposed CES is feasible when $3 \leq N \leq 6$, and the eavesdropping rate and transmission rate of the low-QoS cooperators increase gradually with larger $N$ in this case. The transmission rate of the first user and the high-QoS cooperator is unchanged when CES is feasible. When $7 \leq N \leq 9$, the CES becomes infeasible, and the transmission of the legitimate user will be affected. When $N = 10$, the eavesdropping rate becomes equal to that of the legitimate user, and the CES can be performed even without the help of cooperators.

**Research Challenges**

Although some fundamental works have studied physical layer security issues in IA-based wireless networks in this article, there are still some research challenges as follows.

- In the anti-jamming scheme, the number of antennas equipped at each user will be increased as the number of antennas on the external jammers becomes larger, and as a result, the CSI overhead and computational complexity will also increase accordingly. Therefore, it is essential to guarantee the anti-jamming performance with fewer antennas equipped when plenty of jammers exist in future work.
- In the AN scheme for IA-based wireless networks, AN is generated by each transmitter, and it will disrupt the eavesdropping without affecting the transmission of the legitimate network. Nevertheless, when we consider a much more complex network that consists of several sub-networks, how to manage the inter-cluster interference caused by AN with the minimum number of antennas is a challenging problem to be solved.
- In the previous section, the CES is proposed in IA-based wireless networks by collusive eavesdropping when some hostile users exist. Thus, it
is challenging for us to propose a strategy to prevent internal eavesdropping in IA-based wireless networks in this case. There may be many jammers and eavesdroppers in IA-based wireless networks, and the conventional method to guarantee the security of the legitimate network will become insufficient. Thus, it is interesting to study the distribution model of the jammers and eavesdroppers by exploiting stochastic geometry, and some more generalized methods with low complexity may emerge.

**Conclusions**

While IA is an emerging technology for interference management, its security is still a critical issue to be addressed before widespread deployment of IA. In this article, two primary attacks at the physical layer of IA-based wireless networks, jamming and eavesdropping, were studied. First, an anti-jamming IA scheme is proposed in which the interference among users is aligned into the same subspace as the jamming signal, and thus can be eliminated together perfectly. Then an AN scheme is proposed, and artificial noise is generated at each transmitter, which will disrupt the eavesdropping without introducing additional interference to the legitimate network. Also, to analyze the potential threat of IA-based wireless networks, a CES scheme is proposed in which the some hostile users are cooperating to eavesdrop a legitimate user without its perception. Finally, we point out some research challenges for the physical layer security issues of IA-based wireless networks.

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The authors present a novel cell outage compensation approach using new SHRs added to each cell site in the 5G network. These SHRs operate only in case of fronthaul/backhaul failure of any cell site in the network. A new software defined controller is introduced to handle the self-healing procedures. The article also introduces a high level simulation study that is carried out to assess the proposed approach.

**ABSTRACT**

5G networks are expected to bring the gigabits per second throughput level per user to reality by 2020. This is done using a combination of new and well known technologies such as C-RAN, self-organizing networks, ultra dense networks, massive MIMO, and millimeter waves. In new RAN architectures, C-RAN has been viewed as a promising 5G architecture that centralizes baseband processing units and virtualizes them into a resource pool. The baseband units are connected to the remote radio heads via high speed fronthaul links. Failure of any 5G cell site fronthaul means the loss of hundreds of gigabits, or even terabits. In this article, we present a novel cell outage compensation approach using new SHRs added to each cell site in the 5G network. These SHRs operate only in case of fronthaul/backhaul failure of any cell site in the network. A new software defined controller is introduced to handle the self-healing procedures. The article also introduces a high-level simulation study that is carried out to assess the proposed approach. The simulation results confirm the advantages of the proposed approach in terms of the degree of recovery from failures.

**INTRODUCTION**

The main objective of 5G networks is to achieve the gigabits per second throughput level with high reliability and “limitless” connectivity from everything to everything, anytime, and anywhere. Many new and/or well known technologies are proposed to be integrated in the 5G architecture such as cloud radio access networks (C-RAN), massive MIMO, millimeter waves (mmW), software defined networking (SDN), self-organizing networks (SONs), and dense heterogeneous networks (HetNets).

C-RAN is a cloud computing based newly adapted cellular network architecture that is highly expected to be integrated in future cellular standards. Each conventional base station (BS) consists of a baseband unit (BBU), which is responsible for baseband processing, and radio heads, which are responsible for radio frequency functionalities. By separating the radio heads, which are now called remote radio heads (RRHs), from the BBU, baseband processing is migrated to the cloud and connected to the RRHs via high speed fronthaul links, giving rise to the so called C-RAN. C-RAN provides several advantages compared to conventional RANs, such as faster network deployment and system performance improvement. Even more, by the interworking of SDN with C-RAN in 5G networks, C-RAN can gather more information to perform better global tasks and decisions.

It is expected that 5G data traffic will increase up to 1000 fold over that of 4G, implying that the 5G backhaul/fronthaul capacity has to carry a significant amount of traffic. These high rates pose new requirements for backhaul/fronthaul links [2]. Achieving data rates on the order of 10 Gb/s or 100 Gb/s for access communications or fronthaul links is possible only if the available bandwidth is within 1 GHz, which is available only in the mmW bands [3]. Although 5G requires multiple gigabits per second, tens or hundreds of megabits per second need to be guaranteed with very high availability and reliability in case of failures.

Adding new network elements, optimizing these elements, and mitigating the effect of the different potential failures is one of the most challenging tasks facing any network operator. The automation of these tasks is referred to as SON. SON is defined as the network that has the capability to dynamically adapt changes in the network in order to optimize its performance. The need for SONs arises from the fact that in future networks, the number of nodes will be increasing at a rapid rate. Moreover, it is also because of the introduction of a high degree of heterogeneity and complexity, that such SONs could save significant operational expenditure.

SON defines three areas: self-configuration (plug and play network elements), self-optimization (automatically optimize network elements and parameters), and self-healing (automatically detect and mitigate failures). For the use of SON in 5G networks, the reader is referred to [4].

Self-healing is the execution of actions that keep the network operational and/or prevent disruptive problems from arising. Self-healing is done in two steps: cell outage detection (COD) and cell outage compensation (COC). COD detects and classifies failures, while minimizing detection time. COC executes actions to mitigate or at least alleviate the effect of the failure. If the failure time exceeds a certain threshold, it is con-
Figure 1. The system model.

The proposed HetNet C-RAN Architecture

The new proposed HetNet C-RAN architecture consists of the BBU pool and RRHs of different types of cells where the small cells (SCs), that is, picocells and femtocells, are co-located within the macrocell footprint, as shown in Fig. 1. Each macrocell is supposed to use fractional frequency reuse to avoid interference with other macrocells. The SDN concept was included for network flexibility and the software defined wireless network controller (SDWNC) is used mainly to control the whole process of COC.

The different types of cell sites are front-hauled using different types of links. A macrocell is always connected to the BBU pool using fiber link(s). Picocells are connected to the BBU pool or to the macrocell either by using fiber or wireless point to point links. Femtocells can be front-hauled using any of the mentioned links in addition to coaxial cable, hybrid fiber-coaxial, or copper pairs. If the former links are used, the Internet service provider (ISP) must have a low latency high speed connection to the cloud. This C-RAN architecture is also compatible with the conventional BSs, which are not C-RAN based.

It should be noted that the heterogeneity in this architecture is in two directions. In one direction the C-RAN serves heterogeneous types of cell sites (macrocells, picocells, and femtocells). The second direction includes different wired and wireless types of fronthaul connections.

**THE PROPOSED COC APPROACH**

The conventional and well known approach for COC is to optimize the capacity and coverage of the outage zone by adjusting the antenna gain and transmission power of the neighboring BSs. The disadvantage of this approach is that it will degrade the performance of the users served by the neighboring BSs. Moreover, these BSs will consume considerable time (compared to 1 ms) to change their power and antenna tilt. The proposed COC approach will not change the antenna tilt or the power of the neighboring cell sites. We propose to add a new radio, called a self-healing radio (SHR), to each cell site in the network. When a failure occurs to the fronthaul of any cell site, the failed cell site will acquire its fronthaul connection from the neighboring cell sites using the SHRs. The neighboring cell will help the failed cell if it has available resources. This approach is discussed in detail later.

The network always operates in the normal mode, which may last for days. The network is monitored for detection of any failure by the SDWNC. In case of failure, the COC strategy will be activated by the controller within a few milliseconds. System repair, which will be done by the operator’s maintenance personnel, can typically be done within hours and in this case, the COC approach will provide recovery that satisfies the minimum rate requirements in the failure region. Switching back to normal operation is performed in multiples of milliseconds, and it is triggered by the SDWNC, which will deactivate all SHRs for all cell sites.

There are many challenges and issues that were addressed to provide a reliable network architecture that is able to mitigate any fronthaul failure, such as the bands used for the macrocell tier and the SCs tier, the SHRs bandwidth, how to control the whole COC process, and using pre-planned femtocells to help in the healing process. These issues are addressed below.

**C-RAN ARCHITECTURE**

The C-RAN architecture consists of the following three main components.

- The BBU pool is composed of a set of servers, storage, and switches. The BBU assignment for each RRH could be centralized or distributed. In our architecture we are considering the centralized implementation due to its advantages (flexible resource sharing, energy efficiency, and interference avoidance).
- RRHs are located at the remote cell sites. They transmit the radio frequency signals in the downlink and forward the baseband signals in the uplink for further processing in the BBU pool. They include a radio frequency amplifier, an up/down conversion mixer, and analog-to-digital and digital-to-analog conversion.
• Fronthaul links can be wired or wireless. Although wireless fronthaul is cheaper and faster to deploy than wired, the best choice is high speed optical fiber, but microwave or mmW are also considered (the reader is referred to [9] for an overview of possible wired and wireless backhaul/fronthaul technologies). Fronthaul antennas can be installed on top of roofs to ensure the existence of line-of-site (LOS) propagation. Also, the reader is referred to [10] for more information regarding integrated fronthaul/backhaul architectures in 5G.

By implementing C-RAN, 5G networks will gain many advantages, such as:
• The equipment needed at the cell site is only RRHs, power supply, and backup batteries, which will result in shorter installation and repair time.
• The cell site will consume less energy (no need for air conditioning).
• The ease of communication between BBUs in the cloud will provide better performance for mobility management, interference cancellation, and coordinated multi-point communication, which is expected to provide higher capacity and improve cell-edge coverage.

Backhaul failures of regular BSs is one of many types of failures that can occur in BSs in 4G networks. However, after simplifying the BSs in C-RAN into the RRHs and migrating all processing units (BBUs) to the cloud, fronthaul failures is the failure that has the most impact on the operation of the C-RAN 5G network.

**Millimeter Wave (mmW) Bands**
The mmW bands are typically sub-band free, and high frequency reuse is possible due to very narrow directed beams. The so-called light spectrum licensing scheme provides lower total cost of ownership and lower cost per transmitted bit than microwave bands.

Using mmW bands in 5G networks has already been proposed in the literature. The mmW band is used in short-range LOS communications. It is not used in long distance communications due to the high attenuation and oxygen absorption of these waves [3]. The wireless fronthaul and access links are assumed to be out-of-band, that is, there is no interference between them. This will provide 5G users with the target data rates, and SCs will rely on high-gain beamforming to mitigate pathloss [11].

The macrocell situation is complicated due to its wide area coverage. Using mmW with macrocells is still under intensive research because of the high attenuation associated with wide coverage. As is well known from 4G networks, 80 percent of network traffic is used indoors and only 20 percent is used outdoors [12]. The 20 percent outdoor traffic will be carried out by the macrocell and the outdoor SCs. This trend is expected to continue in 5G, meaning that the traffic carried by the macrocell in 5G networks is much lower than that carried by SCs.

In our proposed solution, mmW is used only in fronthaul connections and in SCs access links between SCs and user equipment (UE). However, the macrocell will use the traditional cellular band (2–6 GHz) for communications with its UEs. The motivations behind using this band are better coverage, lower penetration loss, and eliminating the interference issue between the SCs tier and macrocell tier. The only limitation of this band is its bandwidth, but using massive MIMO, carrier aggregation, and other technologies can facilitate the achievement of macrocell gigabits per second throughput.

**SHRs Band and Cognitive Radio Concept**
The SHRs can use mmW band, the traditional cellular band, or a new dedicated band. For the mmW band, it is difficult to be used because of the NLOS path between SHRs. Dedicating a portion of the traditional cellular band, which is also used by the macrocell, to SHRs communication will affect the band utilization because SHRs are only activated if fronthaul failures occur. Finally, using a dedicated band (licensed for self-healing communications only) will dramatically increase capital expenditures, and also this band will not be fully utilized.

The cognitive radio (CR) concept is one of the promising technologies for solving telecom problems such as spectrum scarcity or in case of disasters [13]. Our proposed solution is to use the CR for SHRs communications, which optimizes the use of the available spectrum. The band used is the same as the second solution, where a portion of the traditional cellular band will be dedicated to the SHRs. The main difference is that when SHRs are inactive, this portion will be available for the macrocell to use as a secondary user. Therefore, if the macrocell is starved for bandwidth, it will sense the SHRs dedicated channels, and if it is free then the secondary user (macrocell) will use the vacant channels until the primary users (SHRs) are activated. Once the primary user (SHR) is active (this means that a failure has occurred), the macrocell will vacate these channels.

Furthermore, in our model the macrocell can avoid wasting time in spectrum sensing by acquiring the SHRs channel occupancy information from the SDWNC. If there is no failure, the macrocell will use the reserved portion of the band without sensing. If a failure happens, the SDWNC will immediately request from the macrocell to vacate the SHRs channels to be used by the primary users (SHRs). This will save sensing power and time and will also increase the reliability of using CR in 5G networks.

**Femtocells Used**
As is known from 4G, femtocells may be deployed by network users. In this case they are randomly placed and are called random femtocells (RFs), or femtocells may be deployed by the operator in pre-planned locations to enhance the capacity or to cover dead zones. In this case they are called pre-planned femtocells (PFs).

The main difference between 4G PFs and the proposed 5G PFs is that the latter are used mainly to self-heal the failed cell sites. In addition, they can also be used for capacity enhancement and dead zone coverage. The only constraint is that they must be located within the SHRs footprint of the target cell site, and it is preferable that the PF be connected to a fiber fronthaul to guarantee the gigabits per second 5G through-
put, which will be used mainly to heal the failed cell sites and also for serving its own users.

**Software Defined Wireless Network Controller**

The software defined wireless network controller (SDWNC) is co-located within the cloud to gather the needed information in a fast and reliable way. The SDWNC is a mandatory component in our architecture as it acts as the supervisor, decision maker, and administrator for all self-healing procedures applied to all network cell sites. The reader is referred to [14] for more details about SDN in wireless networks.

The co-location of the SDWNC in the cloud will help it make optimal decisions in order to decide how to recover from fronthaul failures. This reduces the amount of information needed to be exchanged, which will allow optimal recovery in a very short time.

The SDWNC activates SHRs for all cell sites in the vicinity of the failed cell site, and also deactivates the SHRs after the failure is repaired. In addition, it deactivates the SHRs of the cells that are not participating in the self-healing process to save their power from being wasted.

**Self-Healing Procedures**

While Fig. 1 shows the entire network, in Fig. 2 we focus on one of the macrocell’s coverage, where each macrocell is associated with three PFs (one in each sector) and each picocell is associated with one PF. There are no PFs associated with RFs, because macrocells and picocells are serving a large number of users compared to the RFs. The latter will search for nearby SHRs, in case of failure, in order to heal their failed fronthaul link. This proposed C-RAN architecture is compatible with the conventional BSs, which are not C-RAN based. RF 11 in Fig. 2 shows an example of a BS backhauled from the ISP.

When a certain cell site fronthaul fails, it will automatically activate its own SHRs, but will be totally disconnected from the SDWNC, and in this case the cloud, macrocell, or ISP (depending on which one was fronthauling this cell site) will report the failed cell details to the SDWNC. The SDWNC will then activate the SHRs of all cell sites in the region surrounding the failed cell site. The process of detecting the failure and SHRs activation must be done within a very short time.

The failed cell will try to connect to the healing cell sites via SHRs according to a certain priority order depending on the cell site type, available resources, and the received signal strength (RSS). The failure of a fronthaul link may be permanent or transient. Our COC approach works with both, and the only difference is that in the case of a permanent failure, the SDWNC, after a certain threshold time, will inform the operator that there is a failure that requires maintenance personnel to make a field visit to the failed cell site. It should be noted that our proposed COC scheme can be classified as a hybrid approach according to the definition given in [6].

**Single Failure Scenario**

Single failure means that only one cell site fronthaul failed in the network. In this case, the SC will trigger the self-healing mode, where it will activate its own SHRs and then try to connect to other cell sites’ SHRs, which are activated by the SDWNC. Following a priority order, the SC will first try to connect to a macrocell, then PFs, then picocell, and finally RFs. RFs are assigned the lowest priority because they are personal property and the operator will have to compensate the owners of these RFs.

The flow chart in Fig. 3 shows the SC fronthaul failure mitigation process, where the SDWNC always monitors the network to detect any fronthaul failure. If the fronthaul of any cell site failed, the failed SC will immediately activate its SHRs, and at the same time the SDWNC will activate SHRs of all neighboring cell sites.

![Figure 2. SC failure mitigation process.](image_url)

![Figure 3. SC failure mitigation flow chart.](image_url)
At this point the failed SC will use its SHRs to measure the RSS levels from all neighboring cell sites, and will update a list of healing cell sites with measured RSS levels higher than a certain RSS threshold. The failed SC will use priorities to decide which cell sites’ SHRs it will connect to. The SDWNC will continue to monitor the failed fronthaul link, and when the failed link resumes working properly, the SDWNC will deactivate SHRs of all cell sites in the failure region and the network resumes operating normally.

An example of this failure scenario is shown in Fig. 2, where the fronthaul link of RF 7 failed. The failed RF will apply the priority order described above, but because it is out of SHRs’ coverage of the macrocell, PFs, and nearby picocells, it will connect to the SHRs of RF 6 and RF 8 to mitigate its fronthaul failure. Once the failure is repaired, the SHRs will be deactivated by the SDWNC and RF 7 will return to serve its users using its own fronthaul link.

The failure of the macrocell fronthaul is not considered to be a single failure because this will immediately cause the failure of all SCs fronthauled from the macrocell, causing multiple failures in the network.

**Multiple Failures Scenario**

A multiple failures scenario refers to the situation where two or more fronthaul failures occur at the same time in the same region. There are two cases studied here:

**SCs Fronthaul Failure**: This case can be seen as the failure of two or more SCs in the same region. The procedures in the flow chart above can be used to mitigate these failures when implementing the proposed algorithm for each failed cell site. For example, if two SCs failed, the SDWNC will activate the SHRs of cell sites in their region, and each failed cell site will activate its SHRs and try to connect to the healing cell sites. As the number of failures increases, the probability of healing each failed cell will depend on the number of nearby healing cell sites that can provide the temporary fronthauling connections. Thus, as expected in 5G networks, the dense deployment of SCs will enhance the performance of our self-healing approach, especially in the multiple failures scenario.

**Macrocell Fronthaul Failure**: The macrocell plays a vital role in HetNets, where in addition to its main function, i.e., coverage for outdoor users, it provides wireless fronthaul links to other SCs that cannot directly reach the cloud. In our architecture, two picocells acquire their fronthauling from the macrocell. Figure 4 shows an example of the macrocell fronthaul failure and the mitigation process. The macrocell fronthaul failure (the solid red line) immediately results in the failure of two fronthaul links (Pico 1 and Pico 2). As can be seen in Fig. 4, the self-healing process will mitigate the effect of failure for the fronthauling picocells and the macrocell. The picocells will be recovered first using PFs and RFs in their vicinity. For example, Pico 1 will be recovered by PF 1, RF 1, and RF 3. Similarly, Pico 2 will be recovered by PF 2, RF 9, and RF 11.

The next step, macrocell fronthaul recovery, will be done from two sources. The first source uses the mmW links between the macrocell and recovered picocells. The second source is the recovery from femtocells in its SHRs footprint. As shown in Fig. 4, PF 4, PF 5, PF 6, and RF 4 provide the temporary fronthaul connections. Aggregating all received traffic, the macrocell will be able to guarantee the minimum requirements to its users until the failure is repaired, regardless of how long the failure lasts. A heuristic algorithm for the detailed macrocell self-healing procedures can be found in [15].

**Simulation Model**

Simulations were carried out for one macrocell with a footprint of radius 500m in an urban area. Within the macrocell footprint there are six picocells, nine PFs, and 50 RFs, randomly distributed over the entire area. The fronthaul rate to the macrocell is 100 Gb/s, to the picocells is 10 Gb/s, to the PFs it is 5 Gb/s. Finally, to the RFs it is heterogeneous and distributed among the RFs as follow: 50 percent of RFs with rate of 200 Mb/s, 40 percent of RFs with rate of 500 Mb/s, and 10 percent of RFs with rate of 1 Gb/s.

The performance of our self-healing approach is evaluated in terms of degree of recovery (DoR) from failure. The DoR of a certain BS is defined as:

\[
\text{DoR} = \frac{\text{Sum of recovered rates from other cell sites}}{\text{Original input rate of the failed cell site}}
\]

where the original input rate of the failed cell site is the same as the input rate of that cell site, except for the macrocell because it fronthauls other picocells, so the actual input rate for the macrocell depends on the number of macrocell fronthaul connections.

**Simulation Results**

In this section we study the performance of our proposed COC algorithm by investigating the effect of increasing the number of SHRs on the DoR of the failed cell sites. Two scenarios are considered: single failure and multiple failures. First, we assess the DoR of the macrocell failure, which is the worst case and is considered as mul-
multiple failures. The DoR is evaluated with respect to the number of SHRs in both the macrocell and the picocells. Second, we assess the DoR of the RF in the case of single failure with respect to the number of SHRs in the RFs and picocells.

Figure 5 addresses the failure scenario of macrocell fronthaul. It shows the DoR of the macrocell when the number of SHRs of the macrocell and picocells is increased from 1 to 4. As can be seen clearly from the figure, as the macrocell SHRs increase, the DoR of femtocells increases. However, the number of SHRs in picocells has a slight or even no effect on the DoR of the failed femtocell. This shows that the DoR of femtocells is not directly dependent on the number of SHRs in picocells, but it is dependent on the number of cell sites located within the failed femtocell's SHRs coverage area, regardless of the cell site type.

It is clear that by using two SHRs in femtocells, we can recover up to 20 percent of the original rate of the failed femtocell fronthaul link. Using three SHRs in femtocells will recover approximately 40 percent. Further increasing the SHRs will not increase the DoR by a significant percentage.

The price of RRHs ranges from $2000 for a macrocell to $100 for a femtocell (this is an approximate market price taken from different vendors), and the price of a typical cellular transceiver is much less than these values. So adding more SHRs to macrocells or picocells will not dramatically increase the capital expenditure. For femtocells only one or two SHRs can be added to keep its price low. The operator will pay for the extra cost of the SHRs if the user agrees to be involved in the self-healing process. Otherwise, the user will purchase the regular femtocell (without SHRs). From the DoR results and the approximate cost stated above, it is recommended to equip each macrocell sector with three SHRs, and use four SHRs per picocell. The latter plays a crucial role in recovering the macrocell fronthaul failure. Using two SHRs per femtocell is sufficient to guarantee the minimum requirements in the presence of failures, as confirmed by the results.

CONCLUSIONS

The new technologies proposed for use by 5G networks are C-RAN, massive MIMO, mmW, SDWN, etc. Applying all these and other technologies to 5G will increase the fronthauling load. In this article we propose a scheme to guarantee a minimum throughput to 5G users in the presence of temporary or permanent fronthaul failures. A novel pre-planned reactive cell outage compensation approach is presented to mitigate the effects of fronthaul failure and its main concepts can be summarized as follows:

- A two-tier C-RAN architecture in which SCs use mmW band to connect their users, and macrocell uses traditional cellular bands for macrocell user access.
- SDWN implementation provides flexible network operation where the SDWNC, which is co-located in the cloud, monitors
and implements the self-healing procedures and acts as a database for the macrocell to avoid spectrum sensing during the CR phase.

SHRs are considered to be the main component in the COC process where SHRs are activated only in the case of fronthaul failure. Performance was evaluated using system simulation, and it shows that at least 20 percent of the fronthauling rate can be guaranteed during the fronthaul failure of any cell site type.

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