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Studying Quality of Experience (QoE) over Wireless Networks

Bachelor Thesis on Informatics

by
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Abstract

Quality of Experience is showing strong popularity the last years. More and more developers try to find new products and services, or improve the already existing to make customers' life better. Informatics Engineers working on the wireless networking field, have understood that after the evolution of Quality of Service, it is obvious now that the perceived quality of a service depends in many factors above the network ones that are already, in most cases, handed by the companies. Also basic users most of the times, are not able to understand why their service isn't good and many times are not able to explain what bothers them on this service. This is the spot that Quality of Experience came on the scene, by providing Informatics Engineers with tools to measure the perceived quality of a service and help them find new ways to improve it and give products to users with better quality.

In this thesis, we are focusing on wireless network technologies of IEEE 802.16 (WiMAX), IEEE 802.11, 3GPP LTE and other various Cellular technologies and after a detailed introduction of what is Quality of Experience and how every technology works, we present a literature review of how Quality of Experience is handled and used by researchers and how it is associated with the previous mentioned technologies. More specifically in Chapter 1 we have the presentation of Quality of Experience, in Chapter 2 all the technologies that are found later in this thesis are analyzed, in Chapter 3 we have the summarization of the work of the researches so far and finally in Chapter 4, future work and challenges related to Quality of Experience and previous works are presented.

This work provides an introduction about the Quality of Experience domain and focuses on how QoE pertains to wireless technologies. In addition, the current thesis studies several works that have published by researchers in this field. Finally, although the thesis is focused on the scientific and technical character of QoE, four different groups of challenges are presented and extensively studied.

Contents

CHAPTER 1

Quality of Experience

1.1 Introduction

As Informatics Engineers we tend to accept products that are more based on newer technologies. The situation changes when the product ships to mainstream users. Normal users don't care nearly so much about which technology goes into a product but they care more about the problems that it afflict and the problems that it solves. At given products with the same functionality, the user will choose the product that will make the perceived experience and quality much better. However, the concept of experience is complex. To understand what experience is, users might want to specify more precisely the context in which they are discussing it, for example, the quality of clothes, sound quality, video quality etc.

But experience is *subjective*. As an example, by changing a few colors in an interface will change the effect it has on different people. Moreover, experience is *context-dependent*. The same book can result in a different experience by the same person depending on the context.

At this time Quality of Service comes into foreground. As a definition we can say "On the Internet and in other networks, Quality of Service (QoS) is the idea that transmission rates, error rates, and other characteristics can be measured, improved, and to some extent, guaranteed in advance. QoS is of particular concern for the continuous transmission of high-bandwidth video and multimedia information. Transmitting this kind of content dependably is difficult in public networks using ordinary 'best effort' protocols" [1]. QoS is currently not only a technical issue. It became also a kind of product and marketing subject. Sometimes telecommunication services are advertised as providing QoS control, support and so on, but in fact they have not much in common with a real QoS as meant in communication standards.

To cover different areas and views on QoS, at least the following terms should be distinguished: Class of Service (CoS), Grade of Service (GoS), Quality of Resilience (QoR), and Quality of Experience (QoE) that includes all the previous terms. Sometimes, because the service quality is still growing, the acronym QoX is used and concludes all the terms above.

1.2 Quality of Service (QoS)

There are three notions of QoS defined: *intrinsic*, *perceived* and *assessed*. The formal definition provided by ITU-T in Rec. E. 800 [2] is as follows "*the ability of a network or a network portion to provide the functions related to communications between users*". Also, IETF deals with intrinsic QoS and defines it as "*a set of service requirements to be met by the net work while transporting a flow*". High intrinsic service quality is a technical challenge and is a key for quality perceived and assessed by the customer. The required quality is achieved, among other things, by appropriate selection of transport protocols, the QoS assurance mechanisms, and related values of parameters [3].

Perceived QoS reflects the customer's experience of using a particular service. It is affected by the customer's expectations com- pared to observed service performance. To cover various points of view on QoS, ITU in ITU-T Rec. G. 1000 [bale reference] distinguishes four definitions:

- QoS requirements of the costumer
- QoS offered by the provider
- QoS achieved by the provider
- QoS perceived by the customer

QoS requirements of the costumer are usually expressed in a non-technical language. They are influenced by many factors, like the customer's experience. Service providers take into account these expectations and their own business strategies. All these are written on a Service Level Agreement (SLA). QoS offered by the provider is realized by using several network mechanisms and techniques. The effect is observed as QoS achieved by the provider and expressed in the same, mostly technical, terms as QoS offered. Finally, the service quality is experienced by the customer (QoS perceived by the customer). The costumer decides whether to continue using the service or not (assessed QoS). This decision depends on the quality of the service, the price, or the feedback that to submitted complaints and problems.

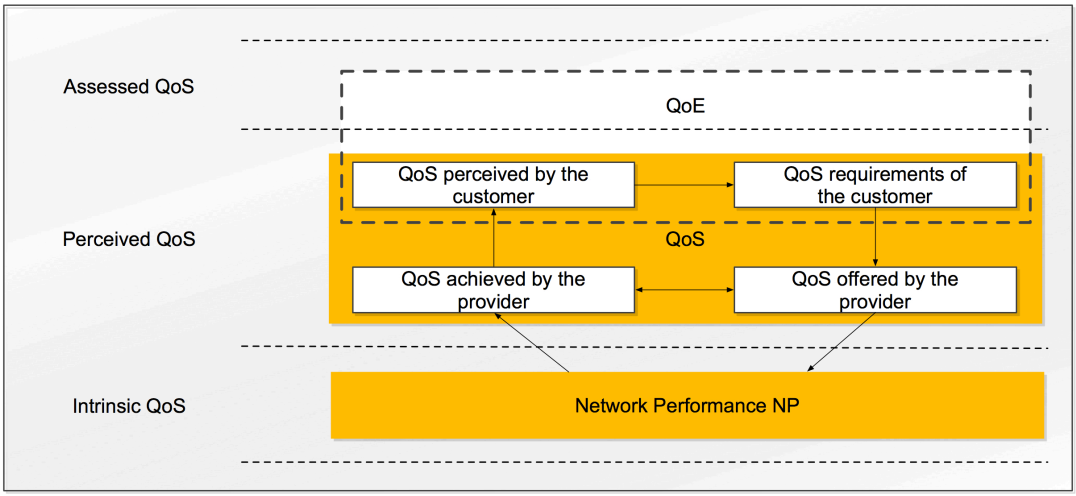

Figure 1 – ITU-T Terminology and standards in relation to the general QoS model

According to QoS level of the general model, different QoS parameters are used. Some of them are: bit rate, delay, jitter and packet loss. These parameters can describe the treatment experienced by packets while passing through the network. Requirements from the customers' perspective are defined in a way meaningful to them and they are specific to a particular service being independent of the networking technology (Quality of Experience).

A defined quality of service may be desired or required for certain types of network traffic:

- Streaming media:
- Internet Protocol Television (IPTV)
- Audio over Ethernet
- Audio over IP
- IP Telephony (VoIP)
- Videoconferencing
- Telepresence
- Safety-critical applications such as remote surgery where availability issues can be hazardous
- Online real-time games where lag can be a factor which may affect the performance

All these applications are called inelastic. We will introduce elastic and inelastic definitions later on this paper.

QoS in the field of telephony was first defined in 1994 in the ITU-T Rec. E.800 [2]. Some wireless standards that support Quality of Service are IEEE 802.11e [4], Worldwide Interoperability for Microwave Access (WiMAX) [5], Long Term Evolution (LTE) [6], Universal Mobile Telecommunications System (UMTS) [7], IEEE 802.11Q, IEEE 802.11p. Some other technologies are Frame-Relay, which is a standardized wide area network technology that specifies the physical and logical link layers of digital telecommunications channels using a packet switching methodology, X.25, which is an ITU-T standard protocol suite for packet switched WAN communication and it is being replaced by Frame-Relay. Also Asynchronous Transfer Mode (ATM) supports QoS. It uses asynchronous time-division multiplexing, and it encodes data into small, fixed sized cells. This differs from approaches such as the Internet Protocol or Ethernet that use variable sized packets or frames. Finally Resource Reservation Protocol (RSVP), Multiprotocol Label Switching (MPLS), Differentiated Services (DiffServ) and some Digital Subscriber Lines (DSL) modes may support QoS.

1.3 Class of Service (CoS)

There are three basic definitions of Class of Service. ITU-T Rec. E.360 [8] defines it as "*characteristics of a service such as described by service identity, virtual network, link capability requirements, QoS and traffic threshold parameters*". Another definition is found in ITU Rec. E.417 [9]: "*any of the network-oriented designations or features that can distinguish between various services, or application-layer uses, of lower-layer telecommunications capabilities for the purpose of more effectively accommodation the specialized network performance needs of specific services*". The third definition provided by IETF RFC 2386 [10], reads as follows: "*The definitions of the semantics and parameters of a specific type of QoS*".

Services belonging to the same class are described by the same set of parameters, which can have qualitative or quantitative values. Usually, the set of parameters within the class is defined without assignment of concrete values, but these values can be bounded [3].

Class of service can be distinguished in two main viewpoints: the applications level and the network level. Applications sharing some features usually generate specific type of traffic and have similar requirements to be met to ensure high performance of the application. At the application level, four types of applications are found:

- Elastic non-interactive (ex. File downloading, p2p file sharing)
- Elastic interactive (ex. Web browsing, telnet)
- Non-Elastic non-interactive (ex. Video on demand, Live TV)
- Non-Elastic interactive (ex. VoIP, Video- Chat)

The applications can be categorized as generating symmetrical (equal bandwidth in both uplink and downlink directions) and asymmetrical (more bandwidth for downloads by sacrificing bandwidth available for uploads) traffic. Another feature that is important is bandwidth. Applications can be distinguished as requiring high or low bandwidth. Applications that require high bandwidth are called inelastic, meaning that they require a certain minimum level of bandwidth and a certain maximum latency to function. By contrast, elastic applications when encounter delay, loss or bandwidth limitations, adapt their rate to maximize throughput.

On the network level, a set of classes must be well defined and characterized by a set of parameters. QoS classes are defined by several organizations, like IEEE or 3GPP. Some examples of QoS classes can be found in Chapter 2, where the wireless networks are analyzed.

1.4 Grade of Service (GoS)

As a small definition for Grade of Service, we can say that is the acceptable level of traffic that a network can lose. It describes all the phenomena occurring during connection set-up, release and maintenance. GoS has been used in the telecommunications industry to indicate components (technical and human), which contribute to overall quality of service what the user receives. The technical components can be measured like connection set up delay or bandwidth of voice. Human components are subjective. There is relation between human and technical components but the exact mapping depends on many factors, for example, the language that is used and other cultural factors [11].

Currently GoS is applied to circuit switched optical services and it can be divided in two standards [12]:

- *Loss Grade of Service*: This standard has component internal loss probability.
- *Delay Grade of Service*: There are several components in this standard depending on technology used for signaling information.

GoS parameters are very important for service differentiation in Wavelength-Division Multiplexing (WDM), Automatically Switched Optical Networks (ASON), and Generalized Multi-Protocol Label Switching (GMPLS) networks and in general, they are even more meaningful that QoS parameters. However, GoS classes in optical networks are under research.

1.5 Quality of Resilience (QoR)

Traditionally the service classes are defined with QoS parameters like delay, jitter, bit error rate, etc. Reliance has been perceived as one of the dimensions of QoS and it concerns mainly the probability that a service is operational.

The goal of a company is to provide a service to its customers. The demand of the customer is to be satisfied from this service. So recently the impact of resilience on service quality has gained more attention and is recognized as an independent field. The motivations are twofold [13]: First, resilience is extremely important to society. Second, a wide range of survivability mechanisms providing variable QoS to a user makes resilience an area independent of transmission/transfer performance evaluation. Thus, Quality of Resilience is emerging.

Approaches relate to QoR are not well formalized as the ones in QoS, CoS or GoS. Approaches like the one that is proposed in [2] define a service as available or unavailable, by determining basic measures such as reliability function, availability etc. Other approaches can be found in ITU-R Rec. G.911 [14] for fiber optic systems, ITU-T Rec. M.1301 [15] for Synchronous Digital Hierarchy (SDH) systems, ITU-T Recs. Y1540-1542 [16-18] for IP networks, or ITU-T Rec. Y.1561 [19] for MPLS connections.

Another area of interest includes frameworks of service class definitions based on resilience properties. Figure 2 shows the relations between QoS

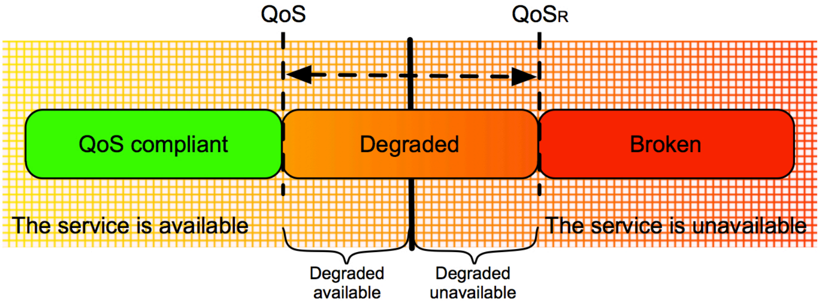

Figure 2 - The service state model

and QoR. If a service meets all QoS requirements defined in its Service Level Agreement (SLA), it is considered to be QoS compliant. If any of those requirements are violated, the service is considered degraded. If they are significantly violated the service is unavailable.

A second set of QoS parameters can be defined in the SLA for the definition of service availability. These parameters are denoted as QoS_R [20]. Choosing QoS_R strongly depends on applications. They are often the same as the QoS parameters defined in the SLA (the service is available if the QoS parameters are fulfilled). Therefore, QoS_R parameters adjustment should be strongly related to service classes' differentiation. For other applications, QoS_R is defined with so loose requirements that only a broken service would be treated as unavailable. The reason in that such an approach does not force an operator to provide a very well service, with consequence resilience-related metrics giving average results for long periods of time, and the requirements of the customers are not being indulged.

Unavailability is generally caused by failure that is not predicted and can't be recovered within a short period of time thanks to resilience mechanisms. For that reason Mean Time Between Failures (MTBF) and Mean Time to Repair (MTTR) where also introduces in SLA (Fig.3).

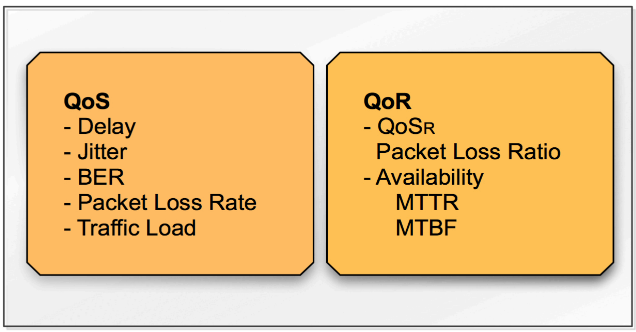

Figure 3 - The service class parameters [20]

1.6 Quality of Experience (QoE)

Quality of Experience (QoE) has made a rapid career recently. The first examples of QoE were introduced by the industry. Nokia introduced this concept as a perception of the end users about a service quality and stated [21]: "*QoE is how a user perceives the usability of a service when in use – how satisfied he or she is with the service*". DSL Forum gave another definition of QoE, and defined it as measure and an indicator of a system in fulfilling the requirements of the customer: "*QoE is a measure of end-to-end performance at the service lever from the user perspective and an indication of how well the system meets the user's needs*" [22]. Finally, another definition may be found in ITU-T Rec. P.10/G.100 [23]: "*the overall acceptability of an Application or service, as perceived subjectively by the end-user*". The quality of a service is influenced by many parameters, like hardware that is used, protocols, techniques etc. Also QoE is influenced by GoS, QoR, and QoS intrinsic parameters. QoE mostly relies on user survey and cores from the user. It is a more "subjective" approach of determining the quality of a service. The overall QoE evaluation is additionally affected by environmental, psychological, and sociological factors, including user expectations and experience with similar services, other opinions, pricing policies, features of the particular location where the service is received, etc. (Fig. 4). In [24], influence factors are grouped in three categories, named *Human IF*, *System IF* and *Context IF*.

A **Human IF** is any variant or invariant property or characteristic of a human user. The characteristic can describe the demographic and socio-economic background, the physical and mental constitution, or the user's emotional state. These types of factors are complex and strongly interrelated. They may influence the perceptual process at two important levels. At the level of low-level processing properties related to the physical, emotional and mental constitution of the user and might play a major role, like the user's gender or age (*dispositional*) or user's mood and personality traits (*dynamic*). At the level of higher-level cognitive processing, interpretation and judgment, other human influencing factors are important. Some examples are the

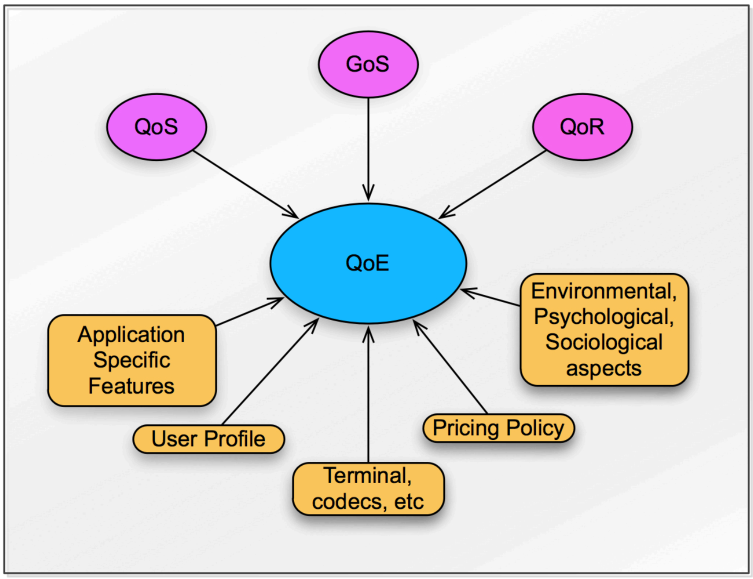

Figure 4 - Factors influencing QoE [13]

educational background, previous experience, emotions, and socio-economic situation.

System IFs refer to properties and characteristics that determine the technically produced quality of an application or service. They are related to media capture, coding, transmission, storage, rendering, and reproduction/display, as well as to the communication of information itself from content production to user. The System IFs may be divided into four sub-categories:

- *Content-related System IFs* referring to the content type and content reliability (color depth, texture, 2D/3D, etc.).
- *Media-related System IFs* referring to media configuration factors (sampling rate, frame rate etc.).
- *Network-related System IFs* referring to data transmission over a network (bandwidth, delay, jitter, packet loss etc.).
- *Device-related System Ifs* referring to the end systems or devices involved along the end-to-end communication path, including *system specifications* (security, privacy etc.), *equipment specifications* (mobility, type, usability etc.), *device capabilities* (display size, headphones, battery lifetime etc.) and *provider specification and capabilities* (server performance and availability).

Finally, **Context IFs** are factors that embrace any situational property to describe the user's environment in terms of physical, temporal, social, economic, task, and technical characteristics. These factors can occur on different levels of magnitude, dynamism, and patterns of occurrence, either separately or as typical combinations of all three levels. The physical context describes the characteristics of location and space, including movements within and transitions between locations. Temporal aspects of the experience, e.g. time of day, duration, and frequency of use (of the service/system), are covered by the temporal context. Costs, subscription type, or brand of the service/system are part of the economic context. The experience can be perceived focused or in a multitasking situation (i.e., task context), alone or with other people present or even involved in the experience (i.e., social context). Finally, the technical and information context describes the relationship between the system of interest and other relevant systems and services including devices (existing interconnectivity of devices), applications (availability of an application instead of just a browser based solution), networks (availability of other networks than the one currently used), or additional informational artifacts (additional use of pen and paper for better information assimilation from the service used).

In fact, factors such as service accessibility, service availability, service usability (ease of use), service integrity (session quality), and service reliability/continuity are very important in QoE evaluation by the user, especially in the case of voice and video services. Some of the side factors are independent, for example the user profile or the price, and others are not (background noise on voice quality). For instance, an end-to-end QoE satisfaction of the end users influenced by their experiences in two stages, the stage before session and the stage in session [25]. Finally, a free service will be evaluated differently if it is free or charged. Customers are more exigent if they pay for a service. On the other hand they will accept some quality degradation of the service, if it is free. (Fig. 5)

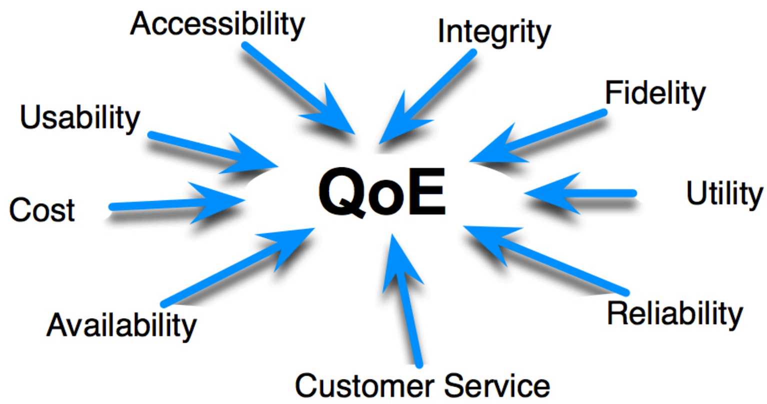

Figure 5 - Examples of Factors

1.6.1 QoS vs. QoE

The foremost difference between QoE and QoS resides in the following points: the former one more focus on what the end user feels, whereas the latter concept is more a measure from the network aspects. Basically, the relationship between QoE and QoS can be concluded in two aspects: firstly, QoE extends the concept of QoS. QoS only encompasses one part of QoE scope. Second, QoE needs the support from QoS, and inversely, QoS performance can impact QoE satisfaction.

End-to-end QoS scope may cover all of the network elements within a traffic flow. Its performance can be evaluated by end-to-end network measures, such as BER (Bit Error Ratio), PLR (Packet Loss Ratio), latency, etc., which usually have little meaning to the end user. On the other side, QoE directly reflects how the end users think about the service. It refers to their experiences during the service utilization, if he/she was satisfied with the service quality etc. For example, a user may experience breaks during the call or he/she may have to wait before talking in order to be sure that the conversation partner has already finished. These speech breaks and long waiting period may be caused by network errors such as network congestion, or due to a wrong software configuration. The user usually does not see the reason behind the bad experiences, but can certainly sense them. And all of these bad experiences can impact his/her overall perception on the service quality.

So, as it obvious, the user's QoE of a service is far beyond a technical metric. It covers a wide scope involving different partners with different responsibilities, e.g., the service/content offering by the service provider, service delivering by the network provider, service utilizing by the end user, etc. Besides of the quality of the transport network, i.e., network QoS, other factors including subjective ones such as user's expectation and user's experience on the service, can also impact the user's perception. For example, a user with experience on a particular service may have higher requirement on the service quality than an inexperienced one. On other hand, since the service has to be delivered through the network, the network QoS can influence user's experience. A poor network QoS usually will result in a user's poor QoE, and a good QoE satisfaction often implies a good QoS of the transport network. The packet loss in the network, for instance, may result in a bad speech quality and disappoint the user. And a good user's experience of the speech quality usually indicates few or even zero packet losses during the conversation. Nevertheless, fulfilling all the QoS requirements cannot guarantee a good QoE satisfaction. In Table 1, the comparisons between these two concepts, QoE and QoS, are concluded.

1.6.2 Classification of QOE Evaluation Methods

Existing quality assessment technologies can be classified into three categories [13]:

- Subjective quality assessment schemes
- Objective quality assessment schemes
- Network planning models

Table 1 – Comparisons between QoS and QoE [25]

	QoS	QoE
Orientation	A description of service quality. Network oriented, from network/providers perspective.	A description of service quality. User oriented, from user's perspective.
Concept	QoS is limited to a technical concept and usually a QoS measurement cannot directly reflect a service problem.	QoE is subjective concept. It directly reflects the user's perception of the service quality.
Metrics	The QoS parameters can be packet loss, jitter, delay, and throughput, etc., which usually has little meaning to an end user.	QoE metric such MOS (Mean Opinion Score) directly expresses the user's satisfaction.
Impact	QoS can impact the service QoE.	QoE satisfaction needs the support of a good network QoS.

Subjective tests are performed with the involvement of human testers. People using the service in a real environment evaluate the service, usually filling in questionnaires. Normally, subjective tests involve a relatively large group of subjects. Also it is the only way to access the psychological and sociological impacts of QoE. Although it is the most credible method to evaluate the quality, they are very costly to perform and can't give us real-time results. On the other side, significant efforts have been devoted to the development of objective quality assessment technologies. They provide QoE evaluation based on the measurement of several parameters related to particular service delivery or service quality indicators in the signal. Objective methods include quality degradation models (like the E-model), instrumental metrics (e.g., Perceptual Evaluation of Speech Quality (PESQ)), or neural network approaches (Rubino's Pseudo-Subjective Quality Assessment (PSQA) Method). Objective methods are intended to overcome the drawbacks of subjective tests. Most objective metrics propose different methods to compare the received sample against the original one. While these metrics lower the cost of quality assessment, their correlation with subjective scores can sometimes be low, mainly when networking parameters are taken into account [26]. On [27] a more detailed approach of subjective and objective measurements is presented.

Depending on the amount of information that is needed for evaluating a service, architectures are classified on three categories: Full Reference (FR), No-Reference (NR), and Reduced Reference (RR).

- Full Reference (FR): The video at the input of the system (which acts as a reference) is compared with the processed signal at the output of the system to determinate quality objectively. But, unfortunately, it can't be applied to packet networks. The ITU use this methodology in order to obtain user requirements for objective perceptual video quality measurements in digital cable television [28]. In [29], the comparison is done on a frame-by-frame basis and, thus, requires precise alignment of the two video sequences, which can be an issue if there is variable delay in the system. Some example metrics are PEVQ (Perceptual Evaluation of Video Quality), and SSIM (Structural Similarity Index) for video, and PESQ for voice.

- Reduced Reference (RR): In this approach, only selected parameters are extracted from the input and output to be compared. It reduces the transmission bandwidth consumption considerably. In [30], the authors only use parameters such as packet loss and related jitter. This work motivates a fundamental relationship between it and the quality impairment factors mentioned before. In [31], the quality measurement techniques have been proposed based on objective. They propose an audiovisual quality assessing method using equivalent Signal-to-Noise (S/N) ratio conversion method. It is used to describe the results of an assessing test carried out with TV telephones. This method also estimates a synthetic quality from individual quality of voice and picture.
- No Reference (NR): Only the received video signal is used to determine the video quality objectively. It is also known as a "single ended" technique. This technique reduces the resembling errors. Some examples of metrics are noise, sharpness/blur, colorfulness, etc. and can be found more in [32].

Network planning models are subcategory of objective models since they do not involve human testers. These models do not require the examination of the real signal, but estimate the expected QoE using a function that maps several measurable intrinsic quality parameters (especially QoS) to the QoE metric. An example of a network planning model is E-Model that is used for analyzing conversational voice quality. E-model is a useful tool for assessing the combined effect of all of the above mentioned parameters, including transmission delay [33].

1.6.3 QoE Measure and Metrics

1.6.3.1 Metric

A metric is defined as *"a system of related measures that facilitates the quantification of some particular characteristic"*, and as such it implies a well defined and measurable component. However, when defining metrics for QoE assessment there are many subjective factors, which might bias the user satisfaction towards a service, hence increasing the complexity of computing objective metrics. As a consequence, there are notable research efforts in objectifying such factors in order to have a measurable value of the user satisfaction in general.

The ITU recommendations [34-35] suggest a 5-point scale, which spans from bad to very good quality. Absolute scales are more commonly used (Fig. 7). Sometimes, especially on subjective tests where people are asked to compare two examples, comparative metrics are used. Except from the quantitative approach (Fig.6), we can find a qualitative approach on metrics, either quality oriented or impairment oriented (Table 2).

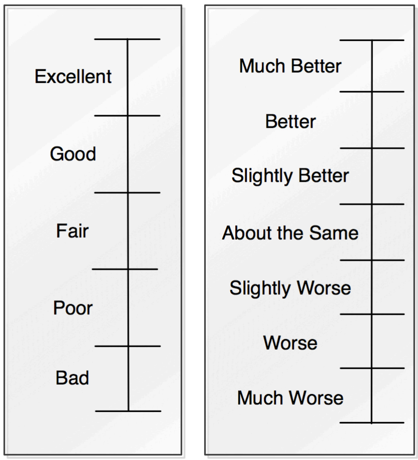

Figure 6 - Quantitative Metrics – Comparative Metrics [34]

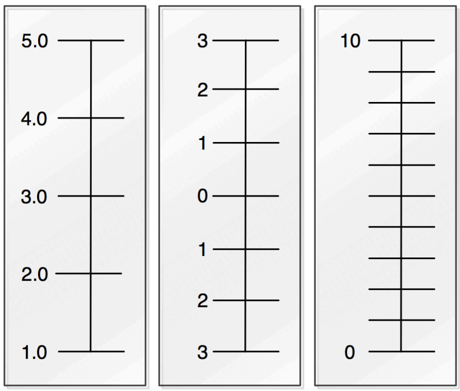

Figure 7 – Qualitative Metrics – The first scale is the recommendation of ITU for MOS [35]

Table 2 – Qualitative Approach (Quality oriented or impairment oriented)

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible
3	Fair	Slightly Annoying
2	Poor	Annoying
1	Bad	Very Annoying

1.6.3.2 Subjective Metrics

Subjective assessment methods are used to establish the performance of systems using measurements that more directly anticipate user perceptions. In order to evaluate those perceptions, a group of people watches a video or hears a sound and gives it a quality score. The results of the tests are treated statically and the output is often an average of Mean Opinion Score (MOS).

MOS is the most popular measure of QoE. It began life as a subjective measure. The basic definition of MOS can be found in [23]: "*The mean of opinion scores, i.e., of the values on a predefined scale that subjects assign to their opinion of the performance of the telephone transmission system used either for conversation or for listening to spoken material.*" ITU, firstly introduced MOS for voice telephone services, but it is currently used for evaluation of other services, especially video and as a more objective approximation of subjective MOS.

MOS scores are rated on a scale of 1 to 5, where 5 is the best possible score. But MOS score can be obtained by different evaluation methods, namely, subjective tests, objective or network planning models.

1.6.3.3 Objective Metrics

As we mentioned above QoE subjective measurements assess how users perceive audio or video streams, i.e., what is their opinion on the quality of particular audio/video sequences. Objective quality metrics are algorithms and formulas that measure, in a certain way, the quality of a stream. With a few exceptions, objective metrics propose different ways of comparing the received sample with the original one, typically by computing a sort of distance between both signals. Below the most well known objective metrics are analyzed (that someone could find in literature).

1.6.3.4 E-Model

The E-model [36], as mentioned above, is a transmission planning tool that provides a prediction of the expected voice quality, as perceived by a end user, for a complete end-to-end (i.e. mouth-to-ear) telephone connection under conversational conditions. The E-model takes into account a wide

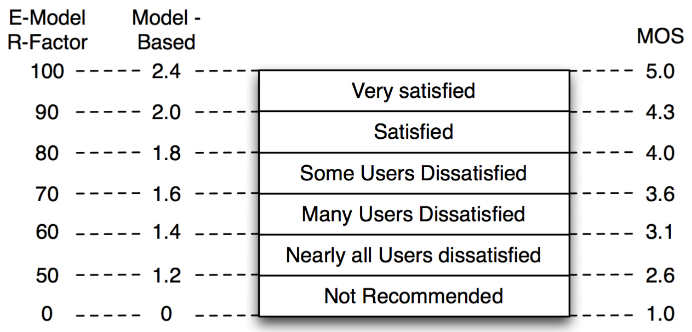

Figure 8 - Comparison of MOS, R-Factor and a Model-Based Approach

range of telephony-band impairments, in particular the impairment due to low bit-rate coding devices and one-way delay, as well as the "classical" telephony impairments of loss, noise and echo. By using the previous impairments and by using a mathematical algorithm, it can be applied to assess the voice quality of wired and wireless scenarios, based on circuit-switched and packet-switched technology. The E-model is based on modeling the results from a large number of subjective tests done in the past on a wide range of transmission parameters. ITU-T Rec. G.108 [37] and G.175 [38] provide detailed guidance for transmission planning using the E-model. The primary output of the E-model is the Transmission Rating Factor R, which can take values from 0-100. It can be easily transformed into other quality measures such as MOS, Percentage Good or Better (GoB) or Percentage Poor or Worse (PoW). On Figure 8 we can see a comparison of MOS, R-Factor and a Model Based approach [26]. However, caution should be exercised when comparing these transformed measures with values from other sources, which may not have been obtained under comparable conditions.

1.6.3.5 Bit Error Rate (BER)

In digital transmission, the number of bit errors is the number of received bits of a data stream over a communication channel that has been altered due to noise, interference, distortion or bit synchronization errors. The measure of that performance is usually the number of bit errors divided by the total number of transferred bits, which quantifies the reliability of the entire radio system from “bits in” to “bits out”, including the electronics, antennas and signal path in between. BER [39] is a unit less performance measure, often expressed as a percentage.

1.6.3.6 Moving Picture Quality Metric (MPQM)

Moving Picture Quality Metric [40] is a video quality rating score for moving picture that is based on a model of the human vision system. It uses a mix of content dependent factors along with a combination of network impairments such as delay and packet loss rate. MPQM incorporates two human vision characteristics, the contrast sensitivity and masking. It first decomposes an original sequence and a distorted version of it into perceptual channels. A channel-based distortion measure is then computed, accounting for contrast sensitivity and masking. Then MPQM uses network impairments such as packet delay, error rate, and packet loss as factors to the quality the score by pooling the data over all the channels to compute the quality rating which ranges from 1 to 5 (bad to excellent).

1.6.3.7 Color Moving Picture Quality Metric (CMPQM)

Color Moving Picture Quality Metric (CMPQM) [41] is an extension of the MPQM metric to consider the effect of color on the quality. The first step is to transform the data into a calibrated space to be device-independent and to be able to apply further color space transformation. The second step converts the linear RGB values into coordinate values in the chosen opponent-colors space. The three coordinates of this color-opponent space correspond to luminance (B/W), red-green (R/G), and blue-yellow (B/Y) channels. Then, each color component of the original and error images is analyzed by a filter bank functions tuned in spatial frequency and orientation. The bank uses the same filters as the MPQM, except in numbers. The remaining of the computation is as in the MPQM. Contrast thresholds are computed and the data is pooled over the channel to yield a distortion measure. Just like MPQM contrast thresholds are computed and the data is pooled over the channel to compute the quality rating which ranges from 1 to 5 (bad to excellent).

1.6.3.8 Peak signal-to-noise ratio (PSNR)

The Peak signal-to-noise Ration (PSNR) [40] is a traditional objective metric used to measure the level of video quality and is based on the maximum possible power of a signal and the power of corrupting noise that affects the fidelity of its representation. Because many signals have a very wide dynamic range, (ration between the largest and smallest possible values of changeable quantity), PSNR is usually expressed in terms of the logarithmic decibel (dB) scale. Typical values for the PSNR in lossy videos are between 30 dB and 50 dB, where the higher value is better. Acceptable values for wireless transmission quality loss are considered to be about 20 dB to 25 dB. PSNR is most commonly used as a measure of quality of reconstruction of lossy compression codecs. The signal in this case is the original data and the noise is the error introduces by compression. The PSNR of a video is most easily defined through the Mean Squared Error (MSE) metric.

1.6.3.9 Mean Squared Error (MSE)

Mean Squared Error is one of many ways to the difference between values implied by an estimator and the true values of the quantity being estimated. MSE measures the average of the squares of the "errors". By the error is defined the difference between the value that was implied by the estimator and the quantity that was finally estimated and like the variance, MSE has the same units of measurements as the square of the quantity being estimated. A minimal MSE often, but not always, indicates minimal variance, and thus a good estimator. A value of zero (0) is the ideal condition but in most situations is never possible. An MSE of 0 means the estimator predicts observations with perfect precision.

1.6.3.10 Structural Similarity (SSIM)

Structural Similarity (SSIM) [40] index is a method for measuring the similarity between two images. SSIM is a full reference metric and it was designed to improve on traditional methods like PSNR and MSE, which have proved to be inconsistent with human eye perception. PSNR and MSE are error based. SSIM differs from them by using the structural distortion measurement instead of the error. The idea behind this is that the human vision system is highly specialized in extracting structural information from the viewing field and it is not specialized in extracting the errors. Thus, a measurement on structural distortion should give a better correlation to the subjective impression. The accepted value lies between [0, 1] and the value 1 is only reachable in the case of two identical sets of data.

1.6.3.11 Video Quality Metrics (VQM)

Video Quality Metrics [42] are developed by the Institute for Telecommunication Science (ITS) to provide an objective measurement for perceived video quality. It measures the perceptual effects of video impairments that arise from the Human Visual System (HVS) including blurring, jerky/unnatural motion, global noise, block distortion and color distortion, and combines them into a single metric. The testing results show VQM has a high correlation with subjective video quality assessment and has been adopted by American National Standard Institute (ANSI) as an objective video quality standard. VQM gets values between 0 (best) and 5 (worst).

1.6.3.12 Perceptual Evaluation of Speech Quality (PESQ) – Advanced Model for PESQ (AdmPESQ)

The Perceptual Evaluation of Speech Quality (PESQ) [43] Algorithm is designed to predict subjective opinion scores of a degraded audio sample as experienced by a user of a telephony system. PESQ is designed to analyze specific parameters of audio, including time warping, variable delays, transcoding, and noise. It is primarily intended for applications in codec evaluation and network testing and returns a score from 5 to 1, with higher scores indicating better quality. PESQ is full-reference algorithm and analyzes the speech signal sample-by-sample after a temporal alignment of corresponding excerpts of reference and test signal. PESQ can be applied to provide an end-to-end (E2E) quality assessment for a network, or characterize individual network components. On [44], Advanced Model for PESQ (AdmPESQ) is proposed that combines the delay impact with the results of PESQ by giving more accurate results.

1.6.3.13 Block Error Rate (BLER)

The Block Error Rate (BLER) is a number, which represents the ratio of incorrect blocks to the total number of blocks occurring within a circuit system, indicative of the fidelity of data extracted from a Compact Disc (CD). The BLER measurement is often used as a quality control measure with regards to how well audio is retained on a compact disc over time.

1.6.3.14 Media Delivery Index (MDI) – MDI based on Frame Classification (FC-MDI)

The Media Delivery Index (MDI) [45] measurement indicates problems generated on the network. It gives an indication of real expected quality video (user's QoE) and is independent on the video encoding. MDI is an objective metric that contains two numbers separated by colon: the Delay Factor (DF) and the Media Loss Rate (MLR). DF is time value indicating how many milliseconds the buffer must be able to contain to eliminate jitter, while MLR is computed difference between number of media packets received during an interval and number of media packets expected during an interval, everything scaling in the value of one second. Because the MLR is a rate, some important information is lost, such as whether the IP packets lost are consecutive or not. It does not consider the quality degradation that suffered some propagated loss from previous temporally related frames, so [46] proposes FC-MDI that takes frame classification into account to improve the performance of the MDI measurement.

1.6.3.15 Normalization Video Fidelity Metric (NVFM)

Normalization Video Fidelity Metric (NVFM) [47] is implemented based on a visibility prediction employing a normalization model. A brief block diagram is presented in Figure 9. The perceptual decomposition is realized in the pixel domain by the steerable pyramid and the bank of IIR temporal filters. The output coefficients of the linear transform are squared to compute a local energy measure that is then normalized by a divisive mechanism. At this stage, inter-channel masking is taken into account. The normalization stage explains the response saturation of VI neurons and cross-orientation

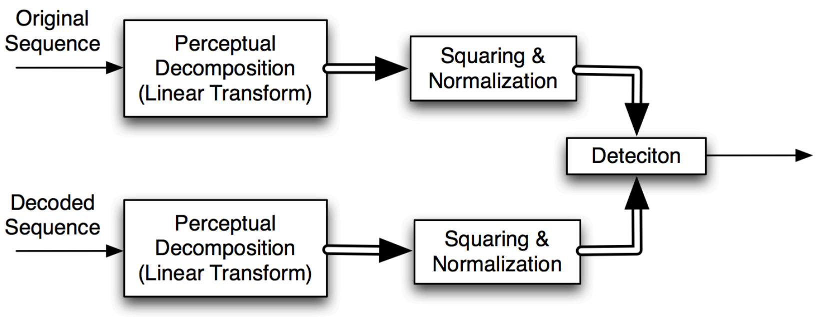

Figure 9 - Block diagram of the Normalization Fidelity Metric [47]

inhibition. Finally, a detection mechanism is computed as the squared vector sum of the difference of the normalized responses.

1.6.4 Hybrid Approach

Finally, a hybrid approach between subjective and objective evaluation has been proposed in [48]. It is a technique that allows approximation of the value obtained from a subjective test but automatically. In more detail, Pseudo-Subjective Quality Assessment (PSQA) metric starts by selecting the factors that may have an impact on the quality, such as: codec, bandwidth, loss, delay, and jitter. Then these factors are used to generate several distorted video samples. These samples are subjectively evaluated by a panel of observers. The results of the observations are then used to train a Random Neural Network (RNN) in order to capture the relation between the factors that cause the distortion (objective approach) and the perceived quality by real-human (subjective approach).

CHAPTER 2

Wireless Network Technologies

2.1 IEEE 802.11

2.1.1 Introduction

In 1997, IEEE released IEEE Std. 802.11-1997 [49], the first IEEE 802.11 WLAN standard, and it was clarified later on 1999. IEEE 802.11 is the most prominent specification for Wireless LAN (WLANs). IEEE 802.11 defines a Medium Access Control (MAC) and several Physical layer (PHY) specifications for wireless connectivity for fixed and moving STAs within a local area. The standard is similar in many respects to the IEEE 802.3 Ethernet standard and is mapped to the OSI reference model as shown in Figure 10.

In particular, the IEEE 802.11 Wireless LAN, also known as Wireless Fidelity (Wi-Fi), provides wireless connectivity for two or more terminals, nodes or STAs (i.e. laptops, tablet PCs, servers, printers, etc.) that may be fixed or portable within a local area. It allows the users to communicate with each other without requiring a physical connection to the network.

2.1.2 Architecture

Basic Service Set (BSS) is the smallest building block that can be set up. It consists a number of stations executing the same MAC protocol and competing for the access to the same shared wireless medium. Different BSS

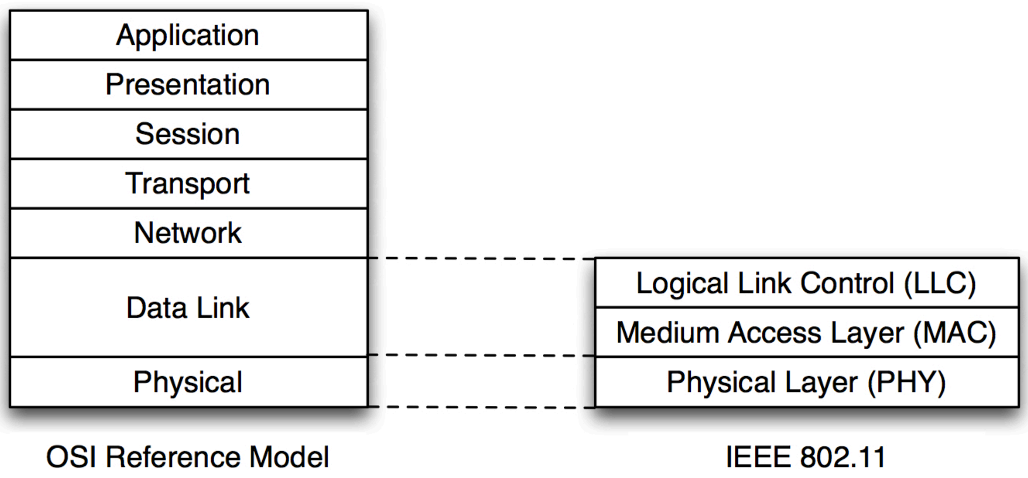

Figure 10 - IEEE 802.11 standards mapped to the OSI reference model

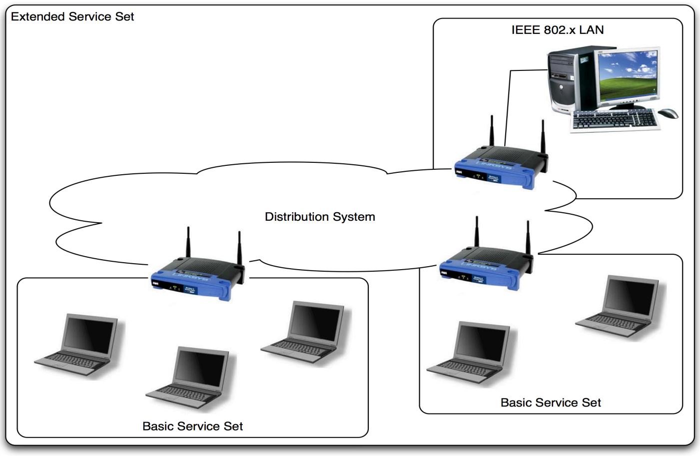

Figure 11 - Extended Service Set, Basic Service Set and Distribution System

can connect to each other through a backbone distribution system (DS) through an Access Point (AP). Inside the BSS, a client doesn't communicate directly with another client. Every frame is being sent first to the AP, and then from the AP to the destination station. Similarly a frame that has a destination another BSS, is sent to the AP first, then through the DS, the frame is being sent to the destination BSS and the AP delivers the frame to the destination.

When all the stations in the BSS are mobile stations, and there is no connection to another BSS, then BSS is called Independent BSS (IBSS). IBSS is an ad-hoc network and all the stations communicate directly, without an AP being involved, but the stations has to be within range.

Because IEEE 802.11 uses as a medium the air, it is possible for to different BSS to overlap and for a station to participate more that one BSS, by dynamically associating to each BSS.

Finally, two or more BSS interconnected by a DS create an Extended Service Set (ESS). Typically, the distribution system is a wired backbone LAN, in which the AP works as a station when it receives the frames from other BSS and sends them to the destination address [50]. In Figure 11 we can see an ESS, that contains two BSS and another LAN network and all are connected through a DS.

2.1.3 Services

IEEE 802.11 defines 13 services [51] that need to be provided by the WLAN to provide functionality equivalent to that, which is inherent to wired LAN. There are two different ways to categorize these services:

The service provider can either be the station or the DS. Station services are implemented in every 802.11 station or AP. Distribution services are provided between different BSS and may be implemented in an AP or another special-purpose device attached to the DS.

Six of the services are used to support Medium access control service data unit (MSDU) delivery between STAs. Three of the services are used to control IEEE 802.11 LAN access and confidentiality. Two of the services are used to provide spectrum management. One of the services provides support for LAN applications with QoS requirements. Another of the services provides support for higher layer timer synchronization. Services are consternated below (Table 3) [52].

2.1.4 Physical Layer

The IEEE 802.11 Physical layer is the interface between the wireless medium and the MAC layer and defines the radio wave modulation and signaling characteristics for data transmission. IEEE Physical layer is divided in Physical Layer Convergence Protocol (PLCP) and the Physical Medium Dependent (PDM). PLCP defines a method of mapping 802.11 MAC Protocol Data Units (MPDUs) into a framing format suitable for sending and receiving user data and management information between two or more stations using the associated PMD sublayer. PMD defines the characteristics and the method of transmitting and receiving user data, through a wireless medium between two or more stations.

Five different Physical layer specifications were defined, entitled Infrared (IR), Frequency Hopping Spread Spectrum (FHSS), Direct Sequence Spread Spectrum (DSSS), Complementary Code Keying (CCK), and Orthogonal Frequency-Division Multiplexing (OFDM) [51].

Table 3 – IEEE 802.11 Services

Service	Provider	Used to support
Authentication	Station	IEEE 802.11 LAN access and confidentiality
Deauthentication		
Data confidentiality		
MSDU delivery		MSDU delivery
DFS		Spectrum management
TPC		
Higher Layer Timer Synchronization (QoS facility only)		Higher layer timer synchronization
QoS traffic scheduling (QoS facility only)		LAN applications with QoS requirements
Association	Distribution System	MSDU delivery
Disassociation		
Distribution		
Integration		
Reassociation		
QoS traffic scheduling (QoS facility only)		LAN applications with QoS requirements

2.1.4.1 Infrared (IR)

Infrared light is part of electromagnetic spectrum that is shorter than radio waves but longer than visible light. It operates between 850 and 950 nm bands and has transmission rate of 1 or 2 Mbit/s. The maximum communication distance is up to 20 m. The IR relies on both reflected IR energy as well as line-of-sight IR energy for communications.

IR is used in indoor environments because doesn't pass through walls, or even in an environment with a few or without reflected surfaces, and where there is not line-of-sight, may suffer reduces communication range.

2.1.4.2 Frequency Hopping Spread Spectrum (FHSS)

FHSS was the first step to the evolution of DSSS and more complex data transmission techniques. It works at 2.4 GHz and operates at 1 and 2 Mbit/s.

At FHSS a station transmits by rapidly switching a carrier among may frequency channels, using a pseudorandom sequence known to both transmitter and receiver. It separates the whole 2.4 GHz band into channels that are spaced of 1MHz and the transmitter changes channel at least 2.5 times per second (every 400 msec or less). The number of channels available ranges from 23 in Japan to 70 in the United States. The hopping patters are described by 3 sets containing 26 hopping sequences each and provide minimum mutual interference.

2.1.4.3 Direct Sequence Spread Spectrum (DSSS)

DSSS is one of the most successful data transmission techniques for today. It operates in 2.4 GHz at data rates of 1 and 2 Mbit/s. The number of channels available depends on the bandwidth allocation by the various national regulatory agencies. This ranges from 13 in most European countries to just one available channel in Japan. DSSS multiples the data being transmitted by a “noise” signal. This noise signal is a pseudorandom sequence of 1 and -1 values, at a frequency much higher than that of the original signal. The STA uses the same center frequency but multiplexing with different spreading codes to reduce the interference between signals and the background noise spreads the signal. The receiver then decodes the original signal using the same code used by the transmitter.

DSSS systems spread transmissions across a relatively wide band by artificially increasing the used bandwidth. A DSSS transmitter converts an incoming data stream into a symbol stream where each symbol represents a group of 1, 2, or more bits. Using a phase-varying modulation technique such as Quadrature Phase Shift Keying (QPSK), the DSSS transmitter modulates or multiplies each symbol with a pseudorandom sequence, which is called a “chip” sequence. The multiplication operation in a DSSS transmitter artificially increases the used bandwidth based on the length of the chip sequence.

2.1.4.4 Complementary Code Keying (CCK)

CCK is a modulation scheme that employs IEEE 802.11b specification. It was adopted to achieve data rates higher than 2 Mbits at the expense of shorter distance. The CCK modulation is based on the use of the polyphase complementary codes. The polyphase complementary codes are complex codes and they are not binary.

2.1.4.5 Orthogonal frequency-division multiplexing (OFDM)

The OFDM system (used in IEEE 802.11a) [53] provides a WLAN with data payload communication capabilities of 6, 9, 12, 18, 24, 36, 48 and 54 Mbits. The support of transmitting and receiving at data rates of 6, 12, and 24 Mbits is mandatory. A large number of closely spaced orthogonal sub-carrier signals are used to carry data. The orthogonality prevents crosstalk between subcarriers. The data is divided into several parallel data streams or channels, one for each subcarrier. Each smaller data stream is then mapped to individual data sub-carrier and modulated using some sorts of Phase Shift Keying (PSK) (i.e. Binary PSK (BPSK)) or Quadrature Amplitude Modulation (QAM) (i.e. QPSK, 16-QAM, 64-QAM). A convolutional code at a rate of ½, 2/3, or ¾ provides forward error correction. The combination of modulation technique and coding rate determines the data rate.

2.1.5 Datalink Layer

IEEE 802.11 is required to appear to higher layers [Logical Link Control (LLC)] as a wired IEEE 802 LAN. This requires that the IEEE 802.11 network handle STA mobility within the Medium Access Control (MAC) sublayer. To meet reliability assumptions (that LLC makes about lower layers), it is necessary for IEEE 802.11 to incorporate functionality that is untraditional for MAC sublayers.

2.1.6 IEEE 802.11 Media Access Control

The MAC is a sublayer of the Date Link Layer specified in the seven-layer OSI model (layer 2). The MAC layer emulates a full-duplex logical communication channel in a multipoint network. The IEEE 802.11 MAC also supports shared access to the wireless medium through a technique called Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), which is similar to the original (shared medium) Ethernet’s Carrier Sense Multiple Access with Collision Detection (CSMA/CD).

Noise, interference and other propagation effects result in the loss of a significant number of frames. By dealing with this problem on a higher layer, we will have great delays. So IEEE 802.11 includes a frame exchange protocol. When a station receives a data frame, it returns an acknowledgement (ACK) frame to the source station. This exchange is treated as an atomic unit, not to be interrupted by a transmission from any other station. If the source doesn't receive an ACK within a short period of time, either because its data frame was damaged or because the returning ACK was damaged, the source retransmits the frame.

To further enhance reliability, a four-frame exchange may be used. In this scheme, a device first sends a Request to Send (RTS) frame to the destination, to introduce that it is ready to send. The destination device answers with a Clear to Send frame (CTS). After receiving the CTS, the source transmits the data frame and waits for the ACK. With the RTS and the CTS, all stations that are within range, are alerted that an exchange is under way, so the transmission is not allowed, in order to avoid collisions. RTS/CTS may be disabled.

Two different type of proposals for a MAC algorithm were made from 802.11 working group: Distributed Coordination Function (DCF) which makes sense for an ad hoc network of peer workstations and Point Coordination Function (PCF) that is employed for infrastructure network configurations. In IEEE 802.11e another MAC algorithm was proposed. The new algorithm is entitled Hybrid Coordination Function (HCF) [51], combines DCF and PCF functions, enhancing QoS management and providing QoS guarantees to QoS aware applications. The HCF uses both a contention-based channel access method, called the Enhanced Distributed Channel Access (EDCA) mechanism, and a contention-free channel access method referred to as the HCF controlled channel access (HCCA) mechanism.

2.1.6.1 Distributed Coordination Function (DCF)

The DCF makes use of a simple CSMA (carrier sense multiple access) algorithm, which functions as follows. If a station has a MAC frame to transmit, it senses the channel. If the medium is IDLE, then the station is permitted to transmit. If the channel is BUSY then the station defers the transmission.

DCF does not include a collision detection function (CSMA/CD) so a set of delays is used to ensure the smooth and fair functioning of the algorithm. On the scenario that the channel is busy, Network Allocation Vector (NAV) is used to determine the time that a station has to wait until the medium is free again. NAV is the duration of the frame that is being transmitted and every frame carries a duration field that is used to update the NAV on every station. [52] When NAV time reaches zero then the station is able to sense again and try to transmit.

Another single delay is the interframe space (IFS). There are three different IFS values. Collision avoidance mechanism uses these values to reduce the probability of collision in the network. The first IFS value is DCF IFS (DIFS), which is the time a station has to sense that the medium is IDLE. To avoid the scenario that all the stations start to send after the DIFS time, there is another delay (backoff time) that is calculated by multiplying the slot time and a random value that is given to the station every time it ends a transmission (successful or not). This value is being chosen between Contention Window values (CWmin and CWmax). Between two frames that have to be transmitted immediately (e.g. RTS and CTS), there is short IFS (SIFS) that is the shortest IFS.

2.1.6.2 Point Coordination Function (PCF)

As mentioned before PCF is used in infrastructure network configurations. The PCF [52] uses a virtual Carrier Sense (CS) mechanism aided by an access priority mechanism. PCF shall distribute information within Beacon management frames to gain control of the medium by setting the NAV in STAs. In addition, all frame transmissions under the PCF may use an interframe Space that is smaller than the DIFS (PIFS) for frames transmitted via the DCF. The use of smaller IFS implies that point-coordinated traffic will have priority access to the medium over STAs in overlapping BSSs operating under the DCF access method.

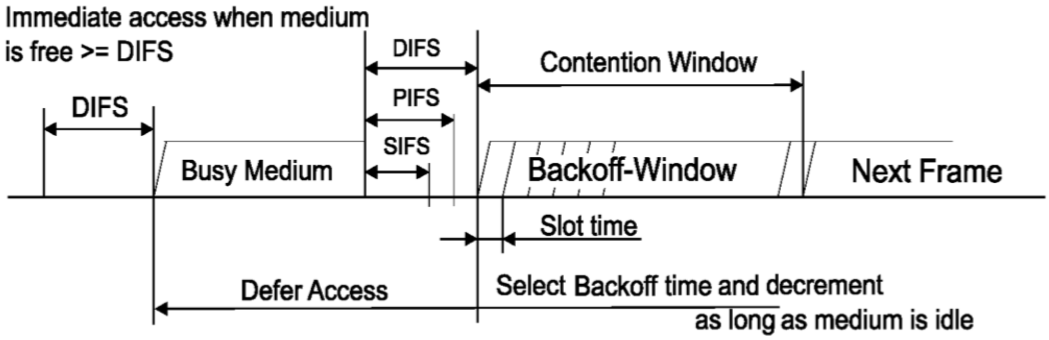

Figure 12 - Delay times and DCF mechanism [51]

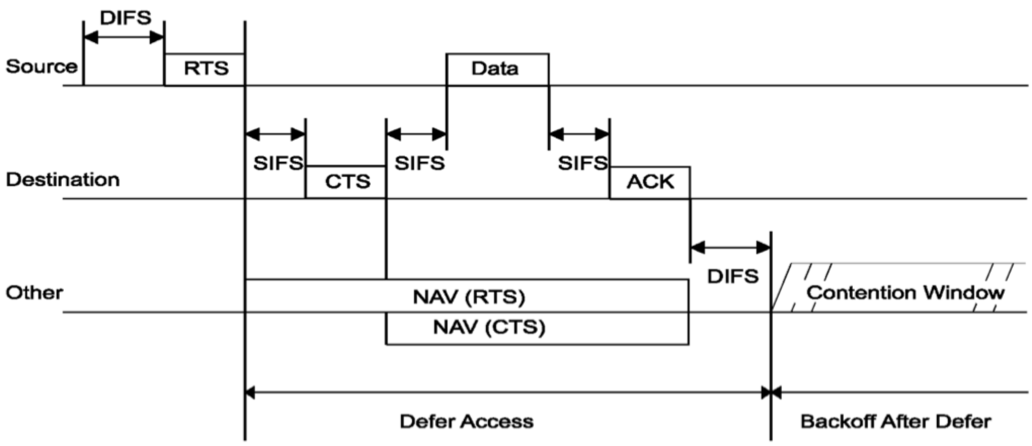

Figure 13 - RTS/CTS/data/ACK and NAV [51]

2.1.7 IEEE 802.11 Standards Family

The original version of the standard IEEE 802.11 was released in 1997. IEEE 802.11 supported a maximum network bandwidth of 2 Mbit/s - too slow for most applications. It specified three alternative physical layer technologies: Infrared (IR remains a part of the standard but has no actual implementations), FHSS and DSSS operating at 1 Mbit/s or 2 Mbit/s. The latter two radio technologies used microwave transmission over the ISM frequency band at 2.4 GHz.

The most major updates in IEEE 802.11 were 802.11 a, b, g and n. Firstly, was released IEEE 802.11b in 1999, it supported a maximum network bandwidth of 11 Mbit/s and was using FSSS as a modulation. It was working in the same frequency as the standard IEEE 802.11. Simultaneously, was released IEEE 802.11a. The big differences were the operating frequency, which was 5 GHz, the maximum bandwidth (up to 54 Mbit/s) and the modulation used (OFDM). Also 802.11a was not compatible with the previous version of 802.11. Later in 2003, IEEE released 802.11b, which supported the same speed as 802.11a, but in was working on 2.4 GHz and it had a backward compatibility with 802.11b. Finally, in 2009 IEEE 802.11n was released, and it supported theoretical bandwidth up to 150 Mbit/s, 4 MIMO antennas and was using OFDM and worked in both 2.4 and 5 GHz frequencies. In 2011, IEEE 802.11ac was announced, but is still under development, will work in 5 GHz and will enable multi-station throughput of almost 1 Gbit/s and up to 8 antennas.

On 2007 and 2012, two main revisions of IEEE 802.11 where published, which contained all the standards and amendments within the working group. The most well known standards are summarized in Table 4 [54].

Except from these in IEEE 802.11 working group we can find the following:

- IEEE 802.11c: Bridge operation procedures; included in the IEEE 802.1D standard (2001).
- IEEE 802.11d: International (country-to-country) roaming extensions (2001).
- IEEE 802.11e: Enhancements: QoS, including packet bursting (2005).
- IEEE 802.11f: Inter-Access Point Protocol (2003).
- IEEE 802.11h: Spectrum Managed 802.11a (5Ghz) for European compatibility (2004).
- IEEE 802.11i: Enhanced security (2004).
- IEEE 802.11j: Extensions for Japan (2004).
- IEEE 802.11k: Radio resource measurement enhancement (2008).
- IEEE 802.11p: WAVE –Wireless Access for the Vehicular Environments (such as ambulances and passenger cars) (July 2008).
- IEEE 802.11r: Fast BSS transition (FT) (2008).
- IEEE 802.11s: Mesh Networking, Extended Service Set (July 2011).
- IEEE 802.11T: Wireless Performance Prediction (WPP) – test methods and metrics.
- IEEE 802.11u: Improvements related to HotSpots and 3rd party authorization of clients, e.g. cellular network offload (February 2011).
- IEEE 802.11v: Wireless network management (February 2011).
- IEEE 802.11w: Protected Management Frames (September 2009).

- IEEE 802.11y: 3650 – 3700 MHz Operation in the U.S. (2008).
- IEEE 802.11z: Extensions to Direct Link Setup (DLS) (September 2010).
- IEEE 802.11aa: Robust streaming of Audio Video Transport Streams (June 2012).
- IEEE 802.11ae: Prioritization of Management Frames (March 2012).

- **Table 4 – IEEE 802.11 Network Standards**

IEEE 802.11 Network Standards

802.11 Protocol	Release Date	Freq. (GHz)	Band. (MHz)	Data Rate (Mbits)	Allowable MIMO Streams	Modulation	Indoor/Outdoor Range (m)
-	Jun 1997	2.4	20	1, 2	1	DSSS, FHSS	20/100
a	Sep 1999	5 (3.7)	20	6, 9, 12, 18, 24, 36, 48, 54	1	OFDM	35/120
b	Sep 1999	2.4	20	1, 2, 5.5, 11	1	DSSS	35/140
g	Jun 2003	2.4	20	6, 9, 12, 18, 24, 36, 48, 54	1	OFDM, DSSS	38/140
n	Oct 2009	2.4/5	20	7.2, 14.4, 21.7, 28.9, 43.3, 57.8, 65, 72.2	4		70/250
			40	15, 30, 45, 60, 90, 120, 135, 150		OFDM	
ac (draft)	Nov 2011	5	20	87.6	8		
			40	200			
			80	433.3			
			160	866.7			

2.2 IEEE 802.16 (WiMAX)

2.2.1 Introduction

IEEE developed the 802.16 in its first version to address Line of Sight (LOS) access at spectrum ranges from 10 GHz to 66 GHz. Although the family of standards is officially called WirelessMAN it has been commercialized under the name Worldwide Interoperability for Microwave Access (WiMAX) by the WiMAX forum industry alliance. WiMAX provides wireless transmission of data using a variety of transmission modes, from point-to multipoint links to portable and fully mobile Internet access. The technology provided up to 10 Mbit/s broadband speed without the need for cables. WiMAX forum describes WiMAX as “a standard-based technology enabling the delivery of last mile wireless broadband access as an alternative to cable DSL” [55].

WiMAX is cheaper that DSL because it does not require wires. As all the wireless technologies, the requirements for WiMAX are basically a transmitter and a receiver. A transmitter is a WiMAX tower, much like a GSM tower, which can provide coverage to an area within a radius of around of 50 km. On the other side, in order to receive the WiMAX waves, you need a receiver for WiMAX for connecting your computer or device.

2.2.2 Architecture

WiMAX has four fundamental architecture components [56]. In Figure 14 we can see two different types of networks and these four components:

- Base Station (BS): BS is a generalized equipment set providing connectivity, management, and control of the Subscriber Station (SS) and governs access to the operator networks. A BS consists of the infrastructure elements necessary to enable wireless communications, i.e., antennas, transceivers and other electromagnetic wave transmitting equipment. BSs are typically fixed nodes, but they may also be used as part of mobile solutions.
- Subscriber Station (SS): The SS is a fixed wireless node. An SS typically communicates only with BSs expect from multihop relay

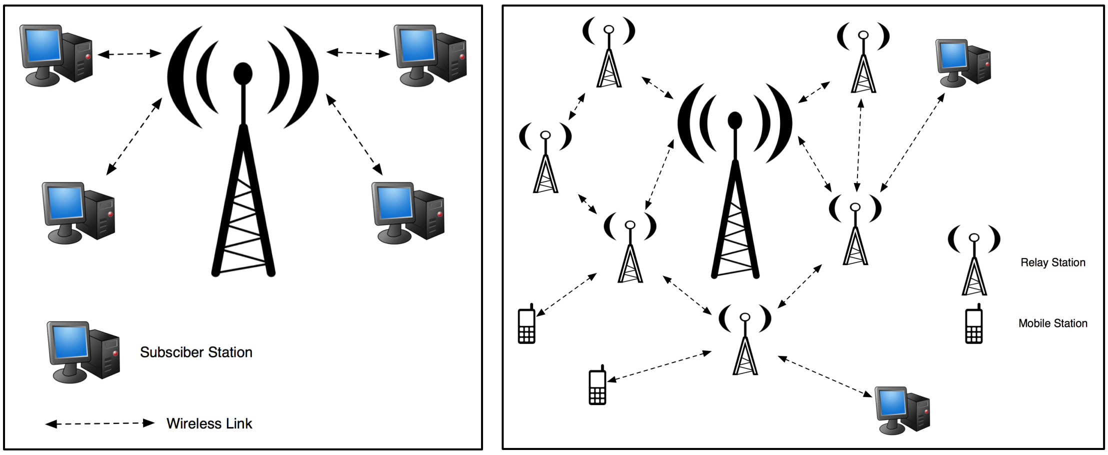

Figure 14 - WiMAX network architectures: (a) Point-to-Multipoint mode, (b) mesh mode

network operations. SS are available in both outdoor and indoor models.

- Mobile Subscriber (MS): Defined in IEEE 802.16e-2005 [57], MSs are wireless nodes that work at vehicular speeds and support enhanced power management modes of operation. MS devices are typically small and self-powered, e.g., laptops, cellular phones, and other portable electronic devices.
- Relay Station (RS): Defined in IEEE 802.16j-2009 [58]. RS is a device configured to forward traffic to other RSs, SSs or MSs in multi-hop network configurations.

2.2.2.1 IEEE 802.16 protocol architecture

The IEEE 802.16 protocol architecture is structured in two main layers (Figure 15): the Medium Access Control Layer (MAC) and the Physical Layer (PHY). MAC layer consists of three sub-layers. The first sub-layer is the Service Specific Convergence Sub-Layer (CS), which maps higher-level data services to MAC layer service flow and connections. The second sub-layer is Common Part Sub-Layer (CPS), which is the core of the standard and is tightly intergraded with the security sub-layer. This layer defines the rules and mechanisms for system access, bandwidth allocation and connection management. The MAC protocol data units are constructed in this sub-layer. The last sub-layer of MAC layer is the Security Sub-Layer which lies between the MAC CPS and the PHY layer, addressing the authentication key,

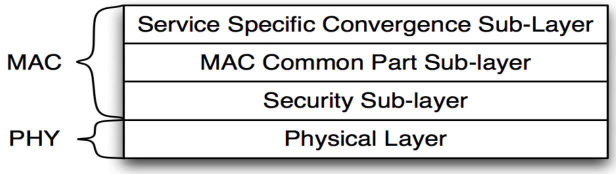

Figure 15- The IEEE 802.16 Protocol Structure

establishment and exchange, encryption and decryption of data exchange between MAC and PHY layers.

The PHY layer provides a two-way mapping between MAC protocols data units and the PHY layer frames received and transmitted through coding and modulation of radio frequency signals.

2.2.3 Services

Scheduling services represent the data handling mechanisms supported by the MAC scheduler for data transport on a connection. Each connection is associated with a single scheduling service. A scheduling service is determined by a set of QoS parameters that quantify aspects of its behavior.

WiMAX is configured to support self-installation and auto-configuration. When customers subscribe to the service, they tell the service provider the service flow information including the number of UL/DL connections with the data rates and QoS parameters, along with the types of applications the customers intends to run. The service provider preprovisions the services by entering the service flow information into the service flow database. When the SS enters the BS by completing the network entry and authentication procedure, the BS downloads the service flow information from the service database [59].

To support a wide variety of applications, WiMAX defines five scheduling services that should be supported by the base station MAC scheduler for data transport over a connection [60]. On Table 5 there is a summarization of services and the applications that support them.

- Unsolicited Grant Service (UGS): This is designed to support fixed-size data packets at a Constant Bit Rate (CBR). The mandatory service flow parameters that define this service are maximum sustained traffic rate, maximum latency, tolerated jitter, and request/transmission policy.

- Real-time Polling Service (rtPS): This service is designed to support real-time service flows that generate variable-size data packets on a periodic basis. The mandatory service flow parameters that define this service are minimum reserved traffic rate, maximum sustained traffic rate, maximum latency, and request/transmission policy.
- Non-real-time Polling Service (nrtPS): This service is designed to support delay-tolerant data streams that require variable-size data grants at a minimum guaranteed rate. The mandatory service flow parameters to define this service are minimum reserved traffic rate, maximum sustained traffic rate, traffic priority, and request/transmission policy.
- Best-Effort (BE) service: This service is designed to support data streams that do not require a minimum service-level guarantee. The mandatory service flow parameters to define this service are maximum sustained traffic rate, traffic priority, and request/transmission policy.
- Extended Real-Time Variable Rate (ERT-VR) service: This service is designed to support real-time applications that have variable data rates but require guaranteed data rate and delay. This service is defined only in [57], not in previous versions of the standard. This is also referred to as Extended real-time Polling Service (ErtPS).

Table 5 – Service Flows Supported in WiMAX

Service Flow Designation	Defining QoS Parameters	Application Examples
Unsolicited Grant Service (UGS)	Maximum sustained rate Maximum latency tolerance Jitter tolerance	Voice over IP (VoIP) without silence suppression
Real-time Polling Service (rtPS)	Minimum reverse rate Maximum sustained rate Maximum latency tolerance Traffic Priority	Streaming audio and video, MPEG encoded
Non-real-time Polling Service (nrtPS)	Minimum reserved rate Maximum sustained rate Traffic priority	File Transfer Protocol (FTP)
Best-Effort Service (BE)	Maximum sustained rate Traffic priority	Web browsing, data transfer
Extended real-time Polling service (ErtPS)	Minimum reverse rate Maximum sustained rate Maximum latency tolerance Maximum latency tolerance Traffic Priority	VoIP with silence suppression

2.2.4 Physical Layer

WiMAX's physical layer is based on orthogonal frequency division multiplexing. We have already introduced OFDM in 802.11 as a technology. As resume, we can say that OFDM enjoys several advantages over other solution for high-speed transmission because:

- It reduces the computational complexity
- It degraded gracefully the performance under excess delay
- It takes advantage of frequency diversity
- It is used as a multi-access scheme
- It is robust against the narrow band interference
- It is suitable for coherent demodulation

Fixed and mobile versions of WiMAX have slightly different implementations of OFDM. Fixed WiMAX, uses a 256 Fast Fourier transform (FFT-based) OFDM in contrast with mobile WiMAX, which uses an Orthogonal Frequency Division Multiple Access (OFDMA-based) physical layer. In the case of mobile WiMAX, the FFT can vary from 128 bits to 2048 bits. For a more extensive description of the two technologies, someone can refer to [60].

WiMAX PHY layer is also responsible for slot time allocation and framing over the air. The minimum time-frequency resource that can be allocated by a WiMAX system to a given link is called a slot. Each slot consists of one subchannel over one, two or three OFDM symbols, depending on the particular subchannelization scheme used. A contiguous series of slots assigned to a given user is called that user's data region. Scheduling algorithms could allocate data regions to different users, based on demand, QoS requirements and channel conditions.

Also WiMAX support both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) operation. TDD is a technique in which the system transmits and receives within the same frequency channel, assigning time slices for transmit and receive modes. FDD requires two separate frequencies generally separated by 50 to 100 MHz within the operation band. TDD provides an advantage where a regulator allocates the spectrum in an adjacent block. With TDD, band separation is not needed. Thus the entire spectrum allocation is used efficiently both upstream and downstream and where traffic patterns are variable or asymmetrical.

Finally in Table 6 there is a summary of the modulation and coding supported in WiMAX.

2.2.5 Medium Access Control Layer

The primary task of WiMAX MAC layer is to provide an interface between the higher transport layer and the physical layer. Packets from upper layer are called MAC Service Data Units (MSDUs) and MAC layer organizes them in MPDUs for transmission over the air [61].

2.2.5.1 Channel Access and Bandwidth Allocation

MAC layer is fully responsible for allocating bandwidth to all users, in both uplink and downlink. Responsible for this is the BS. The only time the MS has some control over bandwidth allocation is when it has multiple connections with the BS. Bandwidth allocation is based on requests of the MSs. Bandwidth requests may be transmitted using a stand-alone bandwidth request MPDU, by piggybacking the request with the data messages or finally by sending a bandwidth request using the ranging channel. The BS executes resources allocation based on the requests and QoS parameters of the connection. This process is called polling. Polling may be done either individually (unicast) or in groups (multicast). Multicast polling is done when there is insufficient bandwidth to poll each MS individually.

Table 6 – Modulation and Coding Supported in WiMAX [60]

	Downlink	Uplink
Modulation	BPSK, QPSK, 16 QAM, BPSK, optional for OFDMA-PHY	BSPK, QPSK, 16 QAM, 64 QAM
Coding	Mandatory: convolutional codes at rate ½, 2/3, ¾, 5/6	Mandatory: convolutional codes at rate ½, 2/3, ¾, 5/6
	Optional: convolutional turbo codes at rate ½, 2/3, ¾, 5/6; repetition codes at rate ½, 1/3, 1/6, LDPC, RS-Codes for OFDM-PHY	Optional: convolutional turbo codes at rates ½, 2/3, ¾, 5/6; repetition codes at rate ½, 1/3, 1/6, LDPC

2.2.5.2 Quality of Service

WiMAX provides an environment for connection-oriented services. Through this architecture, strong QoS control is achieved, where all downlink and uplink connections are controlled by the service BS. A connection is established between a BS and a MS and has a unique Connection Identifier (CID), which serves as a temporary address for data transmissions over the particular link. Except from this, a service flow is also defined, which also has a unique identifier. QoS parameters could include traffic priority, maximum sustained traffic rate, maximum burst, minimum tolerable rate and so on. Earlier in this chapter, we have introduced the five different scheduling services and the applications that they support.

2.2.5.3 Power-saving features and modes

WiMAX's MAC layer has implemented some power-saving features, so the mobile devices to operate for longer duration without having to be recharged. Two modes are introduces for this feature. *Sleep mode* and *Idle mode.* Sleep mode is the primary procedure for power saving. In sleep mode the mobile device becomes unavailable for certain time intervals, normally of exponentially increasing size. During these intervals the mobile remains registered at the base station but can power down certain circuits to reduce power consumption. For a better management of this condition WiMAX defines three power-saving classes (Class 1, Class 2, Class 3) based on the manner in which sleep mode is executed [60].

On the other hand, and if the mobile has no traffic for a long time it can switch to idle mode in which it is no longer registered at any particular base station. To resume traffic between the network and the mobile, a paging procedure may be used by the network. When downlink traffic is arrived for the MS, the MS is paged by a collection of base stations that form a paging group. MS wakes up periodically and updates its paging group. Idle mode saves more power that than sleep mode, since the MS does not even have to register or do handoffs. Idle mode also benefits the network and BS by eliminating handover traffic for inactive MSs.

2.2.5.4 Mobility Support: Handover

In addition to fixed broadband access, WiMAX envisions four mobility-related usage scenarios:

- **Nomadic**. The user is allowed to take a fixed subscriber station and reconnect from a different point of attachment.
- **Portable**. Nomadic access is provided to a portable device, such as a PC card, with expectation of a best-effort handover.
- **Simple mobility**. The subscriber may move at speeds up to 60 km/h with brief interruptions (less that 1 sec) during handoff.
- **Full mobility**. Up to 120 km/h mobility and seamless handoff (less than 50 ms latency and <1% packet loss) is supported.

WiMAX networks will initially be deployed for fixed and nomadic applications and then evolve to support portability to full mobility over time. Also WiMAX has protocols that enable a seamless handover of ongoing connections from one BS to another. To reduce time expenses for the mobile to find the central frequency and acquire parameters of the neighbor base station, the mobile can apply a scanning process when the mobile is away from the serving base station to scan the wireless media for neighbor base stations. Information collected during scanning such as central frequencies of the neighbor base stations can then be used in actual handover. For this purpose, the serving base station periodically advertises information about the central frequency and parameters of the neighbor base stations. There are three specified handover methods. One of them is mandatory (hard handover (HHO)) and the other two are optional (Fast Base Station Switching (FBSS) and Macro Diversity Handover (MDHO)). More information about these three types of handover can be found in [60].

2.2.5.5 Security

Unlike WiFi, WiMAX was designed emphasizing in security. It includes state-of-the-art methods for ensuring user data privacy and preventing unauthorized access, with additional protocol optimization for mobility. The security sublayer provides Extensible Authentication Protocol (EAP)-based mutual authentication between the mobile and the network. Both AES

(Advanced Encryption Standard) and 3DES (Triple Data Encryption Standard) are supported.

To keep the encryption keys fresh, the security sublayer employs an authentication client/server key management protocol (Privacy and Key Management Protocol Ver. 2 (PKMv2)), which supports variety of credentials, such as username/password, digital certifications and smart cards) which allows the base station to distribute keying material to mobiles. Also using message digest schemes such as AES-based CMAC (cipher-based message authentication code) or MD5 (Message-Digest 5 Algorithm)-based HMAC (hash-based message authentication codes) protects the integrity of over-the-air control messages. Finally, WiMAX supports fast handover, in the three-way handshake scheme, which prevents any man-in-the-middle attacks.

2.2.5.6 Multicast and Broadcast Service

Multicast and Broadcast Services (MBSs) allow WiMAX mobile terminals to receive multicast data even when they are in idle mode. The most popular application of this feature is TV broadcasting to mobile terminals.

2.2.6 IEEE 802.16 (WiMAX) Standards Family

A summary of all the standards that belong to 802.16 family can be found in the table below (Table 7) [62].

Table 7 – 802.16 Standards [62]

Title	Publish Date	Description	Comment
802.16-2001	April 2002	Air Interface for 10-66 GHz	Withdrawn. Basic high data links. Data rates up to 134 Mbit/s.
802.16.2-2001	March 2004	Revision of IEEE Std 802.16.2-2001	EEE recommended practice for LAN and MAN
802.16a-2003	April 2003	Air Interface for 2-11 GHz	Withdrawn. Amendment. NLOS. Mesh
802.16b	Never Published	WirelessHUMAN QoS provisioning	Withdrawn. Increased spectrum to include frequencies between 5-6 GHz with also providing QoS
802.16c-2002	January 2003	10-66 GHz Detailed System Profiles	Coexistence and Interoperability
802.16d	2004	Maintenance and System profiles for 2-11 GHz	Merged. Upgrades the 802.16a
802.16-2004		Revision incorporating and obsolescing above 3	Data Rates up to 70 Mbit/s. MIMO. Recommended practice for coexistence.
802.16e-2005		Enhancements to support Mobility	For nomadic and mobile use. Lower data rates 15 Mbit/s. Including handover
802.16f-2005		Amendment for MIBs for fixed Systems	Management Information Base. Extension to support multi hop capabilities (mesh)
802.16g-2007	December 2007	Management Plane Procedures and Services	Suspended
802.16h-2010	June 2010	Improved Coexistence Mechanisms for License-Exempt System	
802.16i		Mobile Management Information Base	Merged to 802.16-2009
802.16j-2009		Multihop Relay	First amendment from 802.16-2009
802.16k	August 2007	Standard for MAC Bridges	Bridging of 802.16
802.16m	May 2011	WiMAX release 2 or WirelessMAN-Advanced (4G systems)	Advanced Air Interface with data rates of 100 Mbits/s mobile and 1 Gbits/s fixed
802.16n	Not published yet	High Reliability Networks	In progress
802.16p	Not published yet	Enhancement to Support Machine-to-Machine Applications	In progress

2.3 Cellular Networks

2.3.1 Introduction

Cellular communications has experienced explosive growth in the past two decades. Today billions of people around the world use cellular phones or other mobile devices. These devices allow a person to make or receive a call from almost anywhere, connect to the Internet, transmit and receive data. Likewise, a person is allowed to continue using the service while on the move. Cellular communications is supported by an infrastructure called a cellular network.

A cellular network is a radio network distributed over land areas called cells, each served by at least one fixed-location transceiver, known as a cell site or base station. In a cellular network, each cell uses a different set of frequencies from neighboring cells, to avoid interference and provide guaranteed bandwidth within each cell.

The cellular network has gone through three generations. The first generation (1G) of cellular networks was analog telecommunications standards. To accommodate more cellular network subscribers, digital TDMA (Time Division Multiple Access) and CDMA (Code Division Multiple Access) technologies are used in the second generation (2G) to increase the network capacity. With digital technologies, digitized voice can be coded and encrypted. Therefore, the 2G cellular networks are also more secure. The third generation (3G) integrates cellular devices into the Internet world by providing high-speed packet-switching data transmission in addition to circuit-switching voice transmission.

2.3.2 1G & 2G

1G technologies were first launched in 1979 in Japan and soon covered a big part of the world. 1G speeds vary between that of a 28k modem and 56k modem, meaning actual download speeds of 2.9 Kbytes/s to 5.6 Kbytes/s.

1G networks were superseded by the newer 2G technologies. The biggest advantage of 2G was the digital signal between the handsets and the towers, which increased the system capacity, by allowing more calls to be packed into the same amount of radio bandwidth. The disadvantages were that in less populous areas, the weaker digital signal may not be sufficient to reach the cell tower so occasionally dropouts should be handled, in areas that with analog signal, users were experiencing static service. But in general the digital signal sounded better than the analog one. The 2G cellular networks were firstly commercially launched on the GSM standard in 1991.

2.3.3 GSM

Global System for Mobile Communications (GSM) was created in 1980s in France and launched in 1991. It uses the 890 to 915-MHz radio band for the upload traffic and the 935 to 960-MHz radio band for the download traffic. GSM is based on FD-TDMA (frequency division–time division multiple access) radio access, which offers a 9.6-Kbit/s rate. Millions of subscribers in the world use the GSM system for their wireless cellular communications. The problem is that GSM will not be able to satisfy news services such as data networks. GSM has applied the frequency-hopping technique, which involves switching the call frequency many times per second for security. A revision of the GSM specifications defines an Extended GSM (EGSM) that extends the original GSM-900 operation band and stipulates lower-power terminals and smaller serving areas. The basic components of the GSM's network structure are the following (Figure 16). More information about GSM and its network architecture can be found in [63]:

- **Base Station Subsystem:** Base station and their controllers.
- **Network and Switching Subsystem:** The part of the network most similar to a fixed network. This is sometimes also just called the core network.
- **GPRS core Network:** The optional part, which allows packet, based Internet connections (from 2.5G technologies).
- **Operations support system:** The system for maintenance of the network.

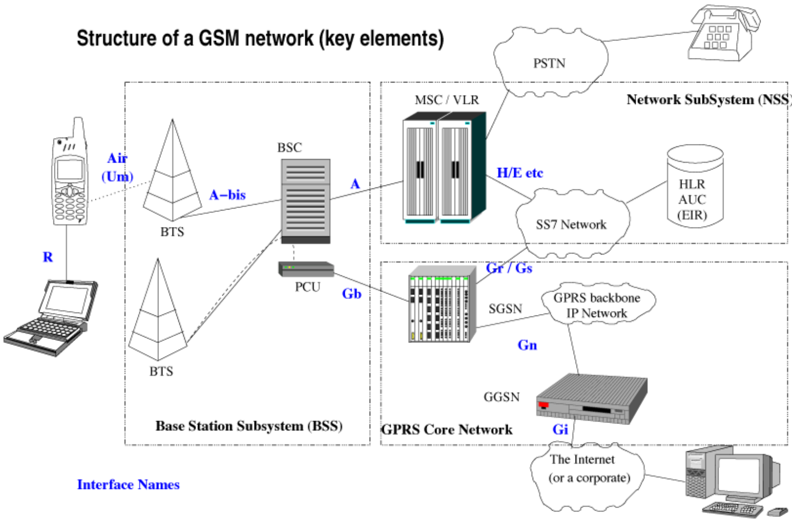

Figure 16 – Key Elements of GSM Network Structure [63]

2.3.4 From 2G to 3G technologies

2G networks were built mainly for voice services and slow data transmission. As an intermediate step in employing full packet-switching 3G systems is the 2.5G and 2.75G wireless systems. They use separate air interfaces – circuit-switching for voice and packet-switching for data – designed to operate in 2G network spectrum. These newer technologies provided data rates up to 115 kbit/s. 2.5G technology is entitled General Packet Radio Service (GPRS) and 2.75G technology is entitled Enhanced Data Rates for GSM Evolution (EDGE).

2.3.4.1 GPRS

GPRS, which was originally standardized by ETSI and is now maintained by 3rd Generation Partner Project (3GPP), is an evolution on GSM networks and is a packet oriented mobile data service on the 2G cellular communication system. It is a best effort service, implying variable throughput and latency that depends on the number of other users sharing the service. In GPRS, the new services that were offered are the: MMS, Push-to-talk, Instant Messaging, WAP, etc. Also two new network elements were introduced to GSM architecture [63]:

Serving GPRS Support Node (SGSN). It provides authentication and mobility management. At a high level, the SGSN provides similar functionality to the packet data network that the MSC/VLR provides to the circuit-switched network.

Gateway GPRS Support Node (GGSN). This provides the interface between the mobile and the backbone IP or X.25 network. The GGSN tunnels packets from the packet data network using the GPRS tunneling protocol. When a mobile wants to send data, it must set up what is referred to as a packet data protocol (PDP) context between the SGSN and the GGSN, which is more or less equivalent (at least in the context of IP) to obtaining an IP address. After setting up a PDP context, the mobile can then begin using GPRS point-to-point or point-to-multipoint services.

Finally GPRS defines three classes of terminals: A, B, and C. A class terminal supports simultaneous circuit-switched and packet-switched traffic. Thus a user of such a terminal can simultaneously talk and browse the Internet. A class B terminal can be attached to the network as both a circuit-switched and packet-switched client but can only support traffic from one service at a time. Thus, when a user of such a terminal receives a call, his Internet connection is suspended. Finally, a class C terminal uses only packet-switched services. Thus, when a user of such a terminal receives a call, his Internet connection is dropped.

2.3.4.2 EDGE

EDGE is a backward-compatible digital mobile phone technology that allows improved data transmission rates, as an extension on top of standard GSM. EDGE was deployed on GSM networks beginning in 2003 and is standardized by 3GPP as a part of GSM family.

EDGE affects the first part of GSM architecture (BSS) as seen in Figure 16. All other nodes and interfaces are not affected at all by the EDGE introduction [64]. Through the introduction of sophisticated methods of coding and transmitting data EDGE delivers higher bit-rates per radio channel, resulting in a threefold increase in capacity and performance compared with an ordinary GSM/GPRS connection. For the GPRS-based packet data services, other nodes and interfaces are already capable of handling higher bit rates and are thus not affected. For circuit-switched services, the A-bis interface can handle 64 Kbit/s per users, which is note by EDGE circuit-switched bearers.

EDGE can be used for any packet switched application, such as an Internet connection. Evolved EDGE continues in Release 7 of the 3GPP standard providing reduced latency and more than doubled performance e.g. to complement High-Speed Packet Access. Peak bit-rates of up to 1 Mbits/s and typical bit-rates of 400 Kbits/s can be expected.

2.3.5 3G

Third Generation (3G) mobile and wireless networks aim to fulfill the demands of future services. 3G systems will offer global mobile multimedia communication capabilities in a seamless and efficient manner. Regardless of their location, users will be able to use a single device in order to enjoy a wide variety of applications, such as voice telephony, mobile Internet access, fixed wireless Internet access, video calls and mobile TV.

3G systems will provide at least 144 Kbit/s for full mobility applications in all cases, 384 Kbit/s for limited mobility applications and 2 Mbit/s for low mobility applications. Two standards are typically branded as 3G: Universal Mobile Telecommunications System (UMTS) and CDMA2000.

2.4.5.1 UMTS

UMTS is based on the GSM standard. It is developed and maintained by 3GPP and IMT. UMTS specifies a complete network system which, covering the radio access network (UMTS Terrestrial Radio Access Network (UTRAN)), the core network (Mobile Application Part (MAP), and the authentication of users via SIM cards (Subscriber Identity Module). UMTS's original and most widespread radio interface is called Wideband Code Division Multiple Access (W-CDMA), which is based on the FDD mode and is described later on this chapter [65].

UMTS integrates the TD-CDMA and the W-CDMA systems. The Radio Network Subsystem (RNS) replaces the BSS in GSM networks and is composed of the following (Figure 17):

- **UE (user equipment):** Is called MS in GSM
- **NodeB:** Is called BTS in GSM

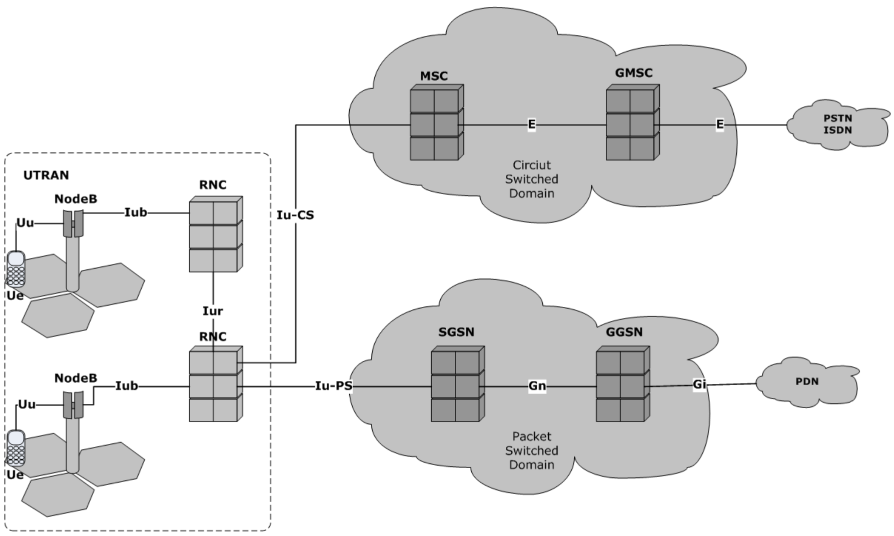

Figure 17 - UMTS Architecture [66]

- **RNC (Radio Network Controller):** Is called BSC in GSM and is the Iub interface (called A-bis in GSM) which is used to connect the RNC to the nodeB.

Finally, UMTS supports maximum theoretical data transfer rates of 42 Mbits/s when HSPA+ is implemented in the network.

2.3.5.2 W-CMDA

W-CDMA is the radio technology of UMTS and is a part of the ITU IMT-2000 family of 3G standards. Both FDD and TDD variants are supported. It increases data transmission rates in GSM systems by using the CDMA air interface instead of TDMA.

W-CDMA is a spread-spectrum modulation technique; one, which uses channels whose bandwidth, is much greater than that of the data to be transferred. Instead of each connection being granted a dedicated frequency band just wide enough to accommodate its envisaged maximum data rate, W-CDMA channels share a much larger band. The modulation technique encodes each channel in such a way that a decoder, knowing the code, can pick out the wanted signal from other signals using the same band, which appear as so much noise. By this, different users can simultaneously transmit at different data rates and data rates can vary in time.

UMTS uses a core network derived from that of GSM, ensuring backward compatibility of services and allowing seamless handover between GSM access technology and W-CDMA. A more detailed presentation of W-CDMA can be found in [63].

2.3.5.3 HSPA

HSPA is a set of mobile telephony protocols. It is defined under the 3GPP Release 99 Standard and is oriented more towards switched circuit operation and was not well suited to packet operation. Some of its basic improvements are the:

- Use of higher order modulation
- Shorter Transmission Time Interval (TTI)
- Use of shared channel transmission
- Fast NodeB scheduling
- NodeB based Hybrid ARQ

It includes three popular standards: HSDPA (High Speed Downlink Packet Access), HSUPA (High speed Uplink Packet Access and HSPA+ (Evolved HSPA) [67].

High Speed Downlink Packet Access (HSDPA): The first step required to upgrade WCDMA to HSPA is to improve the downlink by introducing HSDPA. HSDPA provides packet data support, reduced delays, and a peak raw data rate (i.e. over the air) of 14 Mbit/s. It also provides around three times the capacity of the 3G UMTS technology defined in Release 99 of the 3GPP UMTS standard.

High Speed Uplink Packet Access (HSUPA): The second major step is to upgrade the uplink, which is introduced in 3GPP Release 6 and is HSUPA. HSUPA provides improved uplink packet support, reduced delays and a peak raw data rate of 5.74 Mbit/s. This results in a capacity increase of around twice that provided by the Release 99 services.

HSPA+ or Evolved HSPA: Finally, the evolved HSPA or HSPA+, which was defined in 3GPP release 7 and 8 of the WCDMA specification, provides data rates up to 42 Mbit/s in the downlink and 11 Mbit/s in the uplink (per 5MHz carrier), which it achieves by using high order modulation and MIMO technologies.

HSPA builds on 3G UMTS/WCDMA and is strongly positioned as the leading mobile data technology for the foreseeable future.

2.4 3GPP LTE

2.4.1 Introduction

Long Term Evolution (LTE), marketed as 4G LTE [68], is a standard for wireless communication of high-speed data for mobile and data terminals and is developed by 3GPP. LTE is a step beyond 3G and towards 4G, evolved after EDGE (Enhanced Data Rates for GSM Evolution), UMTS (Universal Mobile Telecommunication Services), HSPA (High Speed Packet Access) and GSM (Global System for Mobile Communications). It enhanced UMTS and GSM network technologies, by increasing the capacity and speed using a different radio interface together with core network improvements. It's contribution make sure that users are able to request more mobile application like interactive TV, mobile video blogging, advanced games or professional services.

The LTE specification provides downlink peak rates of 300 Mbits/s, uplink peak rates of 75 Mbits/s and QoS provisions permitting a transfer latency of less than 5 ms in the radio access network. LTE has the ability to manage fast-moving mobiles and supports multi-cast and broadcast streams. It supports both FDD and TDD, is based on an IP-based network technology and supports a seamless handover with older technologies, such as GSM and UMTS.

2.4.2 Architecture

LTE is designed to support only packet switched services, in contrast to circuit-switched domain from the previous cell systems. Its purpose is to provide an IP connection between the User Equipment (UE) and the Packet Data Network (PDN) without any interrupt in the service during the mobility time.

The LTE system is comprised of two networks [69]: the E-UTRAN and the Evolved Packet Core (EPC). The result is a system characterized by its simplicity, a non-hierarchical structure for increased scalability and efficiency, and a design optimized to support real-time IP-based services.

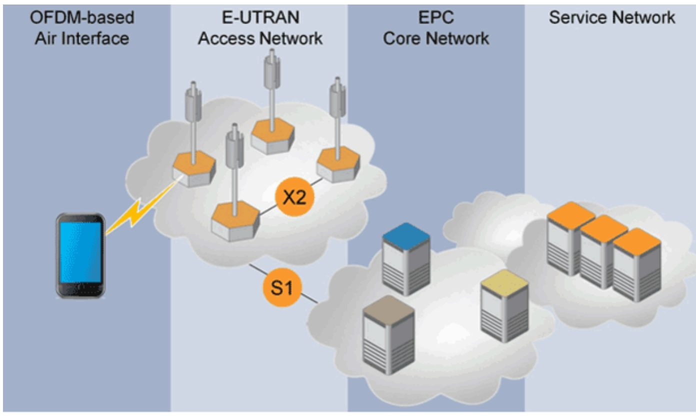

Figure 18 - LTE Architecture [69]

2.4.2.1 E-UTRAN

Evolved Universal Terrestrial Radio Access Network (E-UTRAN) [70] is the air interface and is a part of the Access Network of LTE. Unlike HSPA, which was an upgrade on UMTS, E-UTRAN is an entirely new air interface system. It provides higher data rates, lower latency and is optimized for packet data. It uses OFDMA radio-access for the downlink and Single Carrier Frequency Division Multiple Access (SC-FDMA) on the uplink, and MIMO antenna technology.

Each UE is connected on an Evolved-NodeB (eNB). Each eNB has an IP address, is a part of the all-IP network and all eNBs are interconnected with each other by means of the X2 interface. Dedicated radio network controllers, which were present in earlier generation access network, are not required. The eNBs in LTE collaborate to perform functions such as handover and interface mitigation.

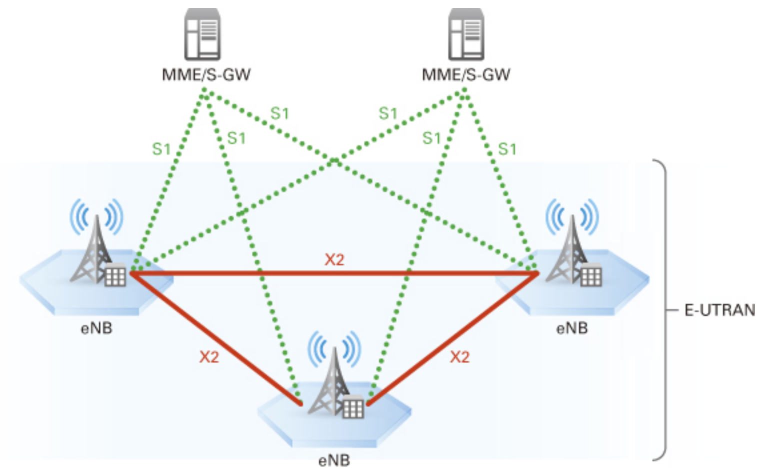

Figure 19 - E-UTRAN and eNodeBs [70]

2.4.2.2 Evolved Packet Core (EPC)

Evolved Packet Core (EPC) [71] is the latest evolution of the 3GPP core network architecture. EPC was first introduced by 3GPP in Release 8 of the standard. It is a smaller part of System Architecture Evolution (SAE), which started from GPRS Core network, but it has some differences (simplified architecture, all-IP network, support for higher throughput and low latency radio access networks, support and mobility between multiple heterogeneous access networks). It was decided to have a “flat architecture”. The idea is to handle the payload efficiently from performance and costs perspective. Few network nodes are involved in the handling of the traffic and protocol conversion in avoided. It was decided to separate the user data and the signaling to make the scaling independent. EPC is composed of four network elements: the Service Gateway (Serving GW), the PDN Gateway (PDN GW), the Mobility Management Entity (MME) and the Home Subscriber Service (HSS). The EPC is connected to external networks, which can include the IP Multimedia Core Network Subsystem (IMS). Below these network elements are briefly presented.

- **HSS**: Is a database that contains user-related and subscriber-related information.
- **Serving GW**: Is the point that interconnects between the radio-side and the EPC. It routes the incoming and outgoing IP packets.
- **PDN GW**: Is the point of interconnecting between the EPC and the external IP networks. The networks are called PDN (Packet Data Network). The PDN GW routes packets to and from the PDNs.
- **MME**: It deals with the control plane. It handles the signaling related to mobility and security for E-UTRAN access.
- **IMS**: Is a complete SIP-based control architecture that includes charging, billing and bandwidth management.

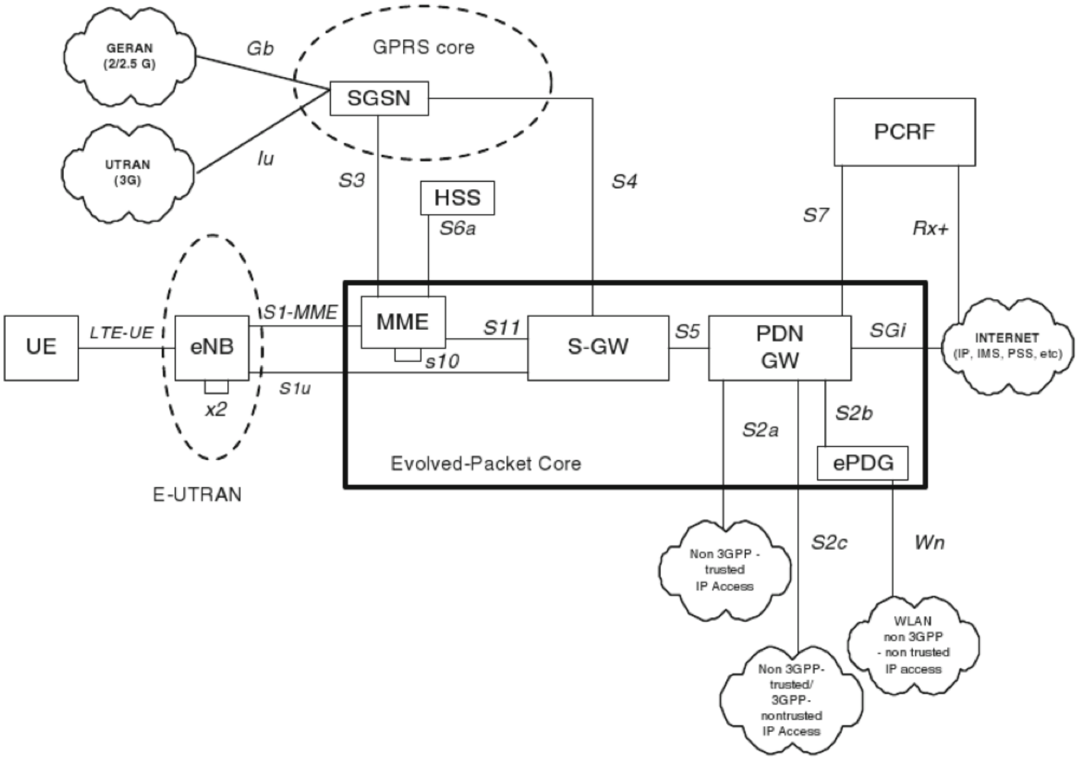

Figure 20 – UE, E-UTRAN, EPC elements and connections with external networks []

2.4.3 Services

LTE supports two different services: Broadcast and Multicast. Broadcast service can be received by any subscriber located in the area in which the service is offered and multicast services can only be received by users subscribed to the service and joined the multicast group associated with the service.

For such a service, only the broadcast service providers can be changed possibly based on the amount of data broadcasted, size of service area or broadcast service duration. Multicast is subject to service subscription, and requires the end-user to explicitly join the Group in order to receive the service. Because it is subject to subscription, the multicast service allows the operator to set specific user changing rules for the service.

LTE also supports QoS. QoS is primarily a layer 3 IP concept. It uses tools that have existed since the early days of IP plus some newer tools and protocols that are designed to aid in the provisioning of precisely defined and predictable data transfers in accordance with certain characteristics. The critical QoS parameter for any EPS bearer (user data flow) is its QoS Class Identity (QCI), which represents the QoS features an EPS bearer should be able to offer for a Service Data Flow (SDF). Each bearer (user data) path in LTE is assigned a set of QoS criteria. Since a user may have services requiring different QoS criteria, additional bearer paths may be added. LTE's identified QCI criteria are listed in Table 8.

2.4.4 Physical Layer

The PHY layer offers data transport to higher layers. The PHY layer is being designed to perform the following functions [73]:

- Error detection on the transport channel and indication to higher layers.
- FEC encoding/decoding of the transport channel.
- Hybrid ARQ soft-combining
- Rate Matching
- Mapping of the coded symbols to physical channels
- Power weighting of physical channels
- Modulation and demodulation
- Frequency and time synchronization
- Radio characteristics measurement and indication to higher layers
- MIMO/transmit diversity beamforming support
- RF processing

Table 8 – Standardized QCI characteristics [72]

QCI	Resource Type	Priority	Packet delay budget	Packet error loss rate	Example services
1	GBR	2	100 ms	10^{-2}	Conversational voice
2		4	150 ms	10^{-3}	Conversational video (live streaming)
3		3	50 ms	10^{-3}	Real time gaming
4		5	300 ms	10^{-6}	Non-conversational video (buffered streaming)
5	Non-GBR	1	100 ms	10^{-6}	IMS signaling
6		6	300 ms	10^{-6}	Video (buffered streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, etc.)
7		7	100 ms	10^{-3}	Voice, Video (live streaming), Interactive gaming
8		8	300 ms	10^{-6}	Video (buffered streaming), TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, etc.)
9		9			

LTE physical layer supports two multiple access schemes: OFDMA with cyclic prefix in the downlink and SC-FDMA with cyclic prefix in the uplink. In addition, both paired and unpaired spectrums are supported, by using full duplex and half duplex FDD and TDD, respectively [74].

The LTE air interface needs to be described in both the time and frequency domains. The frame structure defines the frame, slot and symbol in the time domain. Two types of frame structures are defined: *Type 1* for both FDD modes and *Type 2* for TDD. Although the downlink and uplink utilize different multiple access schemes, they share a common frame structure.

Radio frame length is 10 ms and consists of 10 subframes. Its subframe contains two slots (20 slots the frame, 0.5 ms per slot). In TDD mode there are 7 different configurations defined for the frame structure, according to whether the subframe is an uplink or a downlink subframe or a special subframe with the three fields DwPTS, GP and UpPTS.

To support MBMS, LTE provides the ability to transmit Multicast/Broadcast over a Single Frequency Network (MBSFN), where a synchronized common waveform is transmitted from multiple cells at a given time. The MBSFN transmission enables high performance of MBMS, allowing for the radio interface, combined with broadcasts multi-cells to UE, where the cyclic prefix is used to cover the difference in propagation delays, to appear as a single large cell. The transmission antennas are MIMO with 2 or 4 antennas for transmission and 2 or 4 for reception.

Finally, LTE supports QPSK, 16-QAM and 64-QAM in downlink and QPSK and 16-QAM in uplink as modulation coding scheme, through advanced coding with lower base rate. HARQ is synchronous in uplink and asynchronous in downlink.

2.4.5 MAC Layer

MAC protocol layer exists in UE & eNB and it is a part of LTE air interface control and user planes. The main services and functions of the MAC sublayer include [73]:

- Mapping between logical channel and transport channels

- Multiplexing/demultiplexing of MAC SDUs belonging to one or different logical channel into/from Transport Blocks (TB) delivered to/from the physical layer on transport channels
- Scheduling information reporting
- Error correction through HARQ
- Priority handling between logical channels of one UE
- Priority handling between UEs by means of dynamic scheduling
- Transport format selection
- Padding

A *logical channel* is defined a by the type of information it carries. Generally is classified as: a control channel (used for transmission of control and configuration information necessary for operating an LTE system) and a traffic channel (used for the user data).

2.4.5.1 Scheduling

eNB scheduler controls the time/frequency resources for a given time for uplink and downlink. The scheduler dynamically allocates resources blocks (which are the smallest elements to resource allocation) to users for predetermined amount of time (TTI Transmission Time Interval). Depending on the channel condition, scheduler selects the best multiplexing for UE. The decision can be based on any combination of the following:

- QoS parameters
- Measurements
- Buffered payloads
- Pending retransmissions
- CQI reports from the UEs
- UE capabilities
- UE sleep cycles
- Measurement gaps/periods
- System parameters such as bandwidth and interference level/patterns

LTEs downlink considers the following schemes as a scheduler algorithm: Frequency Selective Scheduling (FSS), Frequency Diverse Scheduling (FDS), and Proportional Fair Scheduling (PFS).

2.4.5.2 HARQ

Hybrid Automatic Repeat Request (HARQ) is a combination of high-rate forward error-correcting coding and ARQ error control. Multiple parallel stop-and-wait processes are used (this can result in data being delivered from the HARQ mechanism out-of-sequence, in-sequence delivery is ensured by the RLC layer). HARQ framework in LTE considers incremental redundancy and special case of chase combining.

HARQ is not applicable for all type of traffic. Broadcast transmissions typically do not rely on HARQ. Also it can be synchronous and asynchronous. Synchronous HARQ requires that transmission occur at known time instants. No explicit signaling is required. On the other hand, for asynchronous HARQ, explicit signaling is required to accommodate HARQ process that happens anytime. HARQ can also be adaptive or nonadaptive. Adaptive HARQ has the ability to change the modulation, resource block allocation and duration of transmission. Note that synchronous operation requires less control signaling and has significant advantage when it is nonadaptive since soft-combining can be performed. This mode is selected for uplink. However, in downlink asynchronous and adaptive HARQ mode is considered.

2.4.5.3 Cell Search

Cell Search is a basic function of any cellular system, during which process time and frequency synchronization between the mobile terminal and the network achieved. Synchronization Channel (SCH) and Broadcast Channel (BCH) are detected during the search. SCH is for timing information such as symbol timing and frequency of the downlink signal. BCH is to broadcast certain set of cell-specific information such as transmission bandwidth, cell id, antenna configuration, etc.

2.4.5.4 Power Control

Power control is considered to mitigate path loss and shadowing. It determines the energy per resource element (EPRE) applied for an uplink transmission. The term resource element energy denotes the energy prior to

CP insertion. It also denotes the average energy taken over all constellation points for the modulation scheme.

2.4.5.5 Intercell Interference Mitigation

LTE standard proposed Intercell interference mitigation to be handled via three different approaches: randomization, cancelation, and coordination and avoidance.

2.4.6 Radio Link Control Layer

Depending on the scheduler decision, a certain amount of data is elected for transmission for the Radio Link Control (RLC) SDU buffer. RLC header and SDU form the RLC PDU. RLC layer is responsible to transfer PDUs between UE and eNB with segmentation if needed and applies error correction through ARQ for received data, although it is capable of handling transmission errors. It applies concatenation, in-sequence delivery, and duplicate detection. RLC layer provides three different reliability modes:

- **AM**: Acknowledge Mode requires acknowledgement and is good for unreal time services such as file download.
- **UM**: Unacknowledged Mode does not require an acknowledgement and is suitable for real time services such as video streaming.
- **TM**: Transparent Mode implement implicit acknowledgement and is used when file sizes are known as in broadcasting.

2.4.7 3GPP LTE Standards Family

3GPP standards are structured as releases. Discussions of 3GPP thus frequently refer to the functionality in one release or another. Each release incorporates hundreds of individual standards documents, each of which may have been through many revisions. Current 3GPP standards incorporate the latest revision of the GSM standards. The documents are available freely on 3GPP's Web Site. While 3GPP standards can be bewildering to the newcomer, they are remarkably complete and detailed, and provide insight into how the cellular industry works. They cover not only the radio part (Air Interface) and Core Network, but also billing information and speech coding down to source code level. Cryptographic aspects (authentication, confidentiality) are also specified in detail. First LTE release was made on 2008 and it was Release 8 of 3GPP standards. In Table 9 we can see a summary of all the 3GPP releases including the pre-LTE released standards from 3GPP too [75].

Table 9 – 3GPP LTE standards

Versions	Released	Info
Phase 1	1992	GSM Features
Phase 2	1995	GSM Features, EFR Codec
Release 96	1997 Q1	GSM Features, 14.4 Kbits/s User data Rate
Release 97	1998 Q1	GSM Features, GPRS
Release 98	1999 Q1	GSM Features, AMR, EDGE, GPRS for PCS1900
Release 99	2000 Q1	Specified the first UMTS 3G networks, incorporating a CDMA air interface
Release 4	2001 Q2	Originally called the Release 2000 – added features including an all-IP Core Network
Release 5	2002 Q1	Introduced IMS and HSDPA
Release 6	2004 Q4	Integrated operation with WLAN networks and adds HSUPA, MBMS enhancements to IMS such as Push to Talk over Cellular (PoC), GAN
Release 7	2007 Q4	Focuses on decreasing latency, improvements to QoS and real-time applications such as VoIP. This specification also focus on HSPA+, SIM high-speed protocol and contactless front-end interface (Near Field Communication enabling operators to deliver contactless services like Mobile Payments), EDGE Evolution
Release 8	2008 Q4	First LTE release. All-IP Network (SAE). New OFDMA, FDE, and MIMO based radio interface, not backwards compatibility with previous CDMA interfaces. Dual-Cell HSDPA.
Release 9	2009 Q4	SAES Enhancements, WiMAX and LTE/UMTS Interoperability. Dual-Cell HSDPA with MIMO, Dual-Cell HSUPA.
Release 10	2011 Q1	LTE Advanced fulfilling IMT Advanced 4G requirements. Backwards compatibility with release 8 (LTE). Multi-Cell HSDPA (4 Carriers)
Release 11	Planned to 2012 Q3	Advanced IP Interconnection of Services. Service layer interconnection between national operators/carriers as well as third party application providers.
Release 12	Planned to 2014 Q2	Content still open (as of January 2012).

CHAPTER 3

QoE over Wireless Networks Technologies

3.1 Introduction

Before we have introduced the wireless networks technologies (WiMAX, LTE, IEEE 802.11 and Cellular) that we will deal with in this chapter and we have done a detailed introduction on QoE. On this chapter we will present the association of QoE over these four wireless network technologies.

By making a literature research, a lot of papers were found that present implementations and ways to improve the perceived QoE. For this thesis we have chosen to present some implementations by taking into account the time that have been proposed, the type of the data streamed and the number of metrics used to evaluate the results. Our target is to make a guide, which will concentrate all this amount of information in a smaller text that will be useful for someone who is interested on this field to read and acquire the necessary background, before starting implementing a new idea.

Our first concern was to present, as many new implementations as possible. Most of the works provided in this thesis, were published the last two years except of some papers that are older. Also we have chosen papers that introduce experiments that use video or audio streams to evaluate the results of each proposed mechanism. Since the volume of the works was really big, we tried to present works that combine more than one metrics, even some times more than one categories of metrics. Finally, all the works presented are related with the four wireless networks technologies described in the previous chapter.

We followed a pattern classification, which comprises on the top with the four wireless network technologies presented, later by taking into account the type of data used for the experiment, and finally the category of the metrics used. Finally, at the beginning of the network subchapters, some tables are created to give a more comprehensive presentation of the efforts that have been made.

3.2 WIMAX

Table 10 – WiMAX and QoE association

File	Publication Date	Type of Streaming	Type of Metrics	Metrics
[76]	April 2010	Video	Objective	MSE, PSNR
[77]	February 2012	Video	Objective	SSIM, VQM
[78]	January 2012	Video	Objective	MDI
[79]	September 2010	Video	Subjective	MOS
[80]	January 2009	Video	Network Planning Models	Delay, Jitter, Packet Loss
[81]	July 2009	Audio	Subjective – Network Planning Models	MOS, Packet Loss, Delay

3.2.1 Video

3.2.1.1 Objective

In this work [76], an effective IPTV channel control algorithm is presented that improves the QoE of mobile IPTV services over WiMAX networks. The proposed algorithm concurrently considers both the distribution state and the bit rate of TV channels as control variables, based on the preferred channels of the subscribers, to achieve an effective settlement between channel zapping time and video quality.

In an engineering sense, if all of the IPTV channels are broadcasting constantly, subscribers are able to change TV channels with a very small channel zapping time because all of the channels are immediately available at the subscriber's side. In this case, the bit rate of each channel must be kept low due to the bandwidth limitation, which means that the received video quality may be degraded. On the other hand, when the bit rates of TV channel streams are high, the video quality becomes better but the channel zapping time is increased since only a limited number of proactive TV channels can be supported because of resource limitations. Thus, we need an effective trade-off between channel zapping time and video quality by adjusting the channel distribution state and the target bit rates of TV channels.

Based on the previous preference information, two methods are presented to determine the optimal solutions. The first one is the full search-based algorithm, and the second is the bisection-based fast algorithm. In the first algorithm the cost function is calculated for all the possible channel distribution state vectors, and then the channel distribution state with the minimum cost is the optimal solution. The problem of the first algorithm was the high computational complexity. To reduce it, a second algorithm was created and is based on the bisection method. Compared to the full search-based algorithm, the fast algorithm significantly reduces the number of candidates, but an extensive subjective test was required to estimate the value of a auxiliary variable that represents the number of the proactive channels, because it was very difficult to define by a mathematic model the human perceptual quality of channel zapping and video quality.

During the simulation, a number of 10 subscribers were set in the cell and 50 channels were used. The maximum bandwidth was 2.5 Mbits/s from which 0.5 was used by proactive channels and the rest of on-demand channels. The algorithm that was used for the simulation was the fast one. As we can see from the results as much as the number of the proactive channels was increased the average distortion (MSE) was increased too because the total bandwidth that was used was 500 Kbits/s. On the other hand the average channel zapping time was decreased because more channels were broadcasting all the time. In this simulation, the average channel zapping time is considered. In a worst case scenario (when channels that are usually not preferred by most subscribers are selected), a subscriber would experience longer channel zapping time although most subscribers would experience a lower channel zapping time. So to avoid this scenario, high order moments should be considered.

According to the value of PSNR, we can see that as the number of proactive channels becomes bigger, the quality of the channels is decreased. So since the average distortion is emphasized in the algorithm when the number of proactive channels small, the proposed system chooses a higher bit rate.

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[77] presents the benefits and the impacts of two different handover policies on a Mobile WiMAX scenario. These two types of handover policies are the following. The first one is when the SS moves to a new BS, it stop the connection with the current BS before establishing the connection with the new BS. This procedure is also known as “hard handover” or “break – before

– make”. The second one is when the SS establishes the connection with the new BS, before it stops the connection with the current BS and it known as “seamless handover” or “make – before – break”.

It is necessary to create seamless mobility schemes to improve Mobile WiMAX handover process. For the experiments and regarding to the position and the speed of the mobile stations, three different mobility profiles were used: high, medium and low. The high mobile node will stay the shortest time inside the cell and in this situation the handover will be triggered before the other mobile nodes. The handover process is triggered according to the speed of the mobile station and the link failure probability. So, for high mobility, three handovers take place, for medium mobility two handovers and for low mobility one handover.

Two types of experiments were performed. The first one was with Case - Based Reasoning (CBR) traffic and the second with video traffic. From the first experiments, it is clear that without a handover policy, the bigger the speed of the mobile node becomes, the lower the throughput of the system is because there are time intervals that the nodes doesn’t receive any data, until the handover with the new BS is finished. On the other hand, with seamless handover, throughput is almost the same for all the mobility profiles.

During the second experiment, there is a difference of 5% of the packets received between the two types of handover, so there is a reduction in the quality of the video. The QoE metrics confirm the previous statement. The video with seamless handover policy has 32dB PSNR. This value describes the video as "good", while the video with a hard handover policy has 29dB PSNR. This value describes the video as "acceptable." Apart from PSNR, another metric that confirms the superiority of the video with a handover policy over the video without it is SSIM. The value 1 means the exact same video. The SSIM for the video with seamless handover was 0.9. For the video with hard handover, the SSIM was equal to 0.7. Finally, according to VQM for the video with seamless handover, the value was 1.4. For the video with hard handover, the VQM was equal to 2.6.

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In [78] can be found a work that improves a previously proposed scheduling algorithm that is responsible to share the allocated capacity to the uplink traffic of an integrated satellite and WiMAX network. This work's target is to improver the previously proposed mechanism in order to make the scheduling of real time connections based on the use of QoE metrics. For these purposes, FC-MDI metric was chosen and used, because it gives a different weight to the loss of different categories of voice and video frames.

The previously proposed mechanism was an interconnection of a satellite and a WiMAX network, assuming that one or more of the Return Channel Satellite Terminals (RCSTs) are also WiMAX BSs serving a number of SSs. The integrated scheduling provision mechanism consisted of three main parts. The algorithm Real Time FIFO Scheduler (RTFS) proposed treated the transmission with the logic of a FIFO queue. The packets of all rtPS connections are inserted in one queue based on the order of their arrival. The third part of the mechanism was responsible for dropping the packet if it had been expired due to delay, and is the part that was improved in this work.

At this point, two different alternatives were studied. The first one, entitled LAQoEG had a greedy logic. It sorted the connections based on mean LA-MLR in ascending way, from the best quality to the worst. The second alternative was named LAQoEF and had a fair logic. In order to be fair and maintain all connections, the connections were sorted in the opposite way, from the worst quality to the best. Also except from these alternates, there are two versions more, those are the previous algorithms combined with rate adaptation (LAQoERAG and LAQoERAF respectively). Also different algorithms to measure the FC-MDI were used.

To measure the performance, a simulator written in C++ was used and was simulating the full operation of WiMAX networks and Satellite networks' return link. For our scenarios only one video connection was used for every SS to present the difference between the greedy and the fair versions.

All versions of the proposed algorithms have the same performance conserving the goodput, mean delay and loss rate, as the logic of the algorithms for sharing capacity is the same. By comparing the QoE results of the simulations we can see that the best QoE performance is achieved by the LAOoERA algorithm. This is due to the rate adaptation of this algorithm, which loses the least of the transmitted information. It may transmit in lower quality but it transmits more information. Also LAQoE algorithm further reduces the mean delay of the connections, and improves the QoE performance of the video connections relatively to the FC_MDI_S algorithm. This is due to the philosophy of this algorithm that serves the sequence of packets with the best QoE metric. Finally, the LAQoERA algorithm has the best mean delay and QoE performance for video connections, as it loses less of the transmitted information due to the rate adaptation that it makes.

3.2.1.2 Subjective

In [79] a packet scheduling algorithm was developed for multihop-wireless networks over WiMAX. The goal of this scheduler was to maximize the overall user perceived quality and fairness among competing flows under resource constraints. The scheduler locates sets of packet combinations across all active flows of all users that pass the node that would satisfy a given buffer reduction. For each of these combinations, an estimation of the user satisfaction expressed in MOS decrease for each flow is calculated. The scheduler then drops the packets whose combination results in the smallest decrease in QoE satisfaction.

Analyzing the impact of packet combinations, instead of just the effect of just one packet at a time, can bring improved performance at the expense of a higher computational effort. By the analysis that was made, was realized that limiting the number of evaluated combinations to a sensible value can provide good scheduling performance with substantial reduction in computational cost.

For the simulation, two scenarios were generated. For the first scenario a growing number of video flows was used (up to 9 flows) on a 9-node grid and the scheduler improved the MOS of each flow more than 0.5 points. The network showed a near-excellent quality (more that 4.0 points in MOS) up to 7th parallel video flow.

For the second scenario where used three different type of flows (video, voice and file transfer services) on the same 9-node grid and on another one of 5 nodes. The flows where increased successively until the network came to saturation point. Scheduler mostly increased MOS on video flows by 0.2 to 0.5 points. On the other hand voice and data flows are less prone to quality degradation and their MOS is decreased when the network saturation is increased. Scheduler reduces the variation in user satisfaction across the different services relative to the case of conventional scheduling

3.2.1.3 Network Planning Model

Except from the classic variables of QoE, some of the most well known variables of QoS can be used to characterize the QoE of a system. That was used in [80]. In this work was proposed a system where several wireless technologies coexisted and the customers have dual or tri-band devices. The wireless networks used where IEEE 802.16, IEEE 802.11a and IEEE 802.11g at any time, but it could be a situation where there are places covered with only two of those technologies.

The proposed algorithm measures the wireless networks' Received Signal Strength Indication (RSSI), when a user wants to see IPTV, and the devices joins the one with the higher value. The IPTV software exchanges information as jitter, packet loss and delay with a QoE test server and the QoE is calculated for this network. QoE in this situation is defined by an equation that takes into account the above information. Later it is compared with other QoE values that are already calculated and saved on server and it is decided if the chosen network is the appropriate for the user. If not the information about the best network are sent back to the user's device and the connection is established again with the new network. The system described allows balancing the network's QoE in by placing the customers to the best network's QoE in that moment.

In order to test the network performance and analyze which features offers, two different scenarios where used. First, a point to multipoint WiMAX system and second, a point to multipoint IEEE WLAN 802.11a/g system, used in both operating modes separately. For every scenario, in order to analyze the performance and the quality of IPTV service, the values of jitter, delay, packet loss, and bandwidth were tested.

According to the delay, all the systems had a delay around 2 msec, which is a really good value if we take in account the maximum of 50 msec for delay. On jitter test, WiMAX and IEEE 802.11a had a better performance of about 35% better than IEEE 802.11g, probably because of the air interference as it is described by the author. While testing packet loss, IEEE 802.11a technology was proved less robust than the other ones because it showed almost double packet loss that the other two systems. Finally, based on the effected bandwidth test WiMAX technology can support up to 5 IPTV channels, IEEE 802.11g 2 channels and finally IEEE 802.11a only one.

By calculating the QoE of the systems using the equation that was proposed IEEE 802.11a technology had the most stable network QoE. WiMAX followed it and the last one was IEEE 802.11g. In all technologies a similar average value of network's QoE was calculated, with IEEE 802.11a to stand, having a slightly better value than the other two technologies.

3.2.2 Audio

3.2.2.1 Subjective – Network Planning Models

The work presented in this paper [81] assesses the VoIP quality of a WiMAX network, using User Datagram Protocol / Real-time Transport Protocol ((UDP)/(RTP)) and Datagram Congestion Control Protocol (DCCP) transport protocols. VoIP quality is measured according to the voice quality experienced by the end users, through the objective calculation of the MOS value, as well as through conventional network parameters, such as one-way delay and packet loss.

For the experiments, two scenarios where used. The first was when the bandwidth reservation was overestimated and the second one when the bandwidth reservation is underestimated. The bandwidth allocated for the first scenario is 9000 Kbit/s. On the second scenario, sets of preliminary tests were performed in the BS in order to establish the minimal bandwidth required for each client set. The number of simultaneous clients using the channel, as well as, the bandwidth reserved to support the respective flows is also tested to evaluate the scalability and the behavior with overestimated and underestimated reservations, respectively.

Analyzing the results, in the first scenario, the average one-way delay increases slightly for both UDP and DCCP with CCID3 as the number of clients' increments but stays below the reference value of 150ms. Also packet loss is below the reference value of 1% in both cases. However, when using DCCP with CCID2, the one-way average delay exceeds the reference value for 60 and 70 clients and the packet loss for 50 clients and more. This is due to the Transmission Control Protocol (TCP) like mechanisms, which require more bandwidth in the uplink channel for delay and adjustment of the window size for the packet loss. According to MOS, system provides very good voice quality for up to 60 clients with UDP and DCCP with CCID3, with UPD providing a slightly better result when the voice quality is poor.

During the second scenario, UDP flows have always one-way delay below the reference value. The similar behavior has the DCCP with CCID2 for up to 30 clients. DCCP CCID2 has always poor performance. According to packet loss, none of the tested protocols was able to grant a value below 1%, so the performance of all was really poor. Finally, calculating the MOS for the system, the MOS value was always equal to one, which means that the voice quality perceived by the clients was very bad.

3.3 LTE

Table 10 - LTE and QoE association

File	Publication Date	Type of Streaming	Type of Metrics	Metrics
[82]	December 2011	Video	Subjective	MOS
[83]	October 2011	Video	Objective	PSNR
[84]	June 2011	Video, Audio	Objective, Network Planning Models	BLER, Throughput

3.3.1 Video

3.3.1.1 Subjective:

In [82] a multi-criteria QoE driven optimization problem for multi-user wireless video delivery was presented. Data rate and network resources that fulfill the criteria are calculated by using parameterized models of application and link layer. Optimization for wireless video delivery takes into account two objectives: utility maximization and utility max-min fairness. The first one (MaxSum) emphasizes achieving a maximum average perceived quality of all users, which can be interpreted as how efficient the network resources are used and distributed to all users. Whereas for the second objective (MaxMin), its goal is to achieve a similar perceived quality among all users. It emphasizes minimizing the quality difference between the user experiencing the highest quality and another user experiencing the lowest quality.

Fairness and system efficiency are partially contrary, so to achieve the desired level for each of the utility, an algorithm that controls the operation point is necessary. So, three tuning algorithms implemented. The first one (Sum-MOS algorithm) enables a full control of resource allocation in order to deliver the desired mean quality of all users that is pre-defined by the network operator. K-algorithm focuses on the quality (un)fairness k. It allows the network operator to apply a strict fairness constraint value that is set in advance and to allocate its network resources accordingly, while maintaining the system efficiency as high as possible under the fairness condition. Finally, on advanced k-algorithm efficiency and fairness are predefined. Since it is possible that both constraints may not be met from any feasible set of resource allocation, a two-step optimization that combines the Sum-MOS algorithm with the k-algorithm is proposed. For the simulations a single LTE base station scenario was used, in which 20 users are watching a video on their terminals and are experiencing different wireless channel conditions.

According to MOS results, by using Sum-MOS algorithm, with MaxSum the differences in perceived quality of all users are larger than applying the MaxMin, however MaxSum has a higher mean MOS of all users in the systems. Using the k-algorithm with different values of fairness, the higher the value, the higher the mean MOS of the system. Finally when applying the advanced k-algorithm with predefined fairness and efficiency values, the requirements cannot be met at the same time so the priority parameters that are set, play an important role. If the efficiency priority is higher, mean MOS is getting closer to Sum-MOS algorithm results and vice versa. For all the simulation results, we can observe that the user fairness comes at a cost of the system efficiency. Also must be mentioned that for all tuning mechanisms do not intend to get a better result than the MaxSum and the MaxMin in terms of the mean MOS and the user fairness respectively.

3.3.1.2 Objective:

A QoS aware packet scheduler for real-time downlink communications was designed for [83]. It has been built on two distinct level that interact together in order to dynamically assign radio resources to UE. They take into account the channel state, the data source behaviors, and the maximum tolerable delays. The upper level exploits an innovative approach based on discrete-time linear control theory. At the lower level a proportional fair scheduler has been properly tailored to our purposes.

At the highest level, an innovative low complexity resource allocation algorithm has been designed (frame level scheduler FLS), which defines frame by frame the amount of data that each real-time source should transmit to satisfy its delay constraint. The lowest layer scheduler allocates resource blocks in each TTI to achieve a trade-off between fairness and system throughput. Lower level scheduler assigns recourses first to flows hosted by UEs experiencing the best channel quality, according to a proportional fair algorithm by considering bandwidth requirements of FLS. Radio resources left free by real-time flows can be used to provide a best-effort service using the proportional fair algorithm, which enforces fairness also for this kind of flow.

An LTE simulator was used to conduct the experiments. The results were compared with well-know scheduling strategies LOG rule and EXP rule. Finally to appreciate the effectiveness of the proposed allocation scheme in realistic settings the impact of QoE perceived by end users for real-time flows has been analyzed. A 19-cell scenario was used. In each sell, there are one eNodeB and a variable number of UEs. Also two different speeds have been used, to analyze both pedestrians and vehicular users. It has imposed that each UE receives at the same time one video flow, one VoIP flow, and one best-effort flow.

MOS on the system has been computed assuming a constant end-to-end delay, equal to the target delay, due to the presence of a playout buffer at the receiver. Then, the transmission rating factor has been mapped to the proper MOS value. All schedulers, as we have seen from the results, are able to provide a good speech quality in all operative conditions. The quality of the received video has been estimated computing the PSNR between the transmitted and the received videos. As expected, PSNR increases as the packet loss rate decrease. Also it is clear from the results, proposed approach is able to greatly outperform the existing ones by guarantee a PSNR gain up of 30 dB with respect to both LOG and EXP, especially in the presence of real-time video flows.

3.3.2 Video & Audio

This paper [84] evaluates the performance of Dynamic Quality Oriented Adaptation Scheme (DQOAS) algorithm in conjunction with a new QoS parameters mapping scheme in case of applications generating VoIP, video streaming and web browsing traffic. DQOAS algorithm is one of the most important factors in the delivery process and is developed to increase end-user quality, while also enabling more that one user to communicate in parallel. It is applied on server side. On the client side there is a Decoding & Playing Module whose role is to decode and play the adapted video stream received from the server. Later the estimated QoE is calculated and the data are passed to the Feedback Module, who is monitoring parameters as loss rate, delay, jitter and accesses the quality of delivery.

On the proposed algorithm, DQOAS is used together with a new prioritization mapping scheme to increase control over the data that are dynamically scheduled. From the data generated by the application only video and Web browsing traffic is dynamically scheduled. DQOAS can update the quality levels for the multimedia stream based on the user preferences, on instantaneous channel conditions and on the recourse allocation scheme, with minimum impact on the VoIP traffic, increasing the overall QoE of the application.

Three scenarios were used for testing the performance of the proposed adaptation mechanism. First scenario uses the standard QoS parameters mapping scheme and the LTE QoS mechanism in order to deliver the three streams that have different priorities. In the second scenario, it is used the proposed mapping scheme, with same priority on video and web browsing. Finally on the third scenario, DQOAS is used as the delivery algorithm in conjunction with the new scheme. For each scenario, three of the most common schedulers are considered. : Maximum Throughput (MT), Round Robin (RR) and Proportional Fair (PF).

Analyzing results, we can see that with DQOAS algorithm, the throughput maintains a stable value but BLER values are reduced slightly. All the streams are kept above the minimum quality level that was expected, something that doesn’t happen when the original mapping scheme is used. Big variations in throughput are unwanted during the multimedia delivery because they rapidly decrease the perceived quality. By observing BLER value, we can see that on the third scenario is lower than 0.7 – 1.4 than the original scheme. Also with MT scheduler, always BLER value is the biggest between the 3 scenarios. The smaller number of satisfied users was achieved on the first scenario with MT scheduler. By using DQOAS there is an increase of 23% on satisfied users number.

3.4 IEEE 802.11

Table 11 – IEEE 802.11 and QoE association

File	Publication Date	Type of Streaming	Type of Metrics	Metrics
[85]	July 2011	Video	Subjective	MOS
[86]	September 2009	Video	Objective	PSNR, VQM, SSIM
[87]	October 2009	Video	Network Planning Models	Delay, Jitter, Throughput
[88]	June 2011	Audio	Subjective, Network Planning Models	MOS, Delay, Packet Loss
[44]	October 2009	Audio	Objective	PESQ, AdmPESQ

3.4.1 Video

3.4.1.1 Subjective

The proposed schemes [85] provide a finer way to allocate resources for video streams. The performance improvement on them can reflect enhanced level of satisfaction for end users. QoE assessment is further applied to control on-going best-effort background traffic. The proposed scheme halts one background connection per second until the desired MOS value for video traffic is reached. If the condition of the system is good then one of the halted connections is squeezed back to the system

For the simulation a WLAN was used, with the BS operating on the default DCF and the background traffic assumed to be Constant Bit Rate. The halting and resuming are applied only on background traffic, so the video connections can have convenient quality improvement without annoying service up and down. The number of video client is 6 and of CBR traffic 12. The quality is quantified by MOS value and goodput of video streams and the supporting number of CBR connections is used as the reference of network throughput.

Running the simulations the system without the proposed enhancement, it is obvious that input rate varies significantly over time. When the system comes to its saturation point, the video quality perceived is really bad. On the second scenario the degradation halt-control is implemented and as we can see from the results 5 CBR connections were halted in the first seconds of the simulation, having as impact the drastically increase on mean and minimum MOS on the system indicating perfect quality video afterwards. But the perfect quality on video traffic indicates that the CBR connections that were halted were more than necessary. So on the third scenario the resuming of background traffic is demonstrated with and without the degradation-halt control. According to the results without the enhancement it is feasible to have more that 6 active background connection throughout the duration of the simulation by having great video traffic quality and with the degradation-halt control more that 9 with acceptable sacrifice of MOS value.

3.4.1.2 Objective

The increasing demand of multimedia applications requires a new behavior of routing protocols for Wireless Mess Networks (WMP). It is necessary to support the minimum requirements for QoS and QoE. In this work [86] Optimized Link State Routing protocol Dynamic Choice (OLSR-DC) is proposed and analyzed. The simulations were performed to demonstrate the performance of OLSR-DC compared against original OLSR and its Expected Transmission Count (ETX) and Minimum Delay (MD) versions considering different performance evaluation metrics and the user perspective for the received video quality.

OLSR protocol is an adaptation of the traditional link-state algorithm for ad-hoc networks. It is a proactive protocol, which uses a routing table obtained through the exchange of messages between nodes about the network and uses the hop count as metric for routing decisions. The limited number of control packets sent by OLSR, makes it suitable for WMP. Restrictions regarding the packet loss rate, delay, jitter and bandwidth are unable to be guaranteed through the selection of routes that, despite of having a low number of hops, can be unstable, so this is the reason that were implemented ETX, MD and finally the DC version that is proposed.

The analysis of the performance of the four different protocols was made in a scenario where data, audio and video flows were sharing the same link. Twenty simulations were performed using different seeds for each protocol. According to network performance metrics OLSR-MD and OLSR-DC achieved the best results by having more or less the same outcomes. By observing VQM, SSIM and PSNR average values these two protocols' video quality is classified as good. OLSR-ETX results were average and finally the legacy OLSR had the worst results of all the protocols. By analyzing further these results, it is obvious that OLSR-DC protocol is the one that is able to better answer the needs of both types of traffic, multimedia and data. However, OLSR-DC has some weak points such as higher memory consumption (increase in routing table) and a longer time for package delivery (determining whether a packet is TCP or UDP).

3.4.1.3 Network Planning Models

Authors in [87] present a QoE management system for wired and wireless IPTV devices to guaranty enough QoE in IPTV service. The system calculates the user's QoE, according to parameters as video quality, zapping time and synchronization time, and proposes the user to roam to another network, which provides better QoE. The system allows providing ubiquity to multi-network devices.

An Internet Service Provider (ISP) has been emulated to perform the experiments that combine a network infrastructure of layer 3 and layer 2 Gigabit and Fast Ethernet switches with an 802.11b/g wireless network. The evaluate parameters have been the delay, the jitter, the bandwidth the packet loss and the zapping time. Regarding the delay, the difference between the mean value for Ethernet and wireless is quite small (8.57 and 9.12 respectively) and is kept in low levels. The jitter measured is higher when IEEE 802.11b/g access network is used. On the other hand with FastEthernet the results are really better and with less variances. According to Bandwidth tests, both networks have the almost the same high average value, but when the user is zapping the bandwidth decreases dramatically. IEEE 802.11 has almost five times more packet loss that IEEE 802.3 and finally regarding to zapping time test, Ethernet has slightly better results than wireless network.

Taking into account these measurements, a formula is proposed that is based on the user's QoE parameters. It allows the network to carry out the appropriate operations. The system calculates some QoE parameters, separated on the network side and the user side, and when it is necessary the users chooses between to change the network parameters, to roam to another access network when the network that is delivering IPTV doesn't have enough resources or to switch to a backup switch or router provided by the ISP.

3.4.2 Audio

3.4.2.1 Subjective - Network Planning Models

[88] examines the behavior of IEEE 802.11s wireless mesh networks considering Nakagami-m fading channels. The effect of fading on the performance of VoIP services in terms of E-Model is investigated. On the simulations, a value of 20 dB SNR is required by severe fading channels when compared to non-fading channels.

Nakagami distribution is selected to examine the performance of VoIP in IEEE 802.11s WMNs under fading effects, because can model different fading channels and has a better fit for experimental data. Nakagami-m model is used as a distribution to represent several distributions of distinct properties simply by changing the m parameter.

During the simulation a homogeneous 3x3 mesh grid topology is used and five different values for m parameter. The tests were made with bidirectional UDP simulations, starting one after the other. 6 and 9 channels were selected to show the effects of delay and packet drop ratio. As it is shown in the results, when SNR value is increased, delay value is decreased for all values of m. Also increasing UDP packet size and VoIP channel number slightly increases the packet drop ratios, thus degrading the system performance. According to MOS values, it is obvious that increasing SNR and m improves the VoIP quality. Low SNR values and severe fading channels cause QoE performance problems in VoIP services. An acceptable voice quality in sever fading channels, is gotten when m>=1 and SNR should be increased to 20 to achieve the same QoE level obtained by an Additive white Gaussian noise (AWGN) channel.

3.4.2.2 Objective

This work [44] studies QoE assessment schemes for quality evaluation of voice calls in Next Generation Networks, by focusing and analyzing E-Model and PESQ. A new QoE metric is proposed, named AdmPESQ, to overcome the limitations on different delays and packet loss, which was not provided by current metrics. Finally a performance evaluation was carried out, based on simulation experiments, to show the benefits on quality level of VoIP services, as well as to enhance pricing schemes.

AdmPESQ is a full reference metrics implemented at end-hosts to produce a final score. It combines important characteristics of both E-Model and PESQ as the impact of the delay and the selective packet loss, which was a limitation on previous proposed metrics. So with AdmPESQ, only one metric is needed to characterize the VoIP quality and can be used by the providers as a manner to optimize network management operations, detect network impairments and define QoE-based pricing schemes.

For the performance evaluation, three different scenarios were used, by varying the end-to-end delay, the packet loss and the load. A wired and wireless network was used of 100 Mb/s and 11 Mb/s respectively. The first one hosted the source and the second the wireless receiver. From delay variation scenario (no loss), we can see that between the ranges of 100ms and 500ms E-Model has a continuous decrease and PESQ, that doesn’t take into account this parameter, has the same value in each simulation. The proposed scheme has similar results with PESQ for low delay values and more than 60% of difference when delay is 500ms and with E-model their variation is similar.

According to selective packet loss variation, it is obvious from the results, that E-Model is not affected by packet loss. On the other hand PESQ has more accurate results and on the scenario where the loss probability is high in period of speeches shows the worst results. AdmPESQ has similar results with PESQ and a significant difference on E-Model presenting a more accurate assessment.

Finally on load variation scenario, both E-Model and PESQ show a degradation when the load increases, but AdmPESQ has more accurate results because the proposed metric combines both the delay and selective packet loss variation that the other two metrics don’t take into account.

3.5 Cellular Networks

Table 12 – Cellular Networks and QoE association

File	Publication Date	Type of Streaming	Type of Metrics	Metrics
[89]	August 2006	Video	Subjective, Objective	MOS, PSNR, MSE, Blur
[90]	January 2011	Video	Objective	SSIM
[91]	March 2010	Video	Network Planning Models	Jitter, Packet Loss
[92]	December 2010	Video	Network Planning Models	Throughput
[93]	April 2010	Audio	Subjective	MOS

3.5.1 Video

3.5.1.1 Subjective - Objective

A QoE model for mobile multimedia services is proposed in this work [89]. Both subjective and objective artifacts, relevant to users quality were considered. According to QoE modeling for 3G streaming services, a billing policy decision is enabled for the streamed content based on post QoE prediction carried out on session logs.

The streaming process is simulated under a W-CDMA network, by using a 3GPP full motion video clip as a test material in four different coding schemes. Tests were carried out for each of the coding profile for 32 sets of impairments during four voting sessions. The audio parameters were not considered because the policy addressed in this paper are relevant with high-motion items in which the video quality is considered as dominant.

A subjective and an objective data analysis were performed by subjective metric MOS and objective PSNR, MSE and Blurriness. With confidence interval of 95%, we can see that most of our samples are categorized between 2 and 4 MOS values. For the pay/not pay policy that is addressed, we have the scales 3-5 and 0-2 respectively. Taking into account the objective metrics, it is obvious that the most sensitive quality metric is Blurriness, but despite the big difference between the experiments for every coding scheme values, the subjective perceptions of the frames are only minor.

3.5.1.2 Objective

[90] proposes a QoE based handover method to maintain experienced quality for a video streaming service. A mobile station that has a dual mode interface needs to transit dynamically from WLAN to an alternative access network (3G) until it finds again a WLAN network with an acceptable RSSI. By this method of handover user does not experience video quality deterioration.

The algorithm that is proposed requires an periodically calculation of the RSSI of the network when it is decided, the 3G network to be used and when the RSSI value is acceptable again, a handover back to WLAN takes place because of the cost of 3G. When the RSSI is unacceptable again the same procedure is followed until a new WLAN that fulfills the requirements is found.

For the experiments, the quality metric SSIM is used simulating two different scenarios. For the first one, the proposed algorithm is not used so the handover between the two WLANs takes place without a supporting 3G network to be used. On the second scenario, the algorithm is implemented on the device. As we can see from the relation between RSSI and SSIM, we have an acceptable QoE when the signal value is better that -80 dBm. Without the proposed algorithm when the handover from one WLAN to the other takes place, we have a significant degradation on quality (from 0.9 that is the acceptable value to 0.3). However, a mobile station can stably receive video streaming by using the proposed scheme. Also the mean SSIM is increased from 0.916 to 0.964 between the two scenarios.

3.5.1.3 Network Planning Models

This paper [91] mainly focuses on the measurement of QoE. The authors try to integrate both objective and subjective measurement approach, related to the use of a mobile video-streaming application in a mobile semi-natural setting. Six different usage contexts were defined: home indoor/outdoor, travelling by train and by bus/outdoor and work indoor/outdoor. The users could decide when they wanted to watch the clip, as long as they respected these usage contexts.

Experiments were performed over a UMTS network by streaming a video from a Darwin Streaming Server over a HTC mobile device. 18 people participated on the experiments and before they were asked to answer a questionnaire about a good experience with a mobile application, their expectations and a list of possible thresholds. Finally they were asked to see a video clip in six different contexts during a normal day and answer some questions about the emotions, current physical and social context and the quality they perceived by using 5/10-point quantitative scales. Because HTC mobile device doesn't support packet capturing, the capture was done on server side and the two important network parameters that were investigated were packet loss and jitter.

Analyzing the results, it is obvious that there is an association between QoE and objective parameters. Also spatial and temporal quality QoE aspects correlate with the objective parameters, since these were also created from technical-quality related question items. By removing the non-relevant objective parameters, the spatial quality QoE aspect is affected only by the audio packet jitter and the RSSI and temporal Quality by video packet jitter and video packet loss. Finally by using 1-way analysis of variation (ANOVA) it is clear that users found the mean quality of the sound better indoor than outdoor and their focus was increased when the watched the video streaming alone.

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In this paper [92] a novel framework for extending utility-based QoS to QoE in wireless networks is introduced. The framework that is proposed, allows users to dynamically and asynchronously express their satisfaction, with respect to the instantaneous experience of their service performance. A network utility maximization (NUM) theory is proposed, that provides the means for reflecting in a normalized and transparent way various services' performance prerequisites, various users' degree of satisfaction and different types of networks' diverse resources, under common utility-based optimization problems.

The users express their preferences from a GUI about the quality of the service. To prevent users' selfish behavior the consequence of users' actions is a change in pricing policy. Following the acceptance of a user's request, the network will allocate its available resources in accordance to the outcome of the corresponding NUM. For the scenario that was simulated, a CDMA cellular network was used that was serving 18 users. For the simulation a user requests for increased QoS in charge of increased cost and two users request degradation in their service quality.

As illustrated via the results all users' preferences are reflected and fulfilled in less than 1000 timeslots (0.62 sec) and after this period users' achieved goodput remains at the desired levels. Because of the NUM framework the resources of the network are reallocated in every change that it has, either on the existing users or are given to the new one that may enter the system. As we can observe at the results, at the third period of the experiment (i.e. after the changes on the system has taken effect) the resources are reallocated and all users enjoy higher values of achieved goodput, revealing the benefits of pricing aware QoE. The total goodput of the system is increased and even when both station request for reduces resources there is only one third reduction on the goodput instead of the value that corresponds to the reduction requests of the two users. Finally, considering a linear pricing scheme, cost-aware QoE enabled behavior would lead to increased profit.

3.5.2 Audio

3.5.2.1 Subjective

A model is proposed in this paper [93] that allocates the network resources to users by trying to decrease overqualified QoE of some users and space and redistribute these resources to users with under-graduated QoE. This is achieved by changing the level of source coding and modulation. On the first situation, higher level of source coding will improve QoE, more radio resource will be consumed and thus degradation of system capacity will be caused. On the other hand, lower level of modulation and coding can ensure the data delivery, but will lead to more consumption of radio resource.

In Adaptive Control Method that is proposed, the source and channel coding adaptation is designed to balance the radio resource according to users' QoE, in which the users with higher QoE will spare some radio resource via Adaptive Multi-Rate degradation or Modulation and Coding Scheme elevation and thereby users with lower QoE can consume more radio resource to improve their QoE via AMR elevation or MCS degradation, or the system can utilize the saved radio resource to increase the capacity. For the experiments were used two different networks, a UMTS with initial AMR of full rate and a GSM with half rate.

With the proposed scheme all the MOS values are kept between 4.0 and 4.1 so all the users to be satisfied from the service they receives. Two scenarios were simulated. The first one had just the AMR enabled and the second one had both AMC and AMR. In both scenarios the satisfaction ratio of users for UMTS network is always in 100%. By using just AMR on our system we have an improvement of 37% in UMTS network's capacity, 2.8% in GSM network's capacity and 16% on ration of satisfied users on the GSM network. On the other hand, using AMC and AMR at the same time we have an increase on system's capacity of 45.3% and 9.3% in both networks respectively and almost 20% more satisfied users in GSM network which is visible more than just using AMR adaptation. The reason about this difference is that the systems can only spare resource from AMR degradation. The spared resource can satisfy AMR improvement but cannot meet the follow-up request of system capacity increase.

CHAPTER 4

Challenges and Future Work

4.1 Introduction

A service must overcome several obstacles to a successful launch and wide use. Also the provisioning of differentiated end-to-end service quality in networks faces a large number of challenges. We can find many challenges in different part of a service. On the stage of the thinking a new service, on the stage of the implementation, or even on the stage of how to promote this service in the market and make it widely used by the costumers. On the field of the QoE in [13], the challenges have been roughly classified into four permeating groups:

- *Scientific – theoretical* related to abstract results and assumptions (in contrast to other three groups that are of practical nature).
- *Technical* dealing with the engineering side of the application of the above.
- *Economical* focused on the QoE market aspects.
- *Legal* encompassing a broad social context of QoE provisioning.

Because of the specialization of the authors and most of the readers of this thesis, the interest will be more centralized to those of the scientific and technical character. But we can't exclude the importance of the last two categories, so we will briefly present them too. Some of the future work of the works, that were previously presented, will also be introduced in this chapter. Also a consternated table is presented (Table 13) for someone that needs a glimpse on the challenges of QoE.

4.2 Scientific – Theoretical Challenges

Today, nobody questions the necessity of QoE, but a good question that we can do is if it's *at all achievable*. The direction of the technology is already driven in packet-switching networks, even when it is possible to provide QoE traded of with high cost in circuit-switched networks, and also IP succeeded because it used to carry traffic associated with services for which best effort operation was sufficient. Also, nowadays there is a need to provide more demanding services and QoE should be achieved without assuming that the resources are always over-provisioned.

Another big part of the today's networks is the statistical multiplexing that is used. QoE designers base their decisions on the *presence of uncertainties*, which make the analytical models created, very complex. Another big deal is that QoE – characteristics are *non – linear*. As a result, the use of standard optimization techniques associated with linear programming is much more difficult. The *scalability* problem is related to it: an attempt to deal with a large number of clients or traffic class types can make the operation impossible.

Finally, another challenge for QoE is the relationship that it has with *risk.* So far, the issue of quantifying the risk of not providing clients with an appropriate level of quality has been somewhat neglected. Nevertheless, with the increased role of the Internet in the public domain, a high level of risk-awareness is necessary.

In [84], authors are thinking to extend their work through the actual prioritization scheme in order to obtain a faster response to the changes that might appear in VoIP delivery. Also they will try to test the proposed scheme with different propagation models. Also in [83] a more challenging problem of scheduling will be considered, at the same time both the uplink and downlink directions using nonlinear controllers.

AdmPESQ will be evaluated in an experimental network and with subjective QoE tests [44]. Also more versions of OLSR will be analyzed in [86], as well as other pro-active, reactive and hybrid protocols. Authors in [79] believe that an integration of a multi-path routing protocol will improve their scheme and also to consider important parameters such as link quality for different neighbors and the route length.

4.3 Technical Challenges

All the services imply at least one wireless link between the source, such as a streaming server for video traffic, and the destination, such as a mobile terminal. Therefore, most of the technical challenges are related to the *wireless link*. The wireless link is vulnerable to physical factors. Even small reflectors and obstacles can degrade the signal quality and cause burst packet losses. For not having degradation in a service, all the devices should be configured and react adaptively to the wireless link's varying conditions.

All the information that is sent in the network passes through several autonomous systems and many different technologies. Also even all the technologies are standardized, there are many differences between devices of different manufactures so it is really difficult to provide a proper service. One of the challenges in this section is the *service coverage,* meaning that a service must be accessible anytime and anywhere. It is clear that it is impossible to deploy a wireless network that covers a geographic area without dead spots. So, *vertical handover* can solve our problem by interconnecting different wireless protocols and provide a service that a device will move across different wireless networks without any significant performance degradation.

Another issue in this field is the *security problems* that arise. QoE provisioning in a multiprovider environment makes it unavoidable that some information on assets and current network state will be revealed.

Also it is not clear from time to time, which *part of the network initializes* the procedures of QoE establishment. Should the required quality level be selected by the network side or suggested by the user? Or even if it should be forced by a client – provider access technique or by a service type. And if the client is finally suggests this level there is the issue if the client will be able to force this level of QoE. Or the *granularity* of the QoE, meaning at which level of the multilayer networking technologies it is provided.

Finally, there is the problem of QoE *monitoring*. How to organize the management plane so that is scalable, and what tools should be provided to a client if it is also responsible for the quality level tracking?

A technical challenge can be found in [85], in which the authors will try to implement the proposed scheme to a different architecture and test it. Because delay is an important issue in wireless networks, they are thinking to decrease the delay and the zapping time [87]. In [80], authors believe that their work can be used as an IP multimedia subsystem to achieve better QoE for the multimedia devices, so they try to improve their IPTV client, in order to achieve lower delays and include other network technologies.

4.4 Economic Challenges

All the above mentioned things, cannot be omitted in the economic context either. Having a free market for QoE, means that we will have many operators competing each other and of course many *market issues* will be address. Some of the problems that we have to be dealing with is how to share costs between different operators passed by inter-domain connections, how QoE can be sold, how to price the clients etc. All these are related to *business models*. Moreover some more questions may arise. To whom should a client pay for the QoE? To the ISP or to the service provider? And who should receive a higher QoE? Someone that already paid or everyone that needs it?

An example of these challenges of trying to address in the future can be found in [92], in which the authors will try to find proficient mechanisms towards enabling dynamic pricing, based on factors as users' type of service, billing policies or other economical factors.

4.5 Legal Challenges

As a last category of challenges we can find the legal ones. The main challenge here is the net(work) neutrality (NN). Due to this, QoE is treated as a public good, like the Internet. There are a lot of supporters of NN, who believe that QoE must be sold as a part of a service, and not separately (double selling) as an additional service supplementing the network connectivity.

Also due to QoE, there are a lot of *violations of agreements* related to the communication service taken into account. An agreed quality is advantageous to operators because it introduces a sort of risk sharing between carriers and clients: the former no longer need to provide every customer with the highest level of QoE because the agreed one is lower. But a practical problem that arises is that is difficult to prove poor quality. However, it is difficult to imagine that this is tough problem left without any *standardization* enforced by national and international regulators.

Table 13 – Challenges for QoE

Type	Challenge	Context
Scientific	Feasibility	General
	Uncertainty	Decision Making, Modeling, Design
	Complexity	Modeling, Design
	Non-linearity	Design
	Scalability	Design, Operation
	Relation to risk	Operation
Technical	Heterogeneity	Control, Transfer, Accounting
	Security	Inter-networking, Confidentiality
	QoE signaling	Network control
	Simplification	Cooperation with clients
	Initiation	Management, Cooperation with clients
	Granularity	Control, Cooperation with clients
	Assurance level	Design, Cooperation with clients
	Monitoring	Management, Cooperation with clients
Economic	Competition	Providers' interface, Portfolio construction
	QoE market issues	Cooperation with clients and peers
	Business models	Accounting between customers and networks as well as service providers
Legal	Net neutrality	Public domain, Cooperation with clients and peers
	Double selling	Cooperation with clients
	Violation responsibility	Public domain, Cooperation with clients
	Standardization	Public domain

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