

Field Test of Mobile Terminals in a Wireless City

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Problem Description

"Wireless Trondheim" is a wireless city where a variety of services can be offered and accessed by different terminals. In this assignment the focus is on small, handheld terminals and on a limited set of services for instance IP telephony (e.g. Skype/VoIP), messaging service, email access. It is expected that the Wi-Fi enabled handheld terminals in the market have different functional capabilities and quality. The main objective of this assignment is to specify benchmark-testing criteria of terminals with respect to the "Wireless Trondheim" configuration. The field test should include testing of quality of service for at least one service under different network and security configurations and different traffic conditions.

The assignment includes:

- identify test service (e.g. IP telephony) and specify performance test criteria
- specify and select wireless terminals and test platform
- conduct series of field experiments for (at least) one service under different conditions
- evaluate and discuss the test results with respect to the quality of the tested service
- compare quality of service results up against results from GSM to answer the question if today's Wi-Fi terminals provide good enough voice quality to compete with GSM

Assignment given: 20. January 2007

Supervisor: Poul Einar Heegaard, ITEM

Preface

This Master's thesis is the final product of a five year study in Communication Technology at the Norwegian University of Science and Technology. The work for the thesis has been carried out from January to June 2007.

I would like to thank my Professor Poul E. Heegaard for our weekly meetings and his valuable input and comments throughout the work.

My supervisor, and manager of Wireless Trondheim, Thomas Jelle deserves a huge thanks for providing me with necessary information and interesting discussions.

I would also like to thank Jardar Leira and Rune Sydskjør at Uninett for their help getting me started with the network analysis tools used in this thesis. Jardar also enabled the tests at Uninett and frequency analysis and deserves a large thanks for valuable input and knowledge.

Malcolm Lee at Ixia also deserves a lot of thanks for outstanding email support with the network analysis tool IxChariot. Thanks to Ground Control Labs for lending me mobile Wi-Fi terminals to test. I would also like to send a thanks to fellow student Sverre Winsnes Rødland for data and information on indoor WLAN coverage in Wireless Trondheim.

Trondheim, June 15, 2007

Petter Stray

Abstract

A rising question today is whether or not wireless networks and terminals are at the border of being able to compete with the cellular phone service. A variety of terminals are available and citywide wireless networks are either already deployed or under deployment in many cities worldwide. Although voice over wireless networks already is up and running in many office buildings and hospitals, few have experimented with and tested the use of voice over IP in public wireless networks.

In this thesis, a series of field experiments are conducted in "Wireless Trondheim", a city-sized wireless network. Through tests gathering information on voice quality, network capacity and other metrics critical for a voice service, differences between terminals and the state of the technology is presented. Using the network analysis tool IxChariot a selection of Wi-Fi enabled mobile terminals from Qtek and HTC are tested under different conditions and network loads.

The tests unveil vast differences among the tested terminals. While some terminals are capable of handling multiple conversations at once (e.g. call waiting and teleconference functionality) others have trouble keeping the quality of a single conversation good enough for it to be of any value. Allover, the achieved voice quality for the tested terminals in the Wireless Trondheim network lies well below the quality of the GSM service. The radios on the mobile Wi-Fi terminals are strongly affected by interference in a densely populated outdoor environment, which makes it difficult to maintain good voice quality. The results obtained in the thesis indicate that the tested mobile terminals are not yet ready to deliver telephony over a shared outdoor wireless network with sufficient voice quality.

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CHAPTER 1

Introduction

1.1 Motivation

Over the last two decades the mobile phone has gone from being a tool for the few to a necessity for the public. More and more people terminate their land line subscriptions, and completely rely on a cell phone for communication. This even though the experienced voice quality on a mobile phone is substantially lower than that on the public switched telephone network (PSTN). People accept this degradation because they value the gain in mobility higher than the loss in quality.

Today many people mean that Voice over the Internet Protocol (VoIP) poses a threat to mobile services as we know them. As entire cities become giant wireless zones it is expected that these also support Voice over Wireless Local Area Networks (VoWLAN). If these large scale networks provide the necessary resources to obtain voice quality at a sufficient level, people are likely to use this as their primary communication network. The advantages are many; "limitless" downloading, free Voice over IP through Skype, GoogleTalk a.o. and the possibility to simultaneously send multiple types of traffic over the same network.

In Trondheim such a citywide wireless network has been deployed on the initiative of the Norwegian University of Science and Technology (NTNU). The network is built with special weight on providing more than enough bandwidth so such services as Voice over Wireless LAN are providable. However, since wireless cities are just starting to open to the public, little research and testing is done on the services and equipment to be used in the networks. Testing different mobile Wi-Fi terminals in

Wireless Trondheim under various network conditions can help give an indication of what works best in a system as the one deployed here. Comparing the quality of a mobile voice service up against the quality of the already ubiquitous GSM service can be useful to see if the potential is present to take voice over citywide wireless networks into use.

1.2 Methodology

The purpose of this Master thesis is to find benchmark testing criteria for mobile Wi-Fi terminals with respect to Voice over WLAN. To establish these benchmarks, tests will be carried out in the Wireless Trondheim configuration using mobile terminals available on the Norwegian market today. The test results will give indicators of what works best in a configuration as the one in Trondheim. The test results will also be compared up against similar results from GSM networks. This is to evaluate if the currently available wireless technologies can be a competitor to GSM and possibly steal market share.

1.3 Scope

This thesis places emphasis on testing wireless fidelity (Wi-Fi) mobile devices in the Wireless Trondheim network. Different results than those obtained here can be the outfall by testing in another network. Testing a different set of terminals may also give different results, but neither another network or different terminals were accessible during the work.

All tests are carried out without any Quality of Service (QoS) mechanisms or traffic shaping. The same security features are used throughout the tests.

1.4 Related Work

Network World is the leading provider of news, analysis, reviews, events and education on information technology [45]. Network World publishes the newsweekly *Network World*, and hosts the active online communities NetworkWorld.com, LinuxWorld.com and JavaWorld.com. Being recognized as the leader in network knowledge [45], the companies reviews are considered as the network markets premier objective and authoritative source of product information.

In late 2005 Network World did a review on voice over WLAN. The goal of the test

was to uncover which of four major infrastructure vendors' WLAN switches and access points provided the best audio quality, QOS enforcement, roaming capabilities, and system features [32]. The systems under test were Aruba Wireless Networks, Chantry Networks (now Siemens), Cisco and Colubris Networks. Among their major findings where [32]:

- All products provided near-toll-quality audio with QOS enforcement enabled, as long as there was only voice traffic in the network.
- When voice traffic had to contend for bandwidth, dropped calls were common and audio quality was poor for the successful calls.
- When roaming succeeded, which was uncommon, it took anywhere from 0.5 to 10 seconds.

Although this test focused on the performance of the different infrastructures instead of different terminals, it has many similarities to the work in this thesis. Both tests:

- look at the same network measures; packet loss, jitter and delay.
- study how an increasing number of calls affects audio quality.
- look at how the introduction of data traffic degrades audio quality.
- evaluate the roaming capabilities of voice over WLAN.

The tests performed by Network World however took place in a lab with identical terminals connected to the system. This thesis' tests are performed in an actual network with realistic noise and background traffic environments, as well as testing a variety of terminals. For further reading about the tests performed by Network World the reader is referred to [32].

Another performance analysis of voice over WLAN was performed at the *National Institute of Information and Communication Technology* and *Japan Advanced Institute of Science and Technology* in 2006. In the tests VoIP using the G.711 codec was studied in an emulated WLAN environment. Emulating the WLAN environment enabled the researchers to study a range of controllable network conditions and measure the voice quality degradations using both the E-model (chapter 4.2) and the Perceptual Evaluation of Speech Quality (chapter 4.3). Their results point out degradations in User Perceived Quality (UPQ) as a result of events in the physical world, such as packet losses and decreased voice quality caused by signal power degradations. For further reading about the tests the reader is referred to [1].

1.5 Readers Guide

This thesis is divided into four main parts; a Prestudy part, a Test Implementation part, an Evaluation part, and an Appendix part.

The prestudy begins with an introduction of benchmarking and the variables therein treated throughout the thesis in chapter 2. Chapter 3 gives an overview of how wireless networks can be tested and evaluated. The following chapter, chapter 4, introduces the most used voice quality evaluation techniques. Chapter 5 describes the voice codecs used in this thesis, while chapter 6 gives an overview over Wireless Trondheim and the infrastructure and possibilities in the network.

The test implementation part starts with chapter 7 giving an introduction to how the tests are carried out. Chapter 8 describes the network analysis tools used in the tests, and chapter 9 gives a quick overview of the hardware components used in the tests. In chapter 10 a description of the various performance metrics used to evaluate voice quality is given.

In the evaluation part a presentation of the results obtained in the tests is done in chapter 11. Chapter 12 discusses the results presented in chapter 11, while chapter 13 concludes the thesis. Further tests that can be done in the future are proposed in chapter 14.

The Appendix contains detailed information on the various terminals tested in the thesis. The digital references disc contains volatile references as well as test setups and results from the tests.

PART I

Prestudy

CHAPTER 2

Benchmarking

Benchmarking is the process of comparing one product's or service's performance up against the performance of other products or services in various areas [3]. Successful benchmarking strongly relies on holding as many variables as possible constant in order to isolate the elements being compared. Benchmarking wireless networks, however, involves a potentially broad range of variables that are not possible to hold constant [10]. For this reason wireless benchmarks are often run in environments where external radio frequency (RF) energy can be blocked out [10]. In the case of this thesis however, the intention is to see how different terminals perform in an actual citywide wireless deployment. The following sections will discuss some of the variables that affect the performance of mobile Wi-Fi terminals in a wireless network.

2.1 Multiple Clients

Mobile Wi-Fi terminals have different performance characteristics and will perform differently in a given wireless environment. Terminals have varying processing power, battery power and radiated power to mention a few terminal specific variables. The terminals tested in this thesis vary in all these areas, and each terminal therefore has the potential to give different results. All terminals tested are so-called hybrid terminals with both Wi-Fi and GSM radios. Using pure Wi-Fi terminals in the tests may give different results.

2.2 Network Infrastructure

The network platform is one of the variables kept constant in this thesis. The type of antenna, and antenna orientation, influences the received signal quality at the terminals location. Since all terminals are tested in the same citywide wireless network in Trondheim, and the terminals are located at the same spot, connected to the same access point during the tests, the network infrastructure can be considered as a constant in the benchmarking.

2.3 Environmental Influences

The environment wireless tests are performed in strongly affects the obtained results. Factors as neighboring walls, the weather and passing crowds influence how signals diverge over the air. Radio propagation is highly dependent upon the physical environment as well as the geometric relationship between the transmitter and receiver [10]. For this reason tests are performed at different locations in the city to have as realistic surroundings as possible. The weather is not possible to control, so tests are carried out under as even weather conditions as possible.

2.4 Security

The use of different security mechanisms can also strongly affect the outcome of wireless tests. For example, application layer security affects the transmitted data differently than data link layer security. For this reason the same security mechanism is used throughout the tests in this thesis.

2.5 Mobility

Mobility is one of the variables in a benchmarking test it is hardest to keep constant, if there is mobility of course. Mobility testing in a wireless network will in most cases involve roaming between access point. Simple roaming tests are performed in this thesis, but are not highly weighted because of repeatability issues. In the majority of the tests the mobility variable is kept constant by having the terminals stay at the exact same location through the course of each test.

CHAPTER 3

Wireless Network Testing

Currently there are no standardized ways to test wireless networks. The closest you come to a standard is Institute of Electrical and Electronics Engineers (IEEE) recommendation 802.11T, also referred to as the Wireless Performance Prediction (WPP) test methods and metrics recommendation.

3.1 IEEE 802.11T

When testing wireless networks and equipment it is necessary to evaluate some specific metrics to be able to compare performance among different solutions. For this reason the IEEE formed the IEEE 802.11T Task Group to develop a test specification document, "Recommended Practice for the Evaluation of 802.11 Wireless Performance". This document defines test metrics, principally for three use cases; data, latency sensitive and streaming media. [30]

3.1.1 Data

Data applications, such as e-mail, Web traffic and file sharing, do not have critical requirements to the network, and the generated traffic is typically considered as low priority traffic. Important metrics for data traffic are; the relationship between throughput and range, number of clients handled by the access point and access point throughput per client [30].

3.1.2 Latency Sensitive

Latency sensitive applications are time-critical and require a lot from the network. For applications such as Voice over Wireless Networks (VoWLAN), certain Quality of Service (QoS) requirements must be met. This includes limits on voice quality vs. range, voice quality vs. network load, and voice quality vs. call load [30].

3.1.3 Streaming Media

Streaming media applications include real-time audio/video streaming, stored content streaming and multicast high-definition television streaming [30]. These applications have stringent QoS demands. Performance metrics include video quality vs. range and video quality vs. network load [30].

This thesis looks at the latency sensitive voice service and will therefore focus on voice quality metrics under different network conditions.

3.2 Testing Alternatives

Wireless testing can be carried out using two different techniques; conducted (controlled RF) or over-the-air (open air). Beneficial with testing in the controlled RF environment is that each test is repeatable and controllable. Interference can be shielded out or filtered to achieve device-to-device isolation [31]. However, tests carried out in controlled RF environments are not completely accurate. They can give an indication of how the technology will work in a real deployment, but it will never be able to correctly imitate all external factors that affect a networks performance characteristics. Such external factors can be the weather, interference from electronic radiation and people walking around the equipment. This thesis will make use of over-the-air testing in the Wireless Trondheim deployment. There are a few different actors that provide software and/or hardware to perform over-the-air testing, two of which are introduced below.

3.2.1 VeriWave WLAN Performance Testing

VeriWave is a provider of performance analysis tools for WLAN-equipment and network testing. Their products give wireless network infrastructure producers a tool to accurately analyze the performance of their products. The tools are also used by carriers and enterprises to help them choose the best equipment for their network

needs.

The cornerstone of VeriWaves solution is the WaveTest Traffic Generator and Performance analyzer. The test system consists of a WaveTest chassis that can contain up to nine independent traffic generators/performance analyzers, called WaveBlades. Each WaveBlade can generate up to 500 stateful WLAN clients across a single or multiple subnets. Each client can generate multiple traffic flows at the data, network or transport layers. Detailed information can be captured down to the packet level and presented to the tester through the WaveManager interface. [42]

In order to use VeriWave's performance testing tool you need the VeriWave WaveTest chassis and at least one WaveBlade card. These are fairly expensive hardware components which makes VeriWaves solution unsuitable for this thesis. Additionally, the mobile clients are the main focus of this thesis' tests, not as much the network infrastructure components.

It seems as if another performance analysis tool is more suited, namely IxChariot.

3.2.2 Ixia's WLAN Performance Assessment Tool

Ixia is a provider of performance test systems for IP-based infrastructure and services. Service providers, system vendors and manufacturers, and enterprises use their tools to test the performance of complex IP networks, devices and applications.

Ixia's major solution is the IxChariot product family. Comprised of an IxChariot Console and Performance Endpoints, IxChariot can offer thorough performance assessment and device testing [27]. By emulating nearly all network protocols, the performance of thousands of network endpoints can be tested. The Performance Endpoints run on most platforms, while the console runs on Windows only [27]. Either by creating your own scripts, or running one of the 140 pre-programmed scripts, real-world application behaviour at the transport layer is achieved while detailed performance data are collected. The built-in scripts include VoIP using different codecs, FTP data, and streaming media.

IxChariot is a \$30,000 tool, where only the software is needed from Ixia. Considering that Wireless Trondheim already has this tool at their disposal, the choice of network analysis tool to use in this thesis is easy. The tool's possibilities to emulate real-world application traffic also supports the choice, enabling the evaluation of individual terminal performance.

Evaluating Voice Quality

Comparing the quality of data networks is a complex task, since there are many factors to consider. Different applications weight a networks performance metrics in different ways. Some applications may require high bandwidth, but are less sensitive to variations in the end-to-end delay. In the telephony world, a single metric has been established to rate call quality - the Mean Opinion Score (MOS). For Voice over IP, the E-model provides a number comparable to the metric from the telephony world.

4.1 Mean Opinion Score

Assessing call quality has traditionally been a subjective task; picking up a telephone and listening to the quality of the voice. The most widespread subjective voice quality metric is the Mean Opinion Score (MOS) described in the International Telecommunications Union (ITU) recommendation P.800 [33]. The MOS for a call service is calculated from letting a large number of people listen to audio and give their opinion of the call quality on a scale from 1 to 5. Each score has a description related to it as illustrated in table 4.1.

The recommendation describes everything from test facilities to detailed test procedures, including script to read. Following these recommendations however, is both costly and time-consuming, and requires a number of people testing every little tuning adjustment. As a consequence the abbreviation is also used for scores that originate from objective models or network planning models [34]. Considerable work has been made to establish objective measurement techniques, one of them being the E-model.

Rating	Definition	Description
5	Excellent	a perfect speech signal recorded in a quiet booth
4	Good	intelligent and natural like (PSTN) telephone quality
3	Fair	communication quality, but requires some hearing effort
2	Poor	low quality and hard to understand the speech
1	Bad	unclear speech, breakdown

Table 4.1: ITU P.800 MOS conversation opinion scale [33].

4.2 The E-model

The E-model was introduced in the ITU recommendation G.107. The E-model uses measured delays and equipment impairment factors to calculate a single scalar, the "R factor". The R factor is given by the equation [4]:

$$R = R_0 - I_s - I_d - I_e(+A) \tag{4.1}$$

where:

- R_0 : unaltered signal, expresses the basic signal-to-noise ratio (SNR)
- I_s : impairments that occur simultaneously with the voice signal, such as too loud speech level
- I_d : delays introduced from end-to-end
- I_e : impairment introduced by the equipment
- A : "advantage factor", willingness to trade voice quality for convenience

The advantage factor is in parenthesis in equation (4.1) because it is not regarded by the network analysis tool used. These delay and equipment impairments are influenced by the data networks one-way delay, jitter and data loss. Implicitly the codec used also influences the delay and impairments, especially if a compression codec is used.

The R factor ranges from 0 to 100, and can easily be translated into a corresponding MOS value. When a voice conversation is converted to a network signal and back there is an inherent degradation. This reduces the theoretical maximum R factor with no impairments from 100 down to 93.2 [26]. The translation from R factor values to MOS values is illustrated in figures 4.1 and 4.2. In figure 4.1 you see the R factor

values from the E-model to the left, the likely opinion of human listeners in the middle and MOS values to the right.



Figure 4.1: Translation from the objective R factor to the subjective MOS value[26].

Figure 4.2 illustrates how a network analysis tool can calculate the MOS value for a given conversation. First raw data metrics such as delay and packet loss are collected. These data are then inserted into the E-model equation (4.1). The obtained R factor from the E-model is finally translated to a MOS value.

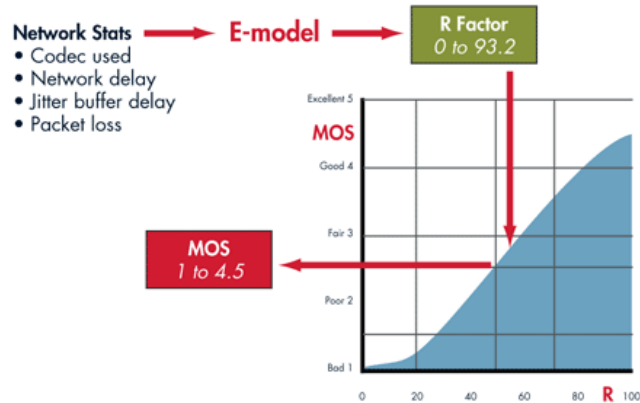


Figure 4.2: Conversion from R factor to MOS [26].

The different analysis tools have their own way of calculating MOS score from R factor values. The tool used in this thesis calculates MOS values in the following way:

$$\begin{aligned}
 MOS &= 1.0 & R < 0 \\
 MOS &= 1 + 0.035R + R(R - 60)(100 - R) \times 7 \cdot 10^{-6} & 0 < R < 100 \\
 MOS &= 4.5 & R > 100
 \end{aligned} \tag{4.2}$$

In the calculations, R is the R factor calculated using the E-model. To illustrate the scale of the MOS, toll quality voice has a MOS of approximately 4.0, while GSM has an approximate MOS of 3.7 [43].

4.2.1 Codec

The choice of voice codec strongly affects the achievable voice quality in a Voice over IP (VoIP) session. On one side you have codecs like G.711 that causes no deterioration of the voice quality, introduces the least delay and is fairly robust to datagram loss [26]. On the other side you have codecs like iLBC that with no packet loss only achieves a MOS of approximately 3.8. Codecs are further discussed in chapter 5. See table 4.2 for an overview over some of the most commonly used voice codecs, and their influence on the voice quality.

Codec	Bit rate (kbps)	Frame time (ms)	Codec impairment
G.711	64,0	10	0
G.729	8,0	10	11
iLBC	13,3	30	20
	15,2	20	

Table 4.2: Overview of common codecs.

The codec impairments in table 4.2 are inserted directly in the I_e portion of the E-model equation (4.1). Using the G.729 codec will therefore without any other impairments reduce the MOS value from 4.42 to 4.13. Indirectly the codec will also affect the one-way delay.

4.2.2 One-way Delay

The one-way delay in a data network is the time it takes a data packet to get across the network. In a voice setting this is the time from one person starts speaking till the person in the other end hears what's being said. Most listeners notice even small delays only exceeding 150ms, and start finding the delay disturbing when it goes beyond 200ms.

The measured delay in a system is made up of four components [26]:

- **Propagation delay:** the time to travel end-to-end across the network. The propagation delay between Oslo and Tokyo is longer than the delay between Oslo and London.
- **Transport delay:** the time to get through the network devices along the path. Networks with many routers and firewalls takes longer to traverse than a simple LAN.
- **Packetization delay:** the time for the codec to digitize the analog signal and build frames, and undo at the receiving end. The iLBC codec has a higher packetization delay than the G.711 codec, because compression and decompression takes time.
- **Jitter buffer delay:** the delay introduced by the receiver to hold one or more datagrams, to damp variations in arrival times

To calculate I_d in equation (4.1) the sum of these delays are processed through the E-model. Since the two streams in a VoIP call can take different paths through the network, it is not sufficient to calculate the round-trip delay and divide this by two [26].

4.2.3 Jitter

Jitter is the variability in datagram arrival time at the receiver. Some packets travel faster through a network than others, and it is these variations the jitter value tries to capture. A VoIP application sends datagrams at a periodic rate, say every 20 or 30 ms. However, at the receiving side the datagrams arrive with more fluctuating periodicity. A way to deal with this variation is using a jitter buffer. A jitter buffer holds the datagrams at the receiving end, and can compensate for variability of arrival times and also deal with datagrams arriving out of order [26]. A buffer will however, as indicated in 4.2.2, increase the delay. Another problem the introduction of a jitter buffer can lead to is the dropping of datagrams in case of a buffer overflow. Jitter is such an important factor in determining voice quality, that already with jitter exceeding 60 ms the quality of the audio starts to suffer [32].

4.2.4 Lost data

In a VoIP call, packet loss is noticed as small gaps in the conversation. A single, or a few consecutive, lost datagrams are seldom noticeable to the listener. If multiple

following datagrams are lost however, the quality is significantly degraded. These "bursts of loss", often considered as five or more consecutive packets lost, can have a devastating effect on the voice quality, and are heavily weighted in the E-model [26]. Voice codecs handle packet losses differently, and will be discussed later on.

4.3 Perceptual Evaluation of Speech Quality

The Perceptual Evaluation of Speech Quality (PESQ) is an objective quality measurement algorithm that reflects momentary effective network degradation[40]. The algorithm was developed to test end-to-end voice quality under real network conditions [35]. The quality evaluation is done by comparing an input audio signal with the output signal of the communication channel. The PESQ algorithm compares the degraded speech signal with the reference speech and computes an objective MOS value [40]. Since the method doesn't need any preparations in the network, and can be used for all voice communication, e.g VoIP, PSTN and GSM, it is a desired tool for many telecom companies. Telenor in Norway for instance use it to evaluate the quality in their GSM network.

CHAPTER 5

Voice Codecs

The choice of codec to use in a VoIP service can be crucial to the experienced quality of the service. As figure 5.1 shows the codec in use both directly affects the perceived voice quality as well as influencing the end-to-end delay which also affects the perceived quality.

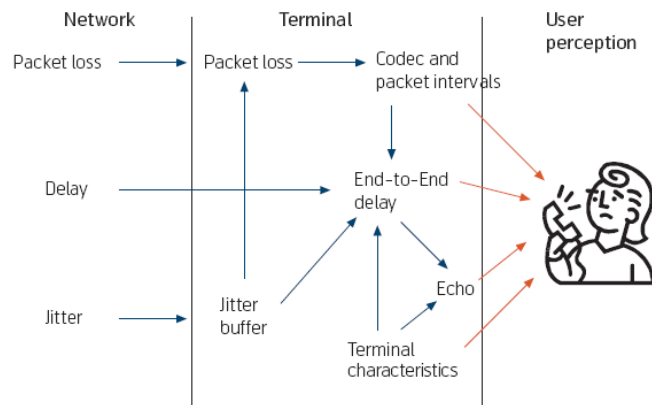


Figure 5.1: Characteristics influencing user perceived voice quality [40].

There are numerous different codecs that provide voice with varying quality and bandwidth needs. Given the span of this thesis, only two of the most used codecs will be studied. The tests will be carried out using the ITU G.711 and G.729 codecs for the following reasons:

- Both codecs are supported by most mobile terminals.
- The codecs have different processing requirements to the devices in use.
- The codecs produce sound reproduction at different bitrates.
- One of the codecs provides compression while the other does not.

Another codec that it would have been interesting to study is the internet Low Bitrate Codec (iLBC). This codec is royalty free and used in free voice applications such as Skype™. Unfortunately the codec is not supported by the analysis software used in this thesis, and will therefore not be studied in detail.

5.1 ITU G.711

The ITU-T Recommendation G.711 is the most widely used codec in PSTN/ISDN [40]. Using 8-bit Pulse-Code Modulation (PCM) encoding and a sampling rate of 8 kHz the codec produces output at 64 kbps. PCM is a digital representation of an analog signal where the signal strength of each of the 8000 samples per second are assigned to the closest of 256 predefined levels [44]. Since an 8-bit expression can represent 256 values ($2^8 = 256$), and the sampling rate is 8 kHz, the digital sound signal to be transmitted has a data rate of $8\text{bits} \times 8000\frac{1}{s} = 64\text{ kbps}$.

There are two "flavors" of the G.711 codec, called A-law and μ -law. The difference lies in how the analog signal is being sampled [24]. While μ -law is used in North America and Japan, A-law is used in the rest of the world. A-law provides a smoother sound due to better suppression of sampling artifacts which causes a more dynamic range [24]. Since the tests in this thesis are performed in Norway, the A-law is used.

Since the G.711 codec merely digitizes the analog signal, it is one of the least processor-intensive codecs [44]. This can be advantageous considering that the terminals that are most likely to use a VoIP service have very limited processing power. The most powerful terminal used in the tests in this thesis has a 520 MHz processor, while the least powerful only has a 195 MHz processor. However, the relatively high bit rate produced by the codec makes it somewhat unsuitable for use with mobile terminals in a wireless network. In a high capacity office WLAN the high bit rate will not necessarily be a drawback, but in a citywide wireless network keeping bandwidth demands low will be of essence.

Because the codec doesn't use compression, it will provide the best voice quality and lowest delay compared to other codecs. These features are attractive, and has resulted in the codec being supported by most VoIP providers. [24]

5.2 ITU G.729

The G.729 codec is an ITU standard codec that produces a low bit rate output. With a sampling rate of 8 kHz and a frame size of 10 ms it only produces an 8 kbps output signal [44]. The codec uses Code-Excited Linear Prediction (CELP), which reduces the number of bits necessary for successful transmission of the sound [44]. Since the frame size is quite small, the transmitted packets are small as well, lowering the risk of packet loss and reducing audible delay [37]. The codec can also deliver the bitrates 6.4 and 11.8 kbps, respectively with lower and higher audio quality than the 8 kbps standard rate.

G.729 is often preferred as a codec for applications traversing the Internet because of the following [37]:

- Many devices offer only 1 or 2 low bit rate codecs, usually G.729 and one other or just G.729.
- Some gateway providers will only allow you to talk to their gateway with G.729.
- A good G.729 implementation uses less bandwidth and less CPU power than other low bit rate codecs such as iLBC. G.729 uses 8 kbps, iLBC uses 13 kbps.
- Few phones implement iLBC. Many, e.g. Cisco 7940 and Swissvoice, only offer G.729.
- Most phones offer G.711 as well - that is actually 64 kbps, eight times the bandwidth required by G.729. It is best suited for use on LANs.

The G.723.1 codec is used for similar reasons to those listed for G.729 above, but gives the benefit of using even less bandwidth, but with a more noticeable degradation of sound quality[37]. G.723.1 is not supported by as many devices as G.729, and is therefore not studied further in this thesis.

There has come two extensions A and B to the G.729 codec. The A extension is a less complex algorithm, but delivers lower audio quality. The B extension provides the same voice quality as the original codec, but at a lower bitrate due to silence suppression.

CHAPTER 6

Wireless Trondheim

The Wireless Trondheim project was actuated by the Norwegian University of Science and Technology (NTNU) in 2005 to with time offer outdoor Internet access to the cities residents. The network would also act as a large scale testbed for research and development. In February 2006 a pilot of the project was ready, and in late September the same year the more or less complete network opened to the public. However, the public, in this case, means civic and county municipality employees and pupils, and NTNU's students. Today the network covers inner-city Trondheim along with the area between the city and NTNU's largest campus, Gløshaugen. In total the network provides continuous coverage in a 1.5 square kilometre area.

6.1 Technology

Essential for the developers of the Wireless Trondheim-network is that it provides "more than enough" bandwidth. To achieve this the access points are fed with either fiber optics or high capacity radio links. Wireless Trondheim demands a capacity of minimum 11 Mbps from each AP, and have therefore placed the access points closer together where there is higher expected traffic. Wireless Trondheim has, for now, only used well known and widespread technologies in their network. The access points communicate with the clients with 802.11b or g, and the radio links use 802.11a. In some areas clients can communicate over 802.11a as well. An advantage from using technologies that already are in widespread use is that a lot of safety features are already implemented. In addition, since most people already have wireless equipment

using IEEE 802.11 the investment cost for a new user would be virtually non-existing.

6.1.1 Wi-Fi

The three network technologies in use in Trondheim all belong under the Wi-Fi umbrella. However, they have different characteristics and provide different bandwidth to the connected terminals. While 802.11a operates in the fairly unused 5 GHz band, 802.11b and g operate in the crowded 2.4 GHz band. Since the b and g standards operate in a much used spectrum, they are more vulnerable towards interference. Both microwave ovens and cordless phones use the same frequency band. If you include the fact that many people have wireless networks installed in their homes, 1200+ private access points in downtown Trondheim alone¹, there is a lot of potentially harmful interference out there.

In addition to the frequency differences, the three technologies have different capacity. While 802.11a and g have theoretically a maximum capacity of 54 Mbps, 802.11b has a theoretical maximum of 11 Mbps. However these rates are only achievable in theory, the approximate capacity achievable is 27 Mbps for a and g, and 6 Mbps for b [41] [44]. While most personal computers (PCs) today have support for the 802.11b and g standards, computer manufacturers have only recently started to add 802.11a support to their network interface cards (NIC). Mobile Wi-Fi terminals usually only have support for 802.11b.

Since 802.11b and g use the same frequency band, the two traffic types affect each other. By allowing an access point with 802.11b and g support to hear 802.11b clients the maximum throughput drops to 18 Mbps even with no 802.11b clients in the presence of the access point [6]. If you add an 802.11b client to the picture, the maximum throughput drops to 9 Mbps on the access point [6].

6.2 Infrastructure

The network in Trondheim is built up with equipment from Cisco Systems, Inc. The end system contains of Cisco Aironet®1010 and 1030 lightweight access points. These access points are controlled by two Cisco 4404 Wireless LAN Controllers that each can support up to 100 lightweight access points [21]. Between the access points and the controller the traffic is tunneled using the lightweight access point protocol (LWAPP). This protocol enables the access points to automatically find the best available wireless controller with no hands-on intervention. The Cisco Wireless Control System (WCS) provides a complete platform for the management of the entire

¹Rogue access points seen by the Cisco Wireless Control System

wireless LAN system. The WCS gives network administrators a single solution for RF prediction, policy provisioning, network optimization, troubleshooting, user tracking, security monitoring, and wireless LAN system management [20]. It also provides a graphical interface that makes the deployment and operation of wireless LANs simple and cost-effective. Figure 6.1 is a screen shot from the WCS. You can clearly see the heatmaps from the access points, as well as clients within the access points' reach. However, the heatmaps are not accurate because they only show the reach of the AP without the external antennas. In reality the access point heatmaps are overlapping.

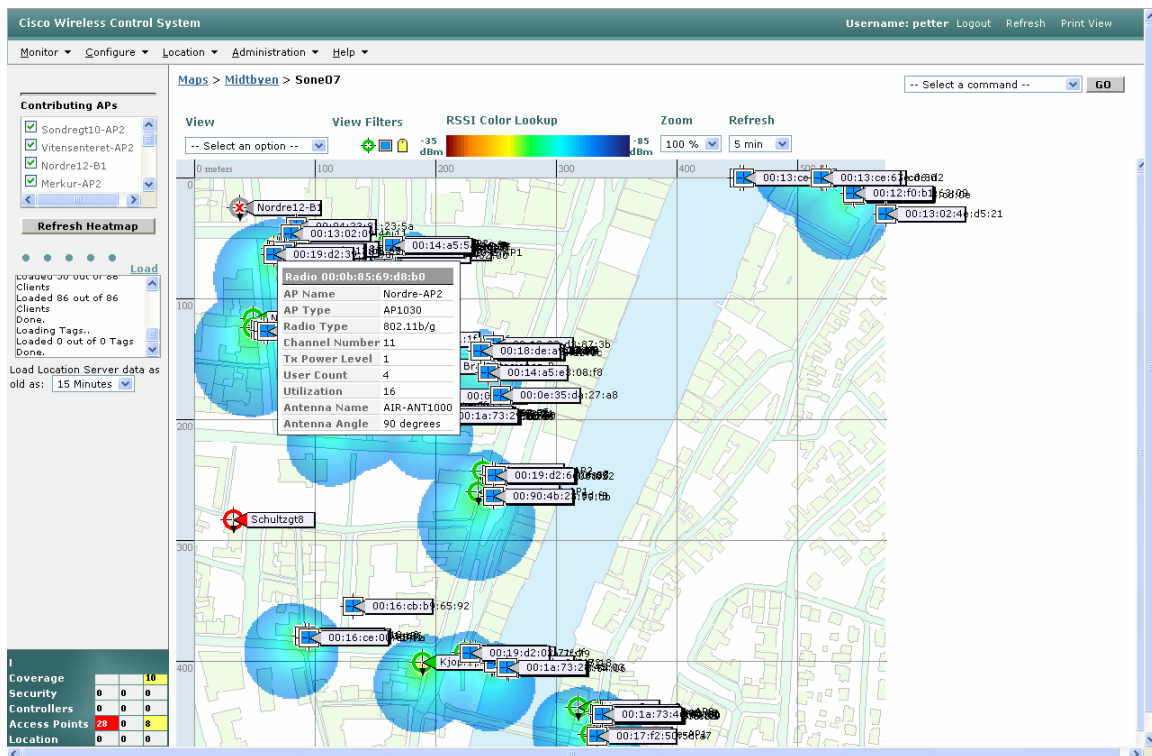


Figure 6.1: Screen shot from the Cisco WCS graphical user interface .

6.3 Voice Support

For a WLAN to provide voice, a pervasive deployment is necessary. Everywhere a client may roam, the network needs to have continuous coverage to avoid gaps in coverage that may cause a call to be dropped [19]. Also, in a network transporting both voice traffic and ordinary data traffic a prioritisation scheme is required to provide the voice traffic with sufficient resources.

6.3.1 Wireless Multimedia Extensions

The Cisco System in Trondheim supports the industry standard for prioritizing traffic, the IEEE 802.11e. Although mechanisms supporting this standard are not yet ready and released, a subset of the standard ratified by IEEE in 2005 under the name Wireless Multimedia Extensions (WMM), is embraced by Cisco [2]. WMM enables differentiated services for voice, video and best-effort data to allow voice traffic to be handled before other traffic on the network. Since voice traffic is bidirectional, prioritization must also be enabled at the client [19]. Terminals with WMM support are not that common, which is why WMM is not used in the tests in this thesis.

6.3.2 Cisco Compatible Extensions

Since there isn't any standard way to implement WMM, Cisco has come up with a proprietary solution in its Unified Wireless Network. Cisco being a major infrastructure vendor, but rather small on the terminal side, has started the Cisco Compatible Extensions (CCX) program to allow terminal vendors to take advantage of their solutions. Through this program Cisco licenses CCX code to client manufacturers, but reserves CCX support on APs for itself [2]. This way the Cisco networks can provide the same quality of service to all terminals without being afraid of losing customers to other infrastructure manufacturers.

6.3.3 Cisco Centralized Key Management

In a large scale network, such as the one in Trondheim, roaming will be critical to the performance of voice applications. Since the density of access points is much higher in certain areas of the city, a user might have to switch access points several times during the same conversation. If each roam causes a noticeable glitch in the conversation, people will find this annoying. To support time-sensitive applications such as voice, the Cisco Centralized Key Management (CCKM) protocol was developed [19]. CCKM provides secure and fast roaming between APs by eliminating the involvement of the centralized authentication server. Instead an access point configured to provide Wireless Domain Services (WDS) takes the authentication servers place, and re-authenticates the client so quickly that there is no perceptible delay in voice or other time-sensitive applications [22]. Figure 6.2 illustrates the handover using CCKM.

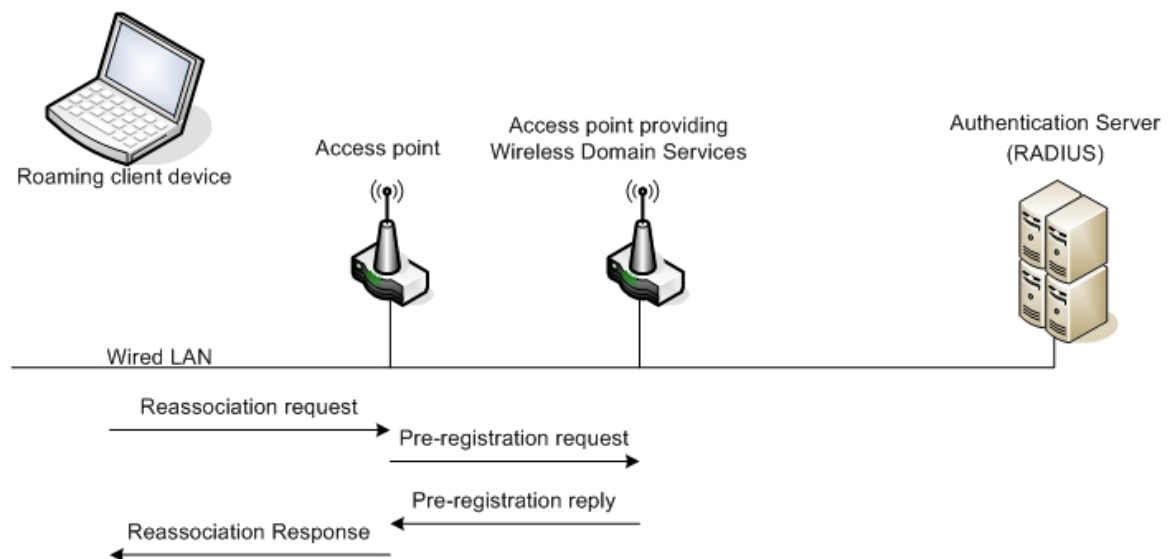


Figure 6.2: Cisco Centralized Key Management [22].

6.3.4 Security

In Wireless Trondheim all roaming takes place at layer 2, so the delay introduced by switching access points should not be very significant unless a vigorous encryption algorithm is used. The Cisco Unified Wireless Network supports all the most common security features such as 802.11i, 802.1X, WEP, WPA, WPA2, AES and TKIP [23]. This enables the use of strong mutual authentication and advanced data encryption using dynamic encryption key management. The introduction of these security measures require processing and additional complexity at the client side and will affect the time spent roaming between APs as well as the overall performance of a time sensitive application e.g. voice.

PART II

Test Implementation

CHAPTER 7

Test Setup

All tests in this thesis are carried out in the wireless deployment in Trondheim. To make the tests as realistic as possible tests are done at different locations in the city, with varying network load. The background traffic consists of both application data traffic and voice traffic, because it is expected that the two traffic types affect the ongoing conversations differently.

Monitoring network traffic generated by the equipment in use is fairly straight forward. This is done by looking at the output from the analysis tools. However, traffic generated by others connected to the network is not that easy to monitor and characterize. To avoid anomalous results as a consequence of others generating traffic, tests are only done connected to access points that don't have other clients associated with them. Through the Cisco Wireless Control System you can see how many clients are associated with any given access point, as well as the MAC addresses of the clients associated. Since the fiber backbone network in Wireless Trondheim has very high capacity, traffic generated at other access points does not affect the performance noticeably.

Throughout the tests different codecs are used. The G.711 and G.729 codecs are used because they generate fairly different traffic. The G.711 codec generates a 64 kbps stream in each direction, while G.729 has a higher compression rate and generates a 8 kbps stream in each direction¹. Both the codecs impair the audio signal, but the G.729 causes a greater attenuation in audio quality, mostly caused by the 10 ms

¹G.711 generates a 64 kbps stream when sampled at 8 kHz with 8 bits per sample. Standard G.729 generates a 8 kbps stream.

compression delay. According to [18] the highest obtainable MOS scores using the G.711 and G.729 codecs are 4.10 and 3.92 respectively. However, according to Ixia, the producer of the test software used in this thesis, the theoretical maximum values are 4.40 and 4.07 for G.711 and G.729 respectively [28].

Throughout the tests all the terminals are connected to the Wireless Trondheim network using Wired Equivalent Privacy (WEP), also called the Wireless Encryption Protocol. WEP provides authentication, privacy and data integrity at the data link and physical layers, which rules out end-to-end security. IEEE 802.1X security is also available in the network, but was not used during the tests.

It is interesting to see what happens with the quality of voice conversations under various network conditions. The following sections stipulate what needs to be tested, and how the tests are planned to be implemented. As far as possible all tests are repeated at least three times, and the results averaged, to give an as exact representation as possible.

7.1 Number of Voice Calls in a Network Cell

If a wireless network is to provide voice services it has to support a certain number of concurrent calls. The bottlenecks will be the individual access points in areas where large numbers of people gather. Producers of wireless network infrastructure, such as Cisco and SpectraLink, do not recommend more than a maximum of six concurrent calls per access point [32]. This constraint can result in a high density of access points in crowded areas of the city. The following test tries to capture the available call capacity in Wireless Trondheim under varying network load scenarios.

7.1.1 Test Methodology

Given the constraints already stated by the infrastructure producers, the capacity tests start with three active voice conversations. Tests with only one or two bi-directional voice streams are also performed, but these tests will mostly be used to point out differences among the tested terminals. The maximum amount of simultaneous voice conversations tested is ten. Since it is expected that the network only can handle six, or perhaps seven concurrent voice conversations, it is unlikely that ten conversations will succeed.

The networks capacity is tested in two ways. The first tests start from three to ten bi-directional voice streams at the same time, and run until all conversations are finished. The tests are run with both the G.711 and the G.729 codecs. The second tests initially start one bi-directional conversation and add one more every

five seconds until ten concurrent conversations are running. This way it is possible to see at which levels audio quality starts to deteriorate. The same tests are repeated with background traffic to see if there is a significant difference in how the two traffic types cause voice quality deterioration.

7.2 Voice Quality Deterioration with Increasing Application Traffic

A citywide wireless network, like the one in Trondheim, is meant to carry both voice and data traffic at the same time. However, the two types of traffic affect the time sensitive voice traffic differently. Without any prioritizing mechanisms data traffic tends to starve the voice traffic for bandwidth, [16]

7.2.1 Test Methodology

These tests are a continuation of the previous test. A series of bi-directional VoIP pairs are set up, and an increasing amount of background traffic is added to the network. For example, three voice conversations are started and the achieved voice quality is measured. Then adding a mixture of background traffic, and after every increase in background load measure the new voice quality. This test is repeated with from one to five voice conversations with various pair combinations, e.g. mobile to mobile, laptop to mobile, etc.

The background traffic is also generated in different ways. In addition to using the built-in scripts in IxChariot, the GenSyn traffic generator is used to generate realistic web browsing traffic.

7.3 Voice Quality under Roaming

Since we so actively use our cellular phones while we are out walking, shopping or driving, the terminals' abilities to roam will be essential to Voice over Wireless LANs success. Since all the tests in this thesis are carried out in the Wireless Trondheim deployment and not in a lab, this might be one of the most difficult tests to carry out. The most difficult part is repeating the same route of movement at the same speed for consecutive tests.

7.3.1 Test Methodology

Two bi-directional voice conversations are started from a laptop to two different terminals. After both conversations have started, one of the terminals is moved out of the coverage area of the access point the terminal currently is associated with, and into the coverage area of another access point. This test is repeated at least three times, walking the same route at the same speed.

7.4 Subjective Voice Quality Evaluation

To get a subjective grasp on how good, or poor, the achievable voice quality in the network is, a few simple tests actually listening to the voice quality are performed. In these tests Skype™ is installed on one of the terminals, and SkypeOut used to call somebody on their cellular phone. The first test is stationary in a position where there is decent signal quality. In the second test the terminal is roaming while having a conversation over SkypeOut.

It is not possible to calculate the voice quality for these conversations, but hopefully insight in what quality is achievable in the network is given. Skype™ uses the internet Low Bit Rate Codec for IP to IP communication, but however, SkypeOut uses the G.729 codec [8]. The conversation from SkypeOut to a cellular phone should then generate roughly the same traffic as the previous G.729 tests.

CHAPTER 8

Test Software

The following sections describe the software used in this thesis.

8.1 IxChariot

The industries leading software for real-world testing of device and system performance is Ixias IxChariot. IxChariot offers thorough performance assessment and device testing by emulating all the most used protocols. The software provides a confident assessment of the expected performance characteristics of any application running on wired and wireless networks [27]. The software enables you to run tests with different traffic types and network settings. It also provides readable output for VoIP quality directly in MOS values, along with delay, loss, throughput, jitter and other metrics. IxChariot can also provide Received Signal Strength Indication (RSSI) measurements, which can be used as an estimate of differences between the different terminals' wireless adapters.

8.1.1 Test Setup

Before being able to test wireless performance with IxChariot all involved terminals need to be set up correctly. The main IxChariot software must be installed on a server directly connected to a Local Area Network (LAN), from now on called the Console. In the case of this thesis the console is placed at Uninett in Trondheim. The

console is accessed through a remote desktop connection while testing in the city. To not affect the results of the tests a separate laptop is used to establish the remote desktop connection and start and stop the tests on the console.

Each of the terminals involved in the tests need to have a small piece of software installed, called Ixia Performance Endpoint. The Endpoint software is platform dependent and is available for most operating systems (OSs) and processor combinations. Before running a test, the Endpoint software must be running on all the involved devices. On Windows terminals the software runs as a process that can automatically start upon switching on the terminal.

On the Console, the IxChariot software is started through the remote desktop connection, and the tests can be set up. In the test setup window one can design how the test is to be carried out, which is shown in figure 8.1. Different communication pairs can be added to the test depending on the type of test that is desired. The pairs can be VoIP-pairs, Multimedia-pairs, Data-pairs etc, and combinations of these can emulate realistic network traffic. Each pair consists of two endpoints, Endpoint 1 and Endpoint 2. If a pair is set up using two mobile terminals as Endpoint 1 and Endpoint 2 the desired traffic will flow from Endpoint 1 to Endpoint 2. If a bi-directional voice conversation is to be tested, two pairs need to be set up with the Endpoints switched in the second pair.

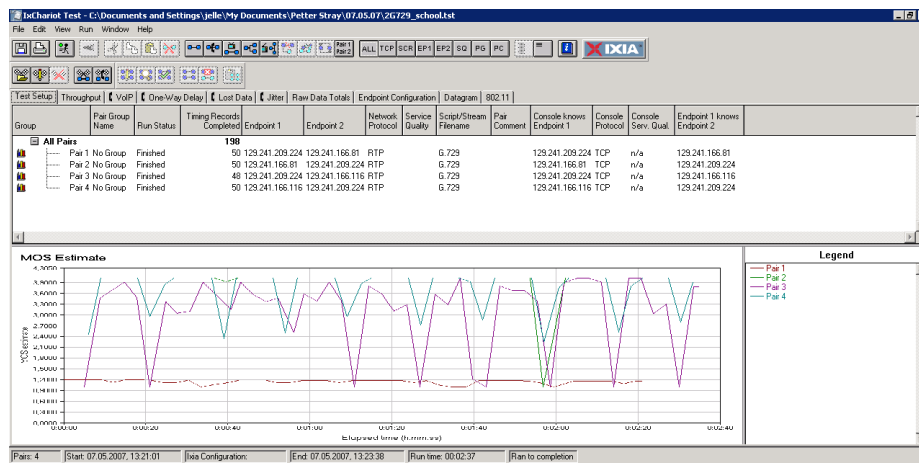


Figure 8.1: Screenshot of the test setup window in IxChariot.

8.1.2 Data Flow

The testing in IxChariot is based on scripts that control the course of the test and set the various parameters for the test. Upon test start, the IxChariot console sends

configuration data and the test script to Endpoint 1. Endpoint 1 extracts its part of the script and sends the rest to Endpoint 2. When Endpoint 1 receives an ACK message from Endpoint 2 it notifies the IxChariot console. The console then tells Endpoint 1 to start running the script. Both Endpoints 1 and 2 run their scripts multiple times, while Endpoint 1 gathers the results. These results can either be forwarded directly to the console (real-time-mode) or gathered and sent to the console in batches. Each run of the test script is called a timing record. When the test is finished the console processes the results and presents them in an analytical and graphical way. The data flow of an example test setup is shown in figure 8.2.

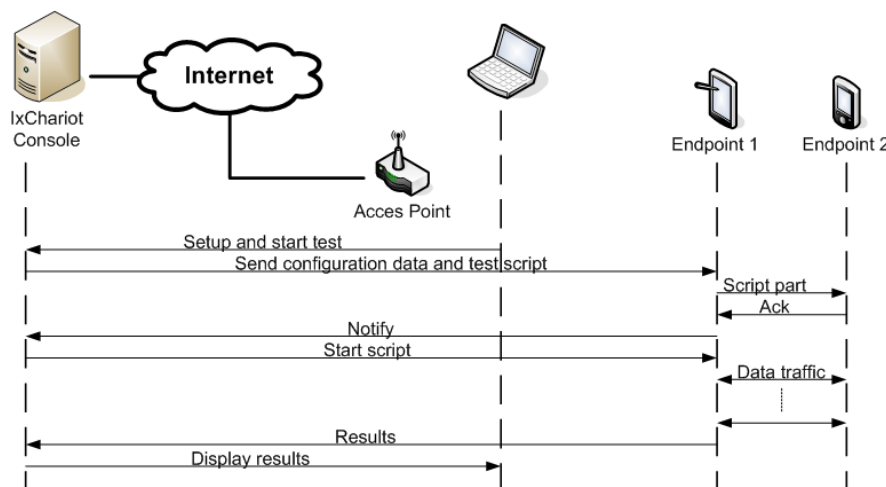


Figure 8.2: Example of WLAN test setup with Ixia’s IxChariot.

If VoIP is tested, the traffic between Endpoint 1 and 2 follows standard VoIP procedure with Session Initiation Protocol (SIP) control plane followed by Real-Time Transport Protocol (RTP) stream and teardown [27].

8.2 GenSyn

GenSyn is a Java-based application that generates synthetic Internet traffic, developed by Poul E. Heegaard at Telenor R&D. The traffic generator is well suited to use when testing new applications and network mechanisms in the Internet. GenSyn uses state diagrams to describe stochastic user behavior which makes it scalable since it allows a composition of users in each state instead of creating a new process instance for every user. Through interface modules, that links GenSyn to the underlying Internet Protocol, the stochastic user behavior model controls the creation of TCP and UDP streams. Each transition between specific states in the stochastic model triggers the

instantiation of an interface process, and IP packets are sent through the network. [11]

GenSyn implements different application models to emulate various data traffic. Built in models include web and FTP for TCP type traffic, and VoIP, MPEG and CBR for UDP type traffic. In this thesis the Constant Bit Rate (CBR) and web traffic models are used. The web traffic model imitates users downloading entire webpages from allover the world. The webpages being accessed can be predefined by the user, or dynamically updated as the experiment evolves [11].

CHAPTER 9

Test Hardware

This chapter briefly describes the hardware used in the tests carried out in this thesis. For a more detailed description of the mobile terminals used, the reader is referred to appendix A.

9.1 Server

The server where the IxChariot Console release 6.25 SP1 is installed has a 2.80 GHz Intel®Pentium®4 processor and 1.0 GB of RAM. The operating system (OS) is Microsoft Windows Server 2003 Enterprise Edition with Service Pack 1 installed. It has a Broadcom NetXtreme Gigabit Ethernet card, with driver version 2.91.0.0. The server is placed at Uninett and is directly connected to a LAN.

9.2 Desktop Computer

The desktop computer used for testing is a Dell Dimension 4550. It has a 2.53 GHz Intel®Pentium®4 processor and 512 MB of RAM. Installed on it is Microsoft Windows XP Professional 2002 with Service Pack 2. It has an Intel®PRO/100 VE Network Connection, with driver version 6.1.3.11. The desktop is placed at NTNU's Gløshaugen campus, and is directly connected to a LAN. The IxChariot Endpoint version 6.4 build 90 is installed.

9.3 Laptop Computers

9.3.1 Acer 5672WLMi

This laptop was actively used in the testing. It has a 1.66 GHz Intel®Core™Duo processor T2300 and 2 GB of RAM. The OS is Ubuntu Linux 6.10 and the wireless adapter is an Intel®PRO/Wireless 3945ABG, driver version ipw3945. The IxChariot Endpoint version 6.4 build 90 is installed.

9.3.2 HP Pavillion dv4153

This computer is only used to start and stop the tests and to run the GenSyn traffic generator. The OS Microsoft Windows XP Home Edition 2002 with Service Pack 2 is installed. It has an Intel®Pentium®M 1.73 GHz processor and 1.0 GB of RAM. It has an Intel®PRO/Wireless 2200BG wireless adapter with driver version 9.0.1.9, and IxChariot Endpoint version 6.4 build 90 is installed.

9.4 Mobile Wi-Fi Terminals

Table 9.1 contains some of the data on the mobile terminals being used. Finding information on the Wi-Fi radios in the different terminals proved to be "impossible". After calling all Qtek and HTC repairs in Norway, as well as contacting HTC headquarters in England, the question was still unanswered. All terminals have the IxChariot Endpoint version 6.4 build 90 installed. All information accessible on the terminals is presented in appendix A.

Terminal:	Processor:	RAM:	Operating System:
Qtek 9000	520 MHz Intel®PXA270	64 MB	Windows Mobile™5.0
HTC TyTN	400 Samsung®2442	64 MB	Windows Mobile™5.0
Qtek 8300 I	195 MHz TI OMAP850	64 MB	Windows Mobile™5.0
Qtek 8300 II	195 MHz TI OMAP850	64 MB	Windows Mobile™5.0
Qtek 8310	195 MHz TI OMAP850	64 MB	Windows Mobile™5.0

Table 9.1: Mobile terminal specifications.

CHAPTER 10

Performance Criteria

If Voice over Wireless LAN is to catch on, the quality of the service must satisfy certain criteria. In a network deployed to serve both voice and data this has proven to be difficult. Pure voice networks can be configured to provide good voice quality for a limited amount of users. Most infrastructure producers don't recommend more than six or seven concurrent voice streams on one access point which results in a dense deployment of access points. In an office environment this may work because the workers mostly are stationary while using the phone. Also, it is easy to predict where large gatherings of people will take place, e.g. a cafeteria or conference room, and increase the density of access points there. In a municipal wireless network however it would be too expensive to deploy a network merely for voice traffic, and both voice and data traffic must be supported in the same network.

10.1 Voice Quality

Generally voice quality is considered to be good if it exceeds a MOS value of 3.5. However, according to [25], a normal GSM network only delivers audio with a MOS score between 2.9 and 4.1. The lower bound of this interval is relatively far down in the "Nearly All Users Dissatisfied" section of the MOS scale in figure 4.1. People clearly value the mobility they get from a cellular phone a lot since they accept such a large degradation in voice quality.

Telenor, Norway's largest GSM provider, has provided some voice quality statistics from their mobile network in six major Norwegian cities. They have measured the

City	MOS down link % over 3.0	MOS up link % over 3.0
Ålesund	98.88	99.40
Bergen	98.36	98.85
Stavanger	99.71	100.00
Stor-Oslo	97.96	98.18
Tromsø	99.41	99.41
Trondheim	98.52	98.71

Table 10.1: MOS values from Telenors GSM network [38].

share of conversations that obtain a MOS score of over 3.0 using the PESQ algorithm. Their results are summarized in table 10.1.

As you can see from the table a very high percentage of the calls in the GSM network qualify for a MOS score over 3, as many as 98.95% of the conversations on average.

Considering that most people find the quality achieved by their GSM phones to be more than good enough, a lower quality than that in GSM will probably be acceptable for the majority of potential users of a citywide wireless network for voice communication. Since the lower MOS value in GSM is around 3, I assume that a value as low as 2.6 on the MOS score might be acceptable for the basically free voice service that is possible over the packet switched network. I have chosen the lower quality limit 2.6 since this also is the lower border of the "Nearly All Users Dissatisfied" group of MOS scores in figure 4.1.

10.2 Delay and Jitter

The delay of audio packets can be especially noticeable to the human ear, and appears as gaps in the conversation. Since VoIP is a time-critical service, packets are quickly dropped if they are delayed. If this happens to multiple consecutive packets, a very annoying gap will appear in the audio stream, as if you are talking to somebody over a walkie-talkie or satellite phone. Already when the one-way-delay rises up towards 150 milliseconds it starts to greatly affect the audio quality [26]. When it comes to jitter, or delay variations, audio quality begins to suffer anytime it exceeds 60 milliseconds [32].

10.3 Packet Loss

In data transmission packet loss is not an especially critical factor since a lost packet can be re-transmitted. This however, is not an acceptable solution for voice packets that have stringent delay requirements. Even a 1 % packet loss can seriously degrade the voice quality of a G.711 conversation, while G.729 is tolerant of a loss of up to 5 % on average for the entire conversation [5] [7]. As previously mentioned, five or more consecutive packets lost appear annoying to a listener [26]. Using the standard G.729 codec this means packet gaps representing over 100 ms of audio are noticed by the human ear.

10.4 Scalability

If the networks limits for active voice conversations are reached, the network should react correctly. Say a user attempts to make a call originating or terminating at an access point under high load. If the call is accepted, the already active conversations will experience poor voice quality and many users will be dissatisfied with the service. However, if the new call is dropped, only one person will be affected. While the caller attempts to reconnect, another call may have been terminated or he may have come within the reach of another access point with available bandwidth/capacity.

Packet loss, delay and jitter are directly linked up against voice quality and constitute a part of the R factor calculations done by the analysis tool. There are therefore not set any special demands to the metrics regarding the evaluation of the results obtained in the tests in this thesis.

PART III
Evaluation

CHAPTER 11

Results

The following chapter presents the results obtained from tests performed in spring 2007. Detailed test setups and results from each individual test are available on the electronic references disc.

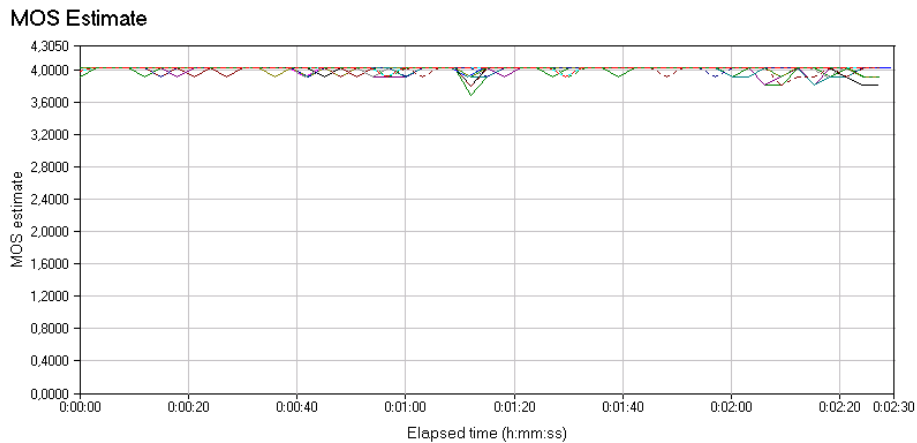
11.1 Network Cell Capacity

Determining the capacity of the network proved to be a difficult task with the equipment available. Having only five terminals at my disposal an accurate evaluation was not possible. The remaining traffic had to be generated by a laptop which has much higher processing power. However, an approximation of the network's capacity was possible.

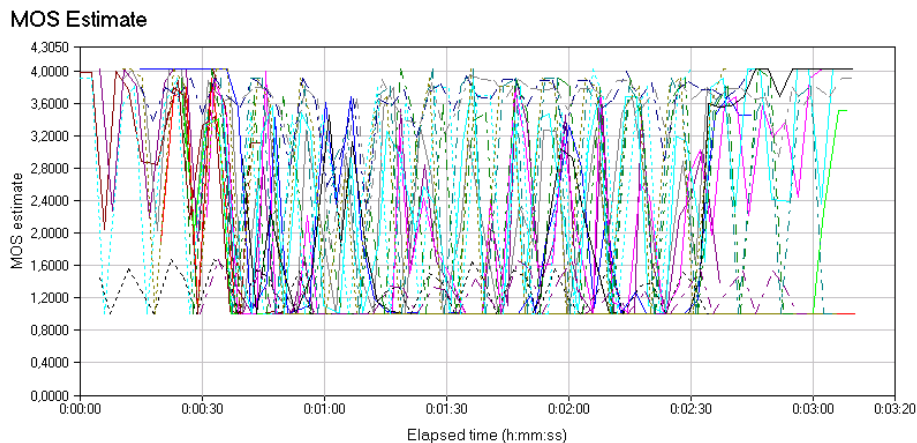
11.1.1 Maximum Number of Voice Calls in a Network Cell

In this test all five mobile terminals, two laptops and a desktop computer were used. Given the capacity constraints already given by infrastructure producers a benchmark test with 10 conversations running between the stationary computer and the Linux laptop was performed. This test was repeated with both the G.711 and G.729 codecs. Afterwards the mobile terminals were brought into the test. Two bi-directional conversations were set up between each terminal and the desktop computer, and repeated with both codecs.

The tests only involving the laptop and desktop computer gave results with MOS values averaging close the maximum achievable values. Adding mobile terminals to the mix however drastically lowered the conversational quality.



(a) Laptop only.



(b) After introducing mobiles.

Figure 11.1: 10 G.729 conversations.

Figure 11.1 shows MOS values for the two cases using the G.729 codec. As you can see from figure 11.1(a) the voice quality when only a laptop is used is quite good. In the case where mobile terminals were introduced three of the voice streams did not manage to complete all the timing records, one of which was not able to get a single timing record through. As you can see from figure 11.1(b) the voice quality is rapidly jumping from 1 to 4 for all the terminals.

11.1.2 Increasing Call Load

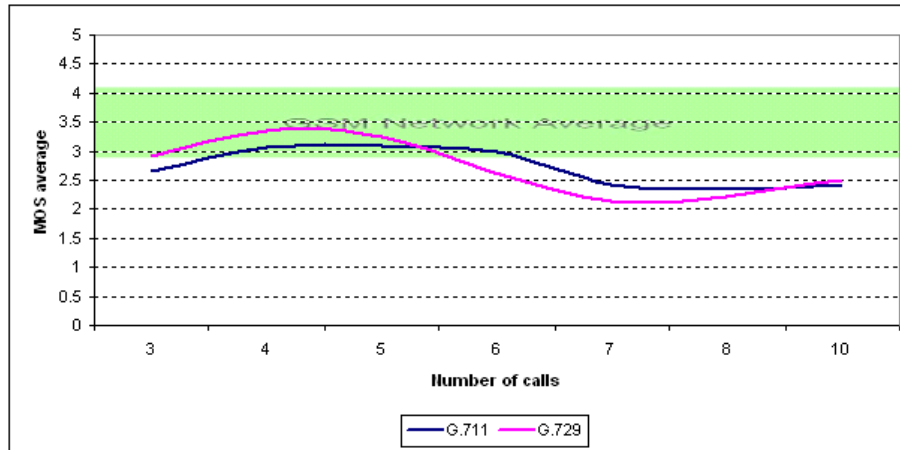


Figure 11.2: Voice quality with increasing call load.

Figure 11.2 shows how the voice quality degrades after more and more conversations are added. As you can see, the voice quality for both codecs drop below acceptable values when 7 conversations are active at once. This strongly relates to other tests such as [32] in that a maximum of six, or sometimes seven, conversations are handled by a single access point.

11.2 Voice Quality Deterioration with Increasing Application Traffic

In a network providing both voice and data traffic, the effect data traffic has on the voice quality needs to be documented. The following tests evaluate how both bandwidth consuming applications and ordinary web browsing traffic affect ongoing conversations.

11.2.1 Voice Quality Deterioration from IxChariots Throughput Script

In this test three mobile Wi-Fi terminals, two laptops, and a desktop computer were used. The three mobile terminals used were Qtek 8300 I, Qtek 8300 II and Qtek 9000. All three mobiles and the laptop with Linux were used as endpoints for voice conversations with the stationary computer. The Windows laptop was used to start

and stop the tests. In the tests where background traffic was present, the background traffic was generated at the stationary computer and sent to the Linux laptop. These tests use the G.729 codec as voice codec for all voice streams. All tests were run three times to rule out any inconsistencies.

First the tests were run with no background traffic. Figure 11.3 shows the voice quality for the four bi-directional conversations. The yellow shaded area indicates the MOS realized in GSM networks.

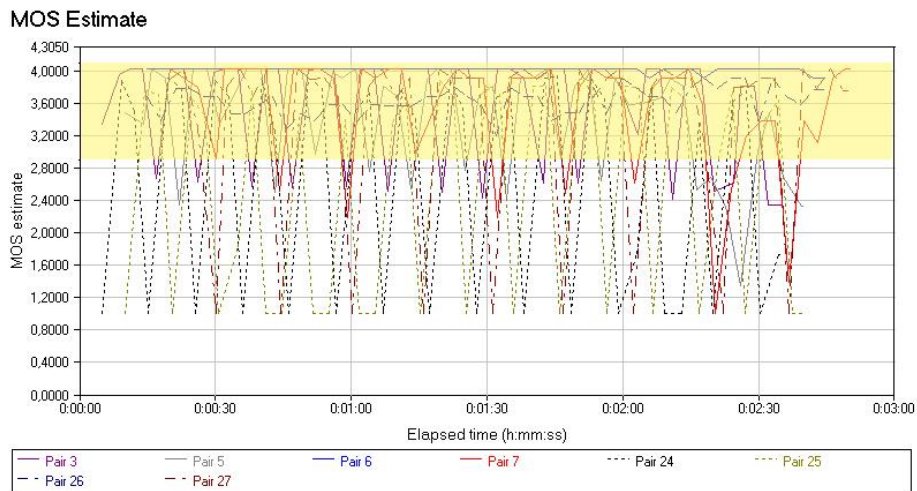


Figure 11.3: 4 G.729 conversations.

Afterwards the tests were repeated with background traffic generated by IxChariots throughput script. The generated background traffic is shown in figure 11.4. As the figure shows the background traffic generated an average throughput of just over 3 Mbps.

The voice quality after the background traffic was introduced is shown in figure 11.5. The yellow area indicates the MOS realized in GSM networks.

As the figures show, after background traffic was introduced the voice quality suffered quite a bit. While nearly 80% of the timing records completed in the conversations without background traffic were within the quality in GSM networks, under 60% of the conversations with background traffic achieved these scores. The one-way delays averaged at 69 ms and 266 ms for the no background traffic and background traffic scenarios respectively, while the packet loss rates were roughly 4.3 % and 6.9 % respectively.

These however, are the results from two single tests. Both test setups were repeated three times. In the two other tests with background traffic one of the voice pairs failed to complete, but the results were otherwise quite similar.

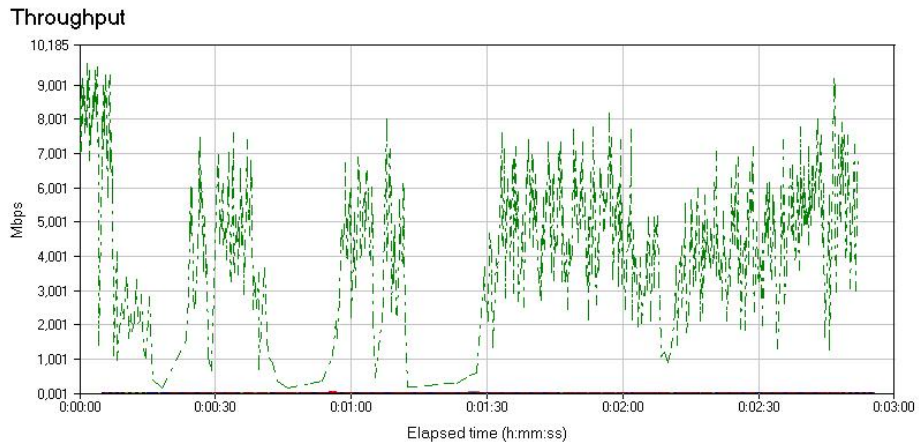


Figure 11.4: The generated background traffic

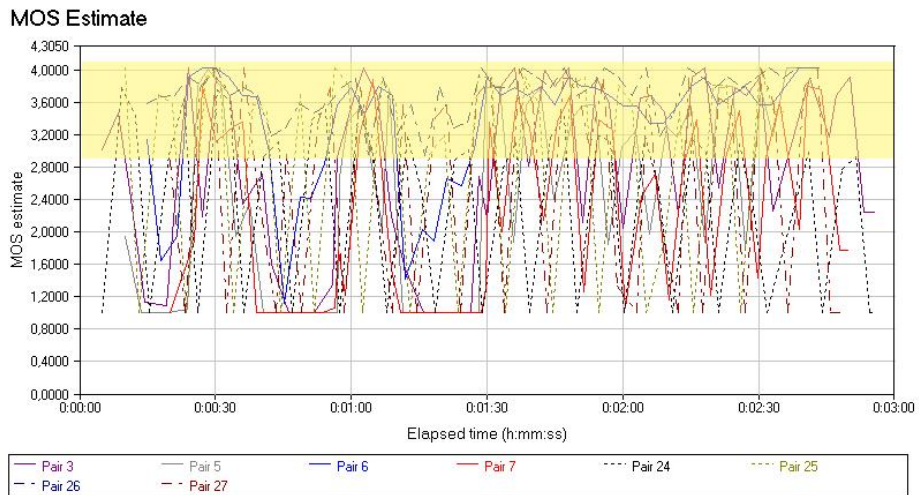


Figure 11.5: 4 G.729 conversations with background traffic.

11.2.2 Voice Quality Deterioration from IxChariots Throughput Script and GenSyn

In these tests a voice conversation between two of the mobile terminals, the HTC TyTN and Qtek 9000, and the Linux laptop was set up. The tests were run with no background traffic, background traffic generated by the IxChariot throughput script and background traffic generated by the GenSyn traffic generator. These tests were performed to see if background traffic generated by IxChariot affected the voice quality differently than GenSyn. This was done to validate the next test, *Voice Quality Deterioration from Realistic Web-Browsing Traffic*.

After testing without any background traffic, IxChariot was set to generate background traffic at approximately 10 Mbps. Such high background traffic load was used to see a significant drop in voice quality in the ongoing conversations. Afterwards the same amount of background traffic was generated using GenSyn's CBR model. Finally the results from the two background tests were compared. The average MOS achieved using IxChariots script was approximately 6% lower than that achieved using GenSyn, but the background traffic generated in the IxChariot case was also slightly larger. The rendition of this test is that background traffic generated by IxChariot and GenSyn have the same affect on the voice quality of ongoing conversations.

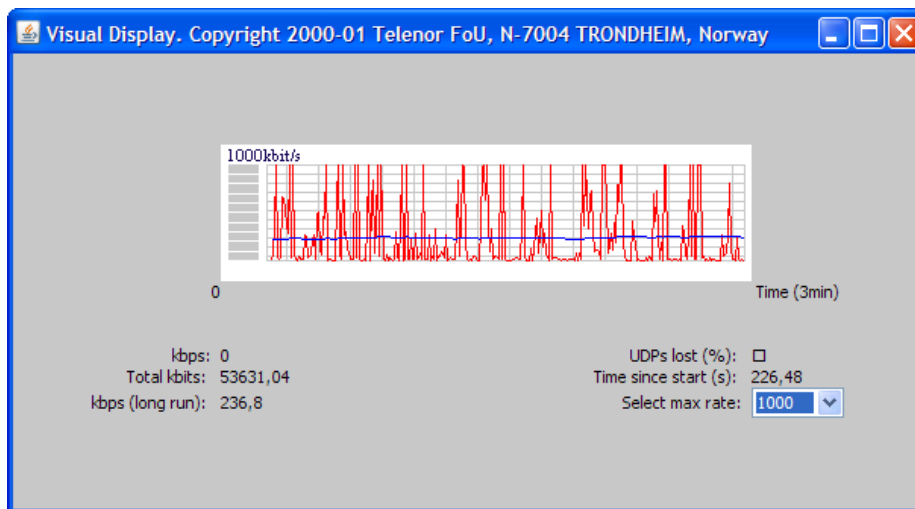


Figure 11.6: Realistic web browsing traffic generated by GenSyn.

11.2.3 Voice Quality Deterioration from Realistic Web-Browsing Traffic

In the nearest future it is most likely that the Wireless Trondheim network will be used for simple web-browsing. Since GenSyn can generate synthetic user-like traffic, this test used the GenSyn web model to generate realistic background traffic.

GenSyn generated traffic by randomly accessing 20 different web pages. On average the generated background traffic was 237 kbps. Figure 11.6 shows the background traffic generated by GenSyn.

The little background traffic generated resulted in a drop of 0.5 in MOS value for the ongoing conversation. As you can see from figure 11.7 the quality experienced at both the TyTN and Qtek dropped a noticeable amount after adding the web traffic. The plotted results are the average values from two tests.

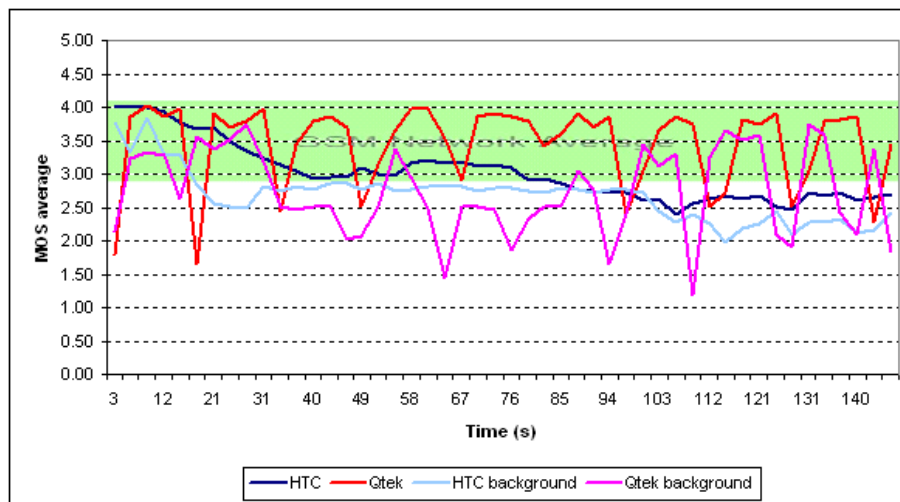


Figure 11.7: The affect on voice quality of adding 237 kbps og web traffic.

11.3 Voice Quality under Roaming

The roaming tests were all performed at the same place in Trondheim, more precisely Trondheim Square. Roaming between access points was achieved by walking with the roaming terminal to the other side side of the square, and back again. This enabled the terminal to leave the coverage area of the associated AP, while moving into the coverage area of a new AP. This test was affected by human factors, in other words the walking speed and path, but this was attempted held as constant as possible. In each test the same elliptical path shown in figure 11.8 was followed.

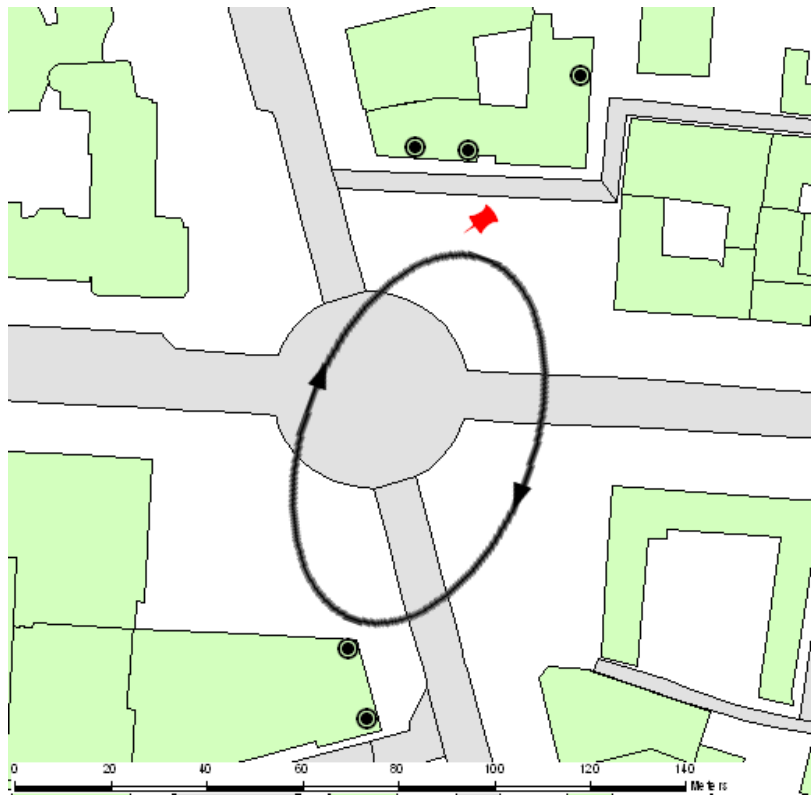


Figure 11.8: Path of roaming tests, Trondheim Square. Map generated in ArcMap.

In the figure the red thumbtack indicates start and stop of the roaming tests. The black circled dots indicate access points, and the ellipse illustrates the path of movement.

11.3.1 G.729

These tests were performed with the HTC TyTN, Qtek 9000 and the desktop computer. A bi-directional voice conversation was set up between the stationary and both the mobile terminals. First the test was run without any roaming and the voice characteristics were captured. Afterwards the test was repeated two times with the Qtek 9000 terminal roaming to another access point. The TyTN was only used to generate some background traffic in the network. The results of all three tests are shown in figure 11.9.

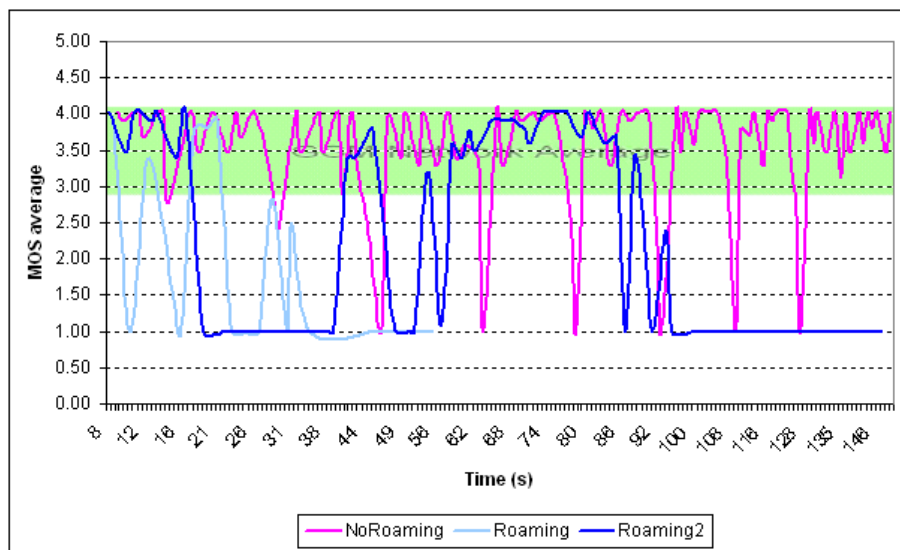


Figure 11.9: G.729 conversations under roaming.

As you can see from the figure, the first roaming test (yellow line) terminated already after one minute. As soon as the terminal started to move away from the access point it was associated with, the connection was lost. In the second roaming test the connection held until the test was supposed to end, but the last 30 seconds of the conversations would have been completely inaudible. A drastic increase in packet loss caused the MOS score to drop to 1, i.e. the "Not recommended" section of the MOS scale.

11.3.2 G.711

The roaming test in section 11.3.1 was repeated with the G.711 codec as well. Results from those tests are shown in figure 11.10. As the figure shows, the G.711 codec

allowed the connection to hold longer than in the G.729 case. On an average however, G.711 performed worse than G.729, providing voice with a MOS score of 2.115 compared to 2.235 using G.729.

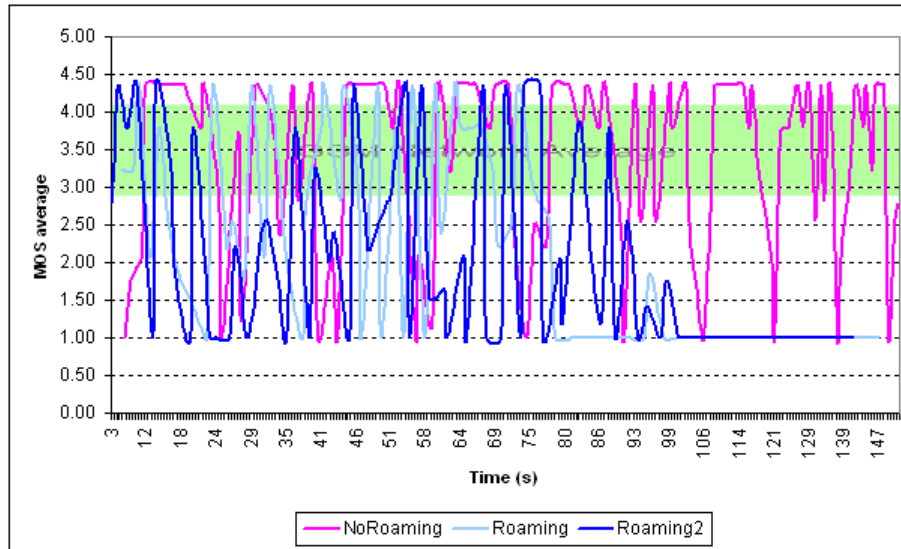


Figure 11.10: G.711 conversations under roaming.

The poor voice quality was caused by severe packet losses as the terminal moved farther away from the access point. At the most 234 consecutive packets were lost which represents over 4.5 seconds of speech using standard G.729 encoding. Clearly this is not desirable.

11.4 Terminal Differences

In order to point out differences between the mobile terminals, each terminal was tested by itself. A G.729 encoded conversation was set up between the Linux laptop and the terminal under evaluation. Some of the tests were also repeated with the G.711 codec to see how the terminals handled the two codecs. This was not done with all the terminals due to both time and battery constraints.

11.4.1 Qtek 9000

The Qtek 9000 is the most powerful terminal tested in this thesis. This fact is strongly reflected in the results. In nearly every test the Qtek 9000 outperforms the other terminals, providing far better voice quality. It even handles multiple conversations at

once. Up to three simultaneous G.729 bi-directional conversations can be active at once. Using the less processor intensive codec G.711 up to six simultaneous conversations can be handled, which clearly can be considered a teleconference. Also when it comes to Received Signal Strength Indication the Qtek 9000 is superior to the other terminals with a measured RSSI of approximately -90 dBm.

With a single conversation on the Qtek 9000 using G.711 the average MOS value was just over 3.0. Using the G.729 codec however, this value rose to 3.5. In Telenors GSM network 98.95 % of the conversations have a MOS value over 3.0. Using the G.729 codec on the Qtek 9000 79.95 % of all the samples experience a MOS value over 3.0. In the same case 86.74 % of the samples have a MOS over 2.6 which is set as a criteria for the acceptance of Voice over WLAN in chapter 10.1.

11.4.2 Qtek 8300-series

The two Qtek 8300's and the Qtek 8310 all have the same processor and radio. As would be expected these three terminals performed almost identically in all the tests. Common for all tests performed on these terminals was that when using the G.729 codec the stream originating at the mobile terminal greatly outperformed the stream terminating at the mobile terminal. This was mostly caused by packet losses at the mobile side. Although 77% of the packet losses were only a single packet, a considerable amount of the remaining losses were over 3 consecutive packets. The total MOS average, both to and from the mobiles, for all individual tests including these three Qtek terminals was 2.36. This score is just below the criteria set in chapter 10.1 for the voice quality to be acceptable. Also regarding signal strength the three Qteks in the 8300-series were similar with RSSIs averaging around -85 dBm.

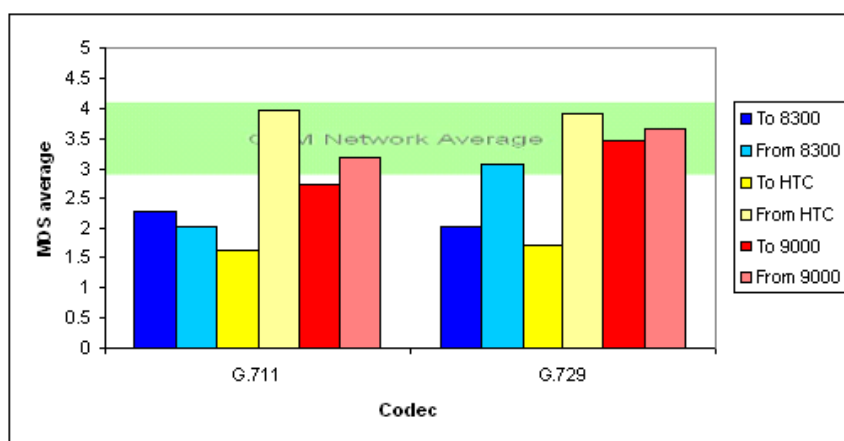


Figure 11.11: Comparison of terminal performance.

Figure 11.11 shows a comparison of the performance of the terminals tested throughout this thesis. Since the two Qtek 8300's and the 8310 performed so evenly their results are combined in the graph. As you can see the Qtek 9000 provided the best voice quality with smallest variance between traffic to and from the terminal. The HTC TyTN performed very well regarding traffic from the mobile terminal, but traffic terminating at the TyTN suffered a great deal. The Qtek 8300-series provided even better voice quality than the TyTN, but as previously mentioned when using the G.729 codec it seems like the terminals have trouble decoding the stream and the conversation suffers large packet losses.

11.4.3 HTC TyTN

The HTC TyTN is the terminal that has performed worst in all the tests. It seems as if it doesn't have the capacity to handle a single two-way conversation. The stream from the phone generally gives a high MOS value, but the stream terminating at the TyTN tends to perform worse and worse as time goes. As figure 11.12 shows, the up-link provides from fair to good voice quality, while the down-link's voice quality has the shape of a converging curve approaching 1.0.

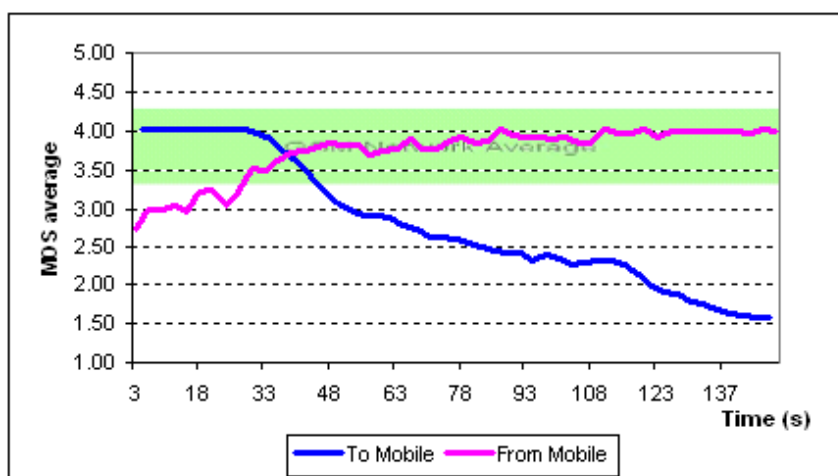


Figure 11.12: Voice quality performance of the HTC TyTN.

Compared to the 98.95 % over 3.0 from Telenors numbers, only 75.76 % of the timing records have a MOS above 3.0 using the HTC TyTN. However, only 47.26 % of the timing records going to the TyTN perform this well, while 91.33 % of the records originating at the TyTN have a MOS above 3.0. The TyTN was the worst of the tested terminals when it comes to RSSI as well with average signal strength of only -80 dBm.

11.5 Subjective Voice Quality Evaluation

Before performing this test Skype™ had to be installed on one of the terminals. The software was installed on the Qtek 9000 which has the "best" capacity and the most memory of the tested terminals. After adding SkypeOut credit to the Skype™-account the actual voice quality achievable with Voice over Wireless could be tested. The quality proved to be worse than expected.

Even in the stationary testing case the perceived conversational quality was far below an acceptable level. The person at the terminating side of the conversation complained over static and jagged audio, while incomprehensible sounds were coming out of the mobile terminals speaker. The audio was both jagged and had gaps several seconds in length. The situation was made even worse when roaming was introduced. The voice quality went from poor to completely inaudible. The small bits of audio that managed to traverse the networks and reach the other terminal were too small and too few to make an understanding of.

11.6 Contra Test at Uninett

To try and get a grip with the rather poor results obtained in the Wireless Trondheim network, tests were performed in the indoor enterprise network at Uninett as well. The purpose with these tests was to try to find the cause of poor performance.

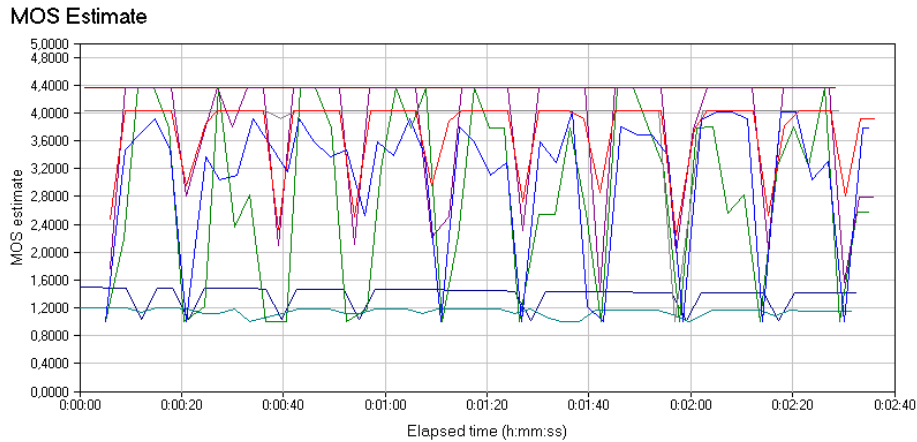
Performing similar tests, using the same terminals, only in another network configuration can help uncover the source of performance weaknesses. It can at least throw light on if the problem lies in the terminal, in the network, or in the compatibility between the terminal and the network.

The network at Uninett differs from the Wireless Trondheim network in several ways. Firstly, the network consists of stand-alone access points rather than light weight access points. This means that there is no central controller that administrates all the access points. Secondly, the network at Uninett has 802.1X compatibility up and running. 802.1X gives a higher level of security than the Wired Equivalent Privacy (WEP) used in the tests in Wireless Trondheim. Using 802.1X will mostly affect the performance during roaming.

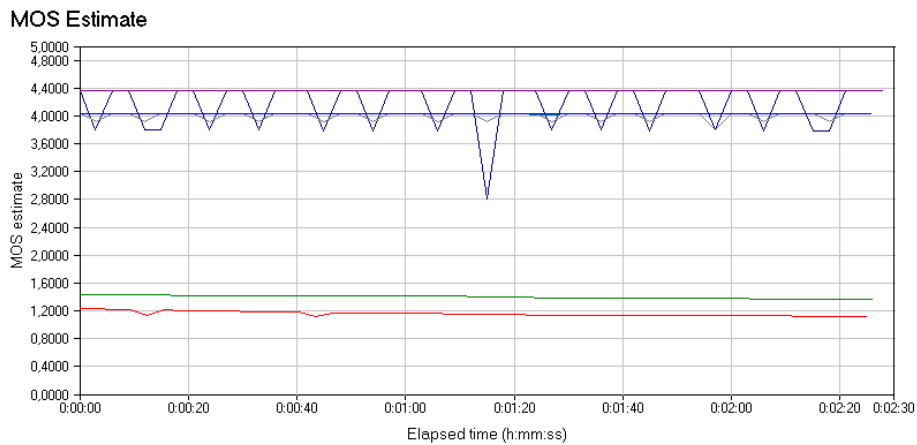
11.6.1 Observations

Both the stationary and roaming tests were performed using the Qtek 9000 and HTC TyTN using the G.711 and G.729 codecs as well as both the WEP and 802.1X security

schemes. The two terminals performed much better in the indoor environment. As you can see from figure 11.13 both terminals maintained more uniform voice quality in the network at Uninett. This can be caused by less interference from external sources or the terminals being more compatible with the infrastructure at Uninett.



(a) Results from Wireless Trondheim.



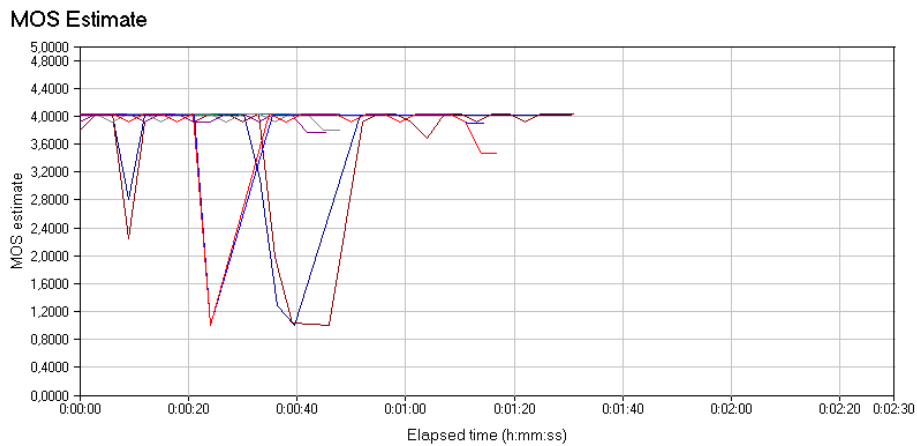
(b) Results from Uninett.

Figure 11.13: Two G.711 and two G.729 conversations.

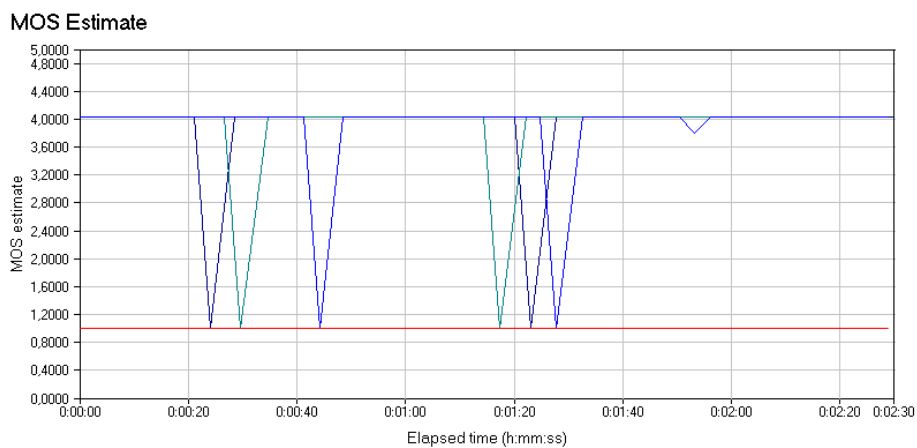
Figure 11.13(a) shows that the voice quality achieved in Wireless Trondheim is constantly shifting from MOS values around 4 and as low as 1.0. In the Uninett network the voice quality is held more constant, as figure 11.13(b) illustrates. In both figures the lines that average between 1.2 and 1.5 are the streams going to the HTC TyTN.

11.6.2 Roaming

Roaming was also handled better in the Uninett network, at least for the HTC TyTN. Using 802.1X authentication which generally will result in longer roaming times, only a single timing record was affected by the switching between access points. While the Qteks connection was broken during the second roam, the TyTN kept the conversation going, and only had a short drop in voice quality exactly when the handover took place. Figure 11.14 shows the voice quality under roaming for the Qtek 9000 and HTC TyTN at Uninett. The horizontal line at 1.0 in figure 11.14(b) is the stream going to the TyTN.



(a) Qtek 9000.



(b) HTC TyTN.

Figure 11.14: Roaming at Uninett.

The roaming tests were performed walking with the terminal through Uninett's offices. The first roam took place after approximately 30 seconds, the second after 90 seconds. Exactly when the roam occurs approximately 75 packets are dropped which causes a 1.5 second gap in the conversation with the TyTN. On the Qtek there was a slightly larger packet loss during the first roam, and the connection was completely lost during the second roam. This could be annoying if roaming occurred very often, but since workers mostly are in their respective offices, roaming wouldn't happen that often.

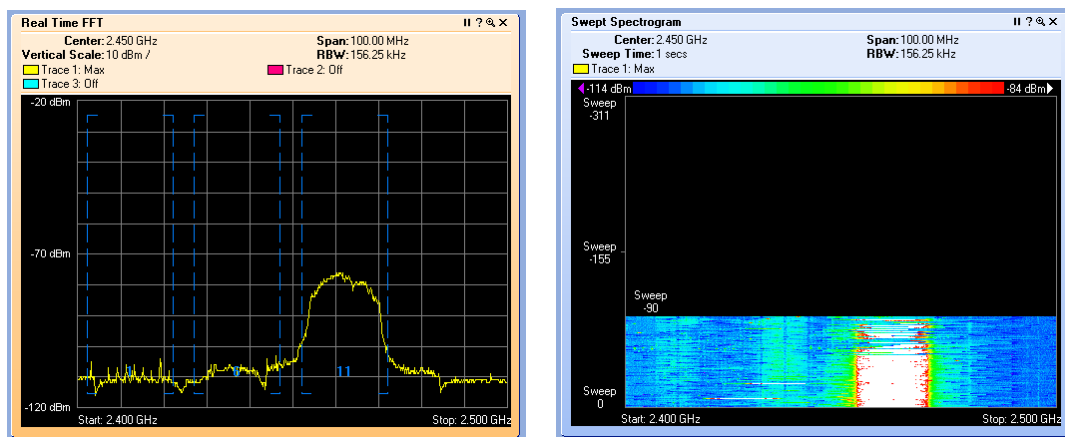
Using WEP security while roaming the results were fairly similar. The main difference was that there were fewer packets lost each roam, approximately 30 packets, representing 0.6 seconds of speech.

11.6.3 Summary

These tests give an indication that interference might be a large problem in Wireless Trondheim since using the same terminals in an isolated environment give far better results. Since laptops perform well in both environments, but the mobiles perform weakly in the citywide wireless network, a possible explanation can be that the hand-held devices' radios tolerate less noise than the laptops'. For instance, in the Wireless Trondheim network the Qtek 9000's MOS scores jump constantly from 4.0/4.4 to 1.0 and back, while in Uninett's network the MOS scores only jump between 4.0/4.4 and 3.8. These even scores will result in a much smoother voice conversation.

11.7 Isolated Access Point Test

After performing the tests at Uninett it became interesting to do a frequency/spectrum analysis in the Wireless Trondheim network. This was done at the most remote location of the network where an access point is alone in a park, far from other access points. A laptop with AirMagnet software and PCMCIA card was used to monitor the activity on the different channels in the 2.4 GHz band. As figure 11.15 shows there was most activity on channel 11, which is the channel the access point uses.



(a) Radio Frequency (RF) power as a function of frequency.

(b) RF power per 1 second sweep.

Figure 11.15: 2.4 GHz band activity.

In figure 11.15(b) you can clearly see that the voice conversation started after approximately 40 seconds where the channel 11 area turns almost completely white.

11.7.1 Observations

A few interesting observations were made during these tests. Firstly, the mobile terminals performed very differently at different distances from the access point. At 40 meters from the AP the HTC TyTN was able to connect to the network, only to drop out a few moments later. The TyTN kept on doing this which made it impossible to use at this distance. The Qtek 9000 had no problems connecting and staying connected at 40 m from the access point. Later on both terminals were moved closer to the AP, approximately 10 m from the external antenna. At 40 m the Qtek maintained a MOS of 2.62, while at 10 m the same conversation scored as high as

3.80 using the G.729 codec. Figure 11.16 illustrates the differences in voice quality obtained at 10 and 40 meters from the access point using the Qtek 9000.

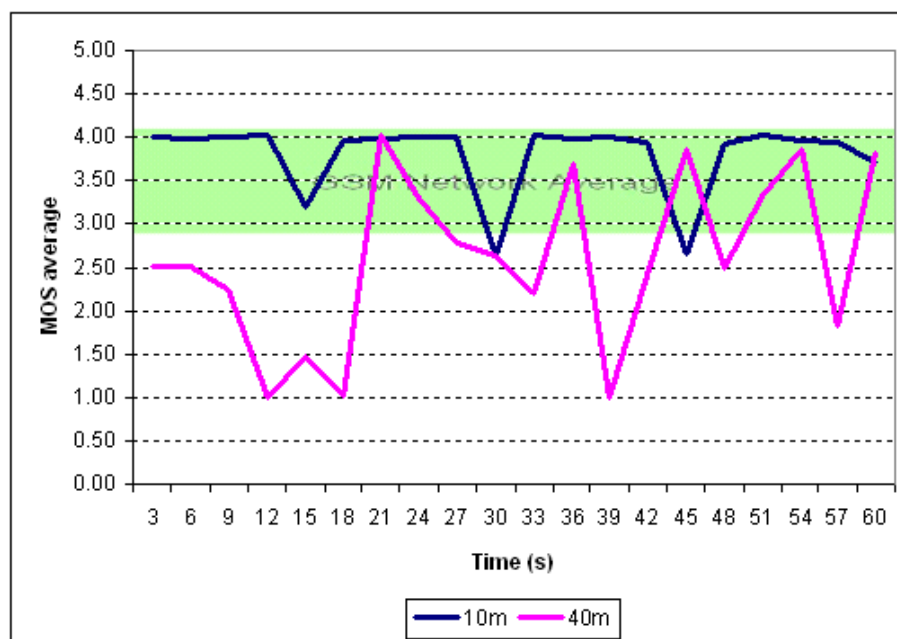


Figure 11.16: Voice quality obtained at 10 and 40 meters from an isolated access point.

Ten meters from the access point as many as 92.5 % of the timing records in the Qtek conversation scored over 3.0 on the MOS scale. This is very close to the 98.95 % that are over 3.0 in Telenors GSM network.

Secondly, compared to the tests performed in the city center the results were much better in the park. At approximately the same distance from an access point the Qtek 9000 scored 0.4 higher on the MOS scale in the park than in the city. The results were also far smoother, only dropping to MOS 1.0 once or twice during a conversation as opposed to the constant jumping between 1 and 4 obtained in the city. These tests also show that there aren't any compatibility problems between the tested terminals and the Wireless Trondheim network.

11.7.2 Subjective Voice Quality Evaluation

A subjective voice quality test was also performed in the park. The audio quality at both ends of the conversation was actually quite good. Indeed there were a few glitches in the conversation, and a few words dropped out, but the overall quality of the conversation was not bad. As the IxChariot tests showed the quality dropped

quickly when walking away from the access point, but the conversation remained audible even at 40 m from the AP.

11.8 Summary

The tests performed with only laptop computers greatly outperformed the ones involving mobile Wi-Fi terminals. While the network had no problem maintaining outstanding voice quality for ten voice conversations between a laptop and a desktop computer, not even six conversations were handled using mobile terminals.

Background traffic significantly reduced the voice quality of ongoing conversations. Adding application data traffic at approximately 3 Mbps caused a 25 % drop in voice quality for four ongoing conversations. Adding background web traffic at approximately 230 kbps lowered the experienced voice quality by a little over 15 %.

The authors subjective evaluation of voice quality using SkypeOut was that the quality was far from acceptable. It was almost impossible to have a sane conversation even in an area with excellent signal reception. However, repeating the subjective test connected to an isolated access point greatly increased the experienced quality.

The tests performed at Uninett and connected to the isolated access point in the park show that interference is a large problem in Wireless Trondheim. The tests performed in the park also show that it isn't compatibility issues that causes the poor voice quality results in the city.

CHAPTER 12

Discussion

The previous chapter presented some of the results obtained from the tests performed throughout this thesis. The following chapter will use these results to appraise the state of voice over citywide wireless networks and the state of mobile Wi-Fi terminals available on the market today.

12.1 Overall Experience

Once again it has to be emphasized that most of the results presented in this thesis are obtained in the Cisco network deployed in Trondheim. Completely different results can be the outfall of testing the exact same terminals under the same conditions in another network. Cisco is known to implement many proprietary solutions in their products which can affect each individual terminals performance significantly. Using Ciscos own terminals in the network would most likely have raised the voice quality quite a bit.

12.1.1 Capacity

The call capacity in each network cell is a constraint. As the test results show each access point can handle approximately six conversations at once. If application traffic is introduced to the access point as well, calls may be dropped and/or the voice quality in the ongoing conversations will drop significantly.

An access point usually has a signaling radius of about 50 meters [29]. This means that when walking down a street in Trondheim you will roam from one access point to another approximately every 100 meters. Walking in one of the main streets in Trondheim on a sunny Saturday afternoon you will most likely see many more than six people every hundred meters talking in their cell phones. Although the density of access points in the city is much higher in such high traffic areas, there will probably be many situations where an access points capacity is breached.

As Network World presents in [32], Cisco isn't the infrastructure manufacturer out there which access points can support the most consecutive calls. The tests carried out in [32] show that Aruba's enterprise WLAN solution provides higher voice quality than Cisco's equipment in nearly every test performed using SpectraLink terminals. One can argue that the terminals from SpectraLink are not comparable to the ones used in this thesis because they are pure Wi-Fi terminals, not GSM/UMTS/Wi-Fi hybrids.

Reducing the transmit power on the access points can be one solution to the cell capacity problem. By reducing the transmit power each access point covers a smaller area, which lowers the risk of overloading an AP. In a wireless network covering an entire city this would be a very expensive solution since lowering the range on each of the access points would require the deployment of a significant amount of new access points.

Another solution could be using a seamless roaming system such as the Generic Access Network (GAN). GAN, also known as Unlicensed Mobile Access (UMA), enables a terminal to seamlessly roam between GSM/UMTS and a LAN if a LAN comes within reach [39]. If this is combined with a capacity check of the LAN before connecting to it, this could greatly prevent access point overloads. Quite a few terminals with GAN/UMA support has already reached the market from large cell phone producers such as Nokia and Samsung. For further reading about UMA the reader is referred to [39].

12.1.2 Terminals

The terminals tested in this thesis had a relatively wide spread of processing power and other specifications for that matter. There also appeared to be significant radio differences between some of the terminals. While all the Qteks had no trouble connecting to the network, and staying connected, the HTC TyTN failed to hold on to a connection unless the signaling power was set to max. Even then the terminal needed straight line-of-sight to an access point to maintain connected. An undesirable consequence from boosting the signal power was that the battery was drained far quicker on the TyTN than the other terminals.

A common characteristic for all the terminals was that the battery power was poor. With the terminals connected to an access point using Wi-Fi the battery didn't hold even a day. Using the terminal for communication as well lowered the battery life to a few hours. If voice over WLAN was your primary network for communication you would have to recharge your phone at least once a day if you wanted to be accessible all the time.

The different processing capacities of the terminals also proved to affect the results, especially using the processor intensive G.729 codec. Interestingly enough it wasn't the terminals with the least processing power that performed worst, it was the fairly new and processor rich HTC TyTN that generally provided the lowest voice quality. The TyTN has a 400 MHz processor, but still the Qtek 8300-series with 195 MHz processors handled the G.729 codec better.

Of course the poor performance of the TyTN can be due to some malfunction in the terminal tested, and completely different results could be achieved using another TyTN terminal.

12.1.3 Interference

As the tests from Uninett show the terminals behave differently in an isolated indoor environment where interference is deliberately counteracted upon. The tests performed connected to an isolated access point in Wireless Trondheim support this as well. It seems obvious that the mobile terminals' radios are far too poor to enable time critical applications to function well in a noisy environment, with interference caused by private access points, cordless phones and microwaves [9]. Data applications may well function because retransmission of lost packets is acceptable, but for the time critical application retransmissions are seldom an option.

A cumbersome radio planning of private access points would likely improve the performance, but is an unfeasible task. The situation will presumably not improve until mobile hand held devices get support for IEEE 802.11a, while, at the same time, private actors do not install private 802.11a networks. If all 802.11a access points are in the control of Wireless Trondheim a mostly noise free environment can exist in the 5 GHz band, hopefully improving the performance of mobile terminals in a voice over WLAN session.

12.2 Comparison with GSM

As previously stated it is unlikely that people will start to use voice applications over the Wireless Trondheim network unless a certain voice quality is maintained. Since

almost everybody uses a cellular phone every day, it is natural to evaluate the quality of VoWLAN up against the GSM phone service.

As you saw in table 10.1 as many as 98.95% of the calls made in Telenors GSM network qualify for a MOS score over 3. Using the G.729 codec on a set of terminals in the Wireless Trondheim network only 77.46% of the conversations score over 3.0 on the MOS scale. This means that almost $\frac{1}{4}$ th of the time the voice quality using voice over wireless is below the quality you get using GSM. If however you are unlucky and somebody is downloading a file while connected to the same access point as you it is likely that only 58.16% of the conversations maintain a voice quality over 3.0. At the same time, only 62% of the time the conversations maintain voice quality above the minimum MOS level of 2.6 set in chapter 10.1.

Compared to GSM voice over a citywide wireless network provides poor and unstable voice quality. It is therefore unlikely that the citywide wireless network in Trondheim will pose a treat to the GSM service providers, but rather act as a supplement for those wanting Internet connection. Some people will probably still use the network for voice traffic, but because of the fiddling required to use VoWLAN and the poor quality achievable this will be a minority of people.

12.3 Comments on Tests

Testing wireless service quality in its real environments proved to be more challenging than initially expected. Not only did the battery life of the test equipment play a role, all tests were also very weather dependent. Since Wireless Trondheims network is designed to give outdoor coverage only, and because electrical equipment seldom handles water very well, all tests had to be performed during a dry spell. Trondheim is a rainy city, so the tests had to be performed whenever the weather permitted it. However, this lead to all tests being performed under fairly even weather conditions so signal depreciatory circumstances such as rain never occured.

As mentioned the battery life of the test equipment played an important role in the tests. The laptops were the most critical units battery life wise, often only being able to operate for a little over an hour. The mobile terminals lasted longer, but not nearly a day of testing. Having to take one to two hour "charging breaks" every hour and a half strongly affected the number of times a test could be repeated and the total number of tests that could be performed i one day.

The battery life problem could have been solved by using an extension cord, but this would have influenced the performance of the terminals. Since as realistic surroundings as possible are desired the mobile terminals had to run from battery power.

Three of the terminals used in the tests were borrowed from Ground Control Labs. They needed the terminals to perform tests themselves, so these were only available for one day. This time constraint limited the repeatability of the tests. While ideally all tests should have been repeated at least three times, this was not possible for all tests involving the Qtek 8300's and the Qtek 8310.

Throughout all the tests the IxChariot batch-mode reporting was used, except when testing roaming where real-time reporting was used. The batch-mode collects all the information from the terminals after the test is complete, while in real-time-mode the terminals constantly report back to the console. Real-time reporting had to be used in the roaming cases because if the roaming terminal completely lost connection with the access point no reporting would be done at all.

12.4 Future of Voice over Citywide Wireless Networks

Voice over wireless LAN is still a young technology, and widespread adoption of it among the general public will probably be long in coming. Using VoWLAN requires some fiddling from the user every time which may scare many users away for now. If however, you could switch to VoIP with the push of a button before placing your call it is more likely that people would put the technology to use. The following sections discuss some of the hurdles VoWLAN needs to overcome.

12.4.1 Indoor Coverage

The Wireless Trondheim network only provides outdoor coverage to its users. If people loose their connection as soon as they walk into a store, this can be a pitfall for the acceptance of voice over the citywide wireless network. A fellow student has performed a series of tests on the indoor coverage in the Wireless Trondheim network in [36]. He measured the signal strength and signal-to-noise ratio (SNR) just inside a window bordering to an area with wireless coverage at 50 random locations in the city. In 40% of the test cases he didn't get a signal at all, while in an additional 20% of the cases the SNR was lower than the acceptable values. His results are presented in figure 12.1, where the results are grouped after Ciscos recommendations on signal strength and SNR for wireless voice cells [17].

As you can see from the figure, only 18% of the tested locations have good enough signal reception to provide data rates up to 12 Mbps. All tests are performed with a laptop which has a far better radio than those in mobile Wi-Fi terminals. The results

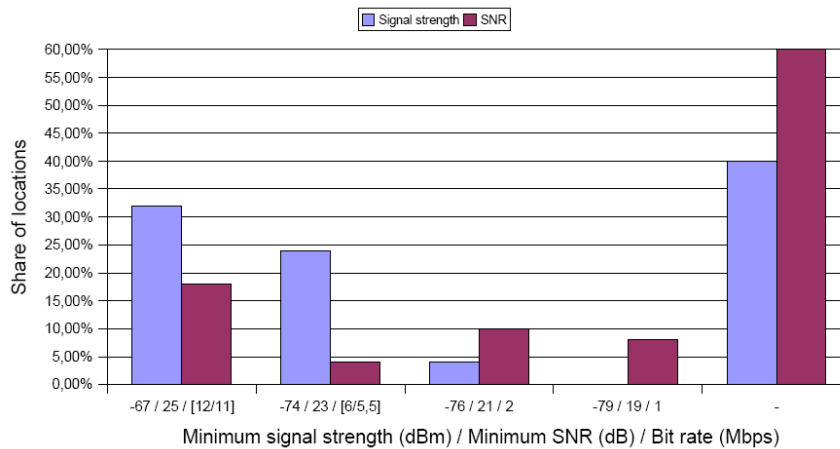


Figure 12.1: Indoor signal measurements [36].

would most likely be quite a lot worse using a mobile Wi-Fi terminal instead, the mobile would probably only be able to get a signal at a very few of the locations.

12.4.2 Terminal Development

Today it seems as if PDA's are approaching mobiles in size and ability to use as a communication device. At the same time mobiles are approaching PDA's in functionality and capacity. In the author's eyes the race will be mostly about getting sufficient battery power. All the mobiles and PDA's tested have been discharged after a few hours actively connected to a wireless network. As more and more people have wireless networks accessible wherever they go, there will be an increasing desire to stay connected to a wireless network all day. If this means recharging once or twice a day, it is unlikely that voice over WLAN will catch on as anybody's primary communication form.

12.4.3 QoS Guarantees

Until a complete implementation of the IEEE 802.11e quality of service enhancement to 802.11 is available, guaranteed quality of a voice service is not achievable. Currently only partial and proprietary implementations are available, prohibiting interoperability between all the terminals and infrastructure available. In the Trondheim network you can achieve a guaranteed service to a certain degree using Cisco terminals. However, it is unlikely that people will purchase a Cisco phone just to use in the wireless network.

CHAPTER 13

Conclusion

The objective of this thesis was to perform a series of field experiments on Voice over IP in the Wireless Trondheim network. Using the network analysis tool IxChariot, Voice over IP was tested on a set of Wi-Fi enabled Qtek and HTC terminals under varying network conditions. Through tests of the available terminals, under different network compositions and loads, the overall quality of the service was assessed. The quality of the service was also compared up against results from GSM in order to see if a voice service over the Wireless Trondheim network can compete with GSM.

The tests and results presented throughout this thesis can be used as a basis for future benchmarking of mobile terminals in Wireless Trondheim as the terminals and network evolves. Slightly modifying the test setups, the test methodology can be used in another wireless network as well.

One of the main findings through the experiments was that in a city like Trondheim there is far too much interference for a mobile terminal to obtain a stable voice conversation with good quality. Today's mobile Wi-Fi terminals operate in the 2.4 GHz band, alongside private wireless networks, cordless phones and microwave ovens. A densely populated city center therefore contains a lot of interfering signals. On average, each of the over 100 access points in the Wireless Trondheim network sees 12 interfering access points not belonging to the network. This naturally influences the clarity of the received signal at the terminals as well.

The obtained voice quality has been measured on a five point scale, called the mean opinion score (MOS), where 1 is poor and 5 is excellent quality. Nearly all calls made in the GSM network score over 3.0 on the MOS scale. Taking this, among other

factors, into consideration the criteria for acceptable quality for the voice over Wireless Trondheim service was set to 2.6 on the MOS scale. The tested terminals have performed differently, however, the obtained quality in the tests has been constantly lower than the criteria set for the service. The voice quality in the network degrades further when adding data traffic to the network. The presence of ordinary web browsing traffic causes the voice quality of ongoing conversations to drop almost 0.5 on the MOS scale.

An indirect factor also affecting the experience of voice over the Wireless Trondheim network was the battery life of the mobile terminals. Connected to a wireless network the terminals' batteries only last a few hours. Considering that most people have wireless networks available both at work and at home, it is likely that a practically free voice over wireless service will be desired to use as ones primary communication medium. If this involves recharging your terminal several times a day it is not likely to catch on in the near future.

Without any prioritizing of latency sensitive traffic the shared wireless network seems unsuitable for VoIP services. After testing several different terminals in the Wireless Trondheim network the conclusion is quite clear, voice over the citywide wireless network is not yet ready for the masses.

CHAPTER 14

Future Work

Testing voice quality under various Quality of Service configurations has not been subject in this thesis, but would be interesting to examine in the future. As complete implementations of the QoS enhancement to the 802.11 standard, 802.11e, hit the shelves it would be interesting to perform the same tests again. The results would most likely be better than those obtained in the current network configuration, at least in cases where background data traffic is present.

Repeating the tests with mobile terminals when mobiles with 802.11a come into the market would be an interesting extension. Since the 5 GHz band 802.11a operates in is much less crowded, it is more likely that sufficient signal quality can be present for achieving good voice quality over WLAN.

It would also be interesting to test roaming from the Wi-Fi network to the GSM or UMTS networks. Using UMA a dual-radio terminal can roam between the networks and provide a continuous service using the best network available.

PART IV

Appendix

APPENDIX A

Terminal data

A.1 HTC TyTN

ROM version: 1.18.259.2
ROM date: 07/22/06
Radio version: 1.05.02.10
Protocol version: 32.36.7010.04H
ExtROM version: 1.18.259.105

CPU: Samsung®2442
Speed: 400 MHz
RAM size: 64 MB
Flash size: 128 MB
Data bus: 32 bits
Storage size: 56.46 MB
Screen resolution: 240 x 320
Colors: 65536

Model nr.: HERM200
Platform: Pocket PC
IMEI: 35771900054308601

Microsoft®Windows Mobile™version
OS 5.1.195 (Build 14955.2.3.0)
Processor: SC32442A
Memory: 49.05 MB



Figure A.1: [12]

A.2 Qtek 8300 I

CPU:	TI OMAP850
Speed:	195 MHz
RAM	64 MB
Flash:	64 MB
ROM version:	1.0.7.0
Operator version:	1.6.311.1
LCD:	240 x 320 TFT
Color:	65536
Model name:	Qtek 8300
IMEI:	35629700005909900
Radio version:	4.0.13.17_01.04.00
Microsoft® Windows OS 5.1.70	Mobile™ Version 5.0 (Build 14406.1.1.1)
Radio version:	4.0.13.17_01.04.00
RIL version:	2.002
Operator/Manufacturer:	5.1.70.14406



Figure A.2: [13]

A.3 Qtek 8300 II

CPU: TI OMAP850
Speed: 195 MHz
RAM: 64 MB
Flash: 64 MB
ROM version: 1.0.7.0
Operator version: 1.6.321.1
LCD: 240 x 320
Color: 65536
Model name: Qtek 8300
IMEI: 35629700004736700
Radio version: 4.0.13.17_01.04.00

Microsoft® Windows Mobile™ Version 5.0
OS 5.1.70 (Build 14406.1.1.1)
Radio version: 4.0.13.17_01.04.00
RIL version: 2.002
Operator/Manufacturer: 5.1.70.14406



Figure A.3: [13]

A.4 Qtek 8310

CPU:	TI OMAP850
Speed:	195 MHz
RAM	64 MB
Flash:	64 MB
ROM version:	1.0.7.0
Operator version:	1.5.321.2
LCD:	240 x 320 TFT
Color:	65536
Model name:	Qtek 8310
IMEI:	35629800722414000
Radio version:	4.0.13.17_01.04.00
Microsoft® Windows OS 5.1.70	Mobile™ Version 5.0 (Build 14406.1.1.1)
Radio version:	4.0.13.17_01.04.00
RIL version:	2.002
Operator/Manufacturer:	5.1.70.14406



Figure A.4: [14]

A.5 Qtek 9000

ROM version: 1.13.64 WWE
ROM date: 09/30/05
Radio version: 1.03.00
Protocol version: 42.36.P8
ExtROM version: 1.13.151 WWE

CPU: Intel®PXA270
Speed: 520 MHz
RAM size: 64 MB
Flash size: 128 MB
Flash chip type: G3
Data bus: 32 bits
Storage size: 43.50 MB
Screen resolution: 480 x 640
Colors: 65536

Model nr.: PU10
Platform: Pocket PC
IMEI: 355195001080547

Microsoft®Windows Mobile™versjon 5.0
OS 5.1.1700 (Build 14354.0.1.1)
Processor: PXA270520MHz
Memory: 50.19 MB



Figure A.5: [15]

Glossary

AES	Advanced Encryption Standard
BSS	Basic Service Set
CBR	Constant Bit Rate
CCX	Cisco Compatible Extensions
CELP	Code-Excited Linear Prediction
FTP	File Transfer Protocol
GAN	Generic Access Network
GSM	Global System for Mobile communications
IEEE	Institute of Electrical and Electronics Engineers
iLBC	internet Low Bitrate Codec
ITU	International Telecommunications Union
LAN	Local Area Network
LWAPP	LightWeight Access Point Protocol
MOS	Mean Opinion Score
MPEG	Moving Pictures Experts Group
NIC	Network Interface Card
NTNU	Norwegian University of Science and Technology
OS	Operating System

PC	Personal Computer
PCM	Pulse Code Modulation
PCMIA	Personal Computer Memory Card International Association
PESQ	Perceptual Evaluation of Speech Quality
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RF	Radio Frequency
RSSI	Received Signal Strength Indication
RTP	Real-Time Transport Protocol
SIP	Session Initiation Protocol
SNR	Signal-to-Noise Ratio
TCP	Transmission Control Protocol
TKIP	Temporal Key Integrity Protocol
UDP	User Datagram Protocol
UMA	Unlicensed Mobile Access
VoWLAN	Voice over WLAN
WCS	Wireless Control System
WDS	Wireless Domain Services
WEP	Wired Equivalent Privacy
Wi-Fi	Wireless Fidelity
WLAN	Wireless Local Area Network
WMM	Wireless Multimedia Extensions
WPA	Wi-Fi Protected Access
WPP	Wireless Performance Prediction

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