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FACULTY OF ELECTRICAL ENGINEERING AND COMMUNICATION

DEPARTMENT OF TELECOMMUNICATIONS

SIMULACE BEZDRÁTOVÝCH A MOBILNÍCH UMTS SÍTÍ S POUŽITÍM QOS

WIRELESS AND MOBILE UMTS NETWORKS SIMULATION USING QOS

DIPLOMOVÁ PRÁCE

MASTER'S THESIS

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NÁZEV TÉMATU:

Simulace bezdrátových a mobilních UMTS sítí s použitím QoS

POKYNY PRO VYPRACOVÁNÍ:

Popište problematiku a možnosti zajištení kvality služeb QoS v mobilních UMTS a bezdrátových sítích. Podrobne popište standard 802.11e. Popište vrstvový model UMTS a mechanizmy, jakými je v systému UMTS implementována a zajišťována kvalita služeb (QoS). Vyberte si některý mechanizmus a podrobně jej rozeberte. V simulačním prostředí Opnet Modeler navrhněte patřičné modely pro oba typy sítí. Vytvořte a nakonfigurujte terminály i protejší uzly pro různé typy služeb z hlediska přenosu dat citlivých na zpoždění (IP telefony, přenos videa, hlasu a multimédií obecně). Implementujte a zaručte kvalitu služeb (QoS) po celé trase spojení. Nasimulujte různé situace provozu a pohybu uživatelů. Porovnejte a zhodnoťte kvalitativní parametry jednotlivých služeb. Získané výsledky prezentujte formou grafů a tabulek. Na základě získaných znalostí a zkušeností vytvořte zadání a ukázkové řešení laboratorního úkolu.

DOPORUČENÁ LITERATURA:

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Poďakovanie

Touto cestou by som chcel poďakovať vedúcemu diplomovej práce, Ing. Michalovi Vymazalovi. Rovnako tak aj svojmu mentorovi na UPV Pla Boscà Vicent Josep, za motiváciu i pripomienky a priebežné konzultácie mojej práce. Ďalej rodičom za všetko úsilie a snahu, no a pochopitelne bratovi za jeho bezmedznú podporu a nekonečnú pomoc, JUDr. Branislavovi Zapletalovi.

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podpis autora

Abstrakt

Diplomová práce je psána v angličtině a věnuje se standardu IEEE 802.11e, obsahujícím podporu kvality služeb a taky pojednává o kvalitě služeb QoS (Quality of Service) v systému UMTS. Rozebrány jsou hlavní teoretické pilíře daných mechanizmů a praktická část se zabývá simulacemi síťových modelů.

Standard 802.11e objasňuje propojení vrstev na fyzické úrovni a zmiňuje metody přístupu k přenosovému médiu DCF, PCF, HCF, EDCA, HCCA. Dále přístupové kategorie a také odlišnosti v MAC podvrstvě, je rozebrána problematika při přenosu prioritních dat na základě identifikátorů. Vzpomenuta je taky struktura formátu rámce a techniky s rozprostřeným spektrem. Časové limity pro doručení prioritních dat a požadavky kladené na tyto data je možné snadno srovnat v tabulkách.

Problematika kvality služeb je velice komplexní záležitost, nicméně jsou rozebrány základní klíčové parametry jako koncové zpoždění, jitter, zahazování dat, propustnost, velikost front a hodnota MOS. Zmíněny jsou i mechanizmy integrovaných RSVP a diferencovaných služeb pro zajištění QoS.

U systému třetí generace UMTS je objasněna architektura a společná kooperace se systémem GSM. Vzájemné propojení obou sítí je zřejmé z obrázků. Zvýšený zájem je věnován vrstvovému modelu a funkcím RRM pro zajištění QoS. Objasněny jsou různé druhy předávání hovorů, tzv. Handover Control a funkce Admission Control. Rozebrány jsou jednotlivé třídy provozu Conversational, Streaming, Interactive a Background.

Praktická část se odehrává v simulačním prostředí programu OPNET Modeler. Byly vytvořeny dva modely s různými scénáři pro srovnání zajištění QoS. Model objasňující princip standardu IEEE 802.11e obsahuje dvě bezdrátové sítě s přístupovými body a stanicemi, na kterých jsou sledovány simulace s různým zatížením přenášených dat. Pro porovnání výsledků je zkoumán rozdíl při použití metody HCF v síti s podporou QoS. Model UMTS sítě obsahuje základnové stanice Node B s možností vysílání do tří sektorů. Mobilní účastník pohybující se po trajektorii představuje princip funkce Softer Handover. Konfrontace mezi scénáři je zastoupena kompresí záhlaví pomocí funkce PDCP a rozlišení Type of Service.

Obzvláště je sledováno chování prioritních dat hlasu a videa u obou sítí. Naměřená data zastupují grafy a průběhy výsledních charakteristik. Analýza diskutuje odlišnosti u sítě bez podpory a s podporou kvality služeb. Rozdíly jsou porovnány a vyhodnoceny dle metodiky QoS. Součástí práce je i řešení problémů při návrhu UMTS sítě a podány jsou taky užitečné tipy a návrhy na jejich odstranění. Projekty v simulačním softwaru jsou popsány dle postupu vyhotovení, avšak hloubka podrobností je potlačena. Detaily nejsou rozebírány, protože se očekává pokročilá znalost mechanizmů a jistá dávka zkušeností s programem.

Klíčová slova

Standard IEEE 802.11e, QoS, UMTS, OPNET Modeler

Abstract

The thesis is written in English and focuses on IEEE 802.11e standard, containing support for Quality of Service (QoS) and also discusses QoS in the UMTS system. It analyzes the main theoretical pillars of the mechanisms while the practical part deals with simulation of the network models.

The 802.11e standard clatifies interconnection among layers on physical level and refers to the access to media DCF, PCF, HCF, EDCA, HCCA, and furthermore, the access category and also differences in the MAC sublayer. It is analysed problems in the transmission of data based on identifiers priority. The structure and format of the framework and techniques of spread spectrum are also discussed. Time limits for delivery of data priority and the requirements for these data can easily be compared in the tables.

The area of quality of service is a very complex issue, and the thesis also analyzes the basic parameters such as end-to-end delay, jitter, dropping data, throughput, queue size and value of the MOS. The mechanisms of integrated (RSVP) and differentiated services to ensure QoS are also mentioned.

In the case of third-generation UMTS architecture is illustrated a mutual cooperation with the GSM system. Interconnection between networks is evident from the pictures. A special attention is focused on layer model and the RRM functions to ensure QoS. The mechanisms of Handover Control and Admission Control are clarified too. It analyzes different traffic classes, such as Conversational, Streaming, Interactive and Background.

The practical part takes place in the software OPNET Modeler programme. The author developed two models with different scenarios for comparison to QoS support. The wireless model explaining the principle of the 802.11e standard includes two wireless network access points and stations, which are monitored by the simulation with different data transmitted loads. For comparison of the results is examined using the difference method HCF in the network with QoS support. The model of the UMTS network includes base stations Node Bs, with the possibility of broadcasting into three sectors. The mobile subscribers moving on a trajectory are to show the principle functions of the Softer Handover. Confrontation between scenarios is represented by using header compression by PDCP and distinguishing the Type of Service.

In particular, it is examined the behavior of priority voice and video data streams in both networks. The measured data are demonstrated by graphs and curves of result characteristics. The analysis discusses the differences in the network without the QoS support and with promotion of quality of services. Differences are compared and evaluated by the methodology of QoS. The work also includes problem solving in the design of the UMTS model and simultaneously gives tips and suggestions for overcoming them. The projects in the simulation software are described according to the procedure of execution, but the depth of details is suppressed. Details are not discussed in this work because some level of advanced knowledge of the mechanisms and a certain amount of experience are necessary.

Keywords

IEEE 802.11e standard, Quality of Service, UMTS, software OPNET Modeler

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Introduction

Wireless networks have become a popular part of life for everyone, because they provide a convenient way to connect to the local network or to the Internet. Undoubtedly, the most widely used family of wireless networks in the world are known as 802.11 standardization institute IEEE (Institute of Electrical and Electronics Engineers). This group includes a well-known standard 802.11e, which emerged after year 2005, and which has the quality of service and support based on its priority preference data and backward compatible with all older accessories.

A decade later GSM brought us to the early stages of the third-generation mobile communications system: the Universal Mobile Telecommunications System (UMTS). The first networks began operations and a new generation of fancy mobile phones appeared by the end of October 2004. The UMTS networks represent a completely new, high bit-rate radio technology Wideband Code Division Multiple Access (WCDMA) for wide area of use. One of the key advantages of UMTS mobile computing and communications devices is the ability to deliver information to users at almost anytime and anywhere. In the UMTS the mobile phone has become to be regarded as a personal trusted device, a life management tool for work and leisure. Among new possibilities for communication, entertainment and business are new kinds of rich call and multimedia data services, fuelled by mobility and personalisation of users and their terminals.

The aim of this thesis is simulation and analysis of an implementation of the quality of service in wireless and mobile UMTS networks. The main intention is put into ensuring and providing the required quality of service (QoS). It aims to provide an overview of these types of networks, especially the systems architecture providing with helpful instructions in the OPNET Modeler environment. It explains and discusses some uncertainties in the field and provides with useful guidance for understanding and awareness activities. The goal is to create an educational tool to deepen the knowledge of potentially interested person in this area.

The content of the master thesis is divided into two parts. The first part consists of three chapters: Chapter 1 discusses the 802.11e standard, deals with an overview of supplements, physical layer modulation and types of transmission, the MAC sublayer and the meaning of fields in the MAC layer. Chapter 2 describes the physical parameters of QoS. Chapter 3 introduces the UMTS technical and service architecture and the key system concept. In addition, the UMTS network is examined as a network for services. It addresses service realisation by describing QoS and giving some examples of services that can be brought by UMTS. The second, practical part deals with the software OPNET Modeler programme, giving the results obtained in the form of graphs in both models, and compares the simulated scenarios. Thanks to those obtained values, it presents the importance of using QoS mechanisms. The last but not least, it offers the frequent troubleshooting.

1 STANDARD 802.11e

Compared with classic metallic networks, wireless networks are less reliable and often behave unpredictably. This is usually caused by interference effects, poor environmental conditions and weather. This is because air is a wireless medium more difficult to coordinate and manage the network than copper lines. The original 802.11 standard also does not define the differences, or prioritizing. Therefore, cannot optimize the network for more efficient and reliable use of applications such as video or voice. This lack of hierarchy in the family following wireless standard 802.11e. The main character is defining the changes and improvements in the management of quality of service QoS. The most important improvement is the efficient use of bandwidth and reduce overhead. Then there is a reduction of response using packet prioritization, according to the type of transmission and distribution of resources as appropriate response.

This standard presents the architectural view, emphasizing the separation of the system into two major parts: the MAC of the data link layer and the PHY. These layers are intended to correspond closely to the lowest layers of the ISO/IEC basic reference model of Open Systems Interconnection (OSI). The layers and sublayers described in this standard are shown in Figure 1-1, taken from [9].

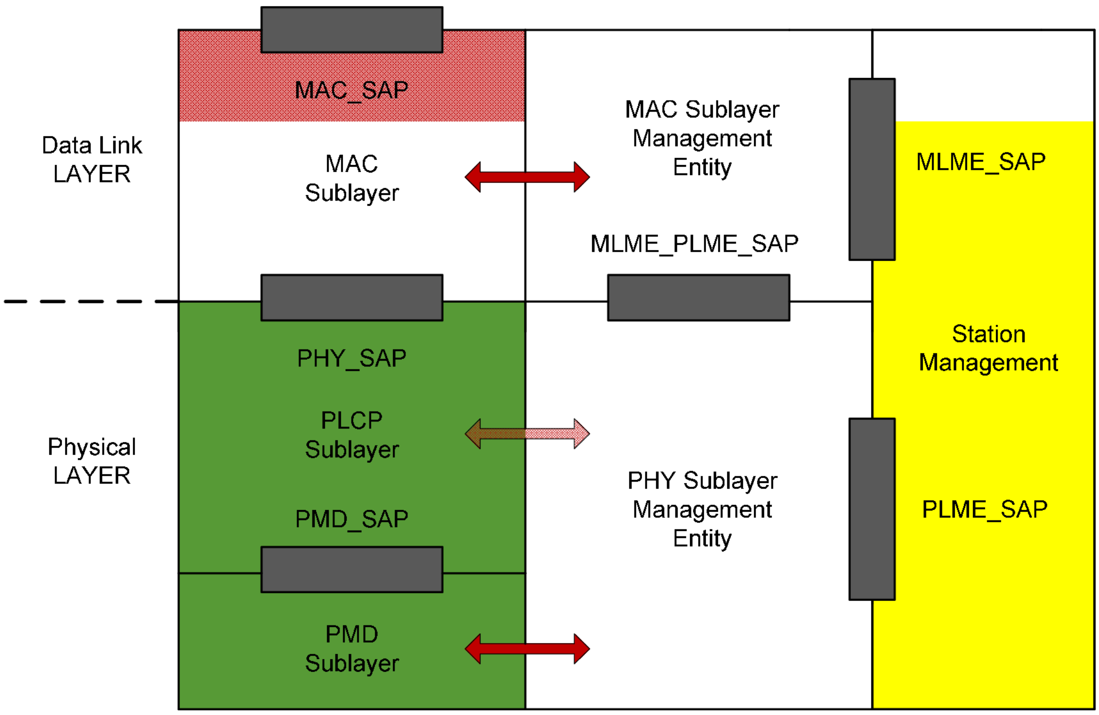


Figure 1-1 Reference model of the MAC and PHY layers

Standard 802.11 provides, in addition the names of active network elements. Access point that supports the management quality of service is called QAP (QoS Access Point) and the station with the support of QoS is QSTA (QoS Station). Also a basic set of BSS is renamed QBSS (QoS Basic Service Set). Starting with these changes, an 802.11e standard also introduced the new method of access to the media to improve the complex MAC layer.

1.1 Medium access control layer

Also know as MAC and deals with asynchronous data transfer, security services and MSDU ordering. This sublayer serve to control access to the media [1]. MAC sublayer separating the upper sublayer of the communication from physical layer. The primary purpose is to transfer data MSDU (MAC Service Data Units Aggregation) between objects of MAC sublayer.

The original protocols of 802.11 MAC layer have no means to distinguish different groups of data or resources, all are treated equally (one priority access to the channel). They can not therefore be guaranteed with the requirements for data bandwidth and delay. Then it could happen that the data traffic with low priority choke bitrate video and worsen the quality of its adoption.

The process of partitioning a MSDU or a MAC management protocol data unit (MMPDU) into smaller MAC level frames, MAC protocol data units (MPDUs), is called fragmentation. Fragmentation creates MPDUs smaller than the original MSDU or MMPDU length to increase reliability, by increasing the probability of successful transmission of the MSDU or MMPDU in cases where channel characteristics limit reception reliability for longer frames. Fragmentation is accomplished at each immediate transmitter. Only MPDUs with a unicast receiver address shall be fragmented. Broadcast or multicast frames shall not be fragmented even if their length exceeds a fragmentation threshold.

The main concern of the MAC sublayer is management and implementation of communication between wireless stations and the use of protocols that provide connection through a wireless medium [2]. Often it is viewing the brain as a whole wireless network, see Figure 1-2 reproduce the [9]. MAC sublayer uses a physical layer to perform tasks such as broadcasting, transmission and reception of frames.

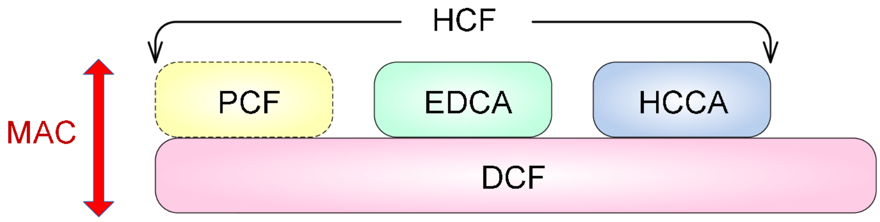


Figure 1-2 Structure of IEEE 802.11e MAC layer

The basic functions of MAC sublayer summarized the following list:

Scanning: 802.11 uses active and passive scanning. By the passive station scans the channels and looking for an access point (AP) with the best signal strength. AP periodically broadcasts beacon frames, which station receives and evaluates the signal strength. Beacon frame contains information about AP, which is the identifier of the wireless network Service Set Identifier (SSID), supported transmission rate, etc. Stations will use this information to estimate the available access points and decide which AP used. Active scanning is similar, but the station begins the process by sending *probe request* frame and all access points within range respond *probe response* frame. Active scanning allows you to receive immediate response from the AP, without waiting (100 ms) to broadcast the beacon[1] frame.

[1] Beacon frame to synchronize the wireless network, carries information about the SSID, the data rates. Furthermore, it indicates what type of services is set (ESS, BSS, IBSS), information security (WEP, WPA) and MAC address of access point BSSID.

Authentication: The process of identification, when 802.11 systems use 2 forms, which are *open system authentication* and *shared key authentication*.

Association: The following is the authentication. Station must associate with the AP before beginning to transmit data frames. Association is necessary for synchronization. Station sends *association request* frame contains the basic parameters and the access point responds by *association response* frame. When AP and station completes the process of association data transmission can begin.

CRC: Cyclic checksum is used to detect errors during transmission and storage. CRC is calculated before sending the frame and is then sent in along with the data.

WEP: Optional security. Station encrypts the body (not header) of each frame before transmission and receiving station decodes the key to its context. WEP data encryption method can now easily break through safer are using WPA or WPA2 encryption.

RTS / CTS: If the network used RTS/CTS prior to each transfer initiated towards RTS frame transmission from the station to the access point. AP answers CTS frame indicating that the station can start data transfer.

Power Save Mode: Mode with reduced consumption. Station receiving data and waking up periodically to receive beacon frame. Beacon frame station shall check on the AP waiting for data for that station. This mode allows to saving the energy.

Fragmentation: Allows stations to divide data packets into smaller frames. It can thus avoid the forwarding of large frames. The user can set the maximum length of a framework for the activated mode of fragmentation. Frames will not be longer than the set threshold. This is an optional operation.

1.2 Physical layer

Most PHY definitions contain three functional entities: the PMD function, the physical layer convergence function, and the layer management function. The PHY service is provided to the IEEE 802.11 wireless LAN MAC entity at the STA through a service access point (SAP). A set of primitives might also be defined to describe the interface between the physical layer convergence protocol sublayer and the PMD sublayer.

Different PHYs are defined as part of the IEEE 802.11 standard. Each PHY can consist of two protocol functions as follows:

- A physical layer convergence function, which adapts the capabilities of the physical medium dependent (PMD) system to the PHY service. This function is supported by the physical layer convergence procedure (PLCP), which defines a method of mapping the IEEE 802.11 MAC sublayer protocol data units (MPDUs) into a framing format suitable for sending and receiving user data and management information between two or more STAs using the associated PMD system.

- A PMD system, whose function defines the characteristics of, and method of transmitting and receiving data through, a wireless medium (WM) between two or more STAs.

Each PMD sublayer may require the definition of a unique PLCP. If the PMD sublayer already provides the defined PHY services, the physical layer convergence function might be null.

When transmitting a packet of data which comes from the application layer, headers has to be added for each layer, making the total transfer size larger, see Figure 1-3.

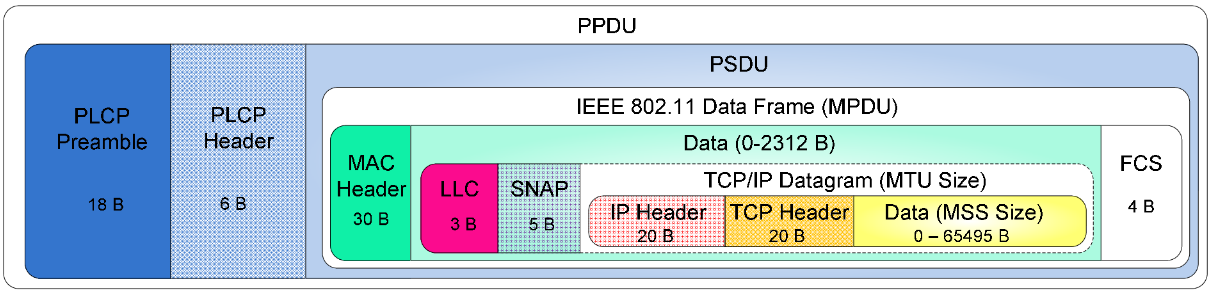


Figure 1-3 Total format frame of standard 802.11

This is an example of how headers are added. On top of the data comes the TCP header of 20 bytes (assuming the option fields are not used). Then comes the IP header of 20 bytes. Together they make the TCP/IP datagram. The size of this the same as the MTU, usually 1500 bytes to get through most router configurations. Then comes the SNAP header (used to encapsulate original Ethernet V2 ethertype values into an IEEE 802.2 frame) of 5 byte and the LLC (Logical Link Control) header of 3 bytes. This is in turn encapsulated by a MAC header of 30 byte and a trailing FCS of 4 byte. This is the MPDU or the PSDU part of the PPDU. Then we get a PLCP header of 6 byte and finally a PLCP preamble of 18 byte ("long preamble"). The preamble is used to signal that "here is a train of data coming" to the receiver. The 802.11b standard gives an option of reducing the size of the PLCP preamble to 9 bytes ("short preamble"), significantly increasing performance on the higher rates.

1.2.1 Frequency-Hopping spread spectrum

Carrier signal containing modulated data is transmitted to the selected frequency for a short period (400 ms), then jumps to another distant enough frequency and this cycle continues to repeat. Zone is divided into 75 or 79 channels each with a range of 1 MHz, the other band functions as a safeguard against interference from neighboring bands. Transmitting and receiving side knows in advance the exact sequence of hop. The advantage of this technology is a higher number of simultaneously operating equipment at 2.4 GHz and reaches bit rate up to 2 Mbps.

A conformant PMD implementation shall be able to select the carrier frequency from the full geographic-specific set of available carrier frequencies. Table 1-1 summarizes these frequencies for a number of geographic locations; values specified in GHz.

Lower limit	Upper limit	Regulatory range	Geography
2.402	2.480	2.400 – 2.4835	North America
2.402	2.480	2.400 – 2.4835	Europe (excluding Spain and France)
2.473	2.495	2.471 – 2.497	Japan
2.447	2.473	2.445 – 2.475	Spain
2.448	2.482	2.4465 – 2.4835	France

Table 1-1 Operating frequency range

The hopping sequence of an individual PMD entity is used to create a pseudorandom hopping pattern utilizing uniformly the designated frequency band. Sets of hopping sequences are used to co-locate multiple PMD entities in similar networks in the same geographic area and to enhance the overall efficiency and throughput capacity of each individual network.

1.2.2 Direct sequence spread spectrum

Technique DSSS operates on the principle that each bit to be transmitted is first replaced by a sequence of bits transmitted (modulated on a carrier signal) is the sequence itself to bits. DSSS divides the band 2.412 GHz – 2.484 GHz for 14 channels, where the width of each channel is 22MHz. However, the interval between the frequency is only 5 MHz so that adjacent channels overlap. Only 3 channels do not overlap at all. Transmission takes place so that the transceiver communicates in 1 channel. For example, at 802.11 for transmission speeds of 1 and 2 Mbps is each bit replaced by a 11 bit sequence duration, called Barker sequence or chip. The signal is distributed to more parts of the spectrum and is less susceptible to interference. Used because of greater transmission reliability and reaches bit rate up to 11 Mbps.

The following spreading sequence 11-chip Barker sequence shall be used as the PN code sequence:

$$+1, –1, +1, +1, –1, +1, +1, +1, –1, –1, –1$$

The leftmost chip shall be output first in time. The first chip shall be aligned at the start of a transmitted symbol.

1.2.3 Orthogonal frequency division multiplex

OFDM, originally used standard 802.11a in the 5GHz band. To 2.4 GHz band, this technology has introduced the 802.11g standard. This technique extends to transfer large portions of the spectrum and thus can achieve higher transmission speeds. Technology divided available spectrum into small parts, called subchannels which is transmitted to a separate carrier signals, the so-called subcarrier. The data can then be modulated to each subcarrier signal. Thus the total data to be transmitted are divided into sub-transmission channels. Given the intensity may be hampering of individual channels used less intensively than the channels showing the deterioration of transmission characteristics. Speeds up to 54 Mbps.

The OFDM subcarriers shall be modulated by using BPSK, QPSK, 16-QAM, or 64-QAM modulation, depending on the rate requested. The encoded and interleaved binary serial input data shall be divided into groups (1, 2, 4, or 6) bits and converted into complex numbers representing BPSK, QPSK, 16-QAM, or 64-QAM constellation points. The conversion shall be performed according to Gray-coded constellation mappings, illustrated in Figure 1-4 BPSK, QPSK and 16-QAM constellation bit encoding.

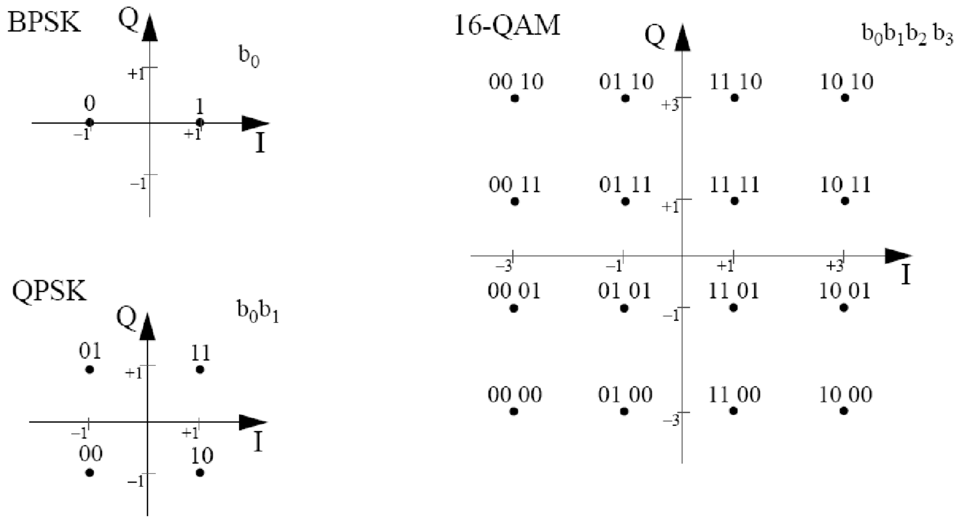


Figure 1-4 BPSK, QPSK and 16-QAM constellation bit encoding

1.3 Access to media

Transmitting the data is not just blasting out a bunch of signals on the air. A strict set of rules is governing the way a transmission should behave (CSMA/CA). First the sender has to wait a period of DIFS (Distributed Coordination Function InterFrame Space) time of 50 Usec (microseconds) before the channel is presumed clear of traffic. Then a data frame or a request to send frame can be sent. The receivers answer to this is with an ACK or Clear To Send, accordingly. The receiver has to wait a SIFS (Short InterFrame Space) time of 10 Usec before this reply is sent.

1.3.1 DCF (Distributed Coordination Function)

The principle scheme is based on the approach to prevent collisions. Before the first broadcast station listens and determines whether the other station does not transmit. Mandatory waiting period, called interval DCF Inter Frame Space (DIFS), each station chooses randomly in the range of values from zero to the size of contention window. There is a possibility of a collision if the broadcast channel is seeking more stations. Value window Contention Window (CW) is doubled with each collision. Station will start a service when the interval expires and a delayed broadcast media is released. Recipient data after receiving a packet waits for an interval of SIFS (Short IFS is less than DIFS) and then send the receipt of the packet. [8] Confirmation of income is important in the wireless data transmission, since the transmission error can occur for many reasons, e.g. because of lack of signal strength, collision data, exceeded the maximum value of the network load, etc.

1.3.2 PCF (Point Coordination Function)

Used for synchronous data transmission, but is rarely used. This is an optional mechanism for access to the media. The access point broadcasts at regular intervals within the type beacon, which communicates stations in the network specific parameters management and identification. Access point divides time between sending these frames into two parts:

• The period of the medium without a fight - *contention free*

• The period of competition for medium - *contention*

Station may to get on the basis of permission to guarantee broadcasting. At this time not to Station may be a call to get a permission to guaranteed broadcast. At this time, do not fight the media with any other station. PCF mode uses interval PIFS (PCF IFS) for the reporting interval without collision with the priority for the stations broadcasting. Interval PIFS is longer than SIFS in DCF mode.

1.3.3 HCF (Hybrid Coordination Function)

This new standard is a hybrid method for coordinating access to the medium HCF (Hybrid Coordination Function), which combines the functions of DCF and PCF methods with extended support for quality of service and new types of frames. This method works in two modes, wich are Enhanced Distribution Coordinate Access (EDCA) and HCF Controlled Channel Access (HCCA). These modes operate on the principles of controversy (*contention-based*) and interviews (*polling-based*). Standard also defines the concept of Transmission Opportunity (TXOP), which specifies the time during which the QSTA can to broadcast. In other words, once the station gets a right of access to the media, is assigned TXOP authorized. TXOP is characterized by the beginning of time and the maximum length of time, which is called TXOPlimit. Time period of this limit is set to access point QAP. [8]

1.3.4 EDCA (Enhanced Distributed Channel Access)

This mode includes a change in the media improving on the DCF method. The basis of the prioritization of different data streams. Defines four access opportunities, which are called access categories (AC). These are allocated to different types of transmission, which are specified for different services and applications. [8]

EDCA is based on the QoS stations competing for access to the media and provides them with a different approach to the distributed medium.

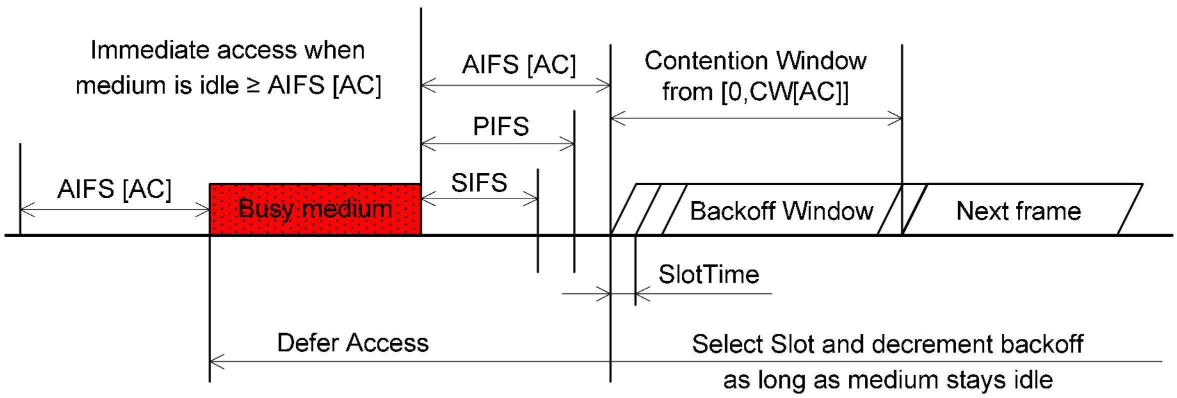


Figure 1-5 Method media access EDCA

From Figure 1-5 Method media access EDCA taken from [1] shows that it is allowed immediate access, when the medium is free, i.e. the interval is greater than or equal to the value of AIFS. If not, the medium is busy producing a suspension of access until further testing of channel occupancy. Selection of the time slot and decreasing backoff window is as long as the media is free. For the idea of bringing the issue 802.11a value of SlotTime is equal to 9 μs. [12]

Defined five different IFS to provide priority levels for access to the media. The following is an extract from the shortest to the longest, with the exception AIFS, it can take different values for different access categories. In Table 1-2 Types of interframe spacesTable 1-2 we can notice the different species [6, 7].

1.	**SIFS**	short IFS	16 μs
2.	**PIFS**	IFS for PCF	25 μs
3.	**DIFS**	IFS for DCF	34 μs
4.	**AIFS**	IFS for QoS selection	
5.	**EIFS**[2]	extended IFS	

Table 1-2 Types of interframe spaces

Different IFS are independent of the bit rate channel. IFS are defined as time gaps into the medium and in addition AIFS are all firmly determined by the attributes specified in the physical layer.

1.3.5 HCCA (HCF Controlled Channel Access)

This access method is defined by the 802.11e standard for parameterization QoS. HCCA method solves major problems PCF method, which resulted in this method to very poor performance management of service quality. HCCA presents a different method of transmission, called traffic streams (TS). This enables design more classes of algorithms for different types of applications. These terms are considered implementation dependent on HCCA and can be constantly improved. Another improvement is that the station QSTA must not send a packet, if before its completion should occur to the collision frame synchronization beacon. This really solves a fundamental problem of delays in beacon frames PCF methods. Other treatment methods of HCCA is use the timeout TXOPlimit to determine the length of time period in which it questioned station allowed to broadcast. QAP during beacon frame can transmit a number of pulses, wich are called controlled access periods (CAP). This can be done at any time after the detection of idle channel, but only for a set period, which is known as PIFS. If the interval is longer than AIFS, choose the QAP instead EDCA the HCCA method.

HCCA method is more flexible than the original PCF and on the ground that the original PCF can be applied only in uncontroversial time the contention-free period (CFP), while HCCA method can be initialized at any time during the duration of the whole beacon interval. PCF method in comparison with the new HCCA becomes unnecessary, but in the 802.11e standard remains optional options.

1.4 Access Categories

For each access category are assigned the various data streams according to the quality of service requirements for certain applications. This is a background application AC_BK, applications requiring maximum power (best effort) AC_BE, applications for voice AC_VO and applications for video AC_VI. Category AC_BK have the lowest priority and vice versa AC_VO greatest. Each frame can carry information about their priority. Its value corresponds

[2] EIFS unlike SIFS, PIFS and DIFS has a variable value and is only used when there has been an error in frame transmission. It is not used to control access onto the radio link.

to the user preference (UP) and the type of communication or application identifies the origin of each frame. There are a total of 8 levels of priorities.

User priority	IEEE 802.1d	Access Category	IEEE 802.1e
0	best effort (BE)	AC_BE	best effort
1	background (BK)	AC_BK	background
2	-	AC_BK	background
3	excellent effort (EE)	AC_BE	video
4	controlled load (CL)	AC_VI	video
5	video (VI)	AC_VI	video
6	voice (VO)	AC_VO	voice
7	network control (NC)	AC_VO	voice

Table 1-3 Types of access categories according to priority

Another theme in the issue of 802.11e is the allocation of priorities in the higher layers. Standard 802.11e does not define a method to allocate these priorities in higher layers than MAC layer. That should therefore take care application that creates a data flow. One solution would be that in future all applications compatible with this standard to use networks as possible. Second possibility would be to adaptively assign priority to the application layer. This would be undertaken on the basis of the size of packets, intervals between packets or the size of the data stream. But that would require major modifications to the higher layers. [13, 19] Each AC is an expanded version of DCF.

1.5 Architecture

Standard 802.11e in addition to the mentioned EDCA and HDCA includes DCF and PCF methods defined by the original 802.11. The reason is backward compatibility. Station management support services QSTA is capable of operating in the basic set of services to support QoS BSS i.e. QBSS, but is also backward compatible and thus can work in the core set of services that do not support the quality of the so-called non-QoS BSS (nQBSS).

It is possible that the workstation is able to associate with the access point do not support QoS so-called non-QoS AP (nQAP) and in the case of the QAP not available for the station. Apart from this station is nQSTA able to communicate with the QAP in QBSS so that it behaves as a classical station as defined by 802.11 and the other side and QAP for communication with the station does not use the framework to support QoS. HCF method used for central management the hybrid coordinator called hybrid controller (HC). It is located on the access point and its front activities is to work with EDCA method. Each method has a slightly different function. EDCA operates during periods of controversy also known as contention period (CP), while HC also operate even during periods of CFP. It can be seen as a control mechanism for EDCA and HCCA methods. [10]

1.6 Frame format

MAC frame format consists of a set of fields that occur in a fixed order. Figure 1-6 shows the general MAC frame format. The first three fields frame control, duration/ID, address 1 and last field FCS are minimal frame format and are present in each of the framework structure, including the reserved types and subtypes. Field address 2, address 3, sequence control, address 4, QoS control and frame body are present only within certain types and subtypes. Frame body field is variable size. The maximum frame body size is determined by the maximum MSDU size (2312 B) plus any overhead from security encapsulation. Each MAC frame contains the MAC header, body frame and FCS. Figure 1-6 and Figure 1-7 are taken from [5, 10].

Frame Control (2 B)	Duration /ID (2 B)	Address 1 (6 B)	Address 2 (6 B)	Address 3 (6 B)	Sequence Control (2 B)	Address 4 (6 B)	QoS Control (2 B)	Frame Body	FSC (4 B)

Figure 1-6 MAC frame format

MAC header consists of:

- *frame control* (2B) includes a frame type (management, control, data), subtype (RTS, CTS), indication of income distribution or broadcast in the system, type, subtype, protocol version, marking more fragments, the indications of re-sending the data, using cost-effective mode and indications using WEP encryption.

- *duration/station ID* (2B) where duration is the time to announce of filling the media and station ID is used to ensure energy saving features.

- *address fields* 1 to 4 contain 4 x 6B (source, destination and more by the value of the frame control).

- *sequence control* (2B) provides support for fragmentation.

The MAC frame of IEEE 802.11e standard differs from the IEEE 802.11 standard in that it is accompanied by support for QoS and correcting errors in support of all physical layer used in 802.11 networks except for ad-hoc networks.

Application Frame Type	Bit 0 – 3	Bit 4	Bit 5 – 6	Bit 7	Bit 8 – 15
QoS CF-Poll Frames sent by HC	TID (UP, TSID)	End of Service Period (EOSP)	ACK Policy	Reserved	TXOP limit (32 Usec)
QoS Data, QoS CF-ACK and QoS Data + CF-ACK frames sent by HC		EOSP			Reserved
QoS data type frames sent by non-AP QoSTAs		0			TXOP duration requested (32 Usec)
		1			Queue size (256 B)

Figure 1-7 QoS Control field

1.6.1 QoS subfield

For correct communication is important information about individual stations, if they work with the support of QoS or not. In other words, if are a QSTA or nQSTA. This information is transmitted in the header field of the frame control field. Name this information is QoS subfield. If the value is 0, it means that the station is a type nQSTA and if the value equals 1, the station supports QoS and therefore QSTA. [5, 11]

1.6.2 Traffic Identifier

The MAC layer assigned each framework into certain priorities. This quantity is defined by the traffic identifier (TID). Information is stored in the field TID Filed, which is included in the box QoS control field. On the basis of this value is defined by the user's priority, which is determined by the numbers 0-7. TID prioritization can be applied only if the station has got value in the QoS subfield set to 1. If it is set to 1, the connection between station and access point with support quality of service operate well, namely QAP-QSTA. If the value is 0, it means that the station does not reach the QAP and TID has no meaning.

Station broadcasts the frameworks with controversial prioritization and the access point handled with them as normal frames. It is similar like in the DCF method. Likewise, when there is a link between the nQSTA and QAP, access point assign the value 0 to each framework, which the station broadcasted, i.e. the lowest value. [10, 11]

1.6.3 Queue size

In addition is in the fieldbox QoS the value of the amount ordered to queue size field. This value defines the number of frames specific priorities, which the station has got included in the queue for transmission by the access categories.

1.6.4 Transmission opportunity

Station with the support of QoS can define the requirement for transmission of multiple frames during the TXOP[3] duration. Will do so by adjusting the value in the duration field for required period for the transmission of other frames. If it is used the HCCA method can QSTA this requirement to define by setting a field for the time required to transmit the TXOP duration requested field, which is located in the field of control of QoS. Subsequently, the QAP can to allocate the required TXOP value. In some cases, be determined less than the required value.

When using the EDCA method are values and parameters defined in the EDCA parameter set. It is regularly in the synchronization beacon frames sent out to the access point. The access point can also be those values depending on environmental conditions to adapt. Parameters are stored in fields AIFSN, TXOP limit, ECWmin and ECWmax. Station that the parameters adopted, will work with them and trying to gain access to the media. In the event that the access point can not set these parameters and distribute, the 802.11e standard defines the basic values of these parameters.

For each newly created EDCA parameters of the method, access point increases the value of a field that carries information on the current broadcast set of parameters and sends it in each frame. This field is called the EDCA parameter set update count field and is stored in the array of information on the promotion of QoS info field. This technique used by stations to always work with the most recent set of parameters and not disadvantaged in the allocation of access to media. [12]

[3] The term TXOP is used in wireless networks supporting the IEEE 802.11e Quality of Service standard. Used in both EDCA and HCCA modes of operation, the TXOP is a bounded time interval in which Stations supporting Quality of Service are permitted to transfer a series of frames. A TXOP is defined by a start time and a maximum duration.

2 QUALITY OF SERVICE

Methods for ensure the design and implementation the quality of services are a trend in development of computer networks. Comprehensive policy models, solutions, protocols and services reflecting the grade of quality of service in telecommunications area. It is collectively refer as QoS and is subject to continuous monitoring. [3] Quality of service was defined by ITU-T in recommendation E.800 as *the collective effect of service performance which determines the degree of satisfaction of a user of the service*. It's aim is to provide users the first class service with a defined quality. A major goal is to define a standard toolkit for providing session QoS end to end independent of access technology

Seeks to meet the QoS requirements used by applications and data management resources. For this purpose uses certain mechanisms that control access and use of wireless media. These methods are based on the fact that each application needs to work bandwidth, a certain level of latency and minimal packet error rate. For example, for voice over internet protocol (VoIP) is a substantial size of the latency, and therefore this application is given high priority for access to the media. On the other hand, the transmission of video is especially prioritized for bandwidth, so the size of response in this case is not crucial. The applications such as text communications (email, ICQ) are an advantage for the smallest error packets.

Ratings QoS is not represented by a single value, because its practical expression is complex. Determination of the quality of services is expensive and time-consuming process. The assessment must be impartial and universal. Followed subclauses described elementary parameters of QoS.

2.1 End-to-End delay

The time between sending a packet from the source and its delivery to the recipient. The packet delay time is influenced by many factors, such as coding, preparing packets for transmission over the media, broadcast packets, packets waiting in lines, transmission equipment, decoding the signal [8]. The highest one-way delay for voice should fluctuate below threshold 180 miliseconds, see Table 2-1. Delays over 250 ms is no longer continuosly ensured and the call has dead spots [9]. Communication is not affected, but incurred sections where one participant listens to one another and talk a while longer.

2.2 Delay variation

Packets in the network are delivered with some delay. This delay may vary depending on the current network load. The delay is variable and is also known as jitter. This phenomenon does not mind too much for data transmission, but for the transmission of video and voice it is a big problem. The maximum delay variation for voice should not exceed 30 ms [8]. But it depends on the current network load and for the network without the support of QoS is very difficult to assure so low jitter. By implementation of the QoS support for priority data result in eliminate the fluctuations to very low level . This phenomenon can also eliminate using the receiving buffer in the VoIP terminal equipment, but this leads to an overall delay in transmission.

2.3 Packet loss

Indicates how many packets arrived from the sending side to the recipient. Loss of packets can cause a lot of factors, such as congestion and network overload, buffer overflow at the network elements, excessive delays and subsequent dropping packet. If the video transmission is lost packets, may blur the picture or short outages. The voice of the loss reflected a short-term failure of communication. Sensitivity to loss of voice packets is less than the data indicates the value of around 1%. Most sensitive to loss of data packets are then followed by voice and video conferencing, a little less sensitive is the streaming video. Data packet transmission algorithm can be used for re-sending the packet, which e.g. voice transmission should not have any meaning.

Datagrams that are lost can not be restored, creating gaps in the conversation. If packet loss is distributed randomly does it to a significant deterioration in voice quality. Small spaces do not mind, but high packet loss or loss of more data units following behind him leads to a marked deterioration in the quality of speech. Just a loss of more data units following behind most notably worse call quality. This effect results in a much larger deterioration in combination with high delay.

2.4 Bandwidth

Bandwidth is transmission capacity, which is related to throughput, which is the volume of data successfully transferred per unit time and affects the transmission speed.

quality network	good	acceptable	poor
delay [ms]	0-150	150-300	over 300
jitter [ms]	0-20	20-50	over 50
loss [%]	0-0,5	0,5-1,5	over 1,5

Table 2-1 Differentiation network quality parameters by transmission VoIP [9].

2.5 Mean opinion score

Testing call quality is very subjective. The most important subjective measurement of voice quality is so-called MOS (Mean Opinion Score) described by the ITU in recommendation P.800. Considerable progress has also been done in the creation of objective methods for measuring call quality. MOS is a widely accepted criterion for assessing the quality of the call. MOS evaluation is done randomly selected group of people who “grades” transmitted voice quality compared to traditional analog connections. For this reason, this method of evaluation of voice quality, very expensive, difficult to implement and can not be automated. Moreover, although this method is relatively accurate, it is still only an estimate. In Table 2-2 are listed as the values of trade should meet certain quality of the call (the recommendation for people who grades).

Value	Call quality	Description
1	worst	Unintelligible speech
2	bad	Bad intelligible speech
3	medium	Communication quality, but requires some effort to listening.
4	good	Signal is clear and natural as the traditional telephone network.
5	excellent	Excellent spoken speech

Table 2-2 Mean opinion score values

3 UMTS

In Europe is the development, research and standardization of mobile communication system known as the universal mobile telecommunication system (UMTS) and is led by european telecommunications standardization institute (ETSI). The system is part of the UMTS third-generation systems, which is presently one of the most common radio interface in the world. Allows the transmission of multimedia services in real time, which is one of the main reasons for the development of 3G radio communication systems. Expedient character of signal code multiplex with spread spectrum in the transmission environment predispose UMTS system into worldwide spread. To minimize interference with the use of Walsh functions.

UMTS system allows the transmission of data with increased speed and is focused on multimedia applications. They are dedicated bandwidth for satellite communications, or access via DECT[4]. In adverse conditions of the radio environment, where the speed of mobile station (MS) is several hundred kilometers per hour, the minimum signal transmission speed 144 kbit/s. Slow movement (walking) of MS is the transfer rate at least 384 kbit/s and in the case, when MS is at rest or in a range of one sub cell, increase transmission speed up to about 2 Mbit/s. [1, 7]

3.1 System architecture

UMTS system architecture can be illustrated via various models (number of releases), but each of them describes the basis of system from a different perspective. The basic parts are the core network (CN), radio network system (RNS), which uses the UMTS terrestrial radio access network (UTRAN) and the end-user, who is representing by user equipment (UE). UE consists of mobile devices and universal subscriber identity module card (USIM), communicates through a radio interface (Uu) with one or more Node B. The transmission of information between the CN and RNS is via interface (Iu) in the circuit switch mode (CS), packet switch mode (PS) or can be connected simultaneously to both domains PS and CS. [2] Network elements and their connections for user data transfer, see Figure 3-1.

[4] Digital Enhanced Cordless Telecommunications is a wireless telephone technology developed specifically for cordless use (relatively large number of users within a small area) instead of cellular use (relatively small number of users over a large area).

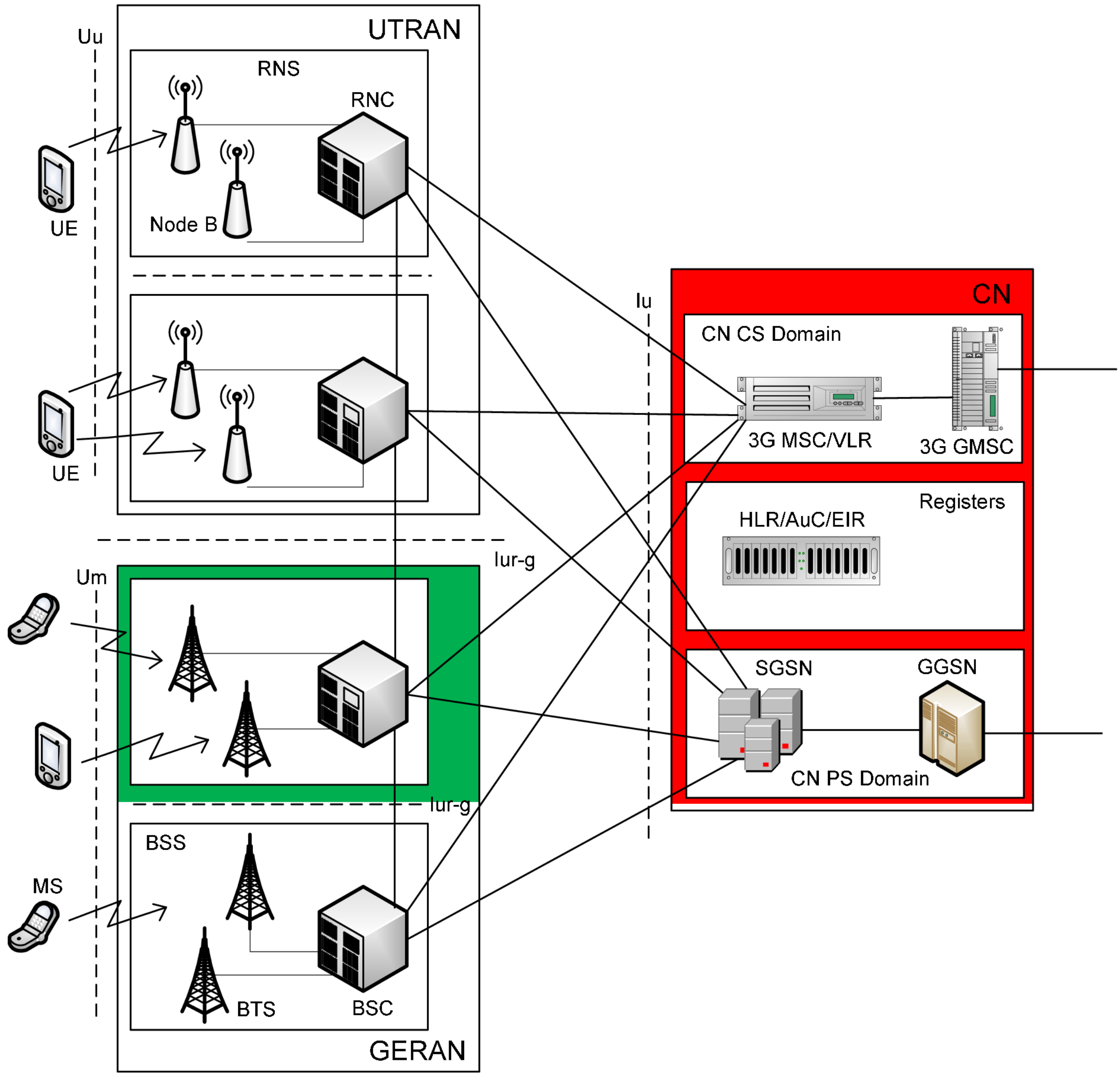


Figure 3-1 UMTS network architecture

Architectury foundation is solid backbone network CN, which manages the operations and connections in the system. It includes a network management service providing control, processing and storage of data and telecommunications network supervision. Towards a participant followed access network RNS, which performs transmission and switching functions. It uses UTRAN interface with which they have access all users. RNS consists of blocks containing the base station in one or more cells that are interconnected with unit of radio network controller (RNC). Multilayer network designing with a macro-, micro- and pico-cell structure brings further enhancements to the cellular and frequency reuse concept to improve system capacity. Base station in 3G systems label Node B includes a radio receiver, transmitter and antenna system covering the cell or multiple cells. The primary functions of Node B units are modulation and demodulation, transmission and reception, coding physical channels, macro diversity, protection against transmission errors and control performance. Among RNSs and intermediate subsystems (domains) in backbone network are interfaces precisely defined, allowing the backbone network used by other radio access technologies, such as the circuit switched (Iu-CS), broadcast (Iu-BC), packet switched (Iu-PS). [17]

The UMTS is due to the multiway spreading of radio waves in atmosphere lead to diversity income in uplink direction (orientation of UE), which increases the probability of successful signal reception. Signal receive a few of Node Bs, but resultant signal arise in RNC combiner by merging. This will reduce the total power stations without the need for synchronization. In the case of diversity in the downlink is situation different, there is increased level of transmission paths from different Node Bs, which operate to the UE and this is reflected as an interference. Therefore growth a rise of interferences.

Although the signals are orthogonal in the donwlink direction due to reflections become these signals non-orthogonal, thereby increasing the size of the interference. Efficient area of cell decreases. This leads to a reduction in the number of users in the cell and then to reduce interference. It then allows remote users to re-establish a connection. Process variable number of users in the cell served by one Node B is called cell respiration.

The UMTS system are used in downlink direction the orthogonal properties of signals to separate of individual participants. For synchronization is used the scramble codes with pulse waveform of autocorrelation. This feature is a random process and by generating the pseudo-random functions is reached to distinguish the different Node Bs.

One of the tasks RNC is to measure interference in the cells under its administration. Based on these measurements, sends commands to control power in the outer loop to reduce. He is in charge of too tender handover (HO[5]), except softer HO, which holds the Node B.

PS domain consists of the network entities to support GPRS (General Packet Radio Service), which are SGSN (Serving GPRS Support Node) and GGSN (Gateway GPRS Support Node). PS domain backbone network operates on the IP protocol. GGSN is in principle classic IP router, which constitute an interface between the GPRS network and external networks. Works on standard IP or X.25. Therefore, it should be included in a network operator an active firewall as a protection against unwanted intrusion from external network. He must be functional also DNS (Domain Name Server) and the allocation of IP addresses to individual devices. Such a function is performed by GGSN or the DHCP server (Dynamic Host Configuration Protocol). The tasks of SGSN are routing packets to the user identification, encryption, and pricing.

The core network also includes a lot of very important databases. HLR (Home Location Register) is one of them is a database of all participants in the network. It also contains data on all the IMSI (International Mobile Subscriber Identity), which are unique identification numbers of individual USIM. The HLR is also information about telephone numbers assigned to the IMSI, set services for the participants, etc. Other database is located right in the heart of MSC (Mobile Switching Center). This is a VLR (Visitor Location Register), which contains the copied records from the HLR of all UE, which are at the moment in operation in the area served by the respective MSC. This serves to more quickly connect calls or SMS, than if all the requirements for the service passed through the HLR. AuC (Authentication Center) is another database that is connected to the HLR. AuC is used to verify the identity of participants in the network entries.

3.2 Layer model

The UMTS Terrestrial Access Network (UTRAN) radio interface is based on Wideband Code Division Multiple Access (WCDMA) radio technology, which has some

[5] Handover is a function of cellular networks providing switched connections between mobile station (MS or UE) and base station (BTS or Node B in UMTS) during a call from one channel to another channel. Occurs when the system assesses that a new channel as a better (smaller value of noise, less delay). This occurs primarily at the border between cells.

fundamental differences from all 2G radio access technologies. The key aspect to be controlled by radio interface protocols is the multiplexing of traffic flows of different kinds and different origins. To ensure effective control of multiplexing a strict layering of duties has been applied, thus resulting in the three-layer protocol reference model illustrated in Figure 3-2, taken from [18].

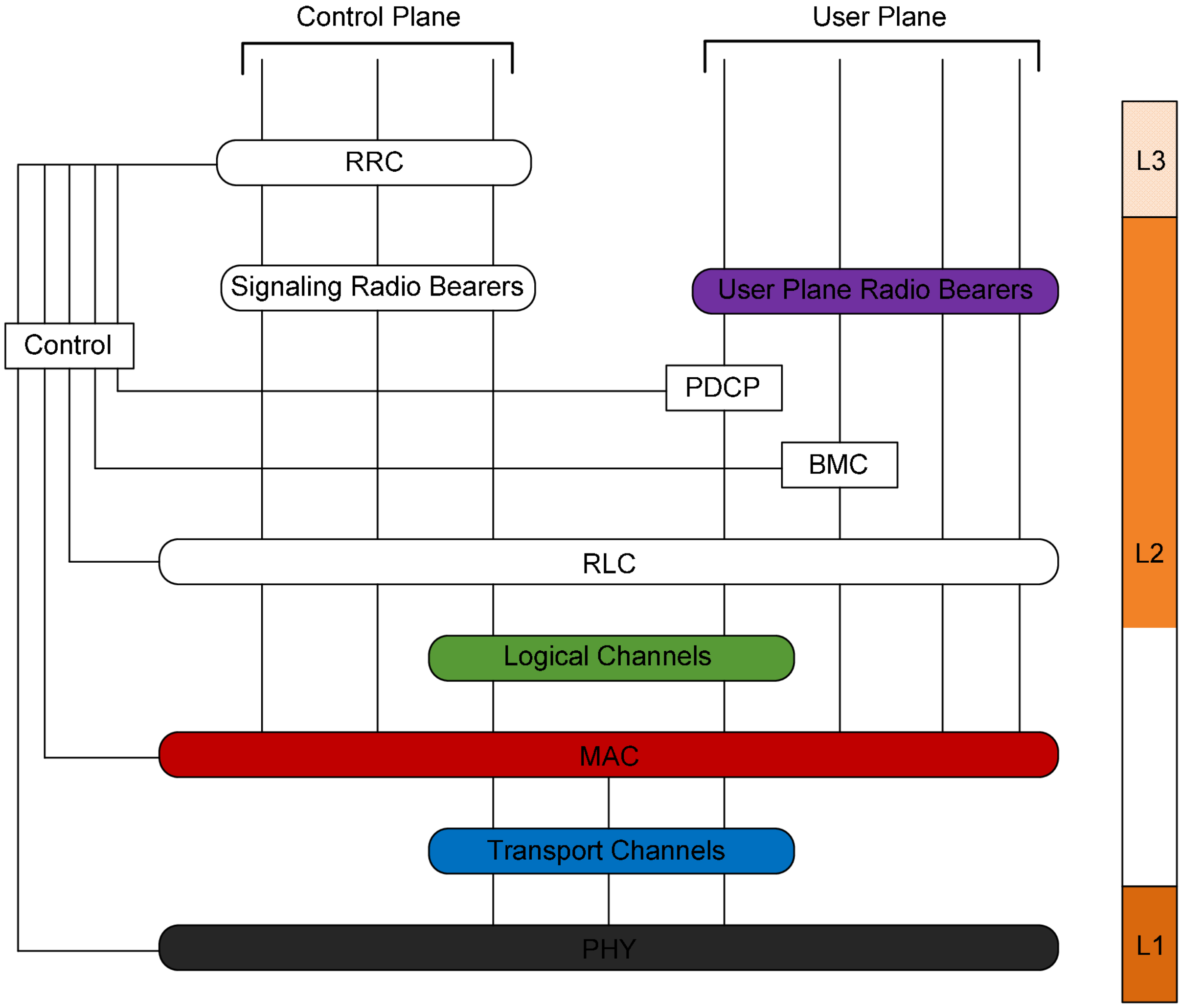


Figure 3-2 Radio interface protocol reference model

Interface UTRAN consists of a set of horizontal and vertical layers. UTRAN requirements are handled in the horizontal level of the radio network through various layers of management and users. Control plane is used to manage connection or interconnection, user plane serve to transparent data user transfer from higher layers. Standard transmission problem, which is independent of the UTRAN are addressed at the transport channels.

The Radio Link Control (RLC) sublayer adds regular link layer functions onto the logical channels provided by the MAC sublayer. Due to the characteristics of radio transmission some special ingredients have been added to RLC sublayer functionality.

For L3 control protocol (signalling) purposes the RLC service is adequate as such, but for domain-specific user data additional convergence protocols may be needed to accomplish the full radio bearer service. For CS domain data (e.g., transcoded speech) the convergence function is null, but for the PS domain an additional convergence sublayer is needed. This

Packet Data Convergence Protocol (PDCP) sublayer makes the UMTS radio interface applicable to carry Internet Protocol (IP) data packets.

As its name suggests PDCP is designed to make WCDMA radio protocols suitable for carrying the most common user-to-user packet data protocol, TCP/IP. PDCP entities can be found at both ends of the WCDMA radio interface: the RNC and the UE. The key functionality of PDCP is its ability to compress the headers of payload protocols, which if sent without compression would waste the invaluable radio link capacity e.g., without header compression the header size for an RTP/UPD/IP header is at least 40 bytes for IPv4 and at least 60 bytes for IPv6. Since payload size in such services as IP voice is about 20 bytes or less, the overhead without compression is obvious. In the 3GPP radio interface protocol model, PDCP belongs to the radio link layer (L2), the topmost sublayer of wich is specially designed for user-plane radio bearers carrying packet data. [4]

The access network, UTRAN, has overall responsibility of WCDMA radio resources. UTRAN carries out radio resource management and control among its own network elements and creates the radio access bearers that allow communication between UEs and the Core Network (CN) across the whole UTRAN infrastructure. This communication is structured according to the generic UTRAN protocol reference model. The model is generic in the sense that the architectural elements though, not the specific protocols are the same for all the UTRAN interfaces: Iu, Iub and Iur. The specific protocols are only using their 3GPP specification numbers, which also underlines the idea of sharing a common protocol architecture.

3.3 QoS in the UMTS

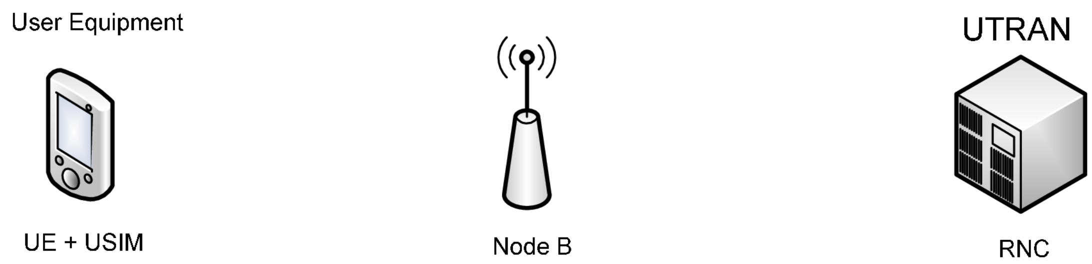
Examples for quality of services at the application level can be video signal of a some resolution and certain of frames per second, the exchange of messages between communicating applications with a certain maximum delay and a maximum loss rates (which must be compensated by repeating the message), or transferring a certain volume of data during a specific time.

QoS is sometimes also used as a term that implies the mechanisms used for different treatment between data types. Here, the QoS can be viewed both from the perspective of end users and also from the perspective of service provider. Individual QoS features enable efficient use of radio interface and transmission resources. RRM (Radio Resource Management) using various algorithms, which aim to stabilize the path of radio waves and also allow QoS to meet the criteria set by the service. RRM algorithms provide QoS in the UMTS network. These algorithms ensure the maintenance of the planned coverage area and offer a high spectral efficiency. One reasonable goal of RRM strategies is, e.g., a balanced traffic load of all base stations. [18] RRM functions can be divided into the following categories:

- *Power control* is the management power of stations with regard to the minimum level of interference.
- *Handover control* handles the handover function.
- *Congestion control* includes algorithms against a network overload.
  - Admission control access of participants.
  - Control regulates the network load.
  - Packet scheduling.

Other RRM algorithms, namely AC, LC, and PS are assisted primarily to QoS guarantee and to maximize throughput for a set of cells of different bit rates, business applications and quality requirements. AC, LC, and PS are so-called *cell-based functions*, so the functions dependent on the current parameters of the cell. RRM algorithms are based on a specific subset of QoS attributes. These attributes are associated with radio carrier, which bear different service applications.

Figure 3-3 is shown typical areas of application QoS functions in cellular networks based UTRAN radio interface. We note that, unlike the GSM system, UMTS dispose with the QoS features in the mobile stations, base station and also RNC. [7]



Power control	Power control	Power control
Handover control	Code generation	Handover control
Radio measurements	Packet scheduling for HSDPA	Admission control
	Radio measurement	Packet scheduling
		Code management

Figure 3-3 Distribution of QoS functions in the UMTS system

3.3.1 Admission control

AC (Admission Control) is an access control participants in the network. The function decides whether to set up and established a new user connections or modify it, or modify an existing connection. Algorithm is run for each cell in the radio setup and changeovers carrier cell parameters separately for uplink and downlink. If the established new or modified existing connections, the algorithm AC determined RRP (Radio Resource Priority). This is a specific priority for service on the QoS profile that take a cell from the core network. RRP value is a parameter that sets the operator, the lower value assigned to a higher RRP determines priority. Requirements of the new radio links are queued according to their priorities and then served by strict principles and based on the time when they arrive. Requirements are rejected when it exceeded the maximum time in queue or the corresponding maximum permissible length of the queue. These parameters can be set differently to the QoS class or so-called ARP (Allocation/Retention Priority) and THP (Traffic Handling Priority), which is a priority for the interactive class. Access is also rejected if the new radio carrier causes excessive interference. Features AC in the system W-CDMA (Wideband Code Division Multiple Access) is particularly important because there is no explicit limit the maximum capacity. Using algorithm is based on measurements or estimates of the current network load.

3.3.2 Handover control

To maintain a communication link between a mobile subscriber and base station Node B serve the handover procedure. It runs in areas where individual cells overlap (mostly in the border cells) and there is a relatively good signal available from multiple stations.

There are three categories of handovers - hard, soft and softer handover, which are divided in Table 3-1 and shown in Figure 3-4.

1.	**Soft handover**		
2.	**Softer handover**		
3.	**Hard handover**	Intra – frequency	UMTS(f1) → UMTS(f1)
		Inter – frequency	UMTS(f1) → UMTS(f2)
		Inter – system	UMTS → GSM a GSM → UMTS

Table 3-1 Types of handovers in UMTS network

Hard handover is rated by a network (Network Evaluated HandOver), when the channel quality measurements made the base station and upon the results of the measurement signals made the decision if to switch. The mobile station is not subject to any requirements. This means that all of the original radio connections are lost even before the new connection is established [1].
Hard handover is seamless (that is unrecognizable by the user) only in the case of non real-time communications. Handover, which require changes to the carrier frequency or technology access are always performed as hard handover. Hard handover can be seamless or non-seamless.

Mobile station evaluate the soft handover (Mobile Evaluated HandOver), the measurement of all channels carried both mobile and base station and the decision to switch made the mobile station, is transmitted to the system and ensure implementation of the switch. The soft handover connection is established and disrupted in such a way that the mobile station always keeps at least one connection to UTRAN. Soft handover is performed, using a macro diversity[6], which refers to the condition of activity of several radio connections at the same time. Soft handover can be used when the user moves between cells operating on the same frequency.

Softer handover is a special case of soft handover, when linked and interrupted radio connections belonging to the same Node B.

[6] With macro diversity, the RRM has to handle the power partitioning between the different links from the base stations to the mobile station.

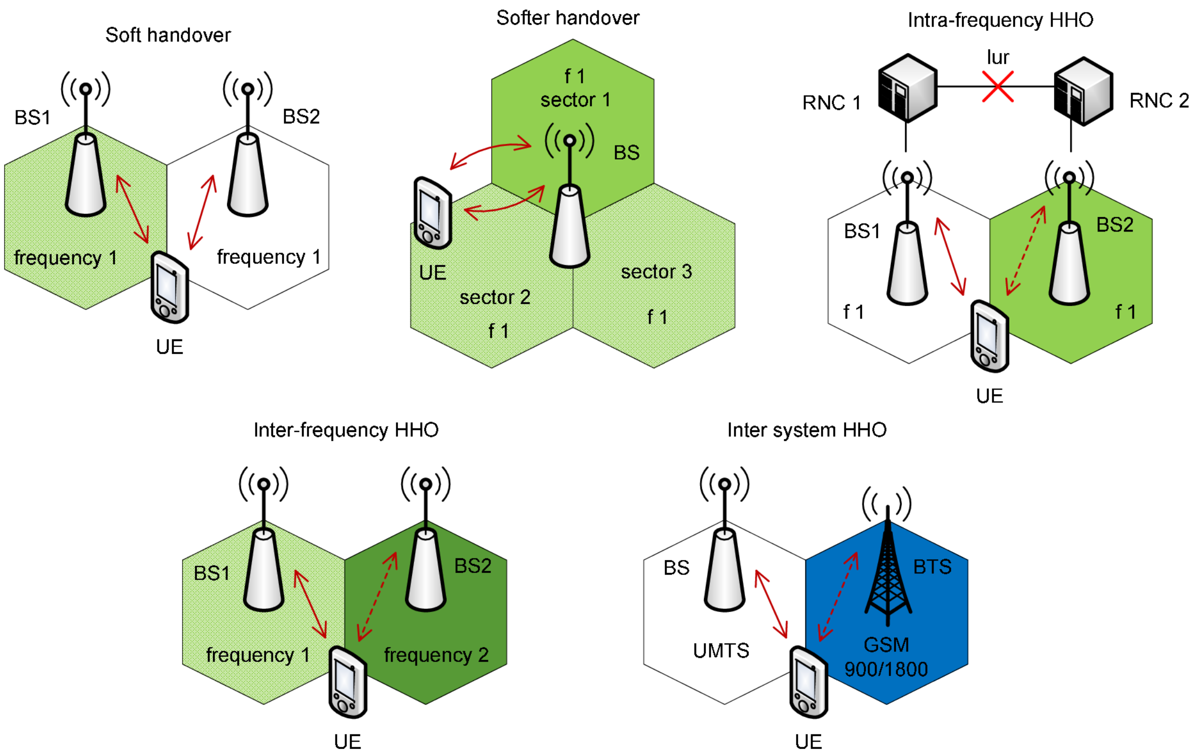


Figure 3-4 Different types of handover trapped in a hierarchical structure

3.3.3 Classes of traffic and QoS attributes

Each service establishes certain requirements for the transmission path for transmitting service information. Some services provide more stringent requirements than others. Networks usually passes many services and service requests from many users simultaneously. Each of these services has its own requirements. Since network resources are limited. The aim is to allocate just enough resources indispensable for each requirement, either too much or not enough.

It is possible to make very complex definition of QoS, but a comprehensive QoS classification leads inevitably to a very complex network, it would have to be added a lot of direction and control.

In UMTS is QoS a relatively simple concept, consisting of four classes of service and some QoS attributes for defining the operational characteristics from traffic classes.

These four service classes are:

- QoS conversational - a phone call, VoIP, videoconferencing, internet, multimedia
- QoS streaming - streaming audio and video in real time
- QoS interactive - normal data traffic web browsing, chat applications
- QoS background - view emails and download files

Traffic class	Fundamental characteristics	Service examples
Conversational	Low delay, Small delay variation	Speech, VoIP, video conferencing
Streaming	Moderate delay and variation (end-user-application-dependent)	Streaming video, streaming audio
Interactive	Round trip delay is a matter of importance Moderate delay variation Request-response pattern	Web-browsing
Background	Destination (end-used application) does not expect a response within a certain time	Email and file-donwloading

Table 3-2 Traffic classes and their characteristics [1,7]

3.4 RSVP

Signaling, dynamic protocol RSVP (ReSerVation Protocol) [3] is a mechanism to ensure QoS. Devised for the reservation of network resources in the IntServ model and its activity is defined in RFC 2205. RSVP is a complex, but offers the possibility to simulate the connection of CS. RSVP is used to launch integrated services. Entity of integrated service created of two alternatives:

• *Guaranteed* is the state of integrated services that emulate a specialized virtual circuits. Provides warranty and a stable limits for the delay in the end-to-end connection based on arranged parameters of operation.

• *Controlled load* is better than "best effort", but this higher level can not be guaranteed in all transmission conditions.

Application requiring a reservation of transmission capacity first of all sends a message Path to the recipient data application. This report contains information on the nature of broadcast data, i.e. specification includes operation TSpec (Traffic Specification) and is here also recorded the path between the transmitter and receiver data stream. Each router with RSVP support will establish the "path-state" in downstream way, which contains source address of the previous PATH report, i.e. the address of the nearest node in the upstream direction to the sender. Following the adoption of the PATH message sends receiver node RESV message, through which is created reservation for the dataflow. RSpec and FilterSpec describe the data flow that routers use to identify all reservations.

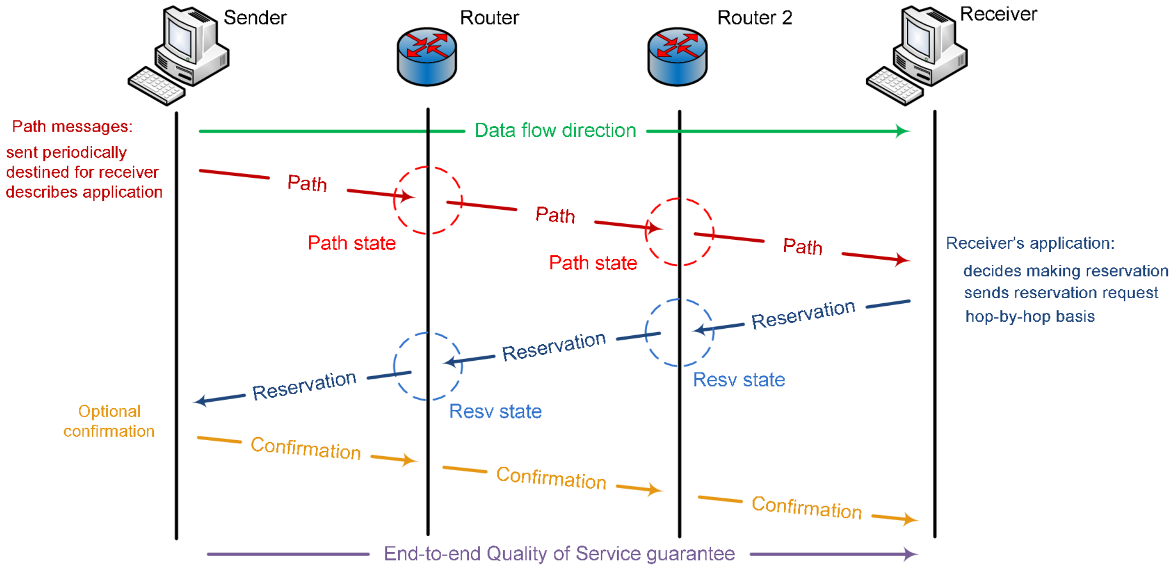


Figure 3-5 System mechanism of RSVP

The following are the sequence of events during resource reservations in unicast traffic in Figure 3-5, taken from [4].

1) Sender's RSVP module sends Path messages regularly soon data communication begins.
2) This report creates a state of each router, which stands in their way. All facilities along the route are aware of the neighboring node in the data stream.
3) When the recipient notified incoming message it can decides to create a resource reservation or not.
4) If decide to meet, the host application creates a request for booking network.
5) The RSVP protocol then carries the request of Resv message to all nodes along the reserved path. Reservation is made on the basis of a "hop by hop" and every intermediate node examines the appropriate resources and consider whether this request can be assigned to the source. When the reservation is successful created the status of a Resv and requirements of the reservation forwarded it to the previous jump in the data path.
6) The transmitter sends approval with status for reserved confirmation. The whole process is repeated when there arrive further action by the transmitter or receiver.

3.5 Differentiated services

Abbreviated as DiffServ is a relatively simple way of classifying services and also allows hadle with different classes in the network. It is possible to have a lot of DiffServ traffic classes, but there are only two important levels of services or classes. Operating classes have predefined profiles behavior and restrictions, known as *DiffServ values* or *codepoints*. [6]

• *Expedited forwarding* class implements DiffServ codepoint parameter. Minimizes delays and jitter, and thus creates the highest level of QoS. Any unsatisfactory operation of the class definition is simply discarded.

• *Assured forwarding* class of service can be divided into four subclasses and three types of priorities. This also makes it easy to allocate a total of 12 DiffServ codepoints.

DiffServ is basically an internal network QoS protocol invisible to end users. If used, is applied at users on the edge of network. Regarding the DiffServ, this bordering entry is called the *network ingress point*. Respectively, split the opposite side of network, called the *network egress point*.

Figure 3-6 is evident colors. For end user's applications or computers can perform DiffServ marking directly. This method can brings benefits, especially when one node consider how to ensure the QoS on end-to-end link. In DiffServ are each traffic flows distinguished, but are collected in a small number of operating classes. Bandwidth and other network atributes are allocated into traffic classes.

By default, DiffServ is assumed that there is SLA (Service Level Agreement) between the networks. SLA defines the technical parameters among networks describing the quality of interconnection. This technical parameter specifies policy collectively. Traffic is policed in the ingress and egress points by the network's political characteristics. If the traffic flow is out of policy, treated according to defined SLA (time, source and destination address, identifier). For example, it simply discards the data or can be supplied extremely at extra cost.

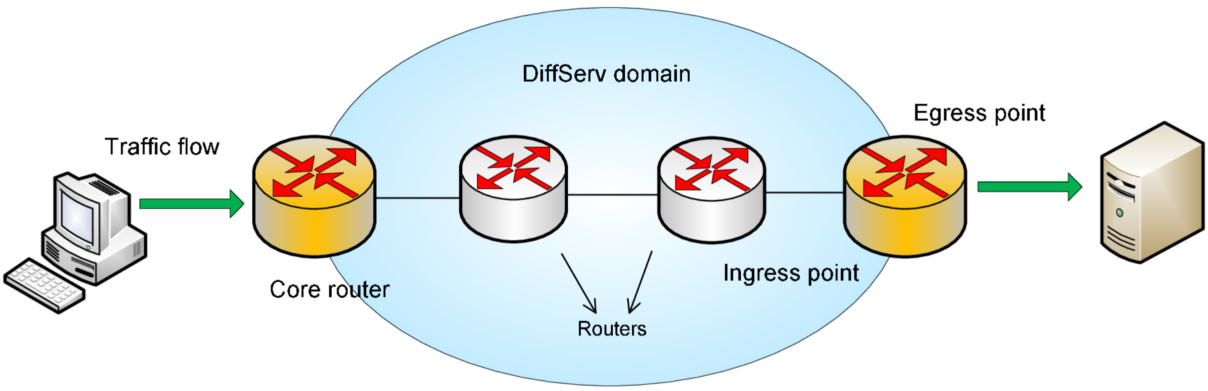


Figure 3-6 Principle diagram of differentiated services

4 MODELS OF NETWORKS

Programme OPNET Modeler is an software for simulation of various networks topologies, protocols, services and nodes. It can be designed any model with a lot of objects (station, server, link, router), traffics and applications (profiles). After simulation it is possible to analyze behaviour of constructed models. Programme also measure every event in the network and provides graphs (with different types of curves) and statistics. We can choose, what we want to examine.

In the programme OPNET Modeler 14.5 educational version are designed two models of the network, which reflect principle of standard 802.11e and architecture of the third generation UMTS mobile networks.

4.1 Wireless model

Using by software described above was modeled final project comprising of two scenarios in which are conducted simulations. Project with scenarios into it is created in campus network scale, where size of X,Y span is set at 2 x 2 kilometers. The range of elements in the project was selected 2x access points, 7x wireless stations, 2x servers, switch, 10 Mbps links, application and profile attributes.

The first scenario is version without QoS assurance based on 802.11b standard. The second scenario was created in accordance with standard 802.11e, which supports quality of service. One network plan is for both scenarios. Scheme on Figure 4-1 represent two basic service set (BSS) operated by access points in the middle of each area. BSS_1 purvey connections for 4 wireless workstations, where all are established to the wireless router. It is connected by link to the ethernet switch. The switch interconnects both regions and likewise to the data servers. The second domain BSS_2 control three stations. Data traffic represent services like file transfer, browsing web pages, email, video streaming and interactive voice. These services load entire network simultaneously by definitions in the applications and profiles configurations.

Supported wireless network configurations in OPNET: Ad-hoc network
Infrastructure BSS
Extended Service Set
Wireless backbone

Designed network in project is an WLAN model type extended service set. Workstations communicate with each other and with nodes outside their LAN through the access point.

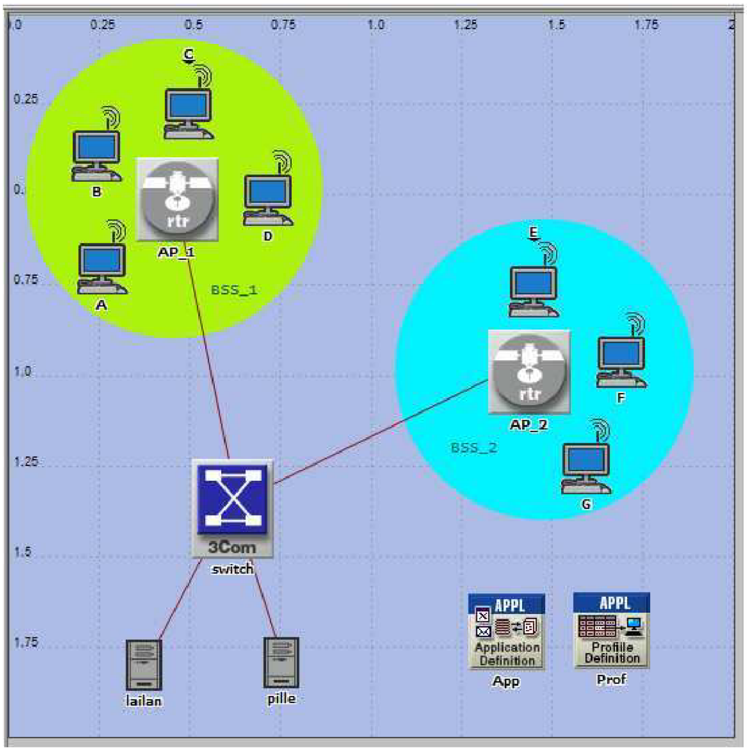


Figure 4-1 Wireless network model for standards 802.11b and 802.11e

Used objects by OPNET terminology: wlan_ethernet_router_adv
wlan_wkstn_adv
ethernet_server
3C_SSII_1100_3300_4s_ae52_e48_ge3
10BaseT
Application Config
Profile Config

4.1.1 Parameters settings

Programme offer miscellaneous parameters and really useful is using default settings. For simplicity just always make changes in followed attributes.

Applications configuration:

Component defines six rows for applied applications. After configuration assign these applications to the servers and it will govern them see Table 4-2.

Database Access (Heavy): Transaction interarrival time *exponential(3)*, transaction size *constant(32768)*.

Email (Heavy): E-Mail size *uniform(1048576, 3145728)*.

File Transfer (Heavy): Inter-Request time *exponential(10)*, file size *constant(500000)*.

Video Conferencing (Heavy): Frame interarrival time information *10 frames/sec* Type of service *Streaming multimedia (4)*.

Voice over IP Call (PCM Quality): Type of service *Reserved (7)*.

Web browsing (Heavy): Page Properties *Large Image*.

Type of service[7] represents a session attribute which allows packets to be processed faster in IP queues.

Type of service:	Service:
Best effort (0)	Database, FTP, email, HTTP
Streaming multimedia (4)	Video
Reserved (7)	Voice

Table 4-1 Type of service to distinguish the application

Server:	Supported application:
lailan	Database, FTP, video
pille	Email, HTTP, voice

Table 4-2 Servers with supported services

Profile configuration:

Profile configuration see **Chyba! Nenalezen zdroj odkazů.** is used to set the parameters by which applications are executed and completed. It can be set different user's activities in different time interval. Each profile created in this object can be assigned to individual stations or servers for data generation and creation of network traffic. For each application is matched one profile and all parameters are identical. It's used also default settings. After completing configuration assigned profiles to the clients by Table 4-3.

Operation mode *Simultaneous*, Start time *constant(3)*, Start time offset *uniform(5,10)*

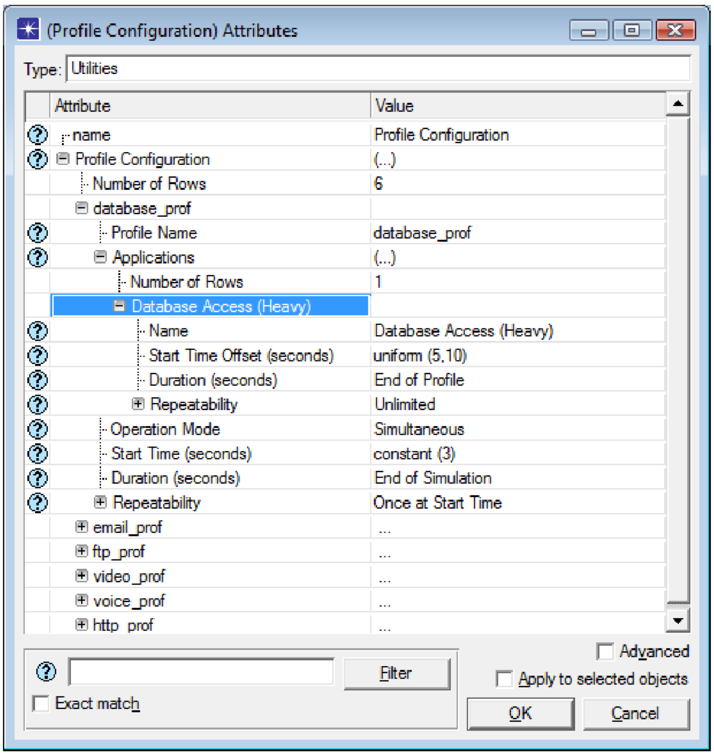


Figure 4-2 Database application setting in profile configuration

[7] ToS assignment at the client is not affected by the ToS value specified at the server.

Operation mode *Simultaneous*, Start time *constant(3)*, Start time offset *uniform(5,10)*.

Domain:	Workstation:	Supported profile:
BSS_1	A	Dafabase, email, voice
	B	FTP, database, HTTP, email
	C	Video, email, HTTP, voice
	D	FTP, email, http
BSS_2	E	FTP, database, HTTP, email
	F	Dafabase, email, voice
	G	Dafabase, video, voice

Table 4-3 Clients with supported profiles

Access points and workstations:

HCF parameters attribute can be used to specify whether MAC supports WLAN QoS facility defined as Hybrid Coordination Function (HCF) in the IEEE 802.11e standard, and to configure the attributes of HCF operation. The model supports Enhanced Distributed Channel Access (EDCA) with the following characteristics:

- Access Categories (ACs) for prioritized contention-based access
- Voice, Video, Best Effort, and Background
- Transmission Opportunity (TXOP) Frame Bursting
- EDCA Parameter Set distribution by Access Point (AP)

Most important data can be found in described check box. Figure 4-2 shows HCF[8] parameters settings.

Wireless LAN Parameters:

- BSS identifier *1* or *2* it all depends on relevant access point.
- HCF parameters *default* or *not supported* depends on QoS or non-QoS warranty.
- Physical characteristics *Direct sequence*.
- Data rate *11 Mbps*.

Priority data are handled according to QoS mechanisms using a Contention Window. If the station didn't support QoS, or lacked the support of QoS access point, the operation would be done under the DCF parameters.

[8] This statistic is not collected if the WLAN MAC is not 802.11e-capable (i.e., doesn't support HCF functionality).

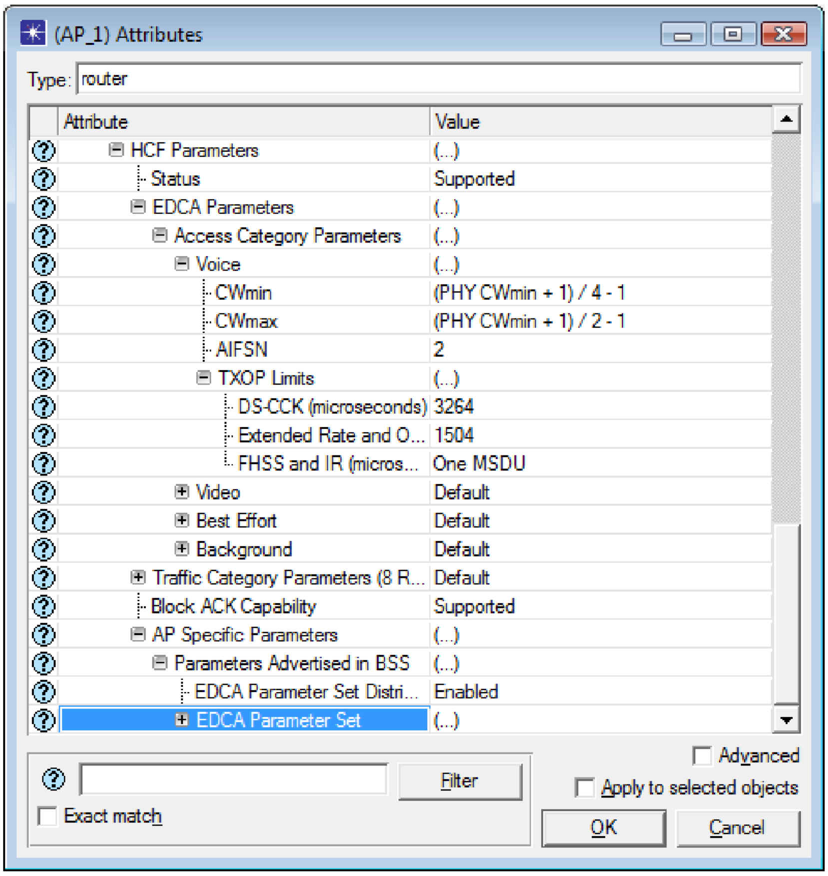


Figure 4-2 HCF parameters settings on AP_1

Duplicate the scenario after completing all the configurations on objects. Accomplish inevitable changes on especially wireless nodes to ensure QoS. Settings of collected paramaters can be found in DES (*discret event simulation)*. Pick required attributes and launch the simulation with *2 minutes* duration. A lot of calculations, measurements and events are in progress so it takes a while.

4.1.2 Results

End-to-End delay:

Fundamental pillar of QoS. Decide whether the network can carry voice and video services. End delay parameter contains a total of successfully delivered packets.

The total voice packet delay in accordance with OPNET Modeler, called “analog-to-analog” or “mouth-to-ear” delay consists by network delay, encoding delay, decoding delay, compression delay and decompression delay. Network delay is the time at which the sender node gave the packet to RTP to the time the receiver got it from RTP. Encoding delay (on the sender node) is computed from the encoder scheme. Decoding delay (on the receiver node) is assumed to be equal to the encoding delay. Compression and Decompression delays come from the corresponding attributes in the Voice application configuration.

Comparing results for both scenarios **wifi_model-802_11e_QoS** and **wifi_model-802_11b**, see Figure 4-3. It can be noticed, that the total end-to-end delay for interactive voice come up to 0.6 seconds into whole WLAN on non-QoS scenario. Value is so high, that can be expected a problems and service will be difficult to understand among clients. The curves are mostly drawed as a spline style.

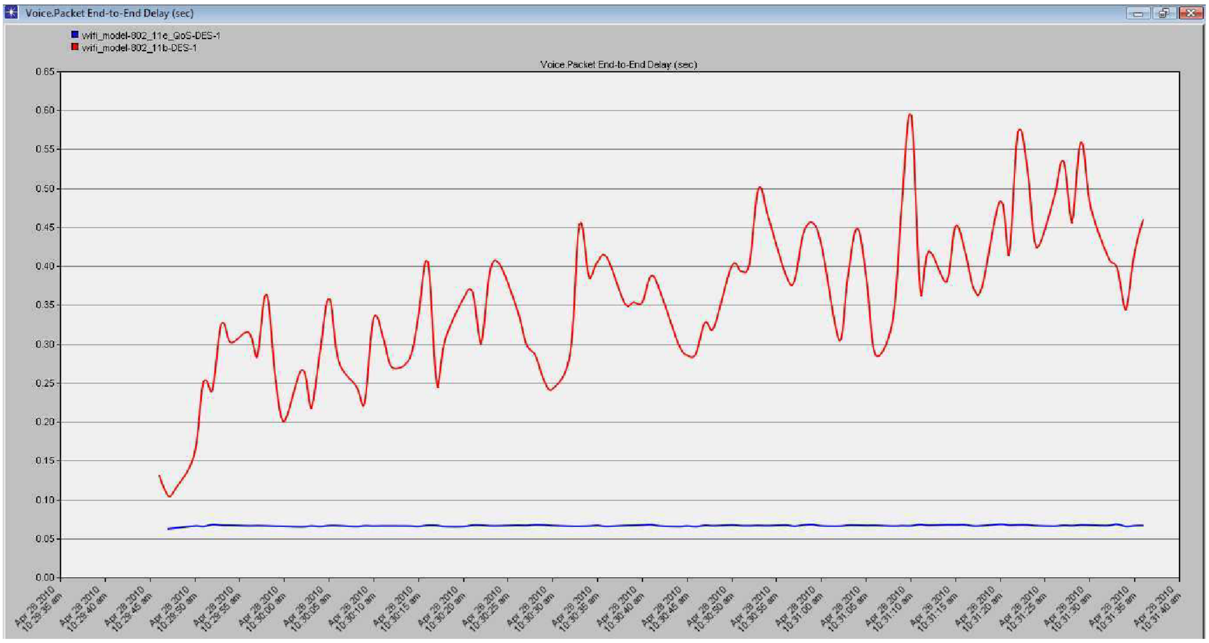


Figure 4-3 Total End-to-End packet delay

Implementation of QoS support to the wireless network will ensure that the transmitted priority voice data will reached constant delay in few milliseconds. They are constant for the duration of whole communication and don't affect them any increasing network load. In the network was achieved for priority data (ToS interactive voice = 7) fluctuations in order units of ms as assessed QoS requirements is a condition where the network can operate multimedia and voice applications without any problems.

See Figure 4-4 end-to-end packet delay for voice application on each workstation A, C in WLAN 1 and F, G in WLAN 2. Delays with QoS support is making the voice approximately 70 ms, which is according to the methodology for voice value of "good".

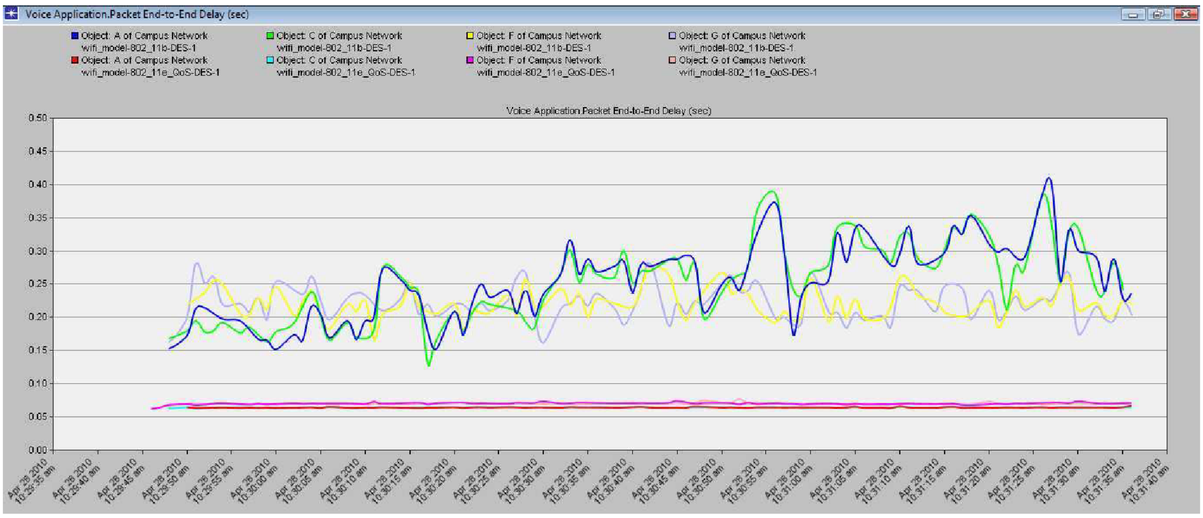


Figure 4-4 End-to-End packet delay for voice application on each clients

Jitter:

If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then resultant jitter will be equal to this equation:

jitter = (t4 - t3) - (t2 - t1)

Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.

The price, see Figure 4-5 reached relatively low values in both scenarios, but to understand a good reflection the principle. Notice, that workstation A and F support the same applications and can be assumed equal activity. However massive load is in WLAN 1, because included more clients and supported services. Graph shows, that curve of this statistics oscilate more in the non-QoS scenario.

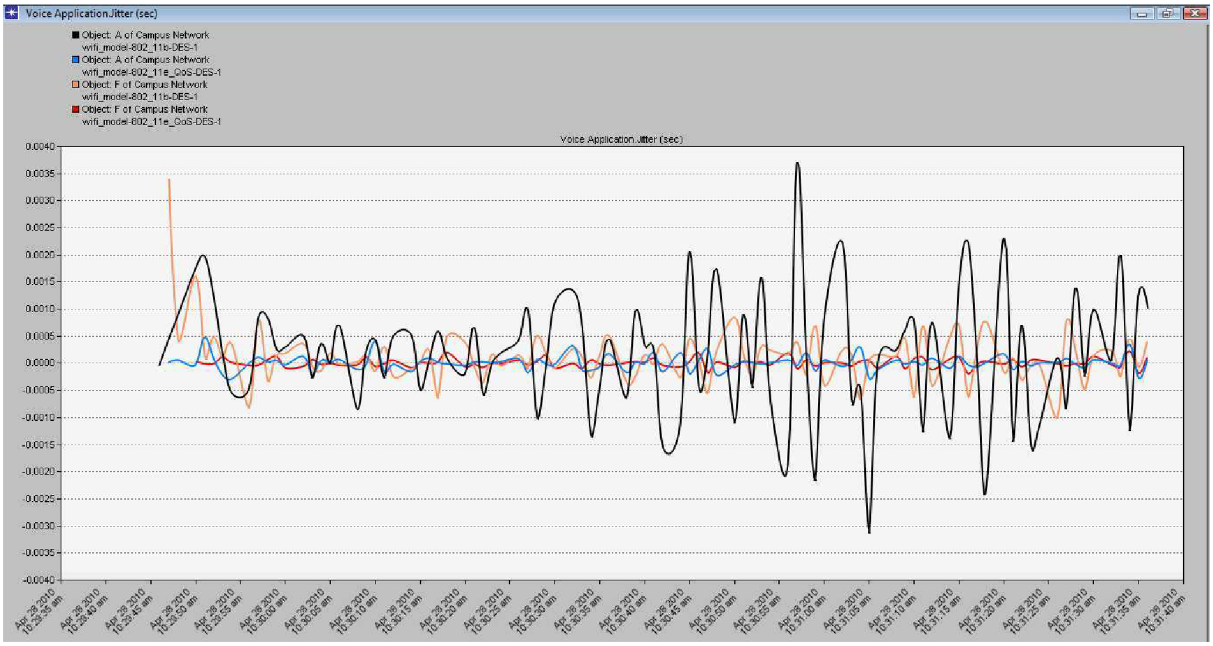


Figure 4-5 Jitter comparing in scenarios

WLAN per HCF access category:

Media access delay is defined as the total of queuing and contention delays of the data, management, delayed Block-ACK and Block-ACK Request frames transmitted by each access category of the WLAN MAC. For each frame, this delay is calculated as the duration from the time when it is inserted into the transmission queue, which is arrival time for higher layer data packets and creation time for all other frames types, until the time when the frame is sent to the physical layer for the first time. Hence, it also includes the period for the successful RTS/CTS exchange, if this exchange is used prior to the transmission of that frame. Similarly, it may also include multiple number of backoff periods, if the initial transmission of the frame is delayed due to one or more internal collisions.

Is evident that type of service *Best Effort (0)* waits longer to access into media because *video* and *voice* are prioritized. Blue curve reaches average 450 miliseconds, see closely Figure 4-6.

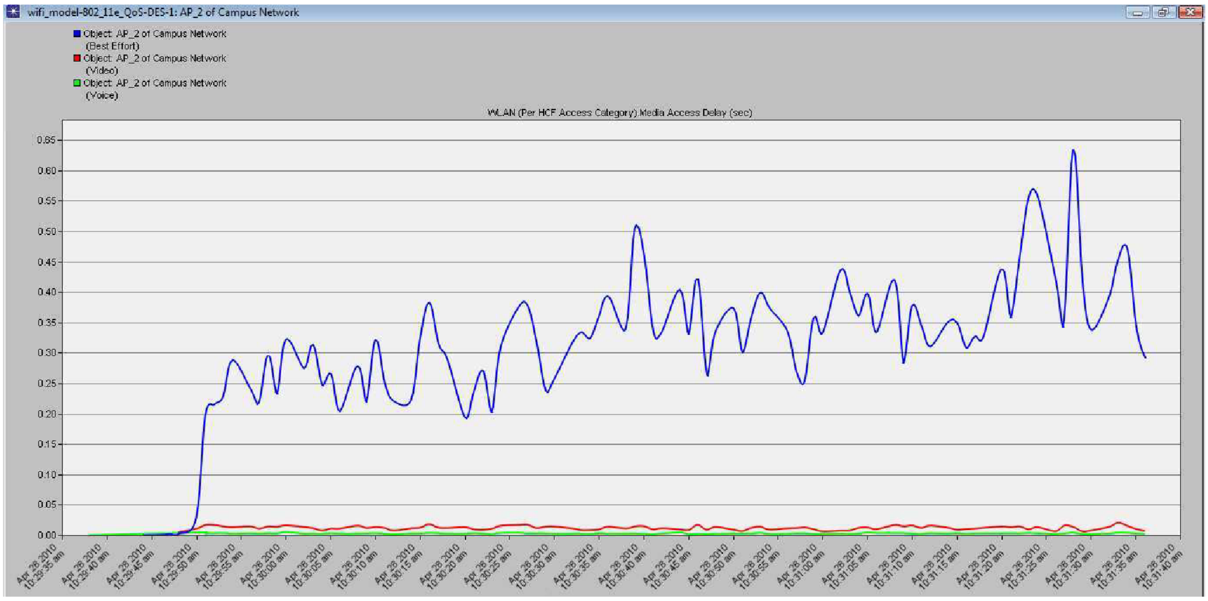


Figure 4-6 Media access delay for types of service

Queue size:

Parameter is specified as total number of data, management, BAR and delayed BA frames in the transmission queue of each access category of this MAC. Data packets received from the higher layer and accepted for transmission are inserted in the access category transmission queues based on their "user priority" values. Access categories of the management, BAR and delayed BA frames are chosen based on the specifications in the 802.11e standard.

Packets that don't require an immediate acknowledgement are removed from the queue as soon as they are transmitted. Packets that require acknowledgement are removed from the queue when all of their fragments are transmitted and acknowledged.

In the MACs that operate as access points, the transmission queues also contain the packets that are received from physical layer and awaiting being forwarded to their final destination within the access point's BSS. Such packets are also included in the number of packets written to this statistic to record the queue size for the MACs with AP functionality.

Differentiation and treatment of various ToS can be seen from Figure 4-7. Queues are formed in the cache access points memory. Access categories with a lower value of the ToS demonstrate inadequate handling. Buffer is filling to full, and therefore may overload and subsequent dropping of packets. Buffer size at the access points is 256 000 bits.

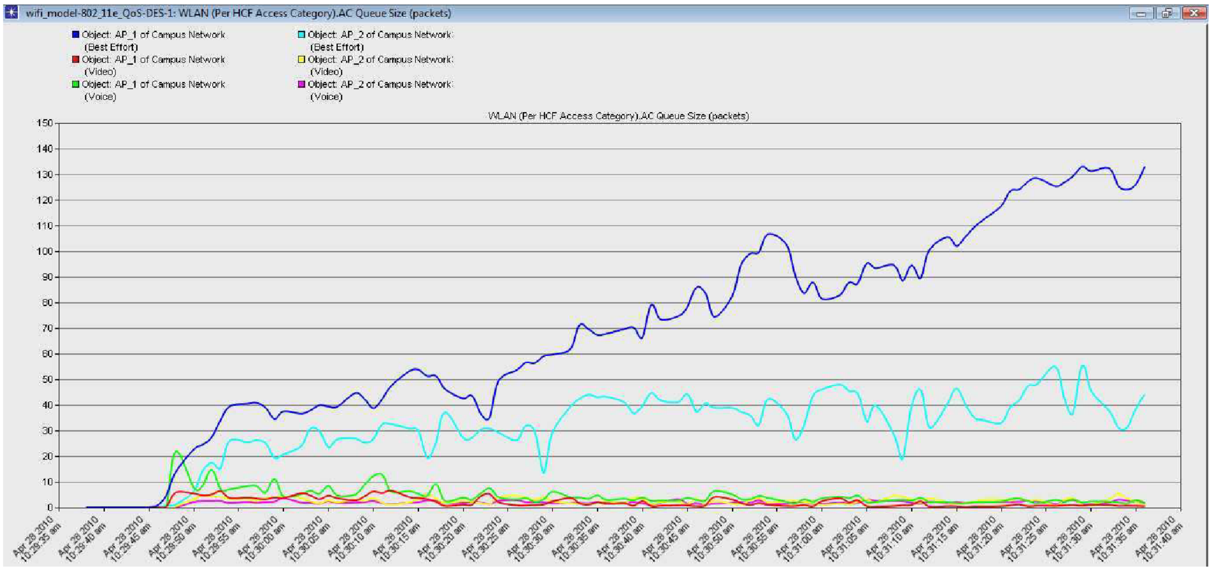


Figure 4-7 Queue size on each access points

Total throughput:

Big numbers sell better. This is also the case for wireless networks. A "11Mbps" or a "54Mbps" wireless network might sound impressive but it is not what you as a user will experience. A lot of the capacity is lost to communications overhead and the data rate given is what can be expected of the physical layer. Wired networks also give their capacity on the physical layer but the overhead is not that large and the experienced throughput is close to the physical capacity.

Global statistic represents the total number of bits (in bits/sec) forwarded from wireless LAN layers to higher layers in all WLAN nodes of the network.

Node statistic symbolizes total data traffic in bits/sec successfully received and forwarded to the higher layer by the WLAN MAC. This statistic does not include the data frames that are:

- unicast frames addressed to another MAC
- duplicates of previously received frames
- incomplete, meaning that not all the fragments of the frame were received within a certain time, so that the received fragments had to be discarded without fully reassembling the higher layer packet

In accordance to standard 802.11e can be reached on physical layer up to 11 Mbps. This value is just theoretical nevertheless truth is different. It shows partial/total throughput on wireless network. Referred to in Figure 4-8, we can see that average throughput on each access point is 2 Mbps more or less (green and red curve). Total throughput amount to 7 Mbps (blue curve).

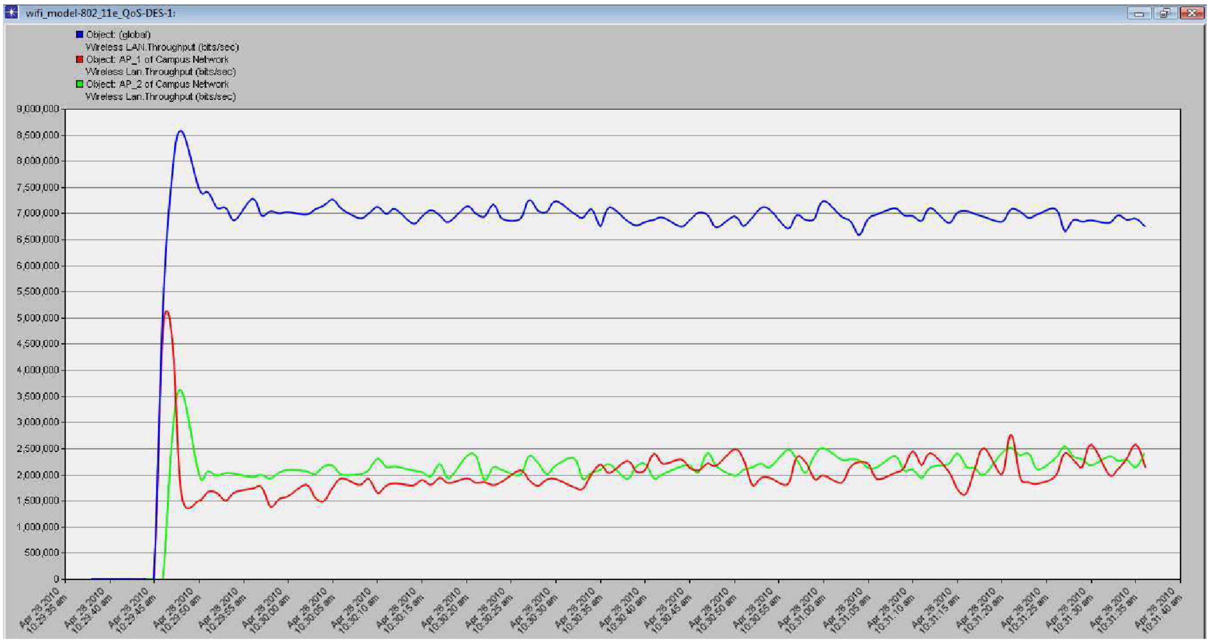


Figure 4-8 Total wireless throughput

4.2 UMTS model

This project contain of two scenarios in which are conducted discret event simulations. Project with scenarios into it is created in world network scale, specifically in Europe city Brno. Use zoom in implement to come near in the map. The list of used components in OPNET Modeler environment includes 2x server, 1x router, 1x GGSN, 1x SGSN, 1x RNC, 3x Node B, 6x UE, application and profile also quality of service attributes.

The first scenario is version without QoS warranty. The second scenario was created in accordance with quality of service support. One network plan is for both scenarios. Scheme on Figure 4-9 is a model diagram of the UMTS network architecture with six clients. One of them is mobile and rest are fixed. Mobile terminal is assigned to the movement trajectory for checking the handover function. Base stations Node Bs containing three directional antennas covering the angle of 160 degrees and the cells are positioned so that they overlapped each other. Users use VoIP calls, online browsing, and file transfer. All of these services are defined in the applications, respectively profiles attributes.

Supported UMTS network configurations in OPNET:

Simple UMTS Network Using Application Traffic

Simple UMTS Network Using Raw Traffic Generation[9]

Designed network in project is an UMTS model type application traffic. Workstations communicate with each other in the UTRAN. Router GGSN interconnect different types of networks, the CN and IP network.

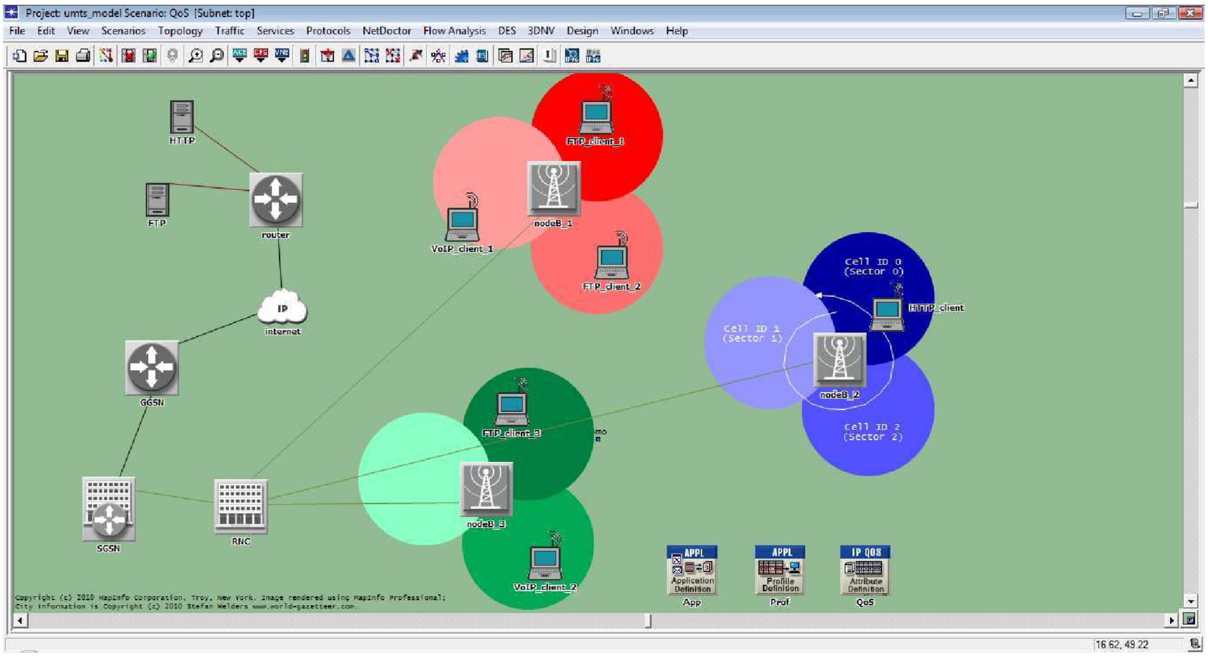


Figure 4-9 Diagram UMTS network architecture

Used objects by OPNET terminology:

ethernet_server_adv
ethernet4_slip8_gtwy
umts_ggsn_atm8_ethernet8_slip8
umts_sgsn_ethernet_ atm9_slip

[9] Using the station and SGSN nodes allow you to configure a traffic generation pattern that is not application-based. This avoids the need to use the application models when you are not interested in application-specific performance in the UMTS network.

umts_rnc_ethernet_ atm_slip
umts_node_b_3sector_adv
umts_wkstn_adv
umts_wkstn_adv (mobile)
Application Config
Profile Config
QoS Attribute Config

Used links in model: 10BaseT, PPP_DS3, ATM_adv

4.2.1 Parameters settings

Programme offer miscellaneous parameters and really useful is using default settings. For simplicity just always make changes in followed attributes. When setting parameters can be labour facilitated using by FILTER in the object palette and via *Edit Attributes* as well. Just always enter required keyword.

Applications configuration:

Component defines three rows for applied applications. After configuration assign these applications to the servers and it will govern them.

File Transfer (Heavy): Command mix (Get/Total) *100%*, Inter-Request time (seconds) *constant (5)*, File size (bytes) *constant (50000)*, Type of service *Streaming Multimedia (4)*.

Voice over IP Call (PCM Quality): Encoder scheme *GSM FR*, Voice frames per packet *2*, Type of service *Interactive Voice (6)*.

Web browsing (Heavy HTTP 1.1): Page interarrival time (seconds) *exponential (10)*, Page Properties *Small Image, Medium Image, Large Image*.

Table 4-4 shows the classification of different classes to QoS model and assignment of the various classes of applications. Interactive bond between ToS and QoS classes can be changed in the configuration of the UMTS logical channels for participating facilities.

QoS class	ToS class	Service
Conversational	Reserved (7)	
	Interactive Voice (6)	Voice
	Interactive Multimedia (5)	
Streaming	Streaming Multimedia (4)	Video
Interactive	Excellent Effort (3)	
	Standard (2)	Web browsing
Background	Background (1)	File transfer
	Best Effort (0)	

Table 4-4 Assigning grades to individual network services

Profile configuration:

Simulation profile created in this object can be assigned to individual user equipment or server for data generation and creation of network traffic. For each application is matched one profile and all parameters are identical. After completing configuration assigned profiles to the each participant or source (FTP, HTTP) by Table 4-5.

Operation mode *Simultaneous*, Start time *constant (5)*, Start time offset *uniform (3,5)*.

Domain:	Workstation:	Supported profile:	Destination:
nodeB_1	FTP_client_1	FTP	FTP server
	FTP_client_2	FTP	FTP server
	VoIP_client_1	Voice	VoIP_client_2
nodeB_2	HTTP_client	HTTP	HTTP server
nodeB_3	FTP_client_3	FTP	FTP server
	VoIP_client_2	Voice	VoIP_client_1

Table 4-5 Clients with supported profiles

Each user equipment supports respective profile, as it's name suggests. Be careful when setting up mutual communication with one another for clients/servers. Change attribute[10] is performed in *Application: Destination Preferences*.

QoS attribute configuration:

Defines attribute configuration details for protocols supported at the IP layer. These specifications can be refrenced by the individual nodes using symbolic names (character strings). For instance *Priority Queuing Profiles*, defines the criteria for the packet in order to be enqueued. Each row represents a queue see Table 4-6. If the packet finds a match (from top to bottom), the packet is enqueued whatever the following queues parameters and criteria.

Priority label:	Maximum queue size (pkts):	Classification scheme:
0 (Low)	70	Best Effort (0), Background (1)
1 (Normal)	50	Standard (2), Excellent Effort (3)
2 (Medium)	30	Streaming (4) and Interactive Multimedia (5)
3 (High)	10	Interactive Voice (6), Reserved (7)

Table 4-6 Priority queuing profile proposed based on Type of Service

A queue which has more packets than its maximum queue size will drop incoming packets according to RED/WRED policy. If an incomimg packet doesn't comply with any of the user-defined criteria, it is put in the queueu configured as the "Default Queue".

RNC configuration:

Radio network controller manages each base station Node_B and communicates with the SGSN node. The RNC received total data traffic flows from individual Node_B and further routed them to the SGSN. At this junction it is necessary to set the proper parameters of data channels for each QoS class.

Firstly notice *Type of Channel Coding*, *Coding Rate* and *PDCP Compression* in Figure 4-10.

[10] You cannot send application traffic to a UMTS station node, nor can you send traffic generated by a station node to a UMTS workstation or server node. When using the UMTS workstation nodes, use the application models to generate traffic as you would for any workstation node.

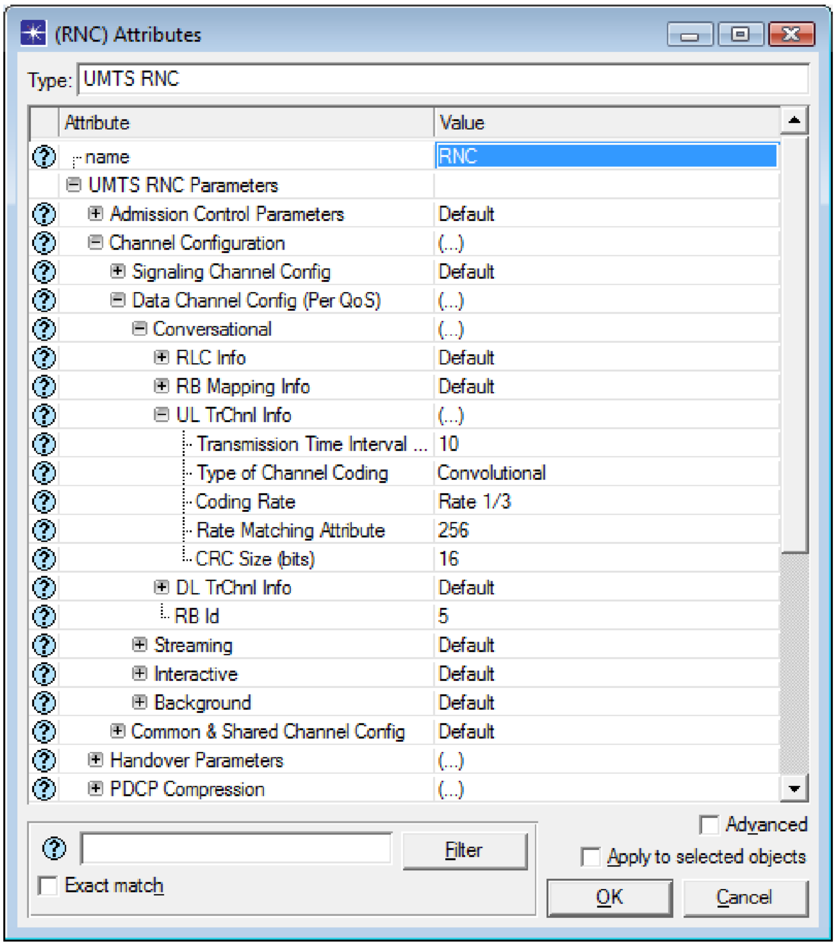


Figure 4-10 RNC data channel configuration per QoS

Timer Discard RLC procedure determine elapsed time in milliseconds before a SDU is discarded. In the transmitter, the timer is activated upon reception of an SDU from higher layer. For the proper functioning of the item set data channels each QoS class timer to the value of *3000* milliseconds.

Each participating user equipment located in the overlap multiple cells will interact with several Node Bs. Use service *Soft Handover* and set *Supported* (*Active Set Size* will be at least 2) value, which indicates that soft handovers are supported by the surrounding RNC. If not supported (*Active Set Size* will be just 1), all handovers will be performed as hard handovers.

Node B configuration:

Base station Node_B on one side communicates with the participating facilities impact on their area and on the other hand delivers data to RNC, which it manages. In both parameters are mainly related to the direct setting of individual cells.

Pathloss model used to compute the received power at the receiver. Use model *Vehicular Environment*, a typical urban and suburban environment is assumed with 10.5 m between the mean building height and the mobile antenna height, 15 m between the mobile and the diffracting edges, and 80 m as the average separation between rows of buildings. The antenna height difference between the base station antenna and the mean building rooftop height can take values between 0 and 50 m.

FACH transmission power is attribute defines the distance of the imaginary UE from the Node-B, which will be used in computing the transmission power of the FACH. In other words, this attribute defines the radius of the cell within which the FACH is expected to be heard at the requested quality level. The value of this attribute is taken into account only if the “Computation Approach” attribute is set to “Compute for Provided Distance”. Set attributes identically by Table 4-7 for each cells, consequently three times at every Node B.

Computation approach	Compute for provided distance
Transmission power (W)	1
FACH cell distance	1 kilometer

Table 4-7 FACH transmission power

<u>*UE's configuration:*</u>

In designed project are situated 6 user equipments. One of them is mobile device represents service handover. It was defined and assigned a circle trajectory to this client. During the simulation mobile facility uses web browsing service HTTP.

UMTS PDCP Compression atribute allows to configure the PDCP Header Compression feature on the UE node for uplink transmissions. Set status *Enable* at all user equipments in scenario with QoS support. If activated, IP/UDP and TCP/IP headers will be compressed for all higher layer packets arriving to the UMTS layer. Header compression ratio is computed based on the configured probability distribution function and its argument(s). Notice that just UDP/IP compression will be used for UMTS station models.

A different compression ratio can be specified for IP/UDP and TCP/IP cases. The compressable header sizes are defined as:

UDP/IP $\rightarrow$ 224 bits

TCP/IP $\rightarrow$ 320 bits

This feature has a specially useful impact on small application packets, due the big overhead IP headers add to them. For example, for GSM voice frames (33 bytes) UDP/IP add an extra overhead of 28 bytes. So, for this case, every packet arriving to UMTS layers carries ~45% overhead due to UDP/IP. Notice that just IPv4 case is covered and RTP overhead is not taken into account.

Increase *Maximum Bit Rate* for uplink and downlink to *200 kbps* at clients with voice service.

<u>*Non-QoS scenario:*</u>

Duplicate the scenario after completing all the configurations on objects. Accomplish inevitable changes on especially UEs, RNC and Application configuration to ensure QoS. The changes mainly concern ToS (best effort), PDCP (disable) and bit rates (default). Pick required attributes and launch the simulation with *320 seconds* duration.

4.2.2 Problem solving

During the networking and setting parameters, we can encounter different problems and error messages. The following list describes the most common mistakes and offers tips to solve them and troubleshooting.

`Simulation terminated by process (umts_gtp) at module (top.Europe SGSN.gtp).`
Error in GTP process model. Wrong Core Network configuration, therefore wrong model was picked. No corresponding GTP[11] module was found in a RNC or SGSN nodes. Use other SGSN node model from object palette.

`Simulation terminated by process (umts_rnc) at module (top.Europe RNC.rnc).`
Error in RNC process model. Simulation stopped due to a wrong UMTS network configuration. The link from the SGSN node to the surrounding RNC node is connected to a port reserved for legacy Node-B connections. Make sure that transceivers whose names have the prefix "nodeb" are used only to connect to Node-Bs. Check transmitter/receiver ports on the links among nodes, for instance for transmitter ports see Table 4-8.

a ↔ b	transmitter a	transmitter b
Node B ↔ RNC	nodeB_1.pt_0	RNC.nodeb_atm_tx_0
RNC ↔ SGSN	RNC.atm_tx_0_0	SGSN.atm_tx_0_0

Table 4-8 Transmitter ports on point-to-point links

`Packets with packet format (ip_dgram_v4) are not supported by link or transceiver channel.` Error in used link model. Wrong link technology, therefore wrong model was picked. Use another link model from object palette.

`Simulation terminated by process (umts_node_b) at module (top.Europe nodeB_1.node_b).` The vehicular environment pathloss model can only support the height difference between the base station and the mean building rooftop between 0 and 50 meters. Adjust the base station height or pick another pathloss model.

`Simulation terminated by process (umts_rnc) at module (top.Europe RNC.rnc).`
Error in RNC process model. Simulation stopped due to a wrong UMTS network configuration. There is an IP-enabled Node-B connected to a port on this RNC reserved for a legacy Node-B. Wrong ports interconnection. Read the simulation log for more details.

`Allocation of memory failed (top.Campus Network.server.ip).`
Virtual memory limits exceeded. Too many allocated object. Kind of software error. Defines indeed a huge operation with a large number of calculations. Redefine the attributes set in the application configuration or prohibit some profiles at each node.

`Process handle is for destroyed process (top.Campus Network.B.app).`
Required service is not supported on client. Execute change via *Edit attributes* on proper client (in this case client B). Check for sure all supported profiles.

OPNET Modeler generates a variety of logs that record errors and other significant events. The following utilities and tools may be helpful in correcting errors and solving problems too. It offer good tips and suggestions for treatment complications.

- **Traffic center** enables you to view, analyze, create, and edit all network traffic in one central window see Figure 4-11.
  During import, OPNET Modeler sometimes cannot associate all imported traffic with the appropriate network objects. This traffic is called unmapped traffic. After an import with unmapped traffic, you have the option to examine the unmapped traffic in

[11] The GGSN node models are similar to the gateway CN node model, except that they do not include the SGSN module and ATM stacks. The GPRS Tunneling Protocol (GTP) runs in the IP module on these nodes and sets up GTP tunnels between the GGSN and SGSN.

the Traffic Center and assign unmapped traffic as needed. Otherwise, if unmapped traffic exists, you can assign it at any time after the import by selecting the Show Unmapped Traffic checkbox in the Traffic Center.

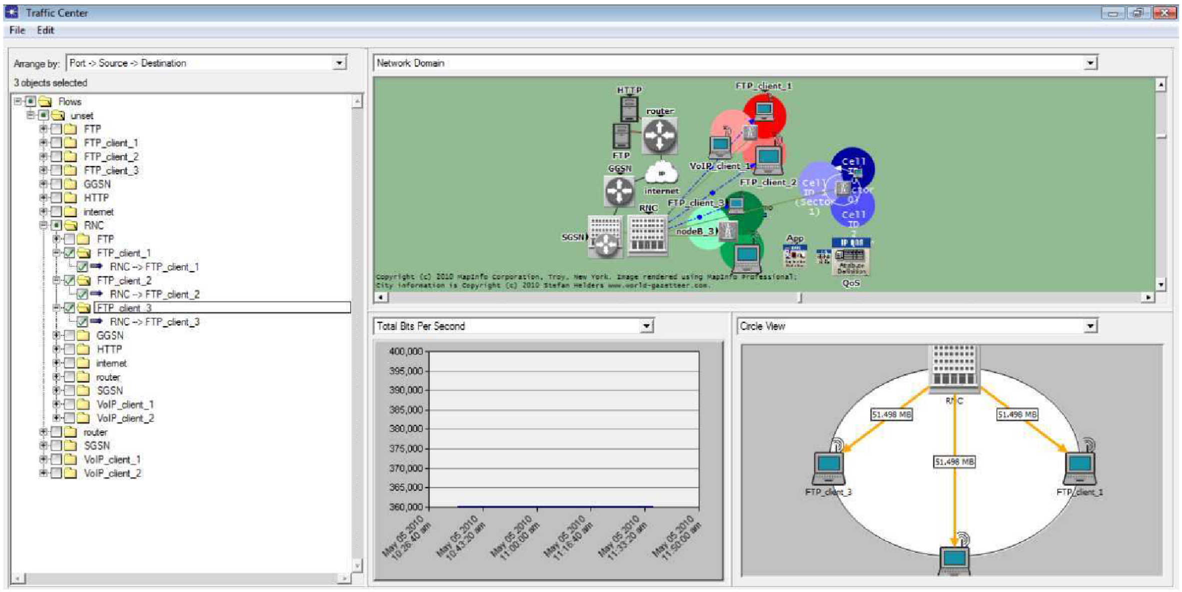


Figure 4-11 Traffic center

- **Animation viewer** assists to view "real time" animation during a simulation run or you can load and view animation history files that were recorded during previous simulation runs; in both cases, you can use different viewer operations to control how animation data is displayed.

- **Log viewer** includes the system and error logs, discrete event simulation (DES) logs, and import logs. The log viewer provides a convenient way to view all of these logs in one place. The log viewer can display both kinds of logs created by OPNET Modeler: text-based and event-based.

4.2.3 Results

Voice application:

See Figure 4-12 for comparison of the total End-to-End packet delay on the VoIP_client_2. We can note that the delay in the scenario without QoS will negatively affect the quality of communication. Although the delay is constant, but reaches a value 1.6 second. Voice service will be incomprehensible. Scenario with QoS support recorded acceptable value 150 ms.

Figure 4-12 End-to-End delay on VoIP_client_2

See on Figure 4-13 value for MOS with QoS assurance on VoIP_client_2. According to the QoS methodology for voice service is value 3 reasonable “medium” and appertain to communication quality, vice versa value 1 (blue curve) belong to unintelligible speech.

Figure 4-13 Mean opinion score value

MOS comprehensively assess the full set of parameters and return one single value. For example some of parameters such as voice performance, quality, performance data, network performance and failure. When using MOS values, call quality is measured on a scale of one to five, with one having the lowest call quality and five the highest. R-factors use a scale of zero to 100, where zero represents the lowest quality and 100 the highest.

File transfer:

Comparing results for both scenarios **umts_model-normal** and **umts_model-QoS**, see Figure 4-14 and for each FTP clients can be seen Figure 4-15. It can be noticed, that the total download response time for file transfer service fluctuate arround 7 seconds (in average) on non-QoS scenario. Value is so high, that can be expected a problems and service might be break off.
Values for QoS scenario reached suitable results approximately 1 second, FTP service can be operated without problems.

Uplink and downlink data rates are the only negotiable QoS parameters. Maximum Bit Rate Uplink/Downlink are the desired data rates for a UE. Guaranteed Bit Rate Uplink/Downlink are the lowest data rates a UE can accept. The UE will accept either maximum bit rate or guaranteed bit rate through a negotiation or renegotiation process initiated by the RNC.

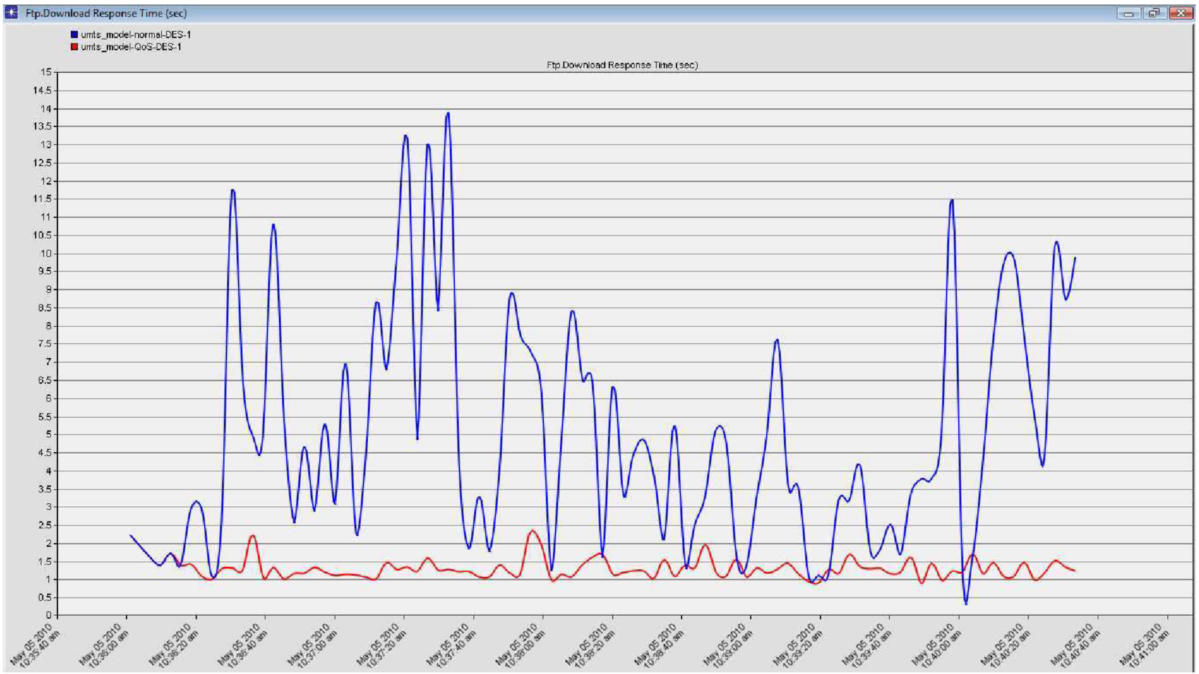


Figure 4-14 FTP total download response

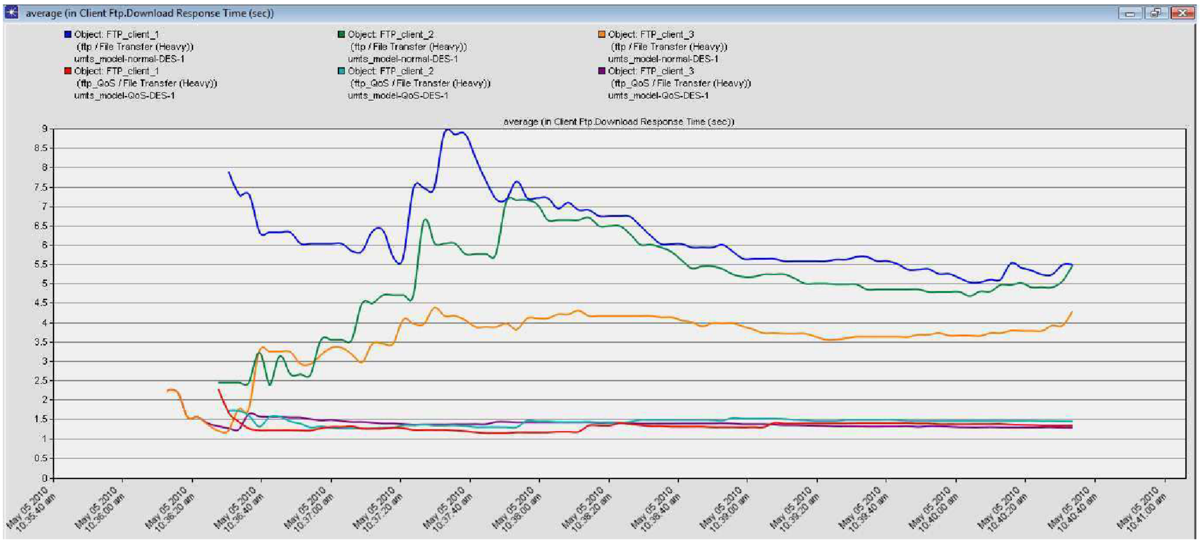


Figure 4-15 FTP download response time at clients

In scenario with QoS support has FTP_client_1 set type of service *Excellent Effort (3)*. QoS negotiation/renegotiation[12] permit whereas non-QoS scenario (in blue) prohibits it. So, when client starts FTP downloading, its RAB request cannot be admitted with the desired QoS (uplink/downlink bit rates). It will be admitted with the reduced QoS, throughput remains the same.

We can notice on Figure 4-16, that client tries to maximize his bit rate in accordance with available radio source and practicable downlink throughput.

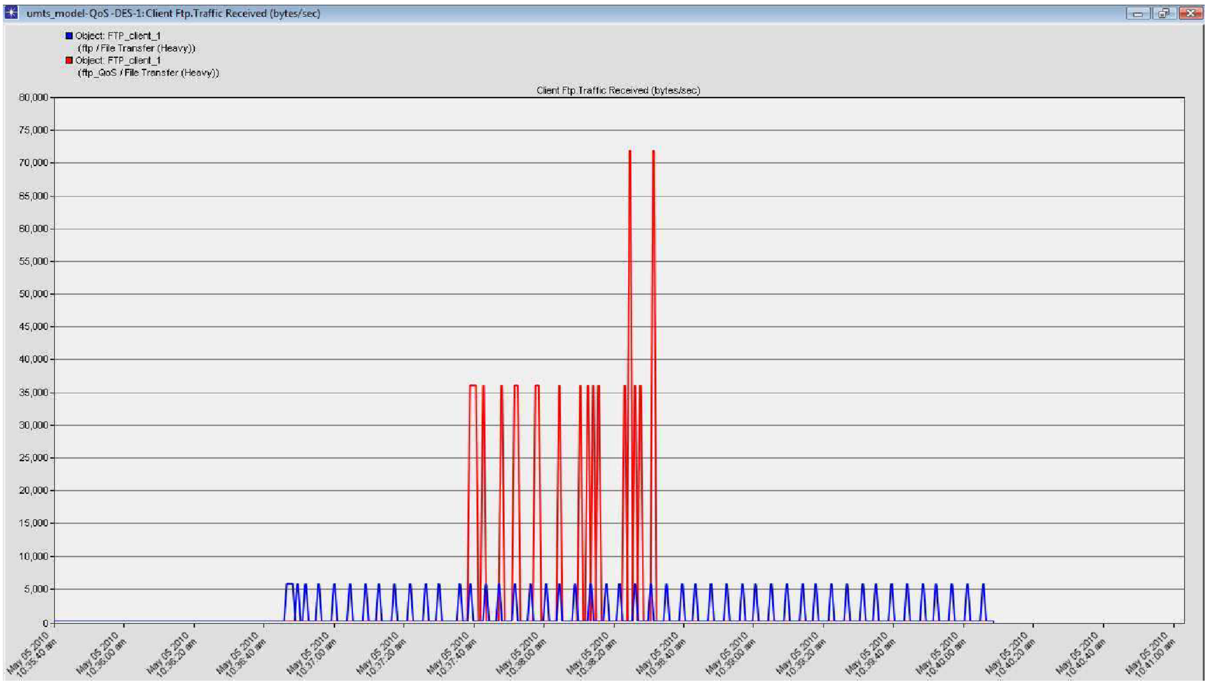


Figure 4-16 File transfer traffic received at FTP_client_1

[12] QoS negotiation/renegotiation are solely initiated by the RNC. The RNC knows all QoS parameters acceptable to the UE. The UE will always accept the negotiated/renegotiated bit rates from the RNC.

Web browsing:

HTTP_client operated by nodeB_2 downloads web pages from HTTP server. Is evident that type of service *Best Effort (0)* in non-QoS scenario shows higher response time nevertheless reached level is still good and acceptable. Blue curve reaches in average 325 miliseconds, see closely Figure 4-17.

Figure 4-17 Web response time at HTTP_client

Softer handover:

The nodeB_2 has three directional antennas covering an angle of 160 degrees and sectors/cells are positioned every 120 degrees starting from 0 degrees (0, 120, 240). So an overlap of 40 degrees is produced between adjacent cells. HTTP_client follows its circle trajectory around the nodeB_2 causing repeating softer handovers[13] from one sector to another sector.

See Figure 4-18 the pilot channel Ec/No measured by the UE for all the sectors that belong to its monitored set. As the mobile UE moves around the Node B, it measures the pilot channel signal strength coming from all three sectors. Based on this measurements it changes its active set.
When the simulation starts, UE gets the strongest signal from sector 1 so it becomes the only member of the initial active set. The UE starts moving at 40 seconds. After some time when it reaches the edge of sector 0 this gets added into the active set, the UE is now in softer-handover between sectors 0 and 1 while it remains in this position. Continues through sector 1, sector 2 and finally returns to incipient position in sector 0.

[13] Softer Handover takes place when the UE is positioned in an area where sectors are ovelapped.

Figure 4-18 Softer handover on mobile client

See Figure 4-19 the total uplink throughput on each sector in nodeB_2 domain. Notice how during softer-handover the sectors 1 and 2 reports at the same time uplink throughput coming from the UE, when that happens all duplicated packets are eliminated at the Node B and are not delivered to the RNC.

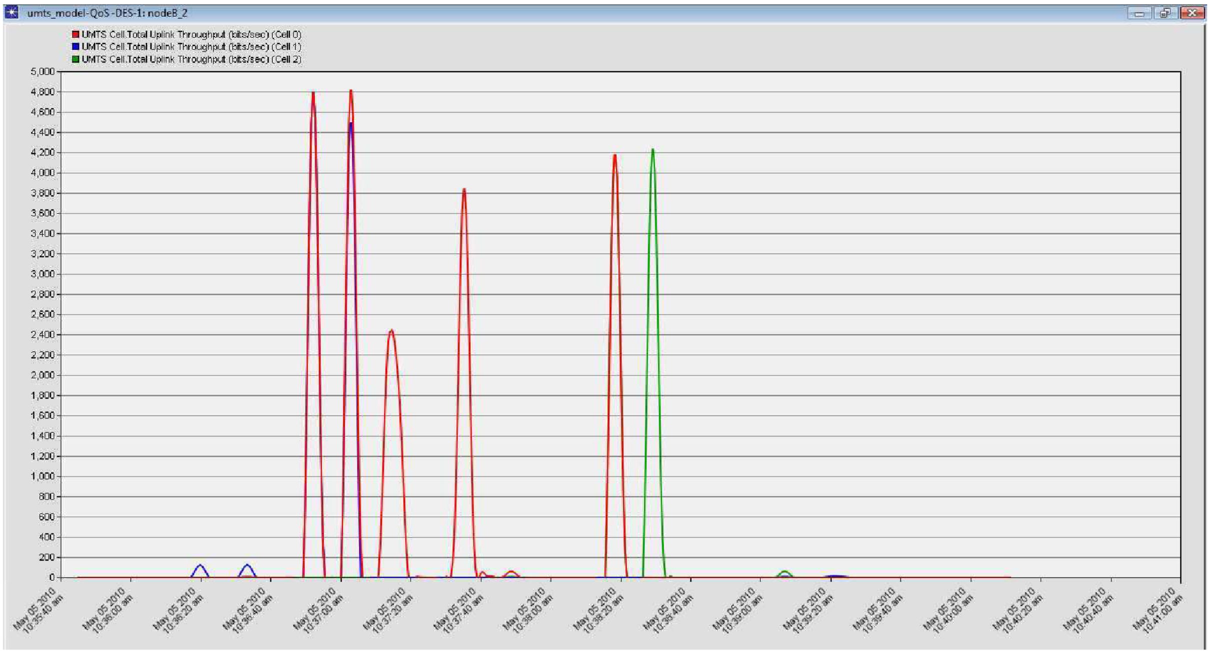


Figure 4-19 Uplink throughput on each cells

5 CONCLUSION

T he wireless network is a breakthrough 802.11e standard, because it contains support for quality of service QoS (Quality of Service), described in the Chapter 1. The well-known family IEEE 802.11 wireless standards, which arose before the 802.11e standard, was unable to distinguish between different types of traffic and all the data are transmitted as Best Effort. The main characteristics of the 802.11e are simplicity and robustness against failures due to distributed access MAC protocol. It is also important to mention that the standard is backward compatible with all earlier standards and can easily choose the method of access to media. This standard became an effective tool for supporting QoS in WLAN for a wide range of applications.

The quality of service, discussed in the Chapter 2 is an important part of all communication systems. Support for QoS can be used in real-time applications (e.g. video, voice) and in non-real-time services. When transferring data sensitive to delays, packet loss and delay variations over a wireless network, you need to ensure for packets individual treating in flows. Ensuring the required quality of service is the most important thing in the field of present data networks and the Internet. Guarantee of QoS is a key factor for further development of communication networks.

The Chapter 3 summarizes the UMTS networks, which belong to the third-generation networks and are successors of the GSM systems. Nevertheless, the core network part of the UMTS system is firmly founded on the successful GSM network, which evolved from the circuit-switched voice network into a global platform for mobile packet data services like short messaging, mobile Web browsing and mobile email access.

The network is rapidly evolving and is also a menu of some operators. It is suitable for data conferencing and provides fast data transfer. Services in this network are focused not only on voice service, but also interactive real-time services and a massive use of the Internet. The rich variety and diversity of service providers forces to ensure proper management in terms of the quality of service and determining their priorities. Manufacturers participating in establishments emphasize compatibility service, because users often use several applications at once and want to avoid the deterioration of communication. To ensure the required parameters of these various mechanisms located in the access and backbone parts procures in association with the required quality of communication along the entire route of transmission. Network design is very important in case of poor design quality of service avoiding interaction of individual applications. Ultimately, it turned out to be impaired transmission of communications features for end users.

The Chapter 4 is the practical part of this work, where the software OPNET Modeler takes place. The programme was devoloped by OPNET Technologies, Inc. for design, simulation and analysis of various network technologies. This and other similar programs, through which it is possible to model and simulate various events outside the network, are nowadays very useful because build a network without any reliable indication of the required functionality is expensive.

The aim is focused on ensuring quality and providing the required services and implement mechanisms to access devices by participants and also to the backbone of the network. The author created two model solutions for demonstrating the principles of both technologies, and also samples of laboratory tasks, which can serve as an educational tool for students. In addition, there were also set parameters with different requirements for transmission of communication network. Finally, the evaluation of the importance of QoS is done by comparing the scenarios and demonstrating for both models to support the quality of service and without it. The values of measured parameters are obvious from the graphs. All monitored parameters reported significantly better results when used with QoS.

The manual in the practical part assumes that you are familiar with mentioned protocols and that you are comfortable with using OPNET Modeler. The work also includes problems solving in the design of the UMTS model, and there are also given tips and suggestions for overcoming them. For your convenience, a brief protocols overview in the theoretical part and a list of common acronyms are included in the supplements. For more detailed information about the topic “Wireless and mobile UMTS networks simulation using QoS”, see one of the documents listed in Reference.

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List of abbreviations and acronyms

AIFS	*Arbitration Inter-Frame Space*
AIFSN	*Arbitration Inter-Frame Space Number*
AP	*Access Point*
CAP	*Controlled Access Periods*
CFP	*Contention-Free Period*
CP	*Contention Period*
CS	*Circuit Switching*
CW	*Contention Window*
DCF	*Distributed Coordination Function*
DiffServ	*Differentiated Services*
DIFS	*DCF Inter Frame Space*
DNS	*Domain Name System*
DSCP	*DS Code Point*
EDCA	*Enhanced Distribution Coordinate Access*
EDCAF	*Enhanced Distributed Channel Access Function*
EF	*Expedited Forwarding*
ETSI	*European Telecommunications Standards Institute*
FCS	*Frame Check Sequence*
GGSN	*Gateway GPRS Support Node*
GPRS	*General Packet Radio Service*
HCCA	*HCF Controlled Channel Access*
HCF	*Hybrid Coordination Function*
IETF	*Internet Engineering Task Force*
IFS	*Inter Frame Space*
LLC	*Logical Link Control*
MAC	*Media Access Control*
MPDU	*MAC Protocol Data Unit*
MSDU	*MAC Service Data Units Aggregation*
MSS	*Maximum Segment Size*
MTU	*Maximum Transmission Unit*
OSI/ISO	*International Organisation for Standardisation/Open Systems Interconnection*
PCF	*Point Coordination Function*
PDCP	*Packet Data Convergence Protocol*
PIFS	*PCF Inter-Framce Space*
PLCP	*Physical Layer Convergence Protocol*
PS	*Packet Switching*
PSDU	*PLCP Service Data Unit*
PPDU	*PLCP Protocol Data Unit*
QAP	*QoS Access Point*
QBSS	*Qos Basic Service Set*
QoS	*Quality of Service*
QSTA	*QoS Station*
RNC	*Radio Network Controller*
RNS	*Radio Network Subsystem*
RSVP	*Resource reSerVation Protocol*
RTC/CTS	*Request to Send/Clear to Send*
SGSN	*Serving GPRS Support Node*
SIFS	*Short Inter-Frame Space*
SNAP	*SubNetwork Access Protocol*
SSID	*Service Set IDentifier*

TCP	*Transmission Control Protocol*
TID	*Traffic Identifier*
TS	*Traffic Streams*
UMTS	*Universal Mobile Telecommunication System*
VoIP	*Voice over Internet Protocol*
VPN	*Virtual Private Network*
W-CDMA	*Wideband Code Division Multiple Access*