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A STUDY OF MOBILE VoIP PERFORMANCE IN WIRELESS BROADBAND NETWORKS

Doctoral Dissertation

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Dissertation for the degree of Doctor of Science in Technology to be presented with due permission of the Faculty of Information and Natural Sciences for public examination and debate in Auditorium AS1 at Helsinki University of Technology (Espoo, Finland) on the 12th of October, 2009, at 12 noon.

Helsinki University of Technology
Faculty of Information and Natural Sciences
Department of Computer Science and Engineering

Teknillinen korkeakoulu
Informaatio- ja luonnontieteiden tiedekunta
Tietotekniikan laitos

Distribution:
Helsinki University of Technology
Faculty of Information and Natural Sciences
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P.O. Box 5400
FI - 02015 TKK
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ISBN 978-952-248-102-3
ISBN 978-952-248-103-0 (PDF)
ISSN 1795-2239
ISSN 1795-4584 (PDF)
URL: http://lib.tkk.fi/Diss/2009/isbn9789522481030/

TKK-DISS-2650

Picaset Oy
Helsinki 2009

ABSTRACT OF DOCTORAL DISSERTATION	HELSINKI UNIVERSITY OF TECHNLOGY P.O. BOX 1000, FI-02015 TKK http://www.tkk.fi

Author Andres Arjona

Name of the dissertation
A Study of Mobile VoIP Performance in Wireless Broadband Networks

Manuscript submitted 8.6.2009	Manuscript revised 23.9.2009

Date of the defense 12.10.2009

☐ Monograph	☒ Article dissertation (summary + original articles)

Faculty	Information and Natural Sciences
Department	Computer Science and Engineering
Field of research	Telecommunications
Opponent(s)	Dr. Ryuji Wakikawa
Supervisor	Professor Antti Ylä-Jääski

Abstract

Voice service is to date still the killer mobile service and the main source for operator revenue for years to come. Additionally, voice service will evolve from circuit switched technologies towards packet based Voice over IP (VoIP). However, using VoIP over wireless networks different from 3GPP cellular technologies makes it also a disruptive technology in the traditional telecommunication sector.

The focus of this dissertation is on determining mobile VoIP performance in different wireless broadband systems with current state of the art networks, as well as the potential disruption to cellular operators when mobile VoIP is deployed over different access networks.

The research method is based on an empirical model. The model and experiments are well documented and based on industry standards for voice quality evaluation. The evaluation provides results from both experiments in a controlled laboratory setup as well as from live scenarios. The research scope is first, evaluate each network technology independently; second, investigate vertical handover mobility cases; third, determine other aspects directly affecting end user experience (e.g., call setup delay and battery lifetime).

The main contribution of this work is a systematic examination of mobile VoIP performance and end user experience. The research results point out the main challenges for achieving call toll quality, and how derive the required changes and technological performance roadmap for improved VoIP service. That is, investigate how the performance and usability of mobile VoIP can eventually be improved to be a suitable substitute for circuit switched voice. In addition, we evaluate the potential disruption to cellular operators that mobile VoIP brings when deployed over other access networks. This research extends the available knowledge from simulations and provides an insight into actual end user experience, as well as the challenges of using embedded clients in handheld devices. In addition, we find several issues that are not visible or accounted for in simulations in regard to network parameters, required retransmissions and decreased battery lifetime.

The conclusion is that although the network performance of several wireless networks is good enough for near toll quality voice in static scenarios, there are still a number of problems which make it currently unfeasible to use as a primary voice service. Moreover, under mobility scenarios performance is degraded. Finally, there are other issues apart from network performance such as energy consumption, hardware limitations and lack of supporting business models (e.g., for WiFi mesh) that further limit the possibility of rolling out mobile VoIP services.

Keywords mobile VoIP, voice quality, performance, end user experience, wireless broadband

ISBN (printed) 978-952-248-102-3	ISSN (printed) 1795-2239
ISBN (pdf) 978-952-248-103-0 (PDF)	ISSN (pdf) 1795-4584 (PDF)
Language English	Number of pages 71 p. + app.

Publisher Department of Computer Science and Engineering

Print distribution Department of Computer Science and Engineering

☒ The dissertation can be read at http://lib.tkk.fi/Diss/2009/isbn9789522481030

Preface

I would like to express my gratitude to my supervisor, Professor Antti Ylä-Jääski, for his guidance during my post-graduate studies and during the preparation of this dissertation. I also wish to thank Professor Petri Mähönen at RWTH Aachen University and Professor Tapani Ristaniemi at University of Jyväskylä for acting as external reviewers of the dissertation.

The research work of this dissertation was carried out with Nokia Networks and Nokia Siemens Networks. I would like to acknowledge all the colleagues that I have had the pleasure of working with during those years. I thank my line managers Nitin Arora, Erkka Ala-Tauriala, Jukka Jokinen, Kari Skog, Antti Vuorinen and Subir Saha for supporting my research. In particular, I thank my colleagues Martin Kristensson, Harri Holma, Cedric Westphal, Jan Berglund, Zheng Li and Outi Hiironniemi for their mentoring, continuous support, guidance and challenging feedback. Additionally, I thank Renaud Cuny, Markku Vimpari, Markus Nyberg, Leo Bhebhe, and the people at the E2E research lab in Säterinportti, which whom I had the pleasure of conducting joint research. Moreover, I am grateful to Kwang-Rok Chang and Tim Cant at NSN Japan for providing me the chance to further pursue new challenges in their organization.

Likewise, I thank Dr. Ryuji Wakikawa at Toyota Information Technology Center, USA, Dr. Chittaranjan Hota, at BITS Pilani, India, Dr. Rod Van Meter at KEIO University, Japan, and Dr. Hyroyuki Morikawa, Dr, Hideaki Imaizumi, and Dr. Shunsuke Saruwatari at Tokyo University, Japan, for giving me the opportunity to present my work in their campuses. Further, I thank Dr. Takasaki at Toyo University, Japan for encouraging me to extend my work, and Dr. Manner at HUT for his support.

I am grateful to many anonymous experts working in the mobile communications business, who have provided me with not only challenging research questions but also insights and feedback on the research results both in blind reviews and International conferences. In particular, I thank Dr. Attahiru Alfa at University of Manitoba, Canada, and the researchers at Magister Solutions.

Carrying out a PhD while working fulltime would not have been possible without support from my good friends Thomas Sandvik, Sakari Laitinen, Hannu Verkasalo, Juha Sørensen, Aki Koskinen, Mika Sundvall, Matti Dahlbom, Kimmo Kokkila, Jaku Avikainen, Sebastian Trzaska, Juha Kupiainen and Baris Boyvat in Finland, and Takashi Okuri and Hiroshi Yanagida in Japan. I thank my family, who although they are remotely located in two different continents from me, have still always given me their full support. Finally, I express my deepest gratitude to Hitomi, for all the happiness and who did not escape even at my most stressful times.

Andres Arjona
Helsinki, Finland, September 2009

Contents

Original Publications

This thesis consists of an overview of the following publications which are referred to in the text by their Roman numerals.

I A. Arjona, C. Westphal, J. Manner, A. Ylä-Jääski, and S. Takala, "Can the Current Generation of Wireless Mesh Networks Compete with Cellular Networks?" Elsevier Computer Communications Journal, Vol. 31, Issue 8, May 25 2008. ISSN 0140-3664, pp. 1564-1578, Elsevier.

II A. Arjona, A. Ylä-Jääski, C. Westphal, M. Kristensson, and J. Manner "Towards High Quality VoIP in 3G Networks: An Empirical Approach", International Journal of Communications, Networks and System Sciences, Vol. 1, No. 4, October 2008. ISSN 1913-3715, pp. 348-359, SciRes.

III A. Arjona, and A. Ylä-Jääski, "Mobile IP as an Enabling Technology for VoIP in Wireless Mesh Networks", In proceedings of IEEE 67th Vehicular Technology Conference, VTC'08-Spring, Marina Bay Singapore, May 11-14, 2008, pp. 2769-2773, IEEE.

IV A. Arjona, A. Ylä-Jääski, and J. Kerttula, "Live Network Performance Challenge: FLASH-OFDMA vs. HSDPA", In proceedings of IEEE 22nd International Conference of Advanced Information Networking and Applications AINA'08, Okinawa Japan, March 25-28, 2008, pp. 918-925, IEEE.

V A. Arjona and A. Ylä-Jääski, "VoIP Call Signaling Performance and Always-On Battery Consumption in HSDPA, WCDMA and WiFi", In proceedings of IEEE International Conference on Wireless Communications, Networking ad Mobile Computing WiCOM'07, Shanghai China, September 21-23, 2007, pp. 2964-2967, IEEE.

VI A. Arjona and H. Verkasalo, "Unlicensed Mobile Access (UMA) – Handover and Packet Data Performance Analysis", In proceedings of IEEE International Conference of Digital Telecommunications ICDT'07, Silicon Valley USA, July 1-6, 2007, 6 pages, IEEE.

The following publications are also related to the topical area but are not included as part of the dissertation for the following reasons: Papers VII, VIII, and IX were extended and included as part of papers I and II. As for Paper X, it reports user experience in mobility scenarios between 2G and 3G. However as 2G is a narrowband radio access it is not directly comparable to the rest of the results in Papers I-VI.

VII A. Arjona, C. Westphal and S. Takala, “Empirical Analysis of the Single-Radio Mesh Architecture Performance, Limitations, and Challenges”, In proceedings of ACM International Wireless Communications and Mobile Computing Conference IWCMC’07, Hawaii USA, August 12-16, 2007, pp. 541-546, ACM.

VIII A. Arjona and S. Takala, “The Google Muni WiFi Network – Can it Compete with Cellular Voice?”, In proceedings of IEEE Advanced International Telecommunications Conference AICT’07, Morne Mauritius, May 13-19, 2007, 6 pages, IEEE.

IX A. Arjona, A. Ylä-Jääski, C. Westphal, M. Kristensson, “Towards High Quality VoIP in 3G Networks: An Empirical Study”, In proceedings of IEEE Advanced International Conference on Telecommunications AICT’08, Athens Greece, June 8-13, 2008, pp. 143-150, IEEE.

X L. Bhebhe, and A. Arjona “Data Outage Across 2G& 3G Wireless Networks”, In proceedings of IEEE World of Wireless Mobile and Multimedia Networks WoWMoM’08, New Port Beach CA, June 23-27, 2008, 6 pages, IEEE.

Author's contribution

None of the mentioned publications I – VI have previously formed part of another thesis. The main results of the thesis and the author’s contributions can be summarized as follows. The first author planned, executed, performed the analysis and reported the test campaigns for all the included publications (Papers I-VI). The other authors advised on the text and provided some corrections.

Publication I. WiFi mesh networks network performance is evaluated in this paper. The study is based on the performance of a representative single-radio mesh network both in a live setup and in a laboratory environment. The study characterizes the performance of different applications including VoIP. Further, the main key challenges of mesh networks such as the fairness in bandwidth allocation and hidden node terminal are also considered. Subsequently, the main key challenges of the WiFi mesh network architecture in regards to performance are characterized. Particularly, the fairness in bandwidth allocation is described in detail. We show that issues in unfairness arise quickly, and that, even in the laboratory, the performance proves disappointing. Finally, the results of the study are compared with traditional cellular networks, and various options to enhance the performance of wireless mesh networks in the future are discussed.

Publication II. This paper studies the performance of VoIP in 3G packet switched networks (WCDMA and HSDPA). When introducing HSDPA in 3G networks the end user experience and systems capacity with VoIP applications can improve considerably. In this paper the performance of VoIP is evaluated via empirical data in both laboratory and live environments. The evaluation includes both static and mobile scenarios as well as different voice codecs. The focus is on determining the main factors limiting VoIP performance, as well as understand in which ways it can be improved.

Publication III. In this paper, the Mobile IP (MIP) technology is evaluated as a potential interworking solution between WiFi and 3G access networks. WiFi networks, and particularly those deployed outdoors often lack the sufficient coverage required in order to provide ubiquitous access and mobility for voice services. However, by the use of Mobile IP, it is possible to roam to other networks with broader coverage once WiFi coverage is unavailable or suffers from coverage gaps. The focus of the evaluation is on the feasibility of using VoIP services over WiFi mesh networks and interwork with 3G networks.

Publication IV. In this paper, a disruptive wireless broadband technology known as FLASH-OFDM is evaluated. FLASH-OFDM is another broadband option operating on licensed spectrum. However, if operated at considerably lower frequencies than 3G, it is particularly interesting for emerging markets, especially rural areas lacking telecommunications infrastructure. In contrast to the 3G standards, FLASH-OFDM standards are proprietary. This paper provides an evaluation based on quantitative measurements of the actual performance of two state-of-the-art live networks in Finland (HSDPA and FLASH-OFDM). The evaluation includes metrics such as throughput, delay and VoIP quality considering performance both on static and mobile scenarios.

Publication V. This paper studies the main signaling delays that take place in a VoIP call (registration and voice call setup delays) and compare them for different wireless accesses: HSDPA, WCDMA and WiFi. In addition, possible optimizations with always-on mode and their drawbacks are presented. In particular, signaling delays and the total battery lifetime affect the perceived user experience directly and thus are of relevance in addition to the overall voice quality.

Publication VI. The performance of Unlicensed Mobile Access (UMA) interworking solution is considered in this article. The paper discusses the UMA technology in detail, after which technical performance measures of the technology are considered. The paper sets out to technically address whether UMA could work in real life situations. The focus is put on the GSM to WiFi (and vice versa) handovers and packet data performance compared to current cellular networks.

List of Abbreviations

2G	2^{nd} Generation cellular telecom network
3G	3^{rd} Generation cellular telecom network
3GPP	3^{rd} Generation Partnership Project
AP	Access Point
ADSL	Asymmetrical Digital Subscriber Line
DECT	Digital Enhanced Cordless Telecommunications
DSL	Digital Subscriber Line
DMB	Digital Multimedia Broadcasting
EDGE	Enhanced Data rates for Global Evolution
FLASH-OFDM	Fast Low-latency Access with Seamless Handoff OFDM
FTP	File Transfer Protocol
GAN	Generic Access Network
GHz	Giga Hertz
GSM	Global System for Mobile communication
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HSUPA	High Speed Uplink Packet Access
HTTP	Hyper Text Transfer Protocol
IP	Internet Protocol
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IMS	Internet Multimedia System
ITU	International Telecommunication Union
kbps	kilobits per second
LTE	Long Term Evolution
Mbps	Megabits per second
MHz	Mega Hertz
MIMO	Multiple Input Multiple Output
MIP	Mobile IP
MOS	Mean Opinion Score
NAT	Network Address Translator
OFDM	Orthogonal Frequency Division Multiplexing
PESQ	Perceptual Evaluation of Speech Quality
PSTN	Public Switched Telephone Network
RoHC	Robust Header Compressions
SIP	Session Initiation Protocol
SR-VCC	Single Radio Voice Call Continuity
UMA	Unlicensed Mobile Access
UNC	UMA Network Controller
VCC	Voice Call Continuity
VoIP	Voice over IP
VoLGA	Voice over LTE via Generic Access
VPN	Virtual Private Network
WCDMA	Wideband Code Division Multiple Access
WiFi	Wireless Fidelity
WiMAX	Worldwide Interoperability for Microwave Access
WPA	WiFi Protected Access

1 Introduction

There will be three certain things driving the future of mobile voice and eventually its evolution towards voice over IP. First of all, there will be a need for a voice communications service. No substitute services could replace the voice service as we know it today, and end users will have to communicate through voice also in the future. Second, voice services will gradually migrate to the mobile world. That is, the fixed to mobile convergence will evolve further. Already today a large number of voice calls in mature telecommunication markets take place in cellular networks rather than the old PSTN networks. Third, voice technologies will be deployed through packet switched technologies in the future. Circuit switched technologies will be replaced eventually by Internet protocols. This movement to Internet technologies will take place in the mobile industry too.

Voice is still the killer mobile service, and most of the revenue of mobile service operators comes currently from circuit-switched mobile voice. However, the evolution of mobile networks has for a long time evolved towards packet-switched technologies, and today's new mobile devices have typically a possibility for packet-switched access. Consequently various kinds of data services from mobile Web browsing to email have emerged in the mobile domain as well (Verkasalo 2007b). The technologies have also been there for mobile packet-switched voice (mobile VoIP) although no services or players have yet emerged on the market seriously utilizing mobile VoIP as a primary voice service.

The motivation to study VoIP performance in wireless broadband networks can be supported by three key trends that will inevitably take place in the mobile service domain, the only uncertainty being the timeline of evolution (i.e. the pace at which the trends become reality). The key trends are:

- There will be a non-decreasing demand for mobile voice services
- Fixed-to-mobile substitution will evolve further
- Packet-switched connectivity and service platforms will emerge

Based on these trends, voice services deployed over Internet technologies in wireless networks will certainly be there in the future. The biggest uncertainties regarding mobile VoIP currently include the business models with which to commercialize the new services and the players who will produce and provision the services.

The current telecom world is characterized by strong vertical orientation in which operators commonly run both the network and services (Vesa 2005). Some phenomena such as handset subsidies directly originate from such closed business models, in which incumbent operators in many cases control the whole value chain. Although this in many cases results in fast ramp up of network infrastructure, open service innovation suffers because of the closed "walled garden" business models.

The world of Internet is much different. This is because of two reasons. First, the end-to-end connectivity results in application-level development in which lower level connectivity issues are irrelevant. Mainly because of this, the services of the Internet are provisioned by totally different companies from the ones who provide the mere connectivity (e.g., DSL operators). Pricing reflects this. DSL operators have typically flat-rate fees, and service-level actors typically explore other sources of revenue e.g., advertising or add-on service based models with innovative business logic. Second, the Internet services have evolved this fast mainly because of network-edge based innovation. Open standards and value networks are much faster in inducing service innovation than the old-fashioned “walled gardens” of incumbent telecom operators.

The potential disruption resulting from the clash of the Internet and mobile telecom world is inevitable (Saarikoski 2006, Funk 2004). Not only do individual companies face a new context in which to operate, but actually the whole business ecosystem faces shocks that might lead to new ways of doing business and generating end-user added value (Verkasalo 2007a). Also mobile businesses should keep an eye on external factors and position themselves either with new innovative value propositions (focus or differentiation strategy) with mobile VoIP, or then move to simpler business models (cost leadership) deploying e.g., bit-pipe strategies (Porter 1985).

The mobile VoIP business presents an interesting playground as both incumbent telecom operators and challenger Internet players have a technical opportunity to leverage IP-based voice. Technically the uncertainty factors are low, and much more interest should be targeted at the potential business impact. Incumbent telecom operators might face significant threats if they do not proactively take account of the technical, business strategic and regulatory trends taking place in the market. The new playground resulting from a technically different service architecture (the closed “walled garden” of the telecom world vs. the OSI-layered Internet model) provides room for technical disruption (for disruptive innovation, see Rogers 2003).

While the outcomes of the mobile VoIP technology might reside far away in the future, business-wise, today’s research should be aligned to investigate how the performance and usability of mobile VoIP can be improved to eventually end up being a perfect substitute for circuit-switched mobile voice. It is a fact that voice will move to the packet-switched domain; the only uncertainties are simply how mature the service is now and how close is mobile VoIP to challenging incumbent service technologies and businesses in the near future.

1.1 Research Problem

In this thesis we focus on determining the performance of Mobile VoIP in different wireless broadband systems with current state of the art networks. The main aspects that we consider for the evaluation are the following: First, the network technology should be able to provide call toll quality voice. Second, the performance of VoIP should be consistent for both static and mobile scenarios. Third, the network technology should provide adequate coverage. Fourth, the network deployment should be commercially feasible and support a valid business case. Fifth, the mobile VoIP service as a whole should be comparable to traditional circuit switched voice. This last requirement in particular brings up additional issues that affect mobile VoIP. For example, audio breaks during mobility, call set-up times and battery lifetime of the devices should be comparable to those of circuit switched voice systems. All of these aspects together lead to the following research questions:

1) Is VoIP over wireless broadband commercially feasible with current state of the art networks?
2) Is VoIP over wireless broadband networks a real contender to cellular circuit switched voice?
3) What are the main technical limitations for high quality VoIP in wireless broadband networks?
4) What is the mobility performance of multi-radio interworking solutions for VoIP services?
5) What is the VoIP signaling performance and battery lifetime when using VoIP with handsets?

Considering the large scope of the research question, we limit the scope of the thesis to three main areas of research that are of importance in order to properly evaluate the performance of VoIP in wireless broadband networks (see Table 1). The first area of research studies each wireless broadband network technology independently. The network technologies selected were based on the latest available release and had an actual commercial deployment available. A commercial deployment is important in order to differentiate the performance in a laboratory environment with that of a real life use case scenario. The study consisted mainly in determining the VoIP performance in static and mobile scenarios. Since the networks were studied independently, the mobile cases are restricted to horizontal handovers. The second area of research investigates mobility cases during vertical handovers**.** Vertical handovers are needed in order to support service continuity across different network technologies (e.g., HSDPA and WiFi). Finally, the third area of research aimed at finding other aspects of performance that can affect user experience. The focus was in regard to signaling performance and the time it takes to set up calls as well as the energy consumed and battery lifetime when using VoIP services. These items are relevant in addition to voice quality since any serious contending technology needs to provide a similar user experience to that of circuit switched voice.

Table 1. Scope of the Thesis

Wireless Broadband	Intersystem Mobility	Other Factors Affecting Quality of Experience
• WiFi Mesh • 3G → 4G Evolution WCDMA (Rel. 99) HSDPA (Rel. 5) HSUPA (Rel. 6) HSPA+ (Rel. 7) • FLASH-OFDM	• UMA • MIPv4	• Energy Consumption • Battery Lifetime • Signaling Performance

1.2 Contributions of the Thesis

Mobile VoIP is a disruptive technology in the telecommunication sector and thus it requires to be further understood. This thesis work considers mobile VoIP quality and performance in wireless broadband networks with an end-to-end approach. By end-to-end it is meant that the application data will traverse all the necessary network elements that it would do in a real life deployment. By doing so it is possible to understand the big picture, actual end user experience and performance. The overall picture is of great relevance in the industry given that it has an impact in regard to business decisions and future developments.

In addition, the thesis considers both laboratory experiments and live use case scenarios. In the laboratory experiments, all the network elements were owned by the organization funding this work and were allocated for use only for the purpose of this research during the tests. These types of experiments are complex and expensive in nature and therefore not easily carried out outside an industrial environment. The set of results provide a substantial contribution with regard to network performance including all the necessary elements end-to-end and under different variables. The assumptions for the tests as well as all configurations in the network are well documented, making the results reproducible.

Moreover, the live use case scenarios give an additional perspective in which the network is not allocated only for our research. The additional traffic and differences in equipment provide an insight regarding the differences between the optimal setup that we had in the laboratory and the real life scenario. With the understanding resulting from our lab experimentation we can correlate the results from the live use cases. Figure 1 shows the area in which each of the papers contributed to VoIP performance and end user quality of experience.

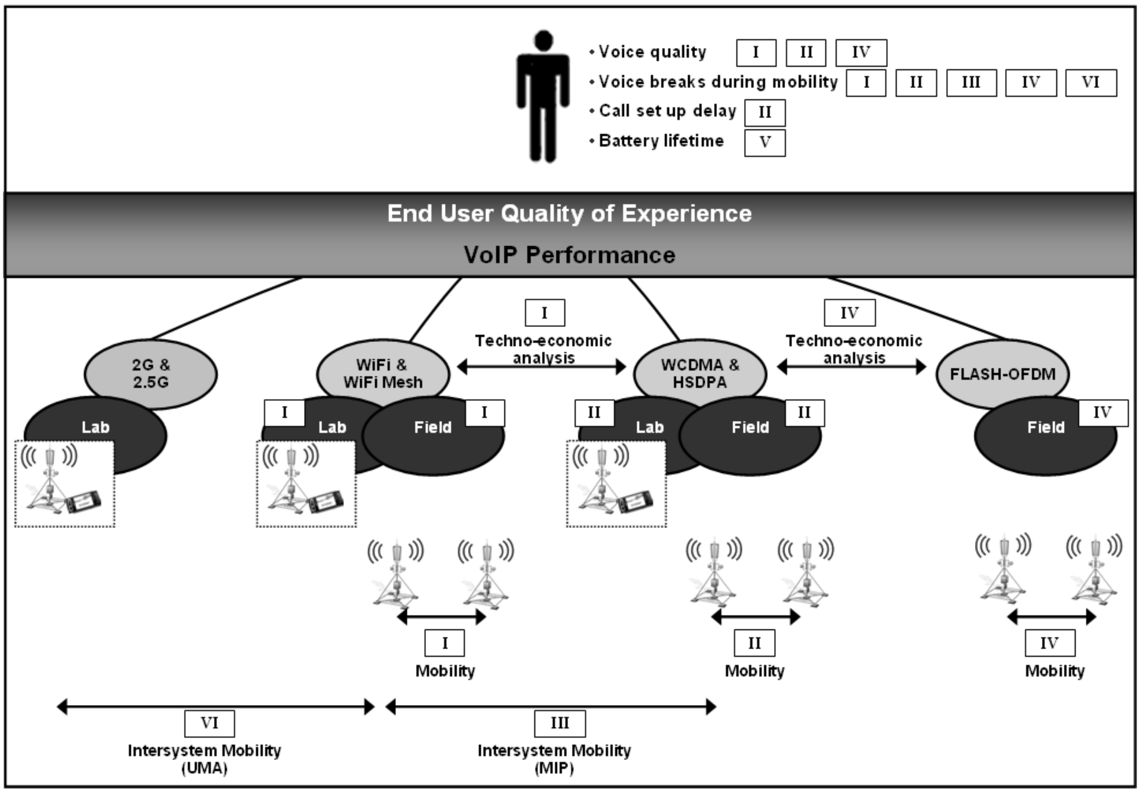

Figure 1. Article Contributions

Although simulations can provide an insight into the potential of the technology, unless end-to-end performance is evaluated, it lacks the necessary proof to consider simulation**s** results as the only means for business decisions. Likewise, the assumptions from simulations as well as results tend to differ one from another. The reason is that regardless of how much scrutiny is paid to the models, there will always be actual equipment behavior and details that cannot be considered. Simulation work needs to be extended with end-to-end performance to give the actual insight into the overall performance.

This study extends the available knowledge due to the following reasons. The majority of research related to VoIP is heavily based on simulations. Although the simulations are useful, they do not always provide a realistic view of the actual performance in the field. The reason is that the deployed equipment will always function differently to some extent. In addition, much of the available material deals only with older 3GPP specifications. Therefore, the performance is based on a different type of network. In our study we will study the actual performance of three different types of networks using the same research methods and tools. Furthermore, in cases like WiFi mesh and FLASH-OFDM, our study provides a first handful of results and performance for these types of networks.

In regard to the different interworking solutions, there are no studies available about UMA performance. As for Mobile IP, the available simulation results are not in line with each other. This thesis provides the first actual VoIP performance measurements for both technologies in a real life scenario.

Finally, for energy consumption and signaling performance the available literature is also not complete. Although there is research in regard to reducing energy in VoIP applications, it has not been emphasized to what extent having always-on connectivity can degrade the overall battery lifetime. This is a very important matter because VoIP is not the only application that uses always-on keep alive messages. Further, overall battery lifetime directly affects end user experience and perception of the service.

The main contribution details are the following: For 3GPP access networks we found some of the most important reasons for poor performance that are not visible from simulations. One of them related to the RLC retransmissions. These retransmissions can create very large delay peaks, around 200ms in static scenarios and even larger in mobility cases. These were pointed out clearly in paper II. In addition, the issues of embedded VoIP client implementation and processing delay were also identified. This, for example, shows us that although Robust Header Compression (RoHC) is important for operators since it can increase capacity significantly, it is not yet feasible. The main reason is that header compression would incur additional client processing delay. Based on this information, we have given estimations for performance improvement in Section 4.2. This kind of analysis extends the available research found in simulations by giving an end user experience view of the overall voice service. This information also aids the operator strategy in regard to mobile VoIP. Since it is possible to understand the overall user experience with this empirical data, we can also understand the potential service quality that VoIP based competing services such as Skype can provide. This way, operators can treat competing services differently and apply a lower quality of service if needed. In addition we provided first hand measurements for MIP, UMA and call setup delay. This latter item allowed us to understand energy consumption in much more detail and identify some points that need to be taken care of; especially network configuration and the importance of increasing CELL_FACH packet size in future releases. In addition, this understanding of energy consumption gave insight into what other things can be thought of in order to increase overall battery lifetime in future research. For instance future research should focus on designing NATs and VPN gateways with different timeout values, and software developers can try to design applications in a way that keep-alive messages are sent simultaneously, or by using a client proxy. With regard to WiFi mesh, we proved that it is not such a disruption for the telecomm industry as was originally thought. When WiFi mesh proposals started, it was seen as a great potential threat. However due to the cost, complexity, and poor performance, this has proven not to be the case. Finally, our analysis of FLASH-OFDM gave us insight into this kind of competing network. Our conclusion showed that this technology is no longer a direct competitor although it was only a few years back. Instead it is a complementary type of network that can provide service in scenarios where 3GPP based networks currently lack the required performance, such as high speed mobility (above 200 km/h).

1.3 Research Process and Methods

This thesis is based on an empirical model. The model and experiments are well documented and based on industry standards for voice quality evaluation. The evaluation provides both results from experiments: (1) in a controlled laboratory setup environment, as well as a correlation of the results with those from (2) live case scenarios in the field.

The chronological line of research in the telecommunications industry is complex and involves multiple steps and iterations (see Figure 2). At the top or initial part we have the standards, which are the basis for products in the telecomm industry. The standards take a considerable time to develop, especially standards such as 3GPP or IEEE. During the development of the standards, it is very common to carry out simulations to evaluate the performance of different approaches and proposals. In this phase proposals are evaluated and with the aid of simulation results, decisions are agreed.

In addition, each of the standards releases has a well defined scope for what is being standardized and which elements are to be considered as different releases or outside the normal scope of these standard releases.

Standards also leave a significant amount of details on implementation open to the different vendors. Already at the standardization and research and development phases, empirical research is required for testing new technologies. Once the standard release is frozen, companies start developing products for that release. Subsequently, when the product is developed and implemented, empirical research comes into play. This research plays a critical role in this phase. Products need to, first of all, be verified and prototyped. As companies, however, develop their products they always realize that the results of simulations differ with the actual empirical data. The reason is simple; simulations cannot account for all the factors involved in the actual product. Simulations are based on assumptions which attempt to be as close to reality as possible. However, simulations alone cannot be used as a basis of performance. During the empirical research, faults, issues with the approaches, additional factors and limitations are encountered. The results from the empirical research also provide an insight into what the performance will really be like in the next few years of the product life. This data is crucial to update and even refocus the strategic intent if necessary. The results of the empirical research also generate many ideas for improvement of the product, develop new algorithms, prioritize tasks, redefine further simulations with more realistic assumptions as well as define new requirements for the future product and standardization releases.

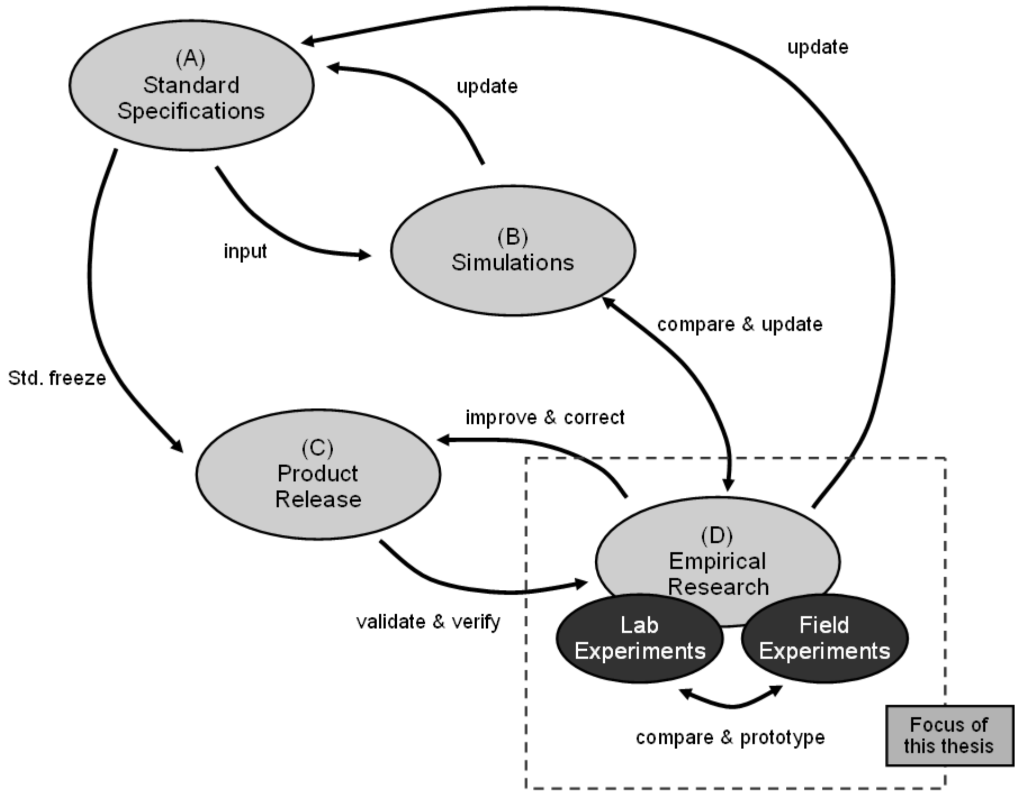

Figure 2. Research Cycle in the Telecommunications Industry

A quite common timeframe between standard freeze and product commercial launch is at least one year, plus an additional year and a half to have terminals ready, and at least two years more to have considerable user penetration. That means that from standard freeze to having a meaningful amount of devices in the networks takes over four years at least. Further, the lifespan of the previous technologies can in many cases slow down the device penetration for several years more (see Figure 3). Taking Long Term Evolution (LTE) technology as an example; the standard was to have been finalized by the end of 2008. However it was not until Spring 2009 that vendors started developing the real products. The reason is that this was the time in which the standard was agreed for interoperability tests with no further changes being made. From that moment on, vendors raced each other to develop their products and decide which features to implement for each hardware and software release. Nevertheless, despite some LTE networks beginning to become available in 2011 (e.g., Verizon in USA), the coverage area will be limited and major coverage is not expected until a few years later. In addition, there will not be availability of devices before 2011. The reason being that it takes at least a year and a half from the moment the chipsets are available to the moment mass market handsets are available. Furthermore, having available handsets does not imply anything for user penetration. Increasing the user penetration requires time as well. Therefore, we can expect to have some countries with LTE handsets and well deployed networks no sooner than 2011-2012 and, a significant amount of users by 2013-2014.

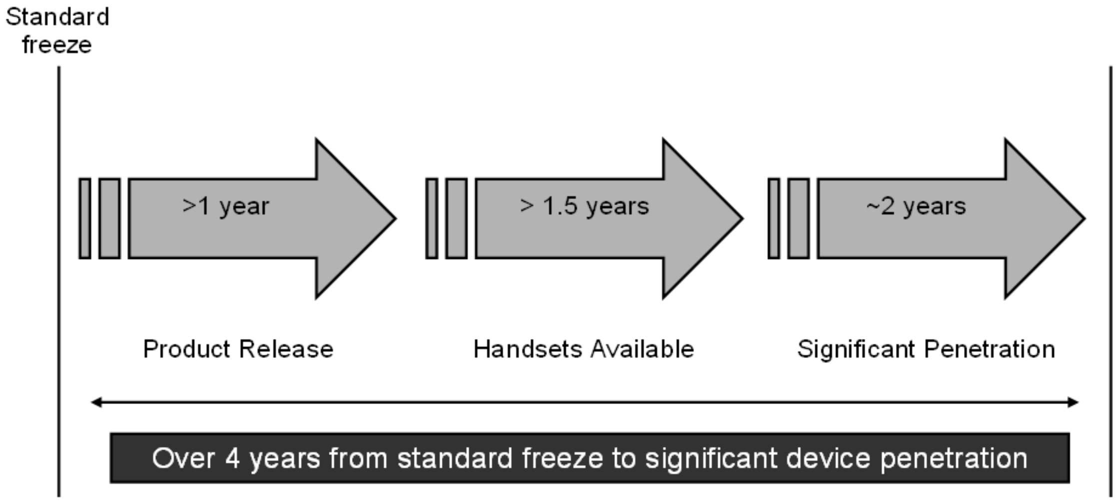

Figure 3. Technology Timeframe

Although in many cases disruptive technologies can be evaluated prior to their release, it is not always the case. Therefore, disruptions can arise already when the products have been developed. Likewise, competitors and new entrants often generate new business models and approaches that can affect the current market. For this reason, empirical research is also triggered due to new technologies, entrants and competitors. It is common for companies to carry out empirical evaluations of their own products against competition, evaluate the technology itself, carry out additional research, improve products, and similarly generate input for consideration in their technology strategies and roadmaps (see Figure 4).

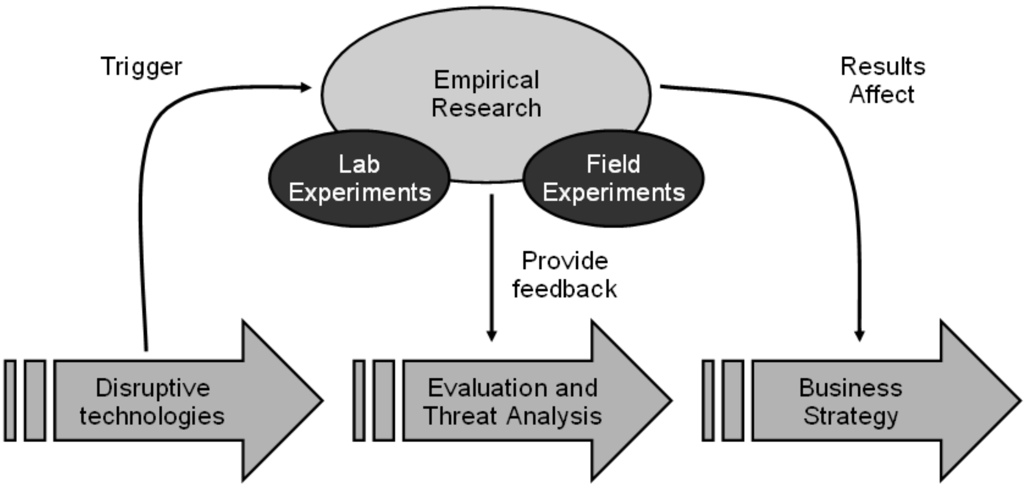

Figure 4. Additional Triggers for Empirical Evaluation

As we can see, empirical research is of great importance for the telecommunications industry, it assists in creating competitive advantages and developing business decisions and technological strategies. Moreover, especially in the context of this thesis, the final end user experience and perception of the voice service is crucial, since voice service is, and will remain, the main source of revenue for telecomm operators for years to come (Roberts & Sims 2008). Despite the massive increase of data traffic in operators' networks, it is unlikely to surpass voice traffic prior to 2011 (see Figure 5).

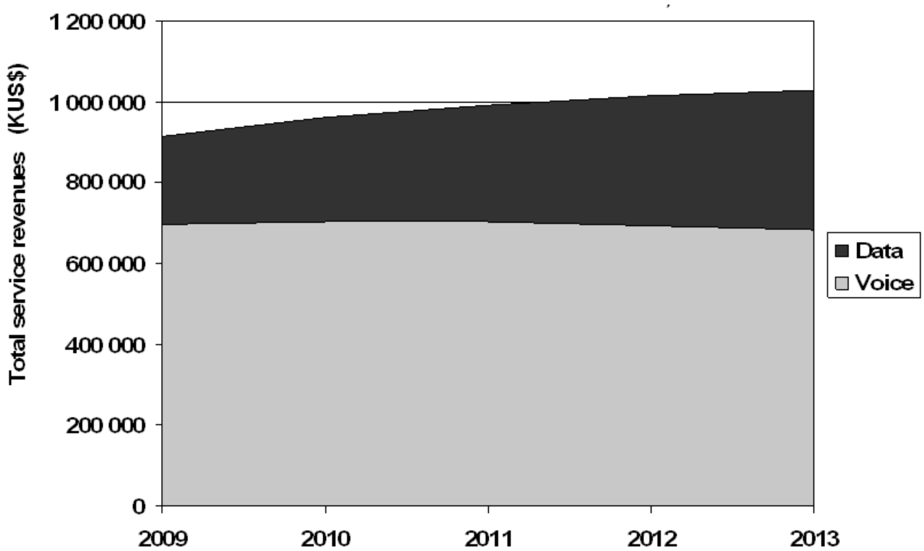

Figure 5. Operator Global Revenue Forecast (source: Informa)

1.4 VoIP Quality Model

In this study we focus on ITU recommendations in order to evaluate voice quality. ITU has several standards. One standard is set to define user satisfaction metrics. The model is based on multiple iterations of voice tests listened to by a large number of users. Based on their opinions a Mean Opinion Score (MOS) metric was defined. Moreover, ITU also has defined different methods by which voice quality can be measured and matched to the original MOS values. The main methods are described in this section.

1.4.1 Subjective Evaluation Methods

There are two ways to evaluate voice over IP, subjectively and objectively. A subjective evaluation is carried out into the way actual audio is transmitted between ends, and a group of people provide an opinion on what they believe is the quality. In contrast, objective evaluation is based on the measurements of key metrics of the transmitted data.

Subjective evaluations provide the closest look at actual end user experience. Since users actually listen to audio samples they can provide their own personal opinion. In general, subjective evaluations are very useful and are required to some extent even if quality has been evaluated with objective methods. However, carrying out subjective tests is not straightforward. Users must be carefully selected in order to provide a real idea of what the end user might be. Likewise, because humans make the grading in subjective tests, the repeatability is very difficult. If a subjective model is to be followed, ITU P.800 provides recommendations for carrying out this kind of evaluation. (ITU-T 1996; Siironen2004)

1.4.2 Objective Evaluation Methods

Objective evaluation methods are based on measurements of key metrics of the transmitted data. Although they do not involve individuals, they are very useful because of their repeatability. The main two objective evaluation methods for VoIP have been developed by ITU and are the following:

The E-Model (ITU-T 2003) is a voice quality evaluation model that is based on network performance metrics. It is based on a mathematical algorithm and provides an “R” performance value based on the sum of four “impairment factors” considered to be cumulative. The algorithm is depicted in equation (1) where, “Is” is voice impairments to the signal, “Id” is delay (ms), “Ief” is packet loss impairment, and “A” is the expectation factor.

$$R = 100 - Is - Id - Ief + A \qquad (1)$$

In practice, ITU-T proposes to use a simplified version of this algorithm. The simplified algorithm considers that noise cancellation is encountered in the network and also dismisses the expectation factor. The expectation variable is intended to provide a balance for some environments in which the user expects a degraded quality, such as satellite connections. However, since this variable is merely subjective it is recommended to ignore it. The simplified algorithm is depicted in equation (2).

$$R = 93.2 - Id - Ief \qquad (2)$$

The R value can be associated with MOS values, which is a subjective grade for voice quality based on studies also carried out by ITU-T. However, even though the R-value can match a MOS value, it cannot predict the absolute opinion of an individual user. To date, the E-Model is the most reliable objective evaluation method for VoIP. The association of R values to MOS is depicted in Table 2.

Table 2. The R-value and the Mean Opinion Score

R-Value	Mean Opinion Score (MOS)	User Satisfaction
90 or higher	4.34 or higher	All users very satisfied
80 or higher	4.03 or higher	All users satisfied
70 or higher	3.60 or higher	Some users dissatisfied
60 or higher	3.10 or higher	Many users dissatisfied
50 or higher	2.58 or higher	Nearly all users dissatisfied

The Perceptual Evaluation of Speech Quality or the PESQ (ITU-T 2001) method has been widely used in circuit switched based communication systems for a long time. In circuit switched systems, the network characteristics are usually constant and therefore a direct comparison of the transmitted audio sample against the received audio sample is convenient.

While this method is well suited for circuit switched communications and systems in which delay is small and constant, it is not suitable for transmission systems with delay variations such as wireless networks. The reason is that the PESQ model does not consider delay. This can be proven with a simple example. Consider a network with a delay of two seconds. Even if the audio is perfectly transmitted from the calling end to the receiving end, the delay by itself makes carrying out a conversation nearly impossible.

The PESQ model is however useful to benchmark the quality of different codecs. With this method it is possible to compare the quality of the same audio sample directly. It is done by storing the original audio sample stored in a lossless file type (e.g., WAV). Subsequently the sample can be transcoded with different codecs and compared directly with the original. This kind of evaluation will show the degradation of the sample after coding and decoding takes place.

In summary, PESQ can provide audio quality, but not necessarily voice or conversation quality. The reason is that the model does not consider delay. Delay is a critical aspect in voice and particularly in conversations. Once delay becomes too large, conversation is not interactive any longer. Moreover, in wireless systems, delay is variable and one of the main aspects affecting voice quality, which makes this method unfeasible.

1.4.3 Codecs for VoIP

There are multiple codecs that can be used to compress voice audio. Each codec provides different maximum voice quality and bitrate required. However, in case of VoIP communications over wireless networks only a handful is relevant for this study. Firstly, the G.711 and G.729 codecs developed by ITU. The former is a high bitrate codec (64kbps) and the latter is a low bitrate codec (8kbps). Both codecs are extensively deployed and supported. In real life most VoIP calls are usually encoded in one of these two codecs. Despite the wide support for these two codecs, the G.711 codec in particular is not well suited for wireless systems. The reason is that it requires a large amount of bandwidth compared to other codecs. Thus, it is recommended to avoid using this codec in wireless systems. Secondly, there are two other low bitrate codecs that are important for VoIP, namely AMR and iLBC. The AMR codec was developed by wireless network vendors and aims at increasing capacity in cellular systems. The AMR codec is widely supported by some of the wireless network manufacturers, but since it is patented it is not fully compatible with other vendor’s equipment. Therefore, in practice this codec will be used between cellular operators, but not necessarily between calls to other systems. In contrast, the iLBC codec was developed by the Internet community IETF. This codec is mainly used in Skype VoIP software for Skype-to-Skype calls. However, it is not as widely supported as the ITU counterparts.

In this thesis we focus mainly on the performance of the G.729 codec. ITU-T has a standardized maximum voice quality for this codec. G.729 is also similar to AMR, which is the main future codec for 3GPP based networks. Therefore, we can make the assumption that the performance values measured with G.729 codec are representative and are a useful basis for our analysis.

1.5 Structure of the Thesis

The remainder of this dissertation is organized as follow. In Section 2 we provide an overview of the wireless broadband and interworking technologies considered in this thesis. Next, in Section 3 the related works are reviewed. Subsequently in Section 4, the summary of the results is given. Finally, Section 5 presents the conclusions, generalization of the results and proposals for future work.

2 Background

Wireless broadband networks are networks able to transport packet based data at high bitrates. It is arguable what is the exact bitrate required to consider a connection as high speed depending on the definition of broadband. However, a simple way is to consider ADSL type of connections as a baseline for broadband. Taking that as a baseline, the bare minimum bitrate to be considered as broadband is 500kbps. Figure 6 shows an overview of the most relevant wireless radio technologies for consumer applications.

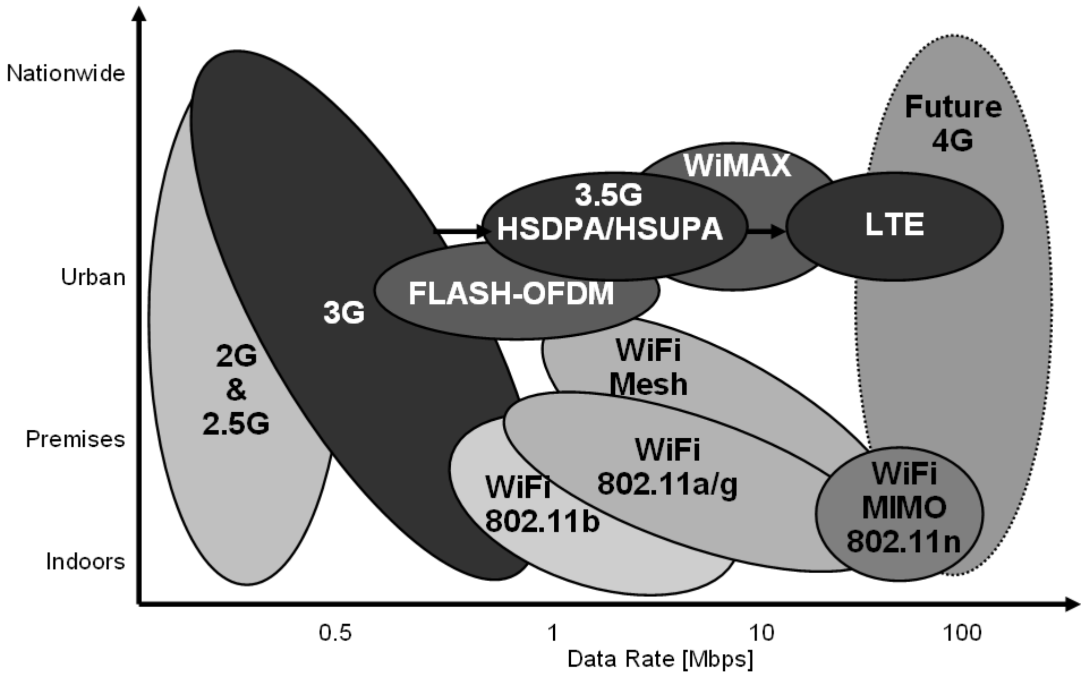

Figure 6. Wireless Radio Technologies

2.1 WiFi and WiFi Mesh

Wireless Fidelity (WiFi) is a wireless technology based on a family of standards developed by IEEE and denoted as 802.11. The 802.11 family contains a large number of standards that regulate the mode of operation and possible improved features.

WiFi can be successfully used for VoIP services, but it has many limitations that range from the number of simultaneous users, problems with mobility, quality of service, etc. However, additional standards from the 802.11 family can improve performance for VoIP. Such features include e.g., quality of service, support for faster mobility handoffs, increased coverage and capacity, etc. However, most of these features are not backwards compatible and are available only to more specialized equipment. This means that both the access point and the WIFi mobile devices need to support them.

If we attempt to consider all the variables and features available in WiFi that affect VoIP, it is very complicated to make a generalization that can be considered as a baseline for the technology. Therefore, in general, the performance and cost of using VoIP over WiFi is more easily characterized based on its deployment case (see Figure 7).

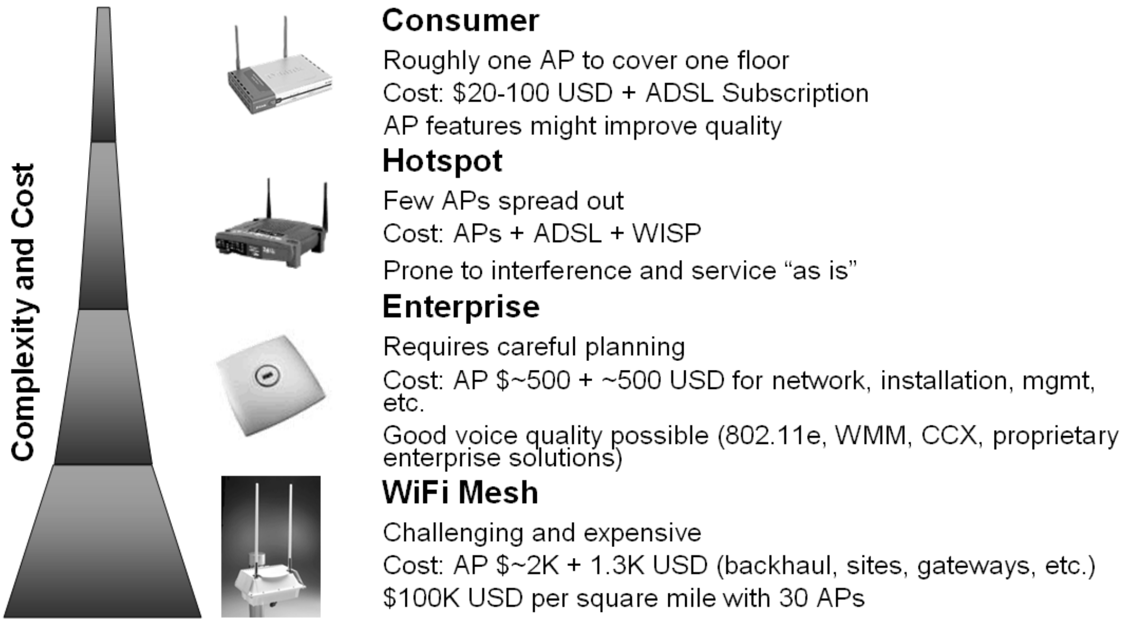

Figure 7. WiFi Deployments

In this thesis we consider the WiFi mesh scenario as the main WiFi contender for cellular networks. The reason is that it is the only unified scenario that can be considered as a competitor to cellular voice systems. Although, in the other scenarios, VoIP over WiFi can take voice minutes away from cellular networks, these cases are scattered and depend on individuals or particular enterprise deployments.

WiFi mesh networks are a special case of WiFi network that is deployed to provide connectivity to city-wide hotspots. Several networks have already been deployed and many cities in the US, and quite a few outside, are either committing to, or studying the possibility of deploying a city-wide WiFi coverage using wireless mesh networks. Several cities are in the planning stages, while smaller cities have already deployed networks which attempt to provide broadband wireless access ubiquitously. The goals in setting up these wireless mesh networks are multiple, and include providing broadband access to underserved communities or supporting emergency services. However, one of the main reasons is to reduce the cost per bit of wireless access to support applications which reduce the expenditures of a city.

The hope is that a lower cost per bit would provide the incentive to use applications on the go, thereby increasing the productivity of city employees. Alternatively, if the network is operated by a provider, the lower cost per bit would provide the margin to compete for mobile applications with cellular operators. Metropolitan WiFi mesh networks are seen by some investors as a potential disruptive technology for legacy cellular operators. A WiFi mesh network combined with a VoIP handheld device could become an alternative to the cellular handset.

2.2 WCDMA and HSDPA

The third generation of wireless telecommunication systems is an evolution of previous wireless systems that enables high bit rate data services. Wideband Code Division Multiple Access (WCDMA) is the main third generation air interface in the world and is most commonly deployed in the 2GHz band. However, in some other countries such as US and parts of Europe, it can also be deployed around 800-900MHz. The standardization efforts are carried out within the 3rd Generation Partnership Project (3GPP). As of 2007, the number of WCDMA subscribers had exceeded 130 million globally in over 150 commercial networks. The next phase of WCDMA, i.e. High-Speed Downlink Packet Access (HSDPA) is currently being intensively deployed worldwide to provide wireless broadband connectivity. When introducing HSDPA in 3G networks the end user experience and system capacity with VoIP applications will improve considerably. When later on also adding high-speed uplink (HSUPA), the system capacity and end user experience will improve even further. HSDPA and HSUPA correspond to 3GPP Releases 5 and 6 respectively. Further HSPA evolution is specified in Release 7 and is known as HSPA+. In addition 3GPP specified a new radio system called Long Term Evolution (LTE) in Release 8, which was completed in 2009. The peak data rate evolution for WCDMA is illustrated in Figure 8. (eds Holma & Toskala 2007)

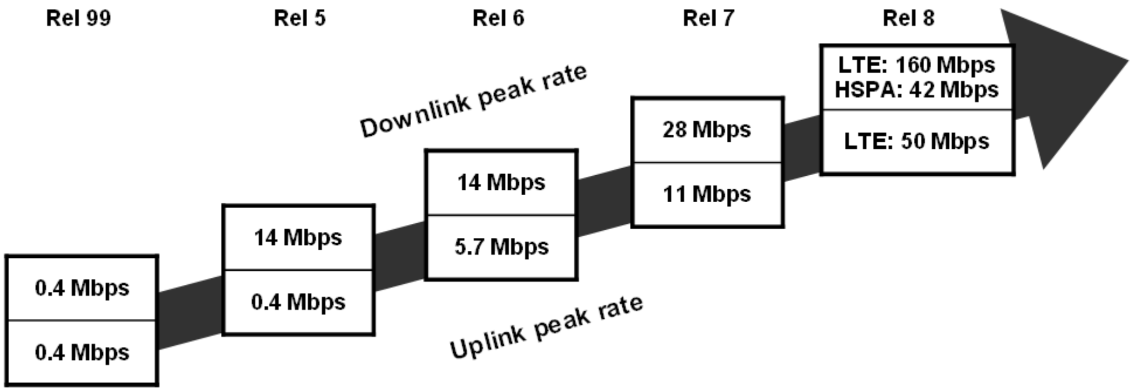

Figure 8. Peak Data Rate Evolution for WCDMA

3G cellular systems are also evolving towards a flat all IP architecture. The evolution is developed step by step based on 3GPP releases **(**see Figure 9**)**. For this reason, VoIP performance in particular is very important because it brings a significant improvement in cellular networks in regard to the capacity of voice users. Therefore, it is envisioned that VoIP will replace circuit switched calls completely in the long term.

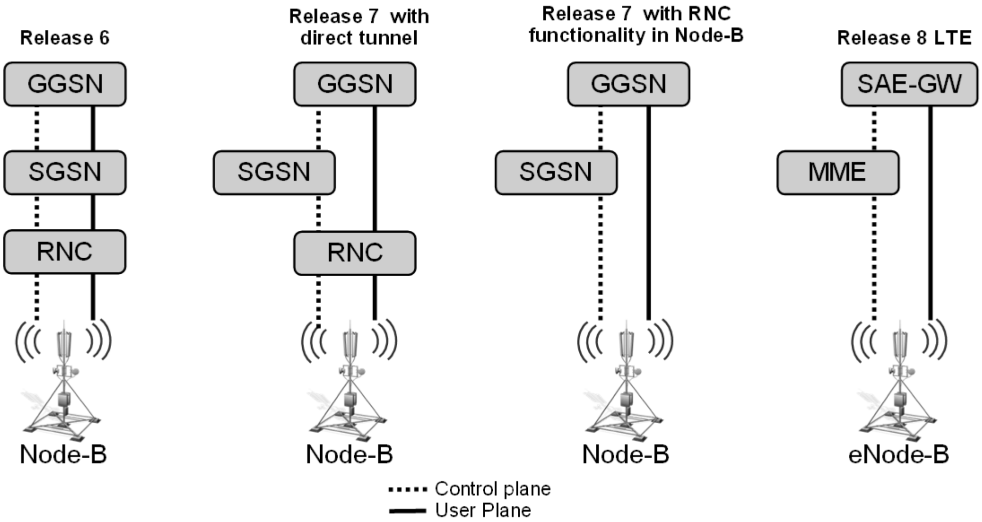

Figure 9. 3GPP Evolution Towards a Flat Architecture

2.3 FLASH-OFDM

FLASH-OFDM (Fast Low-latency Access with Seamless Handoff Orthogonal Frequency Division Multiplexing) is a proprietary system developed by Flarion. Later the technology was purchased by Qualcomm. FLASH-OFDM technology generated a lot of interest as a packet switched bearer which could compete with 3G cellular systems. FLASH-OFDM has been available for several years and at the time of its completion it outperformed 3G networks both in terms of bandwidth, latency and mobility support for data communications. However, despite the early development of the technology, to date there are only a limited number of networks available.

FLASH-OFDM is a wireless broadband technology that can provide nearly ADSL performance. This means that FLASH-OFDM can be a technological rival for ADSL, and in some cases may be the only feasible option to provide broadband access. FLASH-OFDM also operates on a licensed spectrum, but usually at lower frequencies than HSDPA (e.g., 450MHz). Due to the low frequency, a large coverage area can be achieved with a single base station. Thus, it is a particularly interesting option for emerging markets, and especially for rural areas that may lack other telecommunications infrastructure. However, in the 450MHz spectrum there is significantly less bandwidth than at higher frequencies. In Finland, only two 1.25MHz blocks are available for FLASH-ODFM. Therefore, while FLASH-OFDM operating at low frequencies can be feasible for rural and not very densely populated areas, it lacks sufficient capacity for big cities.

2.4 Interworking Technologies

VoIP can be used on different wireless broadband access technologies. These networks are different and are usually independent of each other as well. Hence, they do not provide interoperability. However, there are techniques that can be useful in providing VoIP continuity and interworking across different networks. The most relevant ones are Unlicensed Mobile Access (UMA), also known as Generic Access Network (GAN), Mobile IP (MIP) and Voice Call Continuity (VCC). Whilst MIP focuses on packet data only, VCC provide means for interoperability between circuit switched voice networks and packet switched voice, and UMA supports both packet data interoperability and circuit switched voice to VoIP interworking. In this work we focus on MIP and UMA as VCC was not fully compliant or supported at the time of this thesis. VCC is part of 3GPP Release 7 specifications and is seen as a potential interworking solution for future networks such as LTE (Salkintzis et al. 2009; eds Holma & Toskala 2009).

2.4.1 Unlicensed Mobile Access

Unlicensed Mobile Access (UMA) is a technology that supports handovers to and from WiFi networks (UMA 2004; 3GPP 2006). Originally, UMA was developed to provide an access to GSM/EDGE networks. The technology, however, is nowadays developed in 3GPP under the name Generic Access Network (GAN), which also considers the link to WCDMA. Moreover, a newly created consortium is in the process of extending the standard to include LTE. Despite the official name change to GAN, UMA is the name under which the idea became public and is mostly referred to.

The UMA technology provides a way to access the core cellular network (2G and 3G) through WiFi. This is attractive from the point of view of cellular network operators, who could thus extend their network coverage through WiFi hotspots with minimal additional investment. Improving coverage (especially indoors) is of great importance in some countries and throughout some of the areas in these countries where adequate GSM coverage is not yet available. The United States and Japan are good examples of countries lacking consistent coverage indoors. With UMA, the operator still manages the core network. Radio access technologies on the edge of the network are only used as kind of platforms for packet switched tunneling of either circuit or packet switched connections to the operator’s network The principle on which UMA is based is the possibility to tunnel connections over WiFi back to the operator’s network. The UMA solution does not need much direct investment. The most critical point is a UMA Network Controller (UNC), which provides authentication and a tunneling setup.

Originally when UMA came up, it was thought of as an intermediate step towards IMS. Therefore, many operators dismissed it and it also lost a lot of momentum due to the few operators that deployed it, and to the lack of commercial devices available, which meant there were few compelling reasons for users to buy it. The situation has changed and the amount of UMA devices has increased from 5 in 2007 to 25 in 2009. It is worth mentioning that an additional 10 devices have been retired. Moreover, the UMA concept has also evolved further to include other wireless access technologies. The reason is two-fold; first, because the technology is already available and requires few changes in order to operate with other access networks, and second, because the adoption of femto stations has been very slow (Infonetics 2009). Femto stations are very small 2G/3G base stations intended for extending coverage in subscriber homes. The difference with traditional operator ran base stations is that the backhaul is intended to be the user's broadband connection. However, even though several vendors have released their femto products, the adoption of these stations has been very slow and very few operators have advanced beyond the trial phase. Therefore, UMA has begun to gain momentum again and be considered as an intermediate solution to address the home market and extend indoor coverage. The evolution of UMA now also allows the inclusion of traditional fixed access voice service via the use of a terminal adapter. While this feature does not provide any mobility, it does ease the convergence of old fixed voice networks to newer technologies and allows using the same authorization, billing and subscriber management systems as in cellular networks. Also, it allows subscribers to continue using their older equipment. For the subscriber it does not provide any noticeable difference, but for the operator it brings the advantage of providing the service via a fixed broadband connection that does not need to be operated by the same company (see Figure 10).

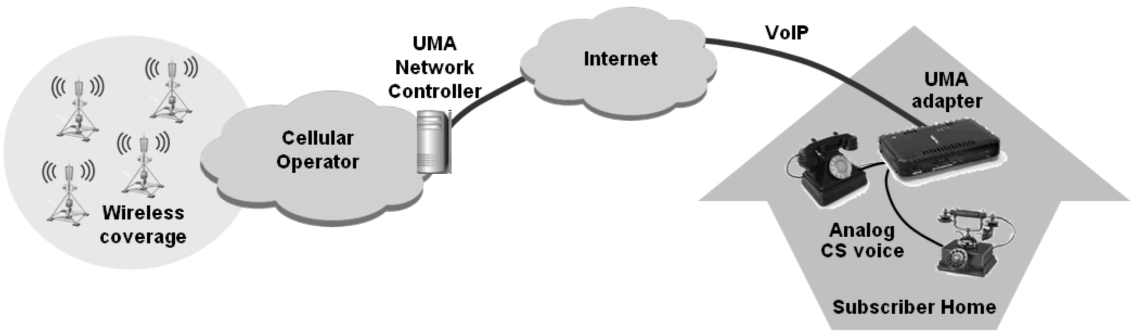

Figure 10. UMA Fixed Access Solution

An additional proposal is to use UMA as the main interworking solution with LTE. The proposal has been brought up by a new consortium known as VoLGA (Voice over LTE via Generic Access) (VoLGA 2009). The founders of this consortium include the majority of major wireless telecommunications equipment vendors and aim at pushing the standard in 3GPP. Particularly since the choice for voice service solution is still under debate and unclear in the LTE case, UMA is being pushed as a proven option for interoperability. Despite this movement there are only two major operators using UMA (T-Mobile and Orange), and a number of smaller ones. Thus, the future of UMA is very likely tied to the technological decisions these two operators take.

2.4.2 Mobile IP

Mobile IP (MIP) is a technology that can be used to provide interoperability between wireless networks that support IP (Perkins 2002). For the VoIP case, MIP allows a user to appear to have the same IP address to the other end user despite the fact that the IP address might have changed. This allows VoIP sessions to continue during mobility. IP addresses can change for a number of reasons. One reason is when a user gets a new address once he joins a new network. The process from moving from one network to another will result in an outage, which for VoIP should be as seamless as possible.

MIP technology is not new and the standard was finalized in 2002. However, it is neither widely supported nor implemented yet. Likewise there are MIP version 4 for IPv4 support and MIP version 6 for IPv6 respectively. In this work we focus on MIPv4 because the majority of networks are still IPv4 based. Likewise, MIPv6 has been proved to be less optimal than MIPv4 in regards to outage time due to the larger amount of signaling involved (Fathi & Chakraborty 2007).

With Mobile IP, the VoIP call can be transferred to one of the newly available networks and enable VoIP continuity. The technology requires a known Home Agent on the Internet and the VoIP handset to have a configured Mobile IP client. The Mobile IP procedure in a VoIP call would be as follows:

1. Clients register to a VoIP server via the Internet using their home IP addresses (i.e. IP-A & IP-B)
2. Clients can call each other via the Internet.
3. The mobile client gets a new IP address (i.e. IP-A*) when it joins a different wireless network (i.e. WiFi to 3G)
4. The mobile client informs the Home Agent that it has been assigned a new IP address (IP-A*).
5. The Home Agent captures all packets destined for the older IP address (IP-A) and forwards them to the new IP address (IP-A*) instead.

This procedure, however, has a drawback. Since mobility is achieved by means of the Home Agent capturing and forwarding packets to the correct address, all packets must pass through the Home Agent. Therefore, the physical location of the Home Agent plays a role in the end user performance, since packets might be routed via a suboptimal route causing additional delays.

3 Related Work

In this section, we will provide a summary of the related works relevant to our study areas, i.e. wireless broadband performance, interworking solutions, VoIP signaling performance, and battery lifetime respectively.

3.1 WiFi Mesh

The performance of WiFi networks have been studied extensively and a large number of works are available studying WiFi deployments in indoor (Cheng et al. 2007, 2006; Tang & Baker 2000), corporate (Balazinska & Castro 2003) and conference meeting environments (Balachandran et al. 2002; Jardosh et al. 2005a, 2005b; Mahajan et al. 2006; Ramachandran, Beldin-Royer & Almeroth 2004; Rodrig et al. 2005). These studies focus mostly on application level performance. In addition, several studies of University WiFi deployments have been published reporting user behavior (Henderson, Kotz & Abyzov 2002; Hernandez-Campos & Papadopouli 2005; Kotz & Essien 2002; McNett & Voelker 2005; Schwab & Bunt 2004; Thajchayapong & Peha 2003). However, in this thesis we focus on city wide outdoor deployments using mesh technology. In regard to mesh technology, there is prior work available investigating several of the limitations of single-radio mesh networks. For instance, several papers (Anastasi et al. 2005; Chen & Zakhor 2006; Xu et al. 2003) investigate the effects of starvation of TCP over wireless networks; which is the reason why the available bandwidth is reduced considerably with multiple users at different hops. However, these works do not demonstrate the effect of starvation via measurements or simulations. Additionally, another group of papers (Chen & Zakhor 2006; Gambiroza, Sadelghi & Knightly 2004; Garetto, Salonidis & Knightly 2008) investigate options (e.g., rate limiting) to counter unfairness and possible starvation. Many studies (Akyldiz & Wang 2005; Kim et al. 2006a; Tobagi & Kleinrock 1975) have also been made on the effect of the hidden node problem and these studies propose several solutions to it. However, most of the solutions require changes in the access nodes and also in the clients. In regard to the number of VoIP calls supported in mesh networks, several papers (Nicolescus et al. 2006; Ting, Ko & Sim 2005; Wei et al. 2006; Gidlund & Ekling 2008) show similar results; other papers (Wei et al. 2006) also analyze optimizations that could increase the VoIP call capacity. Further, there are a few number of measurement studies on the MIT Roofnet mesh network (Aguayo et al. 2004; Bicket et al. 2005; Biswas & Morris 2003; De Couto et al. 2003). These measurements differ significantly from ours since they are based on a single-tier architecture, while most commercial deployments are based on a two-tier architecture (e.g., Google network). Also, some studies of a two-tier architecture are available (Camp et al. 2006, 2008). However, the available research does not focus on the performance of real-time applications such as VoIP, the network capability to carry voice as a primary service, or measure the coverage area of such deployments with voice service in mind. Measurements on live commercial networks such as the one in this study were not available during the time of our study. Nevertheless, recently some groups have published some papers that further extend our work. For instance, (Robinson, Swaminathan & Kinightly 2008) extended the Google network evaluation by developing a framework to determine the location of measurement points to evaluate an outdoor wireless network performance. Likewise, other papers depict the user behavior and usage statistics of public mesh WiFi networks, including the Google network (Afanasyev et al. 2008; Brik et al. 2008). Finally, some recent works evaluating the economical feasibility of deploying city wide WiFi mesh networks also support our conclusions, which are that it is currently too expensive to be considered an alternative to cellular technologies. (Huang 2008)

Although the business case for deploying city wide WiFi mesh has not been successful, there are many networks still in the planning phase and others in the deployment phase (Vos 2009). Likewise, WiFi mesh using off the shelf equipment and open source software has been trialed in rural areas in emerging markets (Johnson 2007). This latter case is very particular, since due to the almost complete lack of Internet connectivity in these areas WiFi Mesh can result in being very attractive despite its limited performance and low capacity at many hops distance. Further, this type of environment is likely to have much less interference compared to urban areas. An example of such **a** network is the Mpumalanga mesh project in Africa (FMFI 2009).

3.2 WCDMA and HSDPA

Although, there is prior work investigating the VoIP performance in WCDMA and HSDPA systems, it is not very extensive and mostly based on simulations. The main difference between the simulations and our study is that simulations are focused on system VoIP capacity while our research focuses on end-to-end voice quality and end user experience as well as the required steps to improve it. Therefore, we extend the available research by providing an additional insight in regard to mobile VoIP performance from an end user perspective.

For instance, (Poppe, Vleeschauwer & Petit 2000, 2001) use an analytical model to study VoIP performance in WCDMA and provide an idea of several of the variables that should be considered. Their conclusion is that if proper parameters are chosen in the air interface and VoIP application, WCDMA is able to provide good voice quality. Another study (Cuny & Lakaniemi 2003), carries out an end-to-end simulation. The major finding of this work is that it notes the importance of RLC UNACK mode for VoIP service. However, the simulation results accounts for the radio interface only and does not account for the potential delay resulting for jitter buffer, header compression or processing delay at the client. Another simulation study (Bajzik et al. 2006), also notices the importance of reducing RLC retransmissions to improve performance in FTP and HTTP browsing. However, this work does not address its importance for VoIP services. In our study we will show that VoIP over WCDMA is possible only with very low quality, thus differing from the conclusions in these other papers.

Some performance simulations are also available for HSDPA and HSUPA. These simulations only provide a capacity figure based on a delay budget. The simulations can be divided in two groups, those that only account for the air interface, and those that account for other delays in the system (network delay, radio access delay, processing delay, compression delay, etc.).

The works that focus only on radio interface delay (Braga, Rodriguez & Cavalcanti 2006; Rittenhouse & Zheng 2005; Chen et al. 2008a, Lunden & Kuusela 2007; Seo & Sung 2006; Wang et al. 2005) commonly use delay budgets of 80, 100 and 150ms and do not account for encoding, processing, jitter buffer implementation or compression delay. The larger the delay budget is given to the radio interface, the higher the capacity. For this reason it is common to find figures based on a 150ms radio interface delay budget. However, if we consider that VoIP quality should be comparable to current circuit switched voice, it means that the total end-to-end delay should be at the most 250ms. Therefore, if 150ms of the total delay is assumed for the radio interface, it means that there are only 100ms left for all the other required aspects in the system. We believe that this is unrealistic in the near future, and that at the most 80ms should be assumed to model representative scenarios. Most of these simulations compare different scheduling algorithms and the possible VoIP capacity improvement with some of the proposed schedulers.

In the case of studies that consider an end-to-end delay budget (Kim 2006b; Hosein 2005; Ericson & Wänstedt 2007; Folke et al. 2007; Chen et al. 2008b), the assumption is between 250-300ms. These studies usually just take the delay budget as a target baseline rather than break it up into different components. Two studies, however, provide values for the processing delay including the jitter buffer implementation. One study (Kim 2006b) assumes 50ms while the other (Ericson & Wänstedt 2007) considers it to be 40ms. As we noticed in our experiments with real handsets and VoIP clients, assuming such short delays is overly optimistic and does not represent actual handset performance. Other works (ITU-T 2003; Cisco 2000) provide some estimated values for processing delays in laptop clients. In addition, a large amount of the studies simulating capacity assume that header compression is utilized. However, doing so will incur additional processing delays that are not modeled either. The available simulations also focus on comparing different schedulers and finding which ones provide higher VoIP capacity. However, some simulations (Ericson & Wänstedt 2007; Folke et al. 2007) point out that VoIP capacity should not be the only focus of study and that it is important to consider prioritization of VoIP packets over other data in the deployed schedulers. Such prioritization of VoIP is even more crucial for limited delay budgets. Finally, there are also simulations of VoIP outage times in mobility scenarios in HSDPA (Lunden et al. 2008) and HSUPA (Wager & Sandlund 2007; Aho, Äijänen & Ristaniemi 2008). The main finding of these works is that the factor that most affects VoIP is velocity. For HSDPA, speeds of 50km/h already have an effect, while for HSUPA the system is able to support VoIP with short outage times at up to 120km/h.

In regard to live network performance, a limited number of works show empirical data from live HSDPA networks (Jurvansuu et al. 2007) and an even more limited number provides empirical data for VoIP (Kim et al. 2008). In this latter work, two live networks in Korea are evaluated (HSDPA and WiMAX). The evaluation of VoIP was performed using Skype on a laptop. The conclusion of this work was that voice quality is only acceptable when no other data traffic is running in the background for both networks, in which case the delay becomes too long.

Even though, it is understandable that the exact encoding, processing delays and jitter buffer playout delays are client specific, unless they are modeled accordingly, or at least to some extent, the differences in performance between simulations and actual deployments will remain very visible. Therefore, results of simulations are only comparable to laptop based performance at its best and not to actual handheld performance, which in the end is the primary use case for VoIP services. Our study extends the available research by providing performance values from environments that provide insight into real user experience. With these results, it is possible to update some of the simulation assumptions and have a better estimation on when the newer features can make the most impact based on the limitations of current hardware equipment.

3.3 FLASH-OFDM

Due to the novelty of FLASH-OFDM network deployments, actual performance measurements in live networks are not widely available. Some of the few live FLASH-OFDM networks are T-Mobile in Slovakia (T-Mobile n.d.), Citizens in Virginia USA (Citizens n.d.), Digita in Finland (Digita n.d.), and DigiWeb in Ireland. While some other networks are planned, most of them are still in trialing phases or in the process of bidding for a spectrum license. Performance metrics found in press releases and technology reports are as follows. One report mentions (3G Newsroom 2005) that expected data speeds are 1Mbps on average in downlink and 300-500kbps on average in uplink. Further, (Unstrung 2004) claims the technology realizes typical downlink speeds of 1.5Mbps, with occasional bursts of up to 3Mbps, and typical uplink speeds of 375kbps, with bursts of up to 750kbps.

After our live network study results with FLASH-OFDM were published, some additional works have been published regarding different tests in a trial network in Sendai Japan consisting of only one cell (Izuka et al. 2008; Oguma et al. 2008, 2009). These works differ from ours in several aspects: They had only one cell rather than a live network, had no background traffic, focused on throughput only rather than application performance, and did not measure delay as a key performance indicator. Additionally, the trial network in Sendai operates at a much higher frequency (2GHz), while the tested network in Finland operates at a much lower one (450Mhz). Figure 11 shows the relative difference in cell size between both frequencies. Therefore, the

Sendai trial results are comparable with 3G networks at 2.1GHz with regard to coverage area.

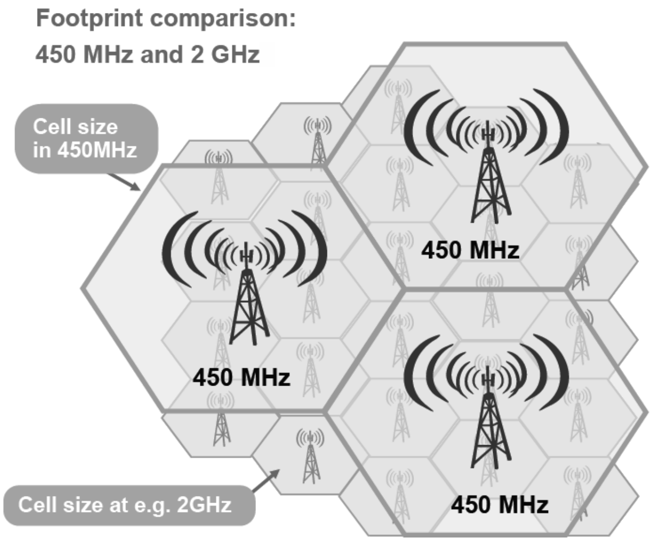

Figure 11. Relative Cell Size Difference at 450MHz and 2GHz

The trial results with one cell (Izuka et al. 2008) show that downlink throughput above 1Mbps is possible within 1km of the base station. However, throughput can be significantly impaired by inter sector-interference when the same frequency band is used in all three sectors, in these cases, the average downlink throughput is only around 300kbps. Their conclusion is that inter-sector interference reduction schemes are necessary. Inter-sector interference issues were also previously studied by (Tsumura et al. 2005). Subsequently, (Oguma et al. 2008) extends the study by testing the effect of frequency reuse factors of 1 and 3, with the former yielding better performance. Finally, (Oguma et al. 2009) carried out further tests aiming at finding the optimal antenna height. The results show that the base stations with antenna heights of 19m, 58m, and 84m can provide coverage of 600m, 800m, and 1.3km respectively with downlink throughput up to 1.5Mbps at cell edge.

FLASH-OFDM has also been developed for other business cases. For instance, in addition to common subscriber use, FLASH-OFDM was used in Finland for updating the GPS location of public buses in real time (Digita@450 2007) (see Figure 12). However, the public transport company operating these buses has recently announced that they will discontinue this service due to high cost (Lehto 2009). Another use case was from TeliaSonera, in which FLASH-OFDM was used as a replacement for fixed lines in very remote rural areas in northern Finland. Further, (Riihimaki et al. 2008) evaluated the feasibility of using FLASH-OFDM in trains. The resulting conclusion is that FLASH-OFDM is a viable pre-stage solution to be used in the meantime until other wireless broadband technologies (e.g., WiMAX, HSPA+ or LTE) become widely available. Denmark and Japan are some of the countries planning trials for the railroad use case.

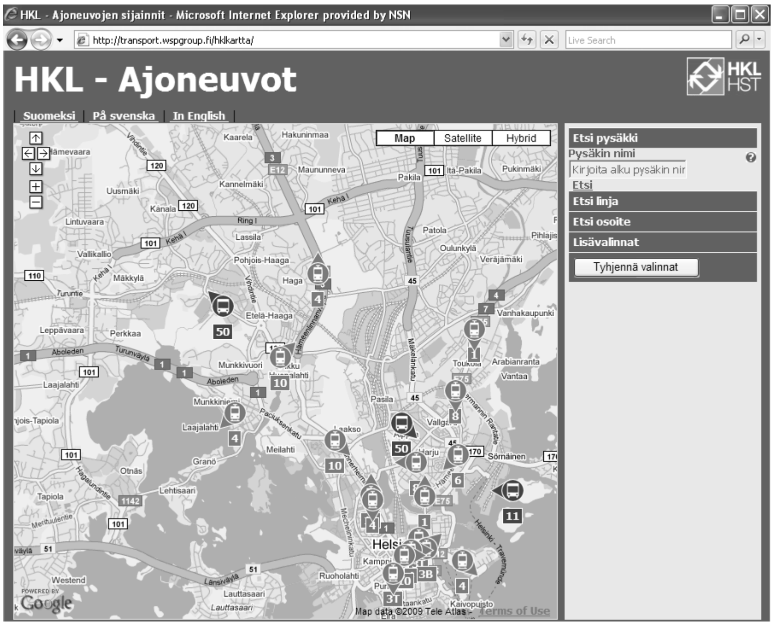

Figure 12. Helsinki Bus and Tram Location Service using FLASH-OFDM (HKL 2009)

3.4 Interworking with UMA

There are very few related works focusing on UMA, and none of them focuses on service performance. To the date of this writing, the results presented in this thesis remain the only available actual performance measures of the solution in the literature. The most closely related work is from (Kale & Schwengler 2009), in which UMA handover mechanism and reliability is compared with VCC. The results of this work show that both technologies are comparable. The main difference in this work when compared to ours are firstly that it is based on simulations, and secondly that it considers 3G as the access technology. However, considering 3G over 2G for the circuit switched side of the call does not matter, since the performance is similar in regard to access delay (Venken & Vleeschauwer 2004). The handover signaling delay metrics of this study were obtained via simulations and are the following: (1) 3G-to-UMA 320ms, (2) UMA-to-3G 290ms, and using VCC (3) 3G-to-WiFi 280ms, and (4) WiFi-to-3G 380ms. Unfortunately, the authors do not show detailed results on what are the possible actual voice outages during the handovers. In this study the handover delay is simulated based on signaling required and the assumptions of general delay of the access networks considered. However, the signaling considered includes messages that do not affect the actual voice outage during the handovers. Nevertheless, the metrics are somewhat comparable to our measurements. Other works have focused on other aspects. UMA feasibility over IMS is evaluated in (Borger et al. 2006), claiming that the UMA solution is much more expensive to develop than IMS. The possible security issues and counter measures are presented by (Grech & Eronen 2005; Zhang, Yang & Ma 2008). Further, (Yaqub, Ul-Haq & Yahya 2008) points out some of the limitations of the UMA architecture, which are mainly that the technology is coupled with GSM/3GPP networks. Other works also propose the use of UMA as an authentication mechanism for other services (Oh et al. 2008) such as the Digital Multimedia Broadcasting (DMB) video. That is, use the already integrated authentication system in UMA for DMB services. Handheld based auctions are another proposed service that could benefit from UMA mobility and service continuity (Tsai & Shen 2007). In addition, an analytical model for estimating 3G UMA energy consumption is presented by **(**Yang, Lin & Huang 2008).

3.5 Interworking with MIP

In regard to Mobile IP and the disruption time during mobility, there are a number of papers based on simulations (Fathi, Prasad & Chakraborty 2005b; Jung et al. 2003; Sharma, Zhu & Chiueh 2004; Musasinghe & Jamalipour 2008), with some focusing on VoIP disruption (Chao, Chu & Lin 2000; Kwong, Gerla & Das 2002; Belhoul, Sekercioglu & Mani 2006). Most of these papers consider the delay between the mobile node and the Mobile IP agents for the calculations, while some also take account of the wireless link delay as well. The published results with a VoIP focus differ greatly between each other and values range mainly between 50ms and 800ms, and in some cases even several seconds. Other papers only consider the signaling involved in MIP to carry out the simulation (Medina, Lohi & Madani 2008). Additional research in (Fathi, Chakraborty & Prasad 2005a,; Fathi & Chakraborty 2007) outlines the differences in performance between the MIPv4 and MIPv6 schemes. However, empirical data supporting the simulations is not widely available. Some papers such as (Zeadally et al. 2004; Tseng et al. 2005) are among the few that provide measurements from an actual implementation, and focus only on mobility scenarios between WiFi to WiFi. In regard to the performance data of VoIP between WiFi and cellular data (i.e. 2G and 3G) the available research is even scarcer. The most closely related work is (Grech, Haverinen & Devaparapalli 2006), which provides mobility measurements between 2G and WiFi. The setup was a laptop with a Mobile IP client streaming packets similar to VoIP packets during mobility, and used the Dynamics Mobile IP implementation (HUT n.d.a, n.d.b). It must be noted though, that since no actual voice packets are transmitted, neither encoding/decoding nor packet loss concealment algorithms are in place either. The results show that seamless mobility (no noticeable break) is possible, except in cases where radio resources from the cellular network are released. In such cases, the data outage is approximately 2.5 seconds. The results from (Grech, Haverinen & Devaparapalli 2006) are faulty since it is well proven that mobility with MIP will always create a disruption, and thus these measurements are unreliable. Our research extends the available MIP research considerably by including actual measurements with commercial handsets, embedded VoIP clients, Internet based VoIP servers and Mobile IP agents, and by comparing the results with the aforementioned available research. Further we estimate the possible performance of MIP for different future wireless access technologies based on our empirical data.

3.6 Signaling Performance

SIP call setup delays and signaling performance have been studied previously mostly for Internet scenarios. The ITU-T E.721 (ITU-T 1999) recommendation and (Lin et al. 1999), provide call setup delay recommendations for circuit switched and Internet Telephony systems respectively. Additionally, (Eyers & Schulzrinne 2003) provides guidelines for Internet Telephony call setup and signaling transfer delays. In regard to 3GPP based wireless accesses, (Kist & Harris 2002) provides simulations for transfer delays with 3GPP signaling, while (Fathi, Chakraborty & Prasad 2006; Pous et al. 2003) modeled signaling performance. Further, (Curcio & Lundan 2002) provides measurements for a WCDMA setup using laptop clients for local, international and overseas calls. Most of the mentioned research focuses on simulations, and does not consider some end user cases such as calls in wireless environments starting from different states. Additionally, performance with different wireless radio accesses and configurations under the same conditions is not available. Also, the available works do not use an embedded VoIP client in a handheld mobile terminal, which yields different delay values than with a PC. HSDPA signaling performance has not been evaluated either. Our research aims at covering these items. The importance of evaluating a mobile terminal relies on the fact that the eventual substitution of CS based calls in 3GPP networks (HSDPA and WCDMA) for VoIP calls will take place with a handheld mobile device and not with a PC or laptop. Likewise, multi radio can provide ubiquitous access via different wireless access technologies that perform differently. Furthermore, the use of VoIP instead of CS calls has an additional impact with regard to battery lifetime. The work most closely related to ours is (Haverinen, Siren & Eronen 2007), which makes an empirical analysis of radio resource control timers and their effect on energy consumption for NAT keep-alive messages. Simulations by (Lee, Yeh & Chen 2004; Yeh, Lee & Chen 2004) have been done on the same topic. Our work extends this research by presenting measurements during actual IMS registrations over multiple accesses.

The lack of actual measurement performance values in literature could be mainly due to the unavailability of integrated VoIP clients in the terminals and available HSDPA networks. However, with the introduction of some Nokia multi radio devices with VoIP capabilities (e.g., N95, 6110), using SIP based VoIP applications without a PC is possible. The VoIP server can be any IETF or IMS based server reachable via one of the wireless accesses.

3.7 Always-On Energy Consumption

The related work in regards to always-on energy consumption in cellular networks is very limited. Amongst the few works available are (Haverinen, Siren & Eronen 2007), in which the effects of different configurations for cellular states are studied. This study focused on WDCMA. Other works are available for WiFi evaluations although energy consumption in WiFi can vary greatly depending on devices, access points, configurations and additional features that might or might not be in place such as 802.11e. For that reason, WiFi energy consumption as a whole is not considered in this literature review. For an overview of the energy consumption variables in WiFi with a VoIP focus refer to (Gupta & Mohapatra 2007). Our work extends the available research by including different configurations, considering WCDMA, HSDPA and also future 3GPP features. Likewise, our data is obtained through measurements from commercial handsets running embedded VoIP clients in different wireless access technologies using the same hardware and software platforms to provide a fair comparison.

4 Summary of Results

4.1 WiFi Mesh Performance

In our evaluation of WiFi mesh, we first studied VoIP performance in an actual, fully deployed state of the art network in Mountain View operated by Google (Google n.d.). The main network characteristics of this network at the time of the study are given in Figure 13**.** Subsequently, we studied in the lab the performance of a mesh network of a single-radio platform against some key performance metrics relevant to VoIP.

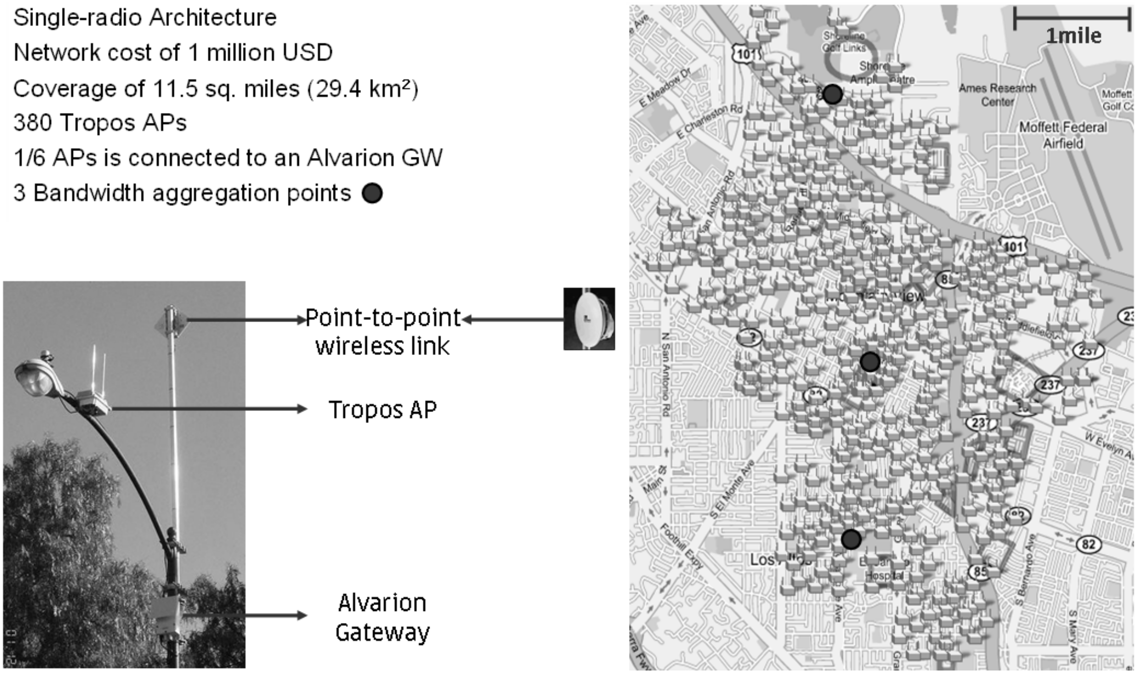

Figure 13. Google WiFi Mesh Network in Mountain View CA

Our study of the Google network had the goal of determining the performance level that can be achieved in different common VoIP calls (e.g., Skype) and data applications. VoIP calls are especially relevant in order to assess whether WiFi mesh networks are a serious alternative to cellular networks. All the scenarios and tests were carried out in outdoor conditions. The performance indoors was expected to be worse than very poor indoors due to signal attenuation and, since we showed a rather poor performance outdoors in the first place, the indoor case was omitted in this study.

The results of this study showed that the voice quality of this network varies significantly. The main reason for variation is the amount of interference caused by other radio devices operating in the unlicensed spectrum, particularly DECT phones and other access points nearby. This is a big issue since it can cause voice quality to greatly degrade making the quality of the voice service unacceptable. Further, our results show that the amount of access points is not enough to provide continuous coverage. Therefore, VoIP calls drop when the user moves. In addition, in a lab environment we evaluated the upper bounds for VoIP user capacity using single radio technology in order to understand the number of users that could be supported per AP cluster. Our results and previous work show that the maximum number of VoIP users with toll quality in single-radio mesh networks is around 7 per cluster under excellent signal conditions (Ting, Ko & Sim 2005; Nicolescus et al. 2006; Wei et al. 2006). With the current average of 5 clusters per square mile, it means in theory that WiFi mesh provides a maximum of 35 VoIP users. However, in practice, with the current cluster configurations, providing good signal coverage is very challenging, and such capacity limits are hardly reached. Furthermore, these calculations are for external outdoor coverage only. The current AP density provides very poor indoor coverage, if any. In contrast, the voice capacity in cellular networks is around 40 users for a cell covering a square mile with very reliable indoor penetration and quality of service assured. Moreover, already standardized Release 7 features will increase capacity considerably to over 100 users (eds Holma & Toskala 2007).

Based on empirical data and radio estimations, we further calculated the amount of access points that would be required to provide good coverage for VoIP service. The resulting amount is at least 81 APs per square mile opposed to the current average of 30 in the deployed networks (e.g.,, Google network). The impact of the cost of and capacity increase of a mesh deployment with 81 access points per square mile is depicted in Figure 14.

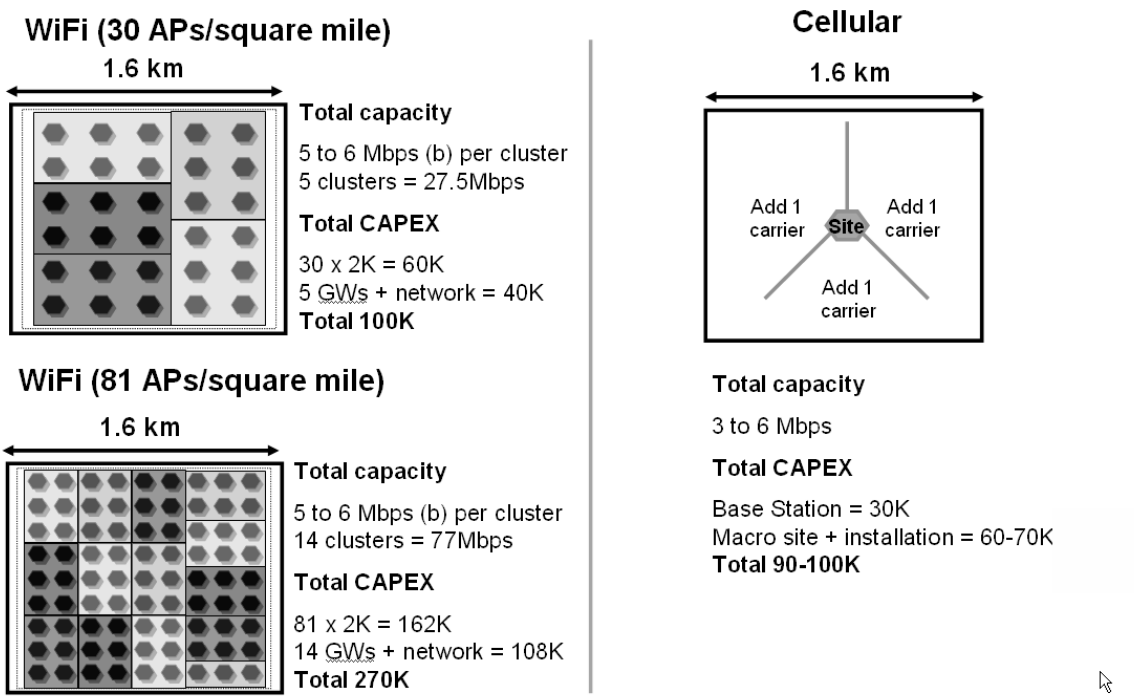

Figure 14. Cost Comparison: TRX CAPEX ($USD)

The cost for providing a WiFi mesh network with proper VoIP coverage risess to roughly 270 thousand dollars per square mile. In contrast, the cost of building a cellular network base station site costs around $30,000 USD (Caiser et al. 2006). However, cellular macro sites and installation costs can push the cost to $100,000 USD in some cases. This shows that the cost of current WiFi mesh deployments is already similar to cellular and the cost of deployment does not provide an advantage over cellular deployments. Cellular deployments also provide much better coverage (e.g., indoors) where WiFi is heavily handicapped due to outdoor-to-indoor wall attenuation. Nevertheless, it is important to note that cellular deployments require an expensive licensed spectrum and relatively expensive sites. However, in the case of an existing cellular operator, spectrum licenses and sites might already be available from previous deployments.

4.2 WCDMA and HSDPA Performance

For our 3G evaluation, we studied the quality of VoIP in WCDMA and HSDPA networks both in the lab and in live network environment setups by conducting a methodic (methodical is more commonplace as an adjective) performance analysis based on the E-Model. Our study takes into consideration both the performance of the network and also the performance of real embedded VoIP clients. In addition, we validated the results of our study by comparing them to the actual performance in a densely deployed HSDPA network in Finland. Our laptop client was modeled with a jitter buffer of 120ms, which is the same size found in the native VoIP client in Nokia devices.

Our results are consistent and show that the achieved quality in the HSDPA system is competitive. Based on ITU-T G.107 (ITU-T 2003), quality was on average medium for HSDPA in both laboratory and live scenarios. The average MOS in the lab environment was 3.7. This is a good figure especially considering that typical PSTN systems provide MOS values around 3.5. In the live network the MOS value was between 3.4 and 3.5 for the majority of scenarios. In the case of WCDMA, quality differed depending on the bitrate supported. WCDMA 128/128kbps provided low quality and WCDMA 64/64kbps gave poor quality. Quality in WCDMA 128/128 is not optimal and is around MOS 3.0 at its best, while WCDMA 64/64 MOS was only 2.25. ITU states that MOS below 2.5 provides an unacceptable voice quality and that nearly all users will be dissatisfied with such a service. Furthermore, we evaluated the additional delay that is incurred when using actual embedded VoIP clients. To carry out this task, we established actual VoIP calls between devices using the same laboratory environment and measured the total end-to-end delay. With the results we estimated the client processing delay by subtracting the average end-to-end delay from our tests based on the E-Model using a laptop client. The result is roughly a 210ms additional processing delay when using a real embedded VoIP client. This value differs considerably from the more optimistic embedded VoIP client processing delay estimations of 50-75ms available in research from (Kim 2006b; Ericson & Wänstedt 2007). A summary of the VoIP results is given in Figure 15.

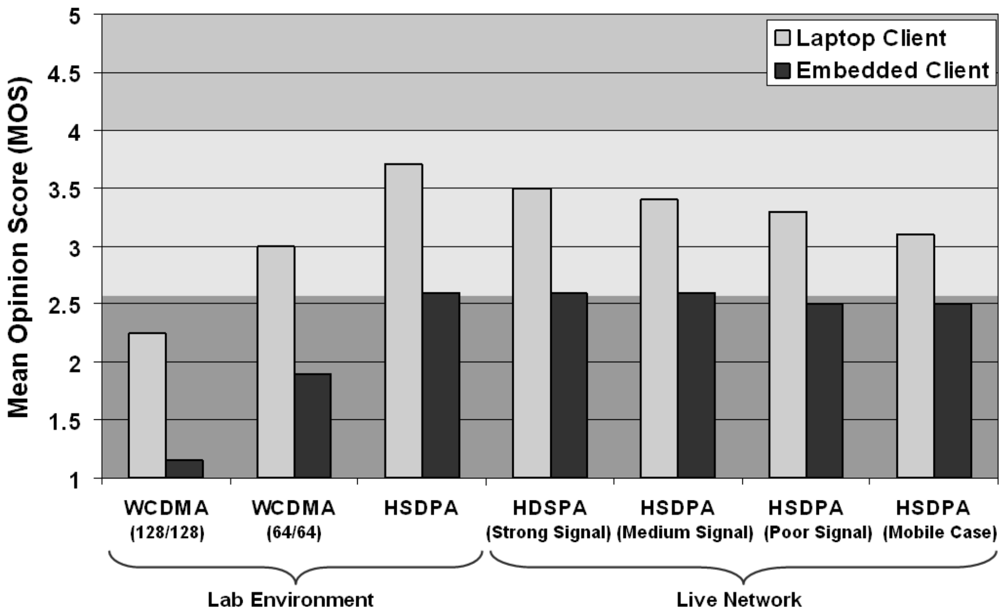

Figure 15. WCDMA and HSDPA VoIP Performance Evaluation

The estimate of embedded client processing delay is representative for generalization purposes since Nokia accounted for 51% market share of smart phone devices at the time of this study (2007). During 2008, with the introduction of the iPhone, Nokia's market share decreased to 39% (Canalys 2008). Nevertheless, Nokia's market share is still significant and the results from the N95 smart phone are still valid as this was Nokia's flagship device. In addition, the newer version of the device (N96) released in 2008 has a very similar processor which should yield similar performance.

The resulting conclusion is that at the current moment, the performance of VoIP in 3G networks is far from optimal. Even though the introduction of HSDPA significantly reduces the user-to-user voice delay, the performance is satisfactory only for selected devices. Overall, the end user experience is still significantly worse than with circuit switched solutions and is not acceptable. End-to-End delay is the main reason for low voice quality. In particular the embedded client processing delay (210ms) and the additional delay from the jitter buffer implementation (130ms) already surpass the recommended delay guidelines. Therefore, it is quite easy to understand why VoIP does not perform well in current systems with handheld terminals, and particularly live networks, even when the round trip time in the wireless interface is low. The final end-to-end delay is just too high. It must be noted though, that in a laptop VoIP client there will also be an additional processing delay. This delay was not modeled in our study since our test software did not encode or decode any actual voice, which makes our results even a little more optimistic than a real case. However, such delay is considerably lower, ~50ms (Cisco 2007). Thus, still ~160ms lower than with the mobile device tested.

Future features such as HSUPA in further 3GPP releases will improve performance slightly. For example, the expected average round trip time for HSUPA networks is roughly 65ms (a reduction of 20ms compared with HSDPA). This reduction however does not improve the VoIP quality when using a laptop. That is, the average MOS with a laptop will still be the same. Contrastingly, the expected quality improvement for an embedded client is about 0.2 points in the MOS score. However, if some HSUPA features like UNACK mode are enabled in the wireless network, it will be possible to reduce the size of the jitter buffer implementation without compromising the VoIP quality. This feature, however, is not yet available in deployed networks. Therefore, a reduction in the client processing delay is extremely important in order to seriously improve the VoIP quality in the mobile environment.

Finally, we provide an estimation of the possible performance roadmap in a static scenario given that the mentioned improvements take place. For this performance model we show values for two scenarios: (a) laptop VoIP client, (b) lab vs. live environment. The reason for these scenarios is that handset VoIP clients require additional processing power and that live networks incur an additional delay (~40ms). In addition, to make the model as realistic as possible, we model the roadmap up to the performance of the reduced, but still accountable processing delay in a laptop client (~50ms) (Cisco 2007). Thus, this will be the case in which a mobile handset has as much computing power as laptop. The model considers a scenario with good signal strength in which packet loss is not an issue. Table 3 summarizes the assumptions.

Table 3. Assumptions in the VoIP Performance Roadmap

	HSDPA		HSUPA		Jitter Buffer	Processing Delay (Mobile)
	Lab	Live	Lab	Live		
Delay	85ms	125ms	65ms	105ms	130ms	210ms

Figure 16 shows the mobile VoIP performance roadmap for embedded clients. The first improvement comes with the use of UNACK mode. This mode will allow delay to be at a stable and lower level than with the ACK mode used in current networks. Since we can assume that there are no RLC retransmissions with this mode, the jitter buffer in the VoIP client can be reduced from 130ms to 80ms. This size should be able to handle the delay variation in the wireless network. Further, as devices mature, we can expect better handsets with more processing power to emerge. It is very hard to estimate the rate for this improvement, but we provide stepped estimations from 210ms in current devices, to 150, 100, and finally 50ms respectively. With this roadmap we can notice that the handset processing delay needs to be at the most 100ms in order to provide VoIP quality at a similar to old analog voice networks. However, to provide toll voice quality, processing delay needs to be reduced further to 50ms, which is unlikely in the near term.

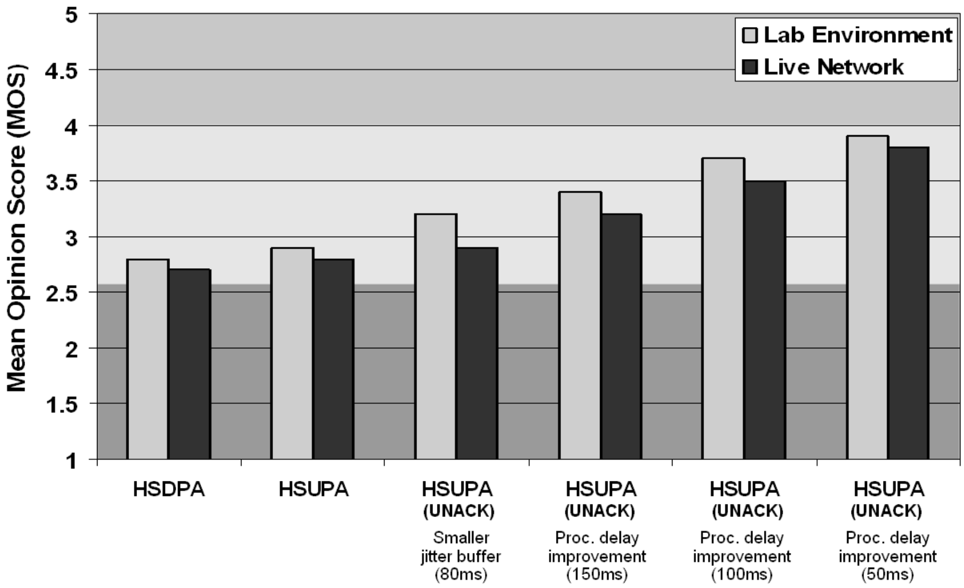

Figure 16. Mobile VoIP Performance Roadmap for Handsets (G.729 codec)

4.3 FLASH-OFDM Performance

In this section we evaluate a live FLASH-OFDM network in Finland deployed by Digita. The FLASH-OFDM network in Finland goes under the commercial name of "@450". It was opened to the public on April 1, 2007 and it is expected to cover the whole of Finland by the end of 2009 (Digita n.d.). This network uses a recently reallocated spectrum (450MHz) which allows covering a large area with a limited number of base stations. Our VoIP quality evaluation was the same as the one used to evaluate the HSDPA network in Finland described in the previous section. We evaluated VoIP quality in different signal conditions (excellent, medium and poor) and in a mobile scenario. The FLASH-OFDM network operator claims to have excellent coverage along the route in their coverage maps. (Digita 2007)

The results from the experiment show that the average delay is low (40-50ms) and does not fluctuate much. Further, the delay is also stable even in mobile scenarios. This contrasts largely with HSDPA in which mobility causes delay variation and packet loss to increase considerably. The FLASH-OFDM system is able to handle VoIP with call toll quality in both static and mobile scenarios (MOS = 4). Only in areas with poor signal is voice quality reduced to medium (MOS = 3.4). However, the coverage in the network is dense enough to provide a high probability of areas with good signal conditions.

As a remark, the VoIP results presented are for laptop based VoIP communication. Therefore, even though jitter buffer delay is considered, the embedded VoIP client processing delay is not included. Since at the time of the study there were no available FLASH-OFDM handsets, it was not possible to estimate processing delays in handsets accurately for this system. Likewise, it would not be fair either to use values from 3G handsets either since the software platform and operating systems can differ.

The main conclusion when comparing FLASH-OFDM with HSDPA for VoIP service is that FLASH-OFDM clearly outperforms HSDPA. However, the capacity for FLASH-OFDM is much lower than HSDPA, 3Mbps opposed to 10Mbps in HSDPA. Therefore, even if deployed at a low frequency spectrum such as 450MHz, a large number of base stations would be needed to support densely populated areas and the lower attenuation would not be an advantage any longer. In contrast, for rural cases a single base station can cover a very large area. If we further generalize the feasibility of deploying one network over the other, we notice also that the spectrum at low frequencies is neither widely available nor cheap. For this reason, other FLASH-OFDM trials are evaluating the 2GHz spectrum (e.g., Japan). This would require the same amount of base stations as HSDPA and provides a much lower capacity. Furthermore, since FLASH-OFDM is a proprietary technology, it has a lock-in factor, which limits the provider choices and terminal availability. Therefore, the main benefit of FLASH-OFDM when considering its limitations are scenarios in which: (1) mobility at high speed is common (e.g., trains), or (2) rural environments lacking telecommunication infrastructure and with a low frequency spectrum available.

4.4 Interworking Solutions Performance

4.4.1 Unlicensed Mobile Access (UMA)

We evaluated the mobility performance of the UMA system between GSM and UMA handovers. Our focus is two-fold. First determine the performance of the whole UMA registration and handover procedure, and second, the voice outage time during the handovers. To measure the procedures, we tracked the signaling data required when establishing a voice call and while moving in and out of the UMA coverage area. The measurements were carried in an interference free indoor laboratory environment.

The results from the registration and handovers measurements carried out are summarized in Figure 17 and Figure 18**.** These figures show average values for the whole signaling procedure. The voice outage that occurs due to the handover is only the area marked with a star. That is, 27ms for GSM-to-UMA handover and 199ms for UMA-to-GSM. The handover performance is good and in line with typical breaks in GSM networks (120-220ms) (3GPP 2003). Finally, although we could not test WCDMA-to-UMA handovers, the performance will be very similar if not better, since the difference in circuit switched voice access delay between GSM and WCDMA is only around 10ms.

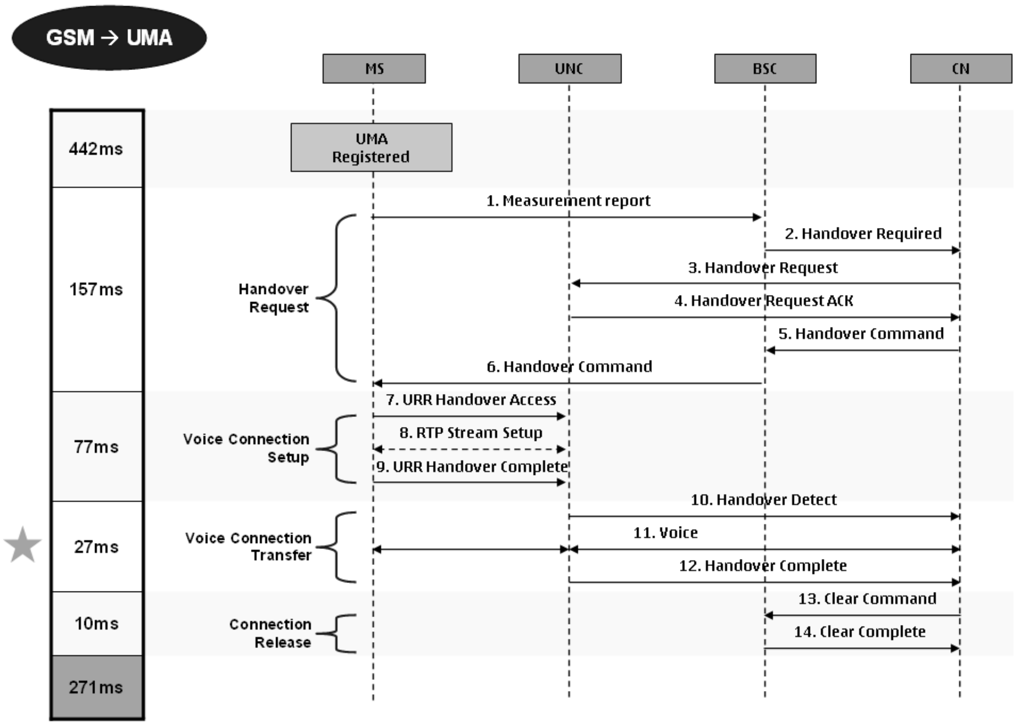

Figure 17. GSM-to-UMA Signaling and Handover Performance Results

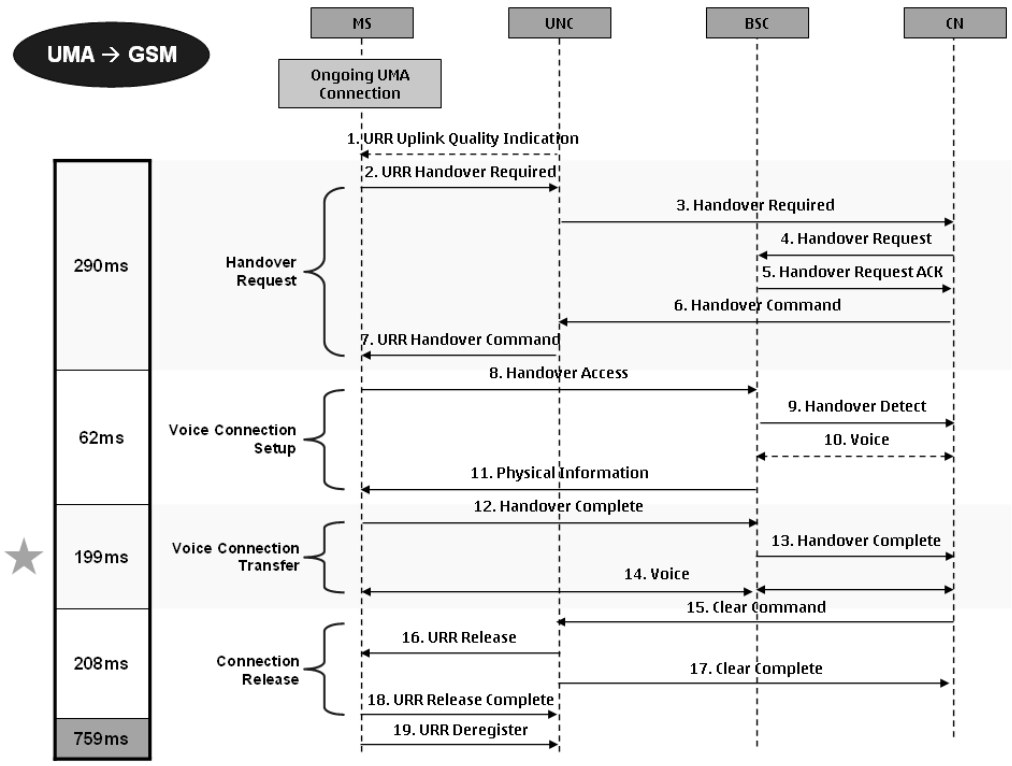

Figure 18. UMA-to-GSM Signaling and Handover Performance Results

4.4.2 Mobile IP (MIP)

To evaluate Mobile IP, we carried out mobility tests during VoIP calls using a commercial 3G network in Finland and a local WiFi link connected to the Internet in an interference free environment. The VoIP client was the default client preinstalled in the Nokia N-series and E-series devices. The Mobile IP client was purchased form Birdstep, and the Home Agent located in Scandinavia. The mobility performance was evaluated at two levels, signaling level, and voice outage time during handovers.

Our results show that signaling is disturbed during mobility resulting in an increased delay and in some cases requiring message retransmissions. The average values for the periods in which this abnormal message flow was observed were 325ms for WiFi-to-3G and 140ms for 3G-to-WiFi. The measurements results for voice outage during mobility show that mobility from WiFi-to-3G results in a longer break (600-700ms), than mobility from 3G-to-WiFi. The effect on user experience from this break is one to two words lost when moving from WiFi-to-3G, and three to four words lost when moving from WiFi-to-3G.

When we compare our results with the available performance values in the literature we notice that the only algorithm that provides similar results is the one presented by (Fathi, Chakraborty & Prasad 2006; Fathi & Chakraborty 2007). The simulation presented by Fathi shows that a wireless link with 20ms delay (similar to WiFi) would result in ~100-150ms disruption time, and a wireless link with 200ms delay (similar to 3G) would result in 600-800ms. This latter break is considerably higher than typical GSM handovers (120-220ms) (3GPP 2003), but it is similar to breaks encountered in WiFi-to-WiFi mobility in deployed networks when WPA encryption is used.

This algorithm (Fathi, Prasad & Chakraborty 2005b) is remarkably close to our measurements. Thus, we consider it as a representative alternative to model Mobile IP performance due to its close estimation values with our empirical results. Therefore, if we further model evolved 3G wireless radio links (HSDPA, HSUPA, and HSPA+) with this algorithm, we notice that the performance can improve considerably. Table 4 summarizes the performance results from our measurements and performance estimations for future 3G releases.

Table 4. Mobile IP Performance Summary

	Handover	Wireless Link Delay	Audio Break	Perceived End User Experience
Measured	WCDMA to WiFi	200ms	~100ms	1-2 words lost
	WiFi to WCDMA	25ms	~600-700ms	Several words lost
Estimated	WiFi to HSDPA	85ms (Rel. 5)	~275-450ms	
	WiFi to HSUPA	65ms (Rel. 6)	~225-350ms	
	WiFi to HSPA+	45ms (Rel. 7)	~175-275ms	

4.5 Signaling Performance and Energy Consumption

Besides VoIP audio quality, signaling and battery lifetime also play a relevant role and can affect end user experience. In this section, we study the main signaling delays that take place in a voice call (registration and voice call setup delays) and compare them for different wireless broadband access technologies. Likewise, we present some possible optimizations such as using always-on mode and their drawbacks. Signaling delays and total battery lifetime affect the perceived end user experience directly, and thus are important items to consider in addition to the overall voice quality.

To evaluate VoIP signaling, we carried out VoIP calls and captured signaling packets to determine performance with different wireless access networks (WiFi, WCDMA and HSDPA). Our terminals were identical Nokia N95 devices using the preinstalled embedded VoIP client and with the same configurations. The measurements took place in an interference-free lab environment.

The registration delay, and call setup delay are the two measurements affecting user experience the most (see Figure 19). The registration delay is the time it takes the user to get access to voice services. The call setup delay (post-dial-delay) is the time it takes from the moment the user pushes the call button until the moment the phone rings at the other end. These delays can be cumulative in case the caller is not yet registered to the service prior to making the call.

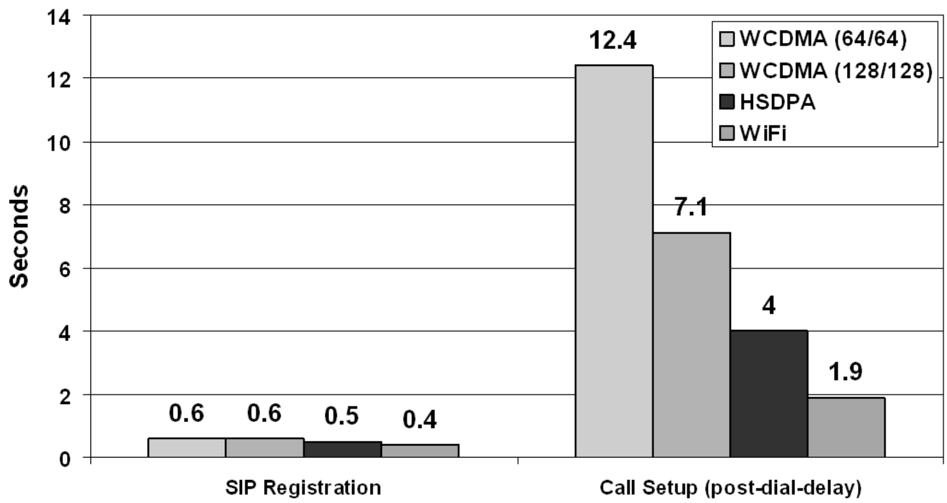

Figure 19. VoIP Signaling Performance

ITU E.721 (ITU-T 1999) recommendation for call setup delay in circuit switched calls is 3 seconds for local, 5 seconds for toll and 8 seconds for international connections, and 6, 8, and 11 seconds with 95% probability. Since all the network elements were located in a private network for our tests, the environment can be thought of as providing local calls. The results show that only HSDPA and WiFi are able to provide performance similar to circuit switched calls. Our results also show that the embedded VoIP client experiences an increased delay compared to a PC client, such as the one measured by (Curcio & Lundan 2002) with a WCDMA network. In addition, our results show the cases in which a 3G data connection (PDP context activation) has been previously established prior to registration or call setup. This latter procedure took ~3 seconds in our tests, while (Pous et al. 2003) simulations give a 2.2 second value. Unless users have always-on connectivity enabled, the three delay values will be cumulative for each call, i.e. PDP context activation, SIP registration, and call setup. Likewise, if always-on is not enabled, the VoIP users are not reachable for incoming calls.

Always-on is a feature that allows mobile devices to keep a packet data connection open even if the user is not actively using any service. With this mode, different services can remain active (VoIP, presence, instant messaging, email, etc.,) allowing the user to be reachable and access services at any given time. In the case of VoIP, a user is not reachable for incoming calls unless he remains registered with the VoIP server. However, maintaining registration to different services has an effect on battery lifetime because the device is periodically sending messages to keep the service connection alive. This is a trivial issue for Laptop clients, but for handheld devices battery lifetime is a critical aspect.

Evaluating the energy consumption and its effect on battery lifetime for always-on connectivity requires understanding two aspects: first, the frequency of keep-alive messages, and second, the time spent in each of the cellular channels to send the keep-alive messages. Each of the cellular channels used to transmit and receive data, CELL_DCH, CELL_FACH and CELL_PCH, consume a different amount of energy. Even though the 3GPP standards allow for different possibilities for transition mechanisms between cellular states, in general only two ways are used (see Figure 20).

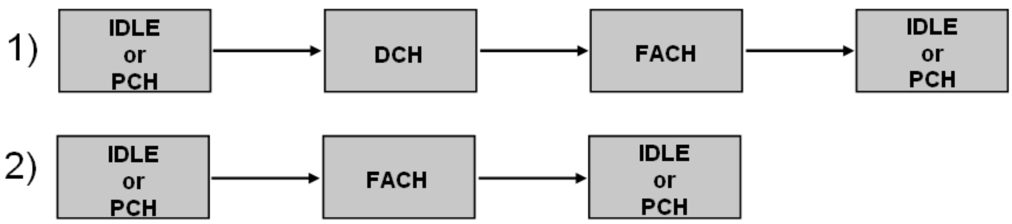

Figure 20. Cellular Cell State Change Transitions

Option 1 in the figure depicts the case in data transmission which uses a dedicated channel (CELL_DCH). The CELL_DCH state achieves maximum throughput and minimum delay, but at the cost of higher energy consumption. Direct changes to CELL_FACH from IDLE or CELL_PCH (option 2 in the figure) are only possible if only a very small amount of data is transmitted (below 500 Bytes). However, SIP messages used for VoIP are usually larger than 500Bytes and thus will trigger changes to CELL_DCH most of the time. Further, the amount of time spent in each of the cell states is defined in the radio access network radio resource control. If settings are set too long the terminal will remain in CELL_DCH or CELL_FACH states longer than what is required for transmission. Operator configurations can vary much. Particularly for broadband networks, it should be emphasized that these parameters should be set as low as possible.

Our experiment focused first on measuring the energy consumption in each of the cellular states and measuring the time spent on each of them as well. Subsequently, we captured signaling VoIP packets to understand the frequency of keep-alive messages, which was every 50 minutes. The time spent in the different cellular states in the network used for our study had default parameters of 10 seconds for CELL_DCH and 5 seconds for CELL_FACH. This setup is not optimal for a broadband network, since due to the high throughput achievable, data can be fully sent in much shorter periods. However, settings with long timeout configurations are quite common, since previous 2G and WCDMA networks required a much longer time to transmit data. We propose operators instead use 3 seconds for both CELL_DCH and CELL_FACH states. This parameter optimization has a great impact on the battery lifetime when always-on applications are used. Figure 21 shows the impact of parameter optimization and its effect on battery lifetime. In addition, a hypothetical scenario in which all messaging could be transmitted in CELL_FACH is also presented. The reason for considering this latter option is that it is proposed that future 3GPP standard releases allow transmitting up to 1Mbps in CELL_FACH state, which would be enough for SIP messages.

Model based on:
Cell state consumption
RRC parameters
Keep-Alive Intervals

Lifetime Decrease

Keep-Alive Frequency	DCH 10s FACH 5s	DCH 3s FACH 3s	FACH 3s
1 min	95%	85%	70%
2 min	89%	75%	53%
5 min	76%	54%	31%
10 min	62%	37%	18%

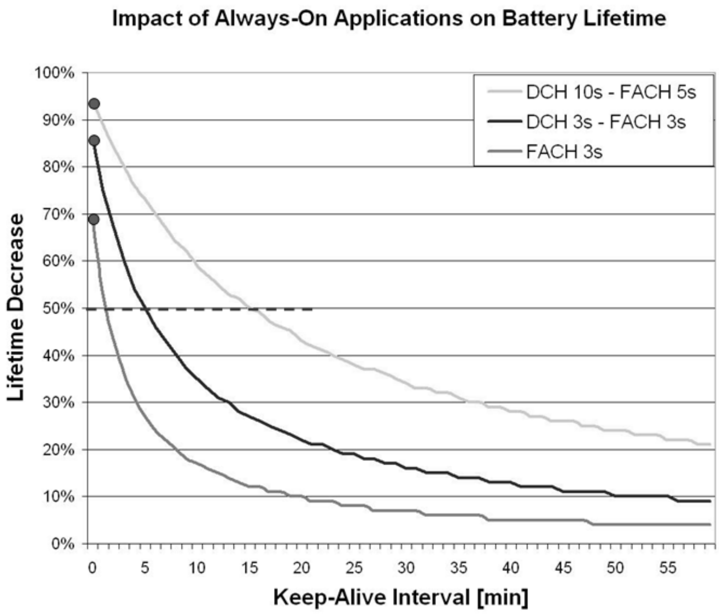

Figure 21. Decrease in Battery Lifetime

When always-on is used for VoIP, it will reduce battery lifetime by 10% in an optimized network. However, if the network is not optimized it could reduce lifetime up to 25-30%**.** This energy consumption is only for keeping the service available, making VoIP calls decrease battery lifetime further. Other applications such as presence, push email, and Push-to-Talk also require periodic messages. Likewise, some network elements such as network address translators (NAT) also require periodic keep-alive messages. Common default timeout settings in NATs are very short, 5-10 minutes for

TCP and 1 minute or less for UDP. Keep-alive messages over UDP are commonly required in connections that involve virtual private network (VPN) gateways, such as enterprise push e-mail. The maximum timeouts possible depend on the NAT manufacturer, e.g., 6min for Cisco (Cisco Systems n.d.; Haverinen, Siren & Eronen 2007). These short intervals have a big impact on energy consumption and decrease battery life considerably. Unfortunately, with such short intervals, battery lifetime will decrease considerably regardless of operator radio resource configuration. Furthermore, when different applications are run at the same time, their required keep-alive messages are not coordinated, which means that energy consumption would be even higher than what we have presented in this study.

5 Summary and Conclusions

This thesis considered the performance evaluation of VoIP in wireless broadband systems. The performance was evaluated using theoretical user experience models developed by ITU-T and using empirical data from field and laboratory measurements. The results concluded that while the network characteristics of several of the wireless networks considered is good enough to provide toll quality or near toll quality voice in static scenarios, there are still a number of problems which make it currently unfeasible to use as a primary voice service. In addition, under mobility scenarios, performance can be degraded further. This study provided a broad perspective on the performance of VoIP as well as the main limitations, challenges and future directions. In this section we provide a summary of the results, followed by a generalization of the results and a proposal for future work.

VoIP over wireless broadband networks is not commercially feasible with current state of the art wireless networks. The reason is that there are still a number of challenges that must be resolved prior to rolling out a VoIP solution over any of these systems. In the case of WiFi mesh the main limitations are coverage, interference outdoors and network deployment price. Improving coverage would require a large investment, and large deployments using multi-radio solutions suffer the same problem. Furthermore, it is still not yet proven if running these type of networks is profitable and under what business model. The current trend is moving towards specialized use cases rather than city wide deployments. In the case of 3G cellular systems, VoIP over WCDMA is possible but at very low quality even for laptop cases. With the use of the RLC UNACK mode it can improve to some extent. However quality remains low. As for HSDPA/HSUPA the radio performance is in general adequate for static scenarios and suboptimal in mobile cases. The RLC retransmissions are one of the main reasons for limited performance. Further, handsets provide many additional challenges, in particular regarding large processing delays. Once the RLC UNACK mode is available, it will be also possible to reduce the jitter buffer implementations in the VoIP clients. These two items alone can improve performance considerably. The next step is to decrease the processing delay. HSPA+ is likely to encounter the same handset limitations as HSDPA/HSUPA and therefore improving handsets performance is very relevant for future networks as well. Some features in Release 7 aim directly at improving VoIP quality; however, most likely VoIP will begin to be a feasible option only for subsidized VoIP. Third party VoIP solutions will not benefit too much from Release 7 compared to subsidized VoIP because operators will be able to restrict the improved features only to their own voice solutions. Finally, in regard to FLASH-OFDM, VoIP performance is fine for laptops, both in static and mobile scenarios. The main limitation is related to capacity and the lack of handsets. The limited capacity makes it unfeasible for densely populated areas. However, it is already deployed for some specific medical and maritime voice communications in rural areas in Finland. This solution should be seen instead as a replacement for ADSL in rural areas rather than as a competitor of cellular networks, or as an alternative for deployments requiring high speed mobility support.

Our research also proved that VoIP over wireless broadband networks is not a real contender for cellular circuit switched voice. WiFi Mesh is not a contender due to the performance issues and very high cost. FLASH-OFDM can be seen as an option only in very few scattered special cases. HSDPA/HSUPA/HSPA+ still require a significant number of improvements before they begin to be considered a substitute for legacy circuit switched voice. Likewise, some additional options such as Release 7 CS over HSPA can delay VoIP deployments even further. In the future, deployments in the 900MHz band are of particular interest due to the very large possible cell areas. However, the advantages of the 900MHz band are not that evident for densely populated areas compared to e.g., 1800 and 2100 frequency bands. Likewise, as mentioned in this thesis, 3GPP and operators are still considering the large use of CS-based voice, even after LTE becomes available. The business dynamics show a trend towards coexistence of both circuit switched voice and VoIP for a long time still. However, eventually, in the very long run, networks will evolve into VoIP service only.

With regard to the performance of multi-radio interworking solutions for VoIP services, UMA proved to be a viable option with performance comparable to circuit switched handovers. MIPv4 performance is still low compared to circuit switched handovers. However, it is similar to performance found in enterprise WiFi VoIP deployments. MIPv4 performance will improve considerably with HSPA+. In addition, MIPv4 enables use cases in which operators own only packet based networks such as WiFi Mesh and HSDPA (e.g., Terrestar network in the USA).

The other performance limitations affecting mobile VoIP that we identified are signaling performance, battery lifetime and capacity. Signaling performance (e.g., call setup delay) for VoIP in cellular systems is not as good as fixed or WiFi. The call setup time is significantly higher than with traditional circuit switched voice. Therefore, the user experience is different. In addition, the lower battery lifetime is an issue. In order to improve battery lifetime it is required to configure networks, modify applications, and apply new features. This is a major issue that affects overall user satisfaction. The performance goal for energy consumption should focus on bringing battery lifetime when using VoIP to a similar level as when using traditional circuit switched voice. Finally, VoIP in cellular does not provide significant higher capacity than circuit switched voice unless header compression is enabled. However, header compression increases processing delays in terminals further. Therefore, its use is not yet feasible.

5.1 Generalization of Results

This thesis research work aimed at evaluating VoIP performance and the opportunities as well as disruption that VoIP could bring to cellular operators' business and networks. Our work started before any generic VoIP client had become commercially available for mass market handsets and the availability of HSDPA networks was very limited. However, in the last year, several changes have occurred in the strategy of network operators, device makers, and software developers and startup companies. In this thesis we showed that the main issue for adoption of VoIP as a replacement of circuit switched voice service was three-fold, first, poor voice quality and performance, second, severe battery lifetime limitations, and third, unavailability of widely deployed HSPDA or WiFi networks. If we consider the current market situation and trends, we can notice that the results and conclusions of this thesis are still valid and are likely to remain valid for the next few years.

First, in regard to the unavailability of HSPDA networks, this has changed dramatically in the last few months and a large number of operators are quickly trying to upgrade and deploy HSDPA and bring higher capacity through future improvements (i.e. HSPA+) in their current networks. However, the reason is not to allow or roll out VoIP services as was planned a couple years before, but to take into account the large increase in data traffic. In many networks, up to 90% of the whole data traffic is driven by computers. Moreover, with the introduction of low cost netbooks (e.g., Asus EePC), using modems for wireless access is increasing considerably. Out of the total traffic use, videos and peer-to-peer data account for up to 60-70% of all data usage in some networks. Furthermore, private consumers are the ones that have driven this increase, by consuming roughly five times more than business users.

Second, VoIP clients are even less available than before. Previously, some Nokia N-Series and E-Series devices had a native VoIP client installed. However, this is not the case anymore since November 2008. Although Nokia did not made public the reason for this, technology analysts believe that it is due to poor VoIP performance and conflict of interest with some operators that are against the use of VoIP in their networks. Additionally, highly expected VoIP clients (i.e. Skype) did not become available. Instead, Skype released a handset client that can provide access to other users in Skype and in cellular and fixed networks seamlessly. However, the supporting technology is not VoIP. The approach actually uses traditional circuit switched voice for the mobile user's access, and dials into a gateway service that is reachable via a local number. Skype has a large number of local numbers for different countries for this purpose. Once the call is placed, the Skype gateway transcodes and packages voice as VoIP to traverse the call through Internet, allowing a low cost service. This is not a new technology and has been available for several years in many countries as "International calling cards". With these cards, users must dial a local number first and the intended International number next. The call is local up to the VoIP gateway, which then transcodes to VoIP in order to traverse the call via Internet. The main difference with the Skype handheld client is that it has integrated the list of contacts and the local access Skype numbers within the client. Therefore, the user is totally unaware of the technology behind the call. Another difference is that registration, login to Skype, and chat services use the data access network. Thus, the Skype client uses both circuit switched access and data access (Skype Forum n.d.). The Skype client implementation differs in some parts between Symbian, iPhone and Blackberry devices; for instance on iPhone the client is run on a browser using Ajax. Nevertheless, all three platform clients use circuit switched for the mobile wireless side and not VoIP. Figure 22 shows the logic behind the Skype service for handsets.

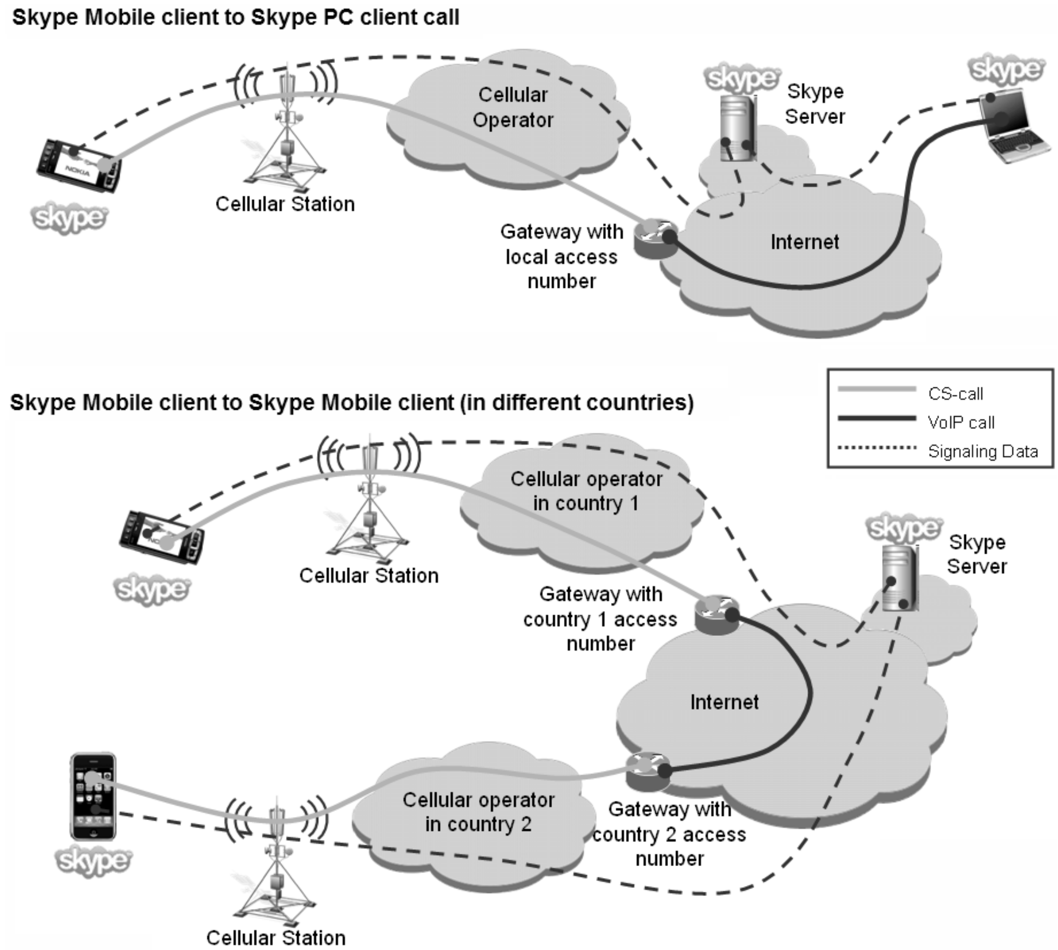

Figure 22. Skype Mobile Calls

Although Nokia does not include a native VoIP client in their devices and Skype clients use circuit switched voice, it does not mean that there are no longer handheld VoIP clients. However, using VoIP end-to-end with handhelds is even more of a niche market than before. Out of the remaining VoIP clients available, Fring, is one of the few handheld clients that allow VoIP calls over 2G/3G/WiFi between Fring and Skype users. However, the performance and voice quality is worse than with the previous native VoIP client in Nokia devices which was evaluated in this thesis.

Third, an additional concern with VoIP clients has been the energy consumption and decrease in battery lifetime. As presented in the thesis results, this is due to the constant keep-alive messages required to keep VoIP users reachable and logged into the voice service. This is still an issue with no clear solution yet. The only improvements have been for operator sponsored VoIP services, in which the 3GPP Release 7 techniques can be used to improve this matter for the operator's own VoIP service. These features however, do not improve the energy consumption of the incumbent or the 3rd party VoIP service provider.

Finally, even with the deployment of LTE, the use of VoIP in handsets is not considered to be a mainstream service. Operators have not yet reached a consensus for their voice strategy. Thus, it is unclear whether VoIP will be even used in LTE from the beginning. For this reason two features to account for both cases were included in the 3GPP LTE standardization. One of the requirements for LTE standards was CS-fallback. This feature allows the full separation of voice and data services. That is, LTE terminals still use 2G or 3G circuit switched voice, and LTE technology only for data traffic. This requirement is due to the fact that handsets' performance is still not optimal and cannot fulfill the requirements for high quality voice, confirming the results presented in this thesis. In addition, ensuring quality of service for packet voice across network elements and radio access is much more difficult than it is for traditional circuit switched voice services. One drawback of this solution is that once the LTE access is replaced with circuit switched voice, data connections are released. That is, simultaneous use of voice and data services is not possible. The other important feature in LTE standards is Single Radio Voice Call Continuity (SR-VCC). The approach in this case is to take account of the lack of coverage of LTE networks. The logic of this feature takes into consideration the possibility of VoIP being used in LTE networks. However, due to the poor performance of VoIP service in 3G and HSDPA networks, the approach is to switch from LTE-VoIP to 3G/HSDPA CS-voice whenever LTE coverage is no longer available. This is very similar to the idea in UMA technology, except that it uses the same radio. Further, an additional approach of bringing UMA to LTE is under discussion, known as the VoLGA forum. Thus, at the current moment the possible strategy for voice in LTE networks is rather unclear.

5.2 Future Work

VoIP over WiFi mesh can have beneficial results under a completely different business case. As was noted, WiFi mesh is neither robust nor capable of competing with cellular voice. However, there have been some studies in which WiFi mesh is aimed at rural deployments in emerging markets (e.g., India and Africa) rather than mature markets. In these cases, the goal is to provide as low a cost network as possible using open software and equipment. In some cases these markets do not have any voice services available. Under these circumstances, the feasibility of VoIP over WiFi mesh takes on a different perspective. For instance, coverage is developed only in certain areas, and access points provide mesh capabilities based on open software. Likewise, VoIP calls are intended to provide communications within the community only. This is possible by using an open source Asterisk PBX. Future studies based on these scenarios are of interest.

Performance of VoIP over HSDPA and HSUPA should be verified once the UNACK mode is available in network equipment. If the feature is available, VoIP clients can be implemented with a shorter jitter buffer. These two items should be able to provide better quality despite the possible processing power limitations in handsets. Likewise, performance with Release 7 features should be evaluated once equipment is available as well.

As for intersystem aspects it is relevant to study ways in which MIPv6 performance could be improved. Eventually, networks will be based on IPv6 rather than IPv4. The low performance of MIPv6 compared to MIPv4 could become problematic once IPv6 becomes more available.

With regard to battery lifetime and energy consumption different techniques should be envisioned for always-on applications. For instance, software applications could try to simultaneously send the keep-alive messages rather than managing them individually. In addition, another option is to develop a software proxy running on the device which would handle all the keep-alive messages on behalf of the other applications. Moreover, network element manufacturers could take into consideration the impact of having such short timers as default in their VPN and NAT equipments.

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ISBN 978-952-248-102-3
ISBN 978-952-248-103-0 (PDF)
ISSN 1795-2239
ISSN 1795-4584 (PDF)