2

UMTS Services and Applications

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2.1 Introduction

2nd generation systems like GSM, were originally designed for efficient delivery of voice services. UMTS networks are, on the contrary, designed from the beginning for flexible delivery of any type of service, where each new service does not require particular network optimisation. In addition to the flexibility, the WCDMA radio solution brings advanced capabilities that enable new services. Such capabilities are:

- High bit rates theoretically up to 2 Mbps in 3GPP Release '99, and beyond 10 Mbps in 3GPP Release 5. Practical bit rates are up to 384 kbps initially, and beyond 2 Mbps with Release 5;
- Low delays with packet round trip times below 200 ms;
- Seamless mobility also for packet data applications;
- Quality of Service differentiation for high efficiency of service delivery;
- Simultaneous voice and data capability;
- Interworking with existing GSM/GPRS networks.

The WCDMA radio capabilities are described in more detail in Chapters 10, 11 and 12. These advanced radio capabilities, combined with the IP Multimedia Sub-system, IMS, allow fast introduction of new services. This chapter presents a few example UMTS services in Sections 2.2–2.5. The services are divided into person-to-person services, content-to-person services and business connectivity. Person-to-person refers to a peer-to-peer or intermediate server based connection between two persons or a group of persons. Content-to-person services are characterised by the access to information or download of content. Business connectivity refers to the laptop access to internet or intranet using WCDMA as the radio modem. The chapter further introduces IP Multimedia Sub-system, IMS, in Section 2.6 and Quality of service differentiation in Section 2.7. The cost of service delivery and the

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maximum system capacity are analysed in Section 2.8. Section 2.9 presents terminal capability classes and Section 2.10 covers location services in WCDMA.

2.2 Person-to-Person Circuit Switched Services

This section considers AMR voice, wideband AMR voice and video. These services are initially provided through the circuit switched core network in WCDMA, but they can later be provided also through the packet switched core network.

2.2.1 AMR Speech Service

The speech codec in UMTS will employ the Adaptive Multirate (AMR) technique. The multirate speech coder is a single integrated speech codec with eight source rates: 12.2 (GSM-EFR), 10.2, 7.95, 7.40 (IS-641), 6.70 (PDC-EFR), 5.90, 5.15 and 4.75 kbps. The AMR bit rates can be controlled by the radio access network. To facilitate interoperability with existing cellular networks, some of the modes are the same as in existing cellular networks. The 12.2 kbps AMR speech codec is equal to the GSM EFR codec, 7.4 kbps is equal to the US-TDMA speech codec, and 6.7 kbps is equal to the Japanese PDC codec. The AMR speech codec is capable of switching its bit rate every 20 ms speech frame upon command. For AMR mode switching, in-band signalling is used.

The AMR coder operates on speech frames of 20 ms corresponding to 160 samples at the sampling frequency of 8000 samples per second. The coding scheme for the multirate coding modes is the so-called Algebraic Code Excited Linear Prediction Coder (ACELP). The multirate ACELP coder is referred to as MR-ACELP. Every 160 speech samples, the speech signal is analysed to extract the parameters of the CELP model. The speech parameter bits delivered by the speech encoder are rearranged according to their subjective importance before they are sent to the network. The rearranged bits are further sorted, based on their sensitivity to errors and are divided into three classes of importance: A, B and C. Class A is the most sensitive, and the strongest channel coding is used for class A bits in the air interface.

During a normal telephone conversation, the participants alternate so that, on the average, each direction of transmission is occupied about 50 % of the time. The AMR has four basic functions to effectively utilise discontinuous activity:

- Voice Activity Detector (VAD) on the TX side;
- Evaluation of the background acoustic noise on the TX side, in order to transmit characteristic parameters to the RX side;
- The transmission of comfort noise information to the RX side is achieved by means of a Silence Descriptor (SID) frame, which is sent at regular intervals;
- Generation of comfort noise on the RX side during periods when no normal speech frames are received.

Discontinuous transmission (DTX) has some obvious positive implications: in the user terminal, battery life will be prolonged or a smaller battery could be used for a given operational duration. From the network point of view, the average required bit rate is reduced, leading to a lower interference level and hence increased capacity.

The AMR specification also contains error concealment. The purpose of frame substitution is to conceal the effect of lost AMR speech frames. The purpose of muting the output in the case of several lost frames is to indicate the breakdown of the channel to the user and to avoid generating possibly annoying sounds as a result of the frame substitution procedure [1] [2]. The AMR speech codec can tolerate about a 1% frame error rate (FER) of class A bits without any deterioration of the speech quality. For class B and C bits a higher FER is allowed. The corresponding bit error rate (BER) of class A bits will be about 10^{-4}.

The bit rate of the AMR speech connection can be controlled by the radio access network depending on the air interface loading and the quality of the speech connections. During high loading, such as during busy hours, it is possible to use lower AMR bit rates to offer higher capacity while providing slightly lower speech quality. Also, if the mobile is running out of the cell coverage area and using its maximum transmission power, a lower AMR bit rate can be used to extend the cell coverage area. The capacity and coverage of the AMR speech codec is discussed in Chapter 12. With the AMR speech codec it is possible to achieve a trade-off between the network's capacity, coverage and speech quality according to the operator's requirements.

2.2.1.1 AMR Source Based Rate Adaptation – Higher Voice Capacity [3]

AMR codec uses voice activity detection (VAD) together with discontinuous transmission (DTX) to optimise the network capacity and the power consumption of the mobile terminal. Active speech is coded by fixed bit rate that is selected by the radio network according to network capacity and radio channel conditions. Although the network capacity is optimised during silence periods using VAD/DTX, it can be further optimised during active speech with source controlled rate adaptation. Thus AMR codec mode is selected for each speech frame depending on the source signal characteristics, see Figure 2.1. The speech codec mode can be updated in every 20 ms frame in WCDMA.

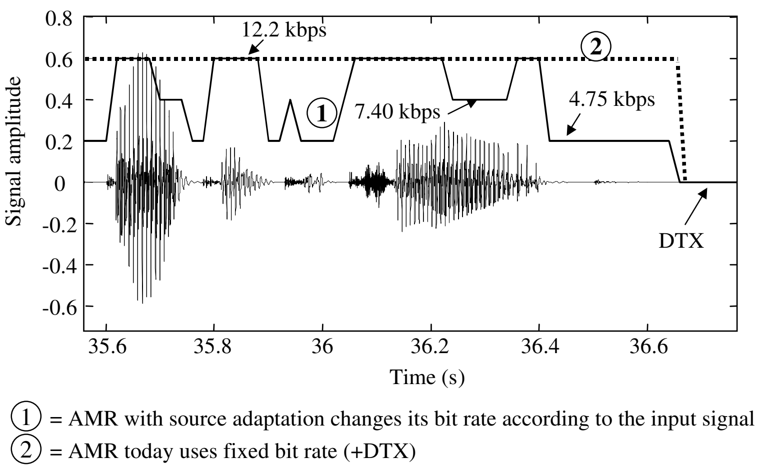


Figure 2.1. AMR source based mode selection as a function of time and speech content

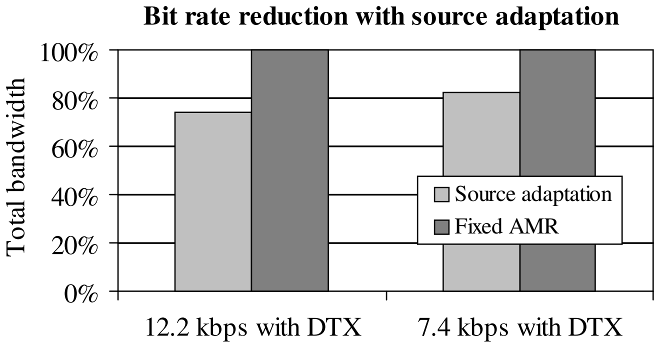


Figure 2.2. Reduction of required bit rate with equal voice quality

AMR source adaptation allows provisioning of the same voice quality with lower average bit rate. The bit rate reduction is typically 20–25% and is illustrated in Figure 2.2. The reduced AMR bit rate can be utilised to lower the required transmission power of the radio link, and it can thus further enhance AMR voice capacity. The WCDMA flexible Layer 1 allows adaptation of the bit rate and the transmission power for each 20 ms frame. The estimated capacity gain is 15–20%. The bit stream format of source adapted AMR is fully compatible with the existing fixed-rate AMR speech codec format, therefore, the decoding part is independent of source based adaptation. The AMR source based adaptation can be added as a simple upgrade to the networks to enhance WCDMA downlink capacity without any changes to the mobiles.

2.2.1.2 Wideband AMR – Better Voice Quality [4]

3GPP Release 5 introduces AMR wideband speech codec, which brings substantial voice quality enhancements compared to AMR narrowband codec or compared to standard fixed telephone line. In the case of packet switched streaming, AMR-WB is already part of 3GPP Release 4. The AMR-WB codec has also been selected by the ITU-T in the standardisation activity for a wideband codec around 16 kbps. This is of significant importance since this is the first time that the same codec is adopted for wireless as well as wireline services. This will eliminate the need for transcoding, and ease the implementation of wideband voice applications and services across a wide range of communications systems. The AMR-WB codec operates on nine speech coding bit rates between 6.6 and 23.85 kbps. The term wideband comes from the sampling rate, which has been increased from 8 kHz to 16 kHz. This allows covering twice the audio bandwidth compared to the classical telephone voice bandwidth of 4 kHz. While all the previous codecs in mobile communication operate on narrow audio bandwidth limited to 200–3400 Hz, AMR-WB extends the audio bandwidth to 50–7000 Hz. Figure 2.3 shows listening test results, where AMR-WB is compared to AMR-NB. The results are presented as mean opinion scores (MOS), where a higher number indicates better experienced voice quality. The MOS results show that AMR-WB is able to improve the voice quality without increasing the required radio bandwidth. For example, AMR-WB 12.65 kbps provides a clearly higher MOS than AMR-NB 12.2 kbps. The improved voice quality can be obtained because of higher sampling frequency.

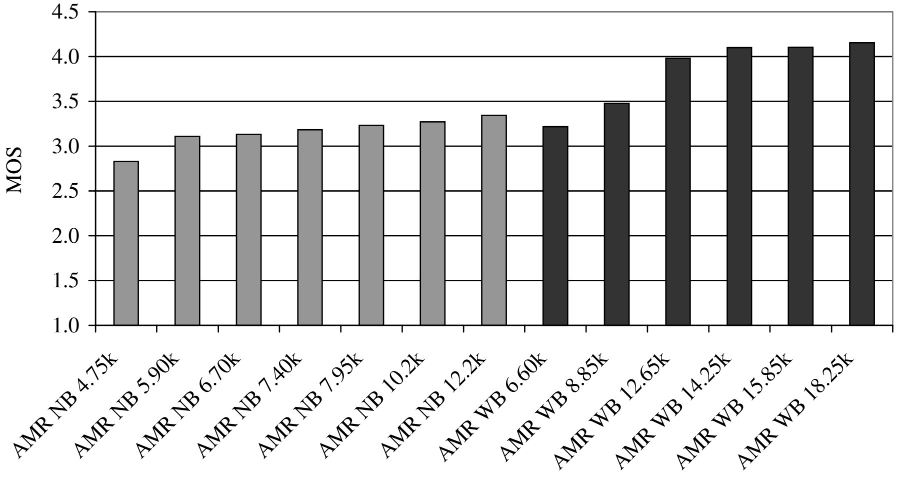


Figure 2.3. Mean opinion score (MOS) example with wideband and narrowband AMR

2.2.2 Video Telephony

Video telephony has similar delay requirements as speech services. Due to the nature of video compression, the BER requirement is more stringent than that of speech. 3GPP has specified that ITU-T Rec. H.324M should be used for video telephony in circuit switched connections and Session Initiation Protocol (SIP) for supporting IP multimedia applications, including video telephony.

2.2.2.1 Multimedia Architecture for Circuit Switched Connections

Originally Rec. H.324 was intended for multimedia communication over a fixed telephone network, i.e. PSTN. It is specified that for PSTN connections, a synchronous V.34 modem is used. Later on, when wireless networks evolved, mobile extensions were added to the specification to make the system more robust against transmission errors. The overall picture of the H.324 system is shown in Figure 2.4 [5].

H.324 consists of the following mandatory elements: H.223 for multiplexing and H.245 for control. Elements that are optional but are typically employed are H.263 video codec, G.723.1 speech codec, and V.8bis. Later, MPEG-4 video and AMR were added as optional codecs into the system. The recommendation defines the seven phases of a call: set-up, speech only, modem training, initialisation, message, end, and clearing. Level 0 of H.223 multiplexing is exactly the same as that of H.324, thus providing backward compatibility with older H.324 terminals. With a standardised negotiation procedure the terminal can adapt to the prevailing radio link conditions by selecting the appropriate error resiliency level.

V.8bis contains procedures for the identification and selection of common modes of operation between data circuit-terminating equipment (DCE) and between data terminal equipment (DTE) over general switched telephone network and leased point-to-point telephone types. The basic features of V.8bis are as follows:

- It allows a desired communication mode to be selected by either the calling or the answering station.

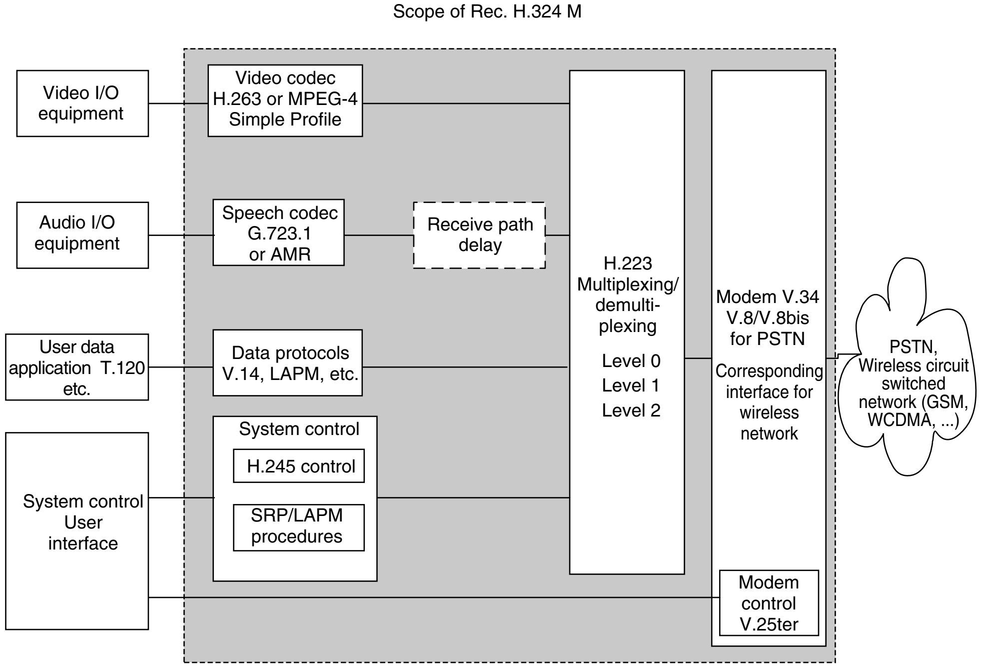


Figure 2.4. Scope of ITU Rec. H.324

- It allows terminals to automatically identify common operating modes (applications).
- It enables automatic selection between multiple terminals that share a common telephone circuit.
- It provides user-friendly switching from normal voice telephony to a modem-based communication mode.

The capabilities exchange feature of V.8bis permits a list of communication modes, as well as software applications, to be exchanged between terminals. Each terminal is therefore able to establish the modes of operation it shares with the remote station. A capability exchange between stations thus ensures, *a priori*, that a selected communication mode is possible. Attempts to establish incompatible modes of operation are thus avoided, which speeds up the application level connection.

As with the mode selection procedure, a capabilities exchange may be performed either at call set-up, automatically under the control of either the calling or the answering station, or during the course of telephony. In the latter case, on completion of the information exchange, the communication link may be configured either to return to voice telephony mode or to adopt immediately one of the common modes of communication.

V.8bis has been designed so that, when a capabilities exchange takes place in telephony mode, and the capabilities exchanged are limited to standard features, the interruption in voice communications is short (less than approximately 2 seconds) and as unobtrusive as possible.

In order to guarantee seamless data services between UMTS and PSTN, the call control mechanism of UMTS should take the V.8bis messages into account. V.8bis messages should be interpreted and converted into UMTS messages and vice versa.

One of the recent developments of H.324 is an operating mode that makes it possible to use an H.324 terminal over ISDN links. This mode of operation is defined in Annex D of the H.324 recommendation and is also referred to as H.324/I. H.324/I terminals use the I.400 series ISDN user-network interface in place of the V.34 modem. The output of the H.223 multiplex is applied directly to each bit of the digital channel, in the order defined by H.223. Operating modes are defined bit rates ranging from 56 kbps to 1920 kbps, so that H.324/I allows the use of several 56 or 64 kbps links at the same time.

H.324/I provides direct interoperability with H.320 terminals, H.324 terminals on the GSTN (using GSTN modems), H.324 terminals operating on ISDN through user substitution of I.400 series ISDN interfaces for V.34 modems, and voice telephones (both GSTN and ISDN). H.324/I terminals support H.324/Annex F (= V.140) which is for establishing communication between two multiprotocol audio-visual terminals using digital channels at a multiple of 64 or 56 kbps [6].

Figure 2.5 shows one of the concept phones for video telephony.

Figure 2.5. 3G concept phone for video telephony

2.3 Person-to-Person Packet Switched Services

2.3.1 Images and Multimedia

It is already common today to send pictures via MMS (multimedia messaging) [7, 8], which, from a user perspective, is perceived almost as an enhanced SMS service. For MMS to be successful it is important that the messages are delivered with a high reliability, while the delivery time is short enough as long as it is roughly below one minute. Since the delivery time is not crucial, it is possible to use a less stringent 3GPP quality of service class for MMS. Another important requirement from an end user point of view is that it should be possible and easy to send MMS messages at the same time as, for example, making a circuit switched call. This requires that both the mobile station and the network are able to handle multiple radio access bearers in parallel.

Although a parallel circuit switched call and MMS transmission is possible, the interactivity and picture information flow of the MMS service is limited. Another imaging service more powerful than MMS is real time video sharing; see Figure 2.6 for an illustration of this service. From a user point of view real time video sharing is about showing to the

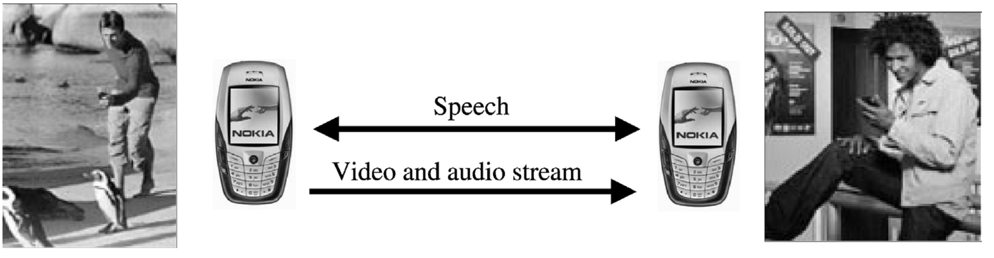


Figure 2.6. Real time video sharing

other end what is going on in your side of the phone connection. A typical usage scenario is that the communication starts out with a normal speech connection and then, when one of the parties has something interesting to show, the one-way video stream is set up. That is, the video stream is only set up when both users feel a clear need to enhance the voice connection with a one-way video connection. This differentiates the one-way video sharing service from full two-way video phone services. The real time video sharing has both professional and private use cases: sharing vacation experiences, showing real estate property for real estate brokers, and explaining what the situation is when there is a need to repair equipment.

The end user performance requirements for the real time video sharing service are that:

- Image quality and update rates should be high enough to enable 'scanning' the environment with the camera.
- Delay between taking a picture and showing it to the other side is low enough to enable true interactivity.
- It is easy and fast to set up the one-way video stream once the voice connection is available.

Note that real time video sharing has different requirements than content-to-person streaming, because in content-to-person streaming there is no or little interactivity and hence no requirements for low delays. For real time video sharing low delays are, on the other hand, crucial. A tolerable delay between taking a picture and showing it to the peer end could be in the order of some seconds (<5 s). When it comes to bit rate requirements it is very much dependent on mobile station display sizes. Based on initial results from video streaming in current networks, a lower bit rate limit for a 3.5 cm times 4 cm mobile phone display is around 40 to 64 kbps. However, note that the required bit rate to use is a non-straightforward function of the tolerable delay, the image update rates and the applied coding schemes.

From a network point of view, the one-way video streaming service has one obvious property that is different from many other proposed services: it requires a fairly high uplink bit rate. The UMTS network must be able to deliver a high and reasonably constant bit rate in order to support the low delay streaming connection as well as the voice connection if the voice connection is mapped over the packet switched domain. These bit rate and delay requirements may be met in a cost efficient way by utilising the QoS differentiation features that are available in UMTS. From a technical perspective, the peer-to-peer connections are in the packet switched domain set up by using the IMS system and by utilising the session initiation protocol (SIP) [9]. From a network and end user perspective this sets requirements on the SIP signalling, that it is fast enough not to disturb the user when setting up the additional video stream connection. Fast SIP signalling may be obtained by supporting one of multiple compression algorithms for SIP.

In Figure 2.7 a simple service evolution path is depicted starting from simple packet switched services like MMS and going towards more demanding services like video telephony. Video telephony has even tighter requirements on the delays than one-way video streaming and a one-way end-to-end delay of less than 400 ms is needed for the connection, while less than 150 ms is preferable [10].

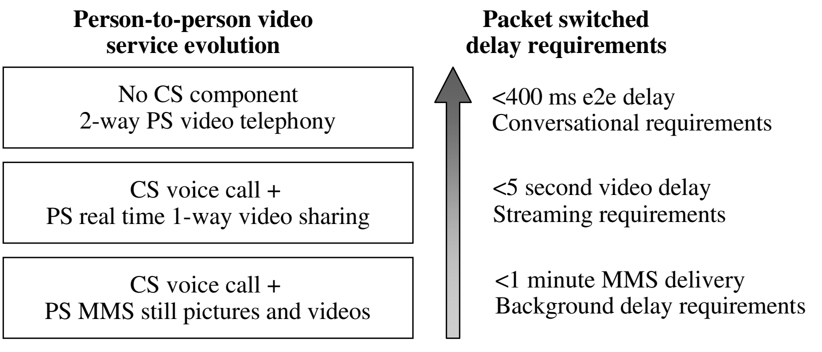


Figure 2.7. Evolution of person-to-person video service

2.3.2 *Push-to-Talk over Cellular (PoC)*

Push-to-talk over cellular (PoC) service is instant in the sense that the voice connection is established by simply pushing a single button and the receiving user hears the speech without even having to answer the call. While ordinary voice is bi-directional, the PoC service is a one directional service. The basic PoC application may hence be described as a walkie-talkie application over the packet switched domain of the cellular network. In addition to the basic voice communication functionality, the PoC application provides the end user with complementary features like, for example:

- Ad hoc and predefined communication groups;
- Access control so that a user may define who is allowed to make calls to him/her;
- 'Do-not-disturb' in case immediate reception of audio is not desirable.

With ordinary voice calls a bi-directional communication channel is reserved between the end users throughout the duration of the call. In PoC, the channel is only set up to transfer a short speech burst from one to possibly multiple users. Once this speech burst has been transferred, the packet switched communication channel can be released. This difference is highlighted in Figure 2.8.

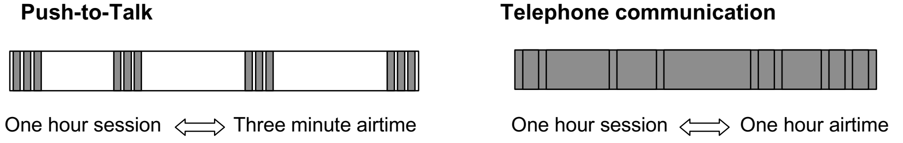


Figure 2.8. Push-to-talk versus ordinary telephone communication

The speech packets are in the PoC solution carried from the sending mobile station to the server by the OPRS/UMTS network. The server then forwards the packets to the receiving mobile stations. In the case of a one-to-many connection, the server multiplies the packets to all the receiving mobile stations. This is illustrated in Figure 2.9. The PoC service is independent of the underlying radio access network. However, as we will see later in this section as well as in Chapter 10, the characteristics of the PoC service also set tight requirements on the underlying radio access network.

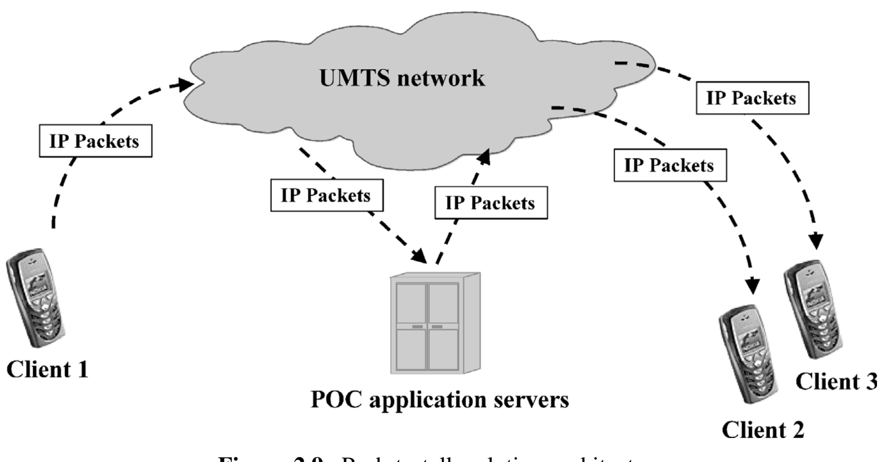


Figure 2.9. Push to talk solution architecture

In order for the PoC service to be well perceived by the end users it must meet multiple requirements. Some examples of end user requirements are:

- Simple user interface, for example, a dedicated push-to-talk button;
- High voice quality and enough sound pressure in the speaker to work also in noisy environments;
- Low delay from pressing the push-to-talk button until it is possible to start talking, called ‘start-to-talk time’;
- Low delay for the voice packets to receive the peer end, called voice through delay.

The end user is expected to be satisfied with the interactivity of the PoC service if the start-to-talk delay is around or below two seconds, while the speech round trip time should be kept lower than 1.5 seconds. The voice quality is usually evaluated by the mean opinion score (MOS) and is naturally dependent on both the mobile station and the network characteristics. A radio network that hosts PoC connections must, for example, be able to:

- Provide always on packet data connections;
- Reserve and release radio access resources fast in order to keep start-to-talk and speech round trip times low;
- Deliver a constant bit rate with low packet jitter during the duration of one speech burst.

Chapter 10 includes an investigation of the PoC service performance in a WCDMA network.

2.3.3 Voice over IP (VoIP)

The driver for Voice-over-IP, VoIP, in fixed networks has been access to low cost long distance and international voice calls. The driver for VoIP in cellular networks is rather to enable rich calls. A rich call can be defined as a real time communication session between two or more persons which consists of one or more media types. VoIP connection can be complemented with 2-way video, streaming video, images, content sharing, gaming etc., see Figure 2.10. VoIP and rich calls can be carried over WCDMA as the end-to-end network

Figure 2.10. VoIP as a building block for rich calls

delay is low enough to meet the conversational service requirements. The QoS differentiation and IP header compression are important to make an efficient VoIP service in WCDMA.

2.3.4 Multiplayer Games

We first group the existing multiplayer games into key categories based on their end user requirements. Three reasonable categories are, according to the study in [11, 12], real time action games, real time strategy games and turn based strategy games, see Figure 2.11. The

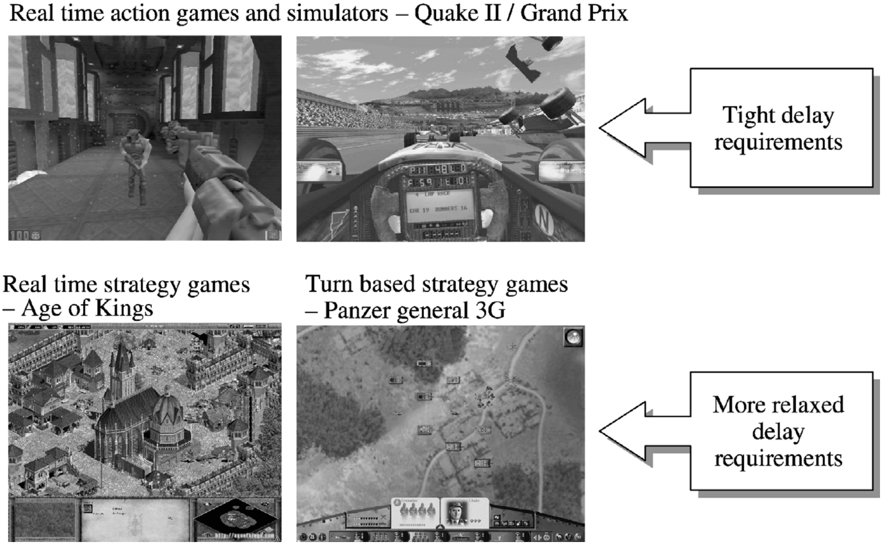


Figure 2.11. Multiplayer game classification

different categories are characterised by the properties and requirements given in Table 2.1. Note that these requirements have been derived from studies using a fixed network connection and not a cellular network connection. Although cellular networks behave somewhat differently than fixed networks, and although mobile station displays are much smaller than computer displays, the results give indications for what the maximum delay may be in order to generate a nice gaming experience for the end user.

It can be noted that for experienced players it is an advantage to have significantly lower end-to-end network delays than what is given by the requirement in Table 2.1; end-to-end network delays down to as low as 70 to 80 ms are needed to satisfy the most demanding

Table 2.1. Multiplayer game delay and bit rate requirements [11, 12]

Gaming category	End user delay requirements for average player
Real time action games	End to end network delays < 300 ms
Real time strategy games	End to end network delays < 900 ms
Turn based strategy games	End to end network delays < 40 s

users. The end-to-end network delay is particularly noticeable for the users if some users have low delays, like 70 ms, while others have higher delays, like 200 ms. Bearing in mind that today's WCDMA networks provide round trip times of 150–200 ms it is possible to provide real time strategy and turn based strategy games, and even real time action games over WCDMA.

The real time action games are constantly transmitting and receiving packets with typical bit rates of 10–20 kbps. Such bit rates can be easily delivered over cellular networks. However, these packets must be delivered with a very low delay which sets high requirements for the network performance. For real time strategy and turn based strategy games both the requirements on the bit rate and the end-to-end network delays are looser and there is more freedom on how to map these services to radio channels. This mapping is discussed in detail in Chapter 10.

2.4 Content-to-person Services

2.4.1 Browsing

During the early launch and development of WAP for mobile browsing, there were huge expectations that browsing via the mobile station would take off rapidly. Because of several reasons the take off did not happen as fast as expected. However, with better mobile station displays – resolution and colour – and with higher bit rates and increased content, the browsing experience on mobile station devices is increasing rapidly and service usage is going up. There have been several releases of the WAP protocol stack, of which the most important releases are WAP1.1 and WAP2.0; see further Figure 2.12. The WAP version

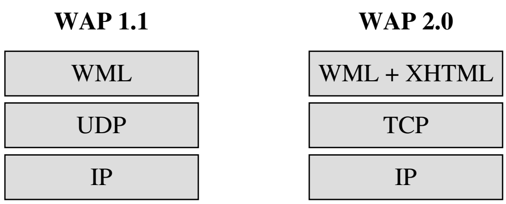


Figure 2.12. Evolution of the WAP protocol stacks

denoted WAP1.1 was approved in June 1999 and the first products based on this version were launched later in the same year. The WAP2.0 version was released in July 2001 by the WAP forum, which is currently part of the Open Mobile Alliance (OMA). The most important difference between WAP1.1 and WAP2.0 is that WAP2.0 is based on the standard Internet transport protocols (TCP/IP, HTTP/XHTML), while the WAP1.1 release utilises WAP1.1 specific transport protocols. From an end user point of view, the TCP/IP protocols provide faster download of large content size.

The focus in WAP1.1 development was to make browsing perform well in systems with large packet round trip times and with limited bit rates. That is, WAP1.1 enables the transmitter to send the packets almost at once, without waiting for connection establishment between the communication peers. This makes WAP1.1 fast for small packets over unreliable links. The weakness is that the link will usually not be fully utilised if the file to transfer is large. The decreased link utilisation lowers the end user bit rate for large files if the air interface bit rate is high.

WAP2.0 introduces standard Internet protocols to the WAP protocol stacks. Because the TCP/IP protocols have well developed link and congestion management algorithms, this makes WAP2.0 more efficient when transferring large files over radio links with high bit rates. To make TCP even more efficient for mobile systems, a particular flavour called wireless TCP (wTCP) has been defined. The wTCP protocol is based on standard TCP features, but in wTCP the support of certain features is mandatory and recommendations for parameter values have been aligned to cope with the higher packet round trip time in wireless networks. The higher link utilisation with TCP/IP for large files is illustrated in Figure 2.13 assuming WCDMA 128 kbps connection. The difference between WAP1.1 and WAP2.0 download times is quite small for small page sizes because of the low round trip time in WCDMA. A low round trip time helps standard Internet protocols perform satisfactorily over WCDMA without special optimisation.

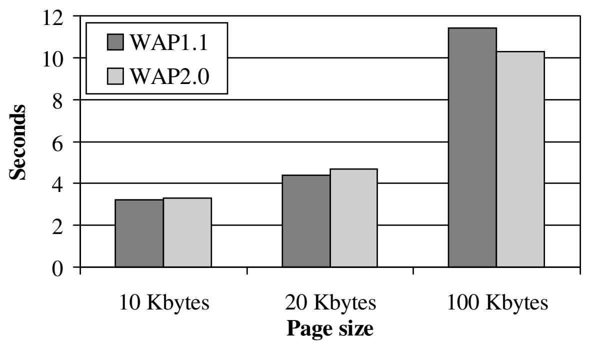


Figure 2.13. Page download time with WAP1.1 and WAP2.0

From a user perspective it is crucial that browsing is easily accessible and fast. Rough performance requirements for browsing are that the first page download time is lower than 10 s and for the second page download, lower than 4 to 7 s is preferred [10]. However, bear in mind that end user service requirements are different from market to market and also in different market segments within the same market. Another user requirement is that it should be possible to use browsing smoothly when travelling by car, train or bus. This requires efficient handling of cell reselections in order to prevent connection breaks. Because WCDMA utilises handover for packet switched data, there are no breaks at cell reselection.

From a network perspective the first page download is different from the second page download. The reason is that the first page download time may include GPRS attach, security procedures, PDP context activation and radio bearer set-up times depending on how the network and the mobile station have been configured. For the second and consecutive pages the download time will be lower because the initial set-up messages have already been sent. The second page download time is mainly limited by the basic packet round trip time, the radio channel bit rate, TCP/IP efficiency, HTTP versions and possibly also the radio bearer set-up time depending on the idle period from the last page download.

2.4.2 Audio and Video Streaming

Multimedia streaming is a technique for transferring data such that it can be processed as a steady and continuous stream. Streaming technologies are becoming increasingly important with the growth of the Internet because most users do not have fast enough access to download large multimedia files quickly. Mobile station memory may also limit the size of the downloads. With streaming, the client browser or plug-in can start displaying the data before the entire file has been transmitted.

For streaming to work, the client side receiving the data must be able to collect the data and send it as a steady stream to the application that is processing the data and converting it to sound or pictures. Streaming applications are very asymmetric and therefore typically withstand more delay than more symmetric conversational services. This also means that they tolerate more jitter in transmission. Jitter can be easily smoothed out by buffering.

Internet video products and the accompanying media industry as a whole are clearly divided into two different target areas: (1) Web broadcast and (2) video streaming on-demand. Web broadcast providers usually target very large audiences that connect to a highly performance-optimised media server (or choose from a multitude of servers) via the actual Internet. The on-demand services are more often used by big corporations that wish to store video clips or lectures to a server connected to a higher bandwidth local intranet – these on-demand lectures are seldom used simultaneously by more than hundreds of people.

Both application types use basically similar core video compression technology, but the coding bandwidths, level of tuning within network protocol use, and robustness of server technology needed for broadcast servers differ from the technology used in on-demand, smaller-scale systems. This has led to a situation where the few major companies developing and marketing video streaming products have specialised their end user products to meet the needs of these two target groups. Basically, they have optimised their core products differently: those directed to the ‘28.8 kbps market’ for bandwidth variation-sensitive streaming over the Internet and those for the 100–7300 kbps intranet market.

At the receiver the streaming data or video clip is played by a suitable independent media player application or a browser plug-in. Plug-ins can be downloaded from the Web, usually free of charge, or may be readily bundled to a browser. This depends largely on the browser and its version in use – new browsers tend to have integrated plug-ins for the most popular streaming video players.

In conclusion, a client player implementation in a mobile system seems to lead to an application-level module that could handle video streams independently (with independent connection and playback activation) or in parallel with the browser application when the service is activated from the browser. The module would interface directly to the socket interface of applied packet network protocol layers, here most likely UDP/IP or TCP/IP.

Example terminals supporting streaming services are shown in Figure 2.14.

2.4.3 Content Download

Content download examples are shown in Figure 2.15: application downloads, ringing tone downloads, video clips and MP3 music. The content size can vary largely from a few kB

Figure 2.14. Example streaming terminals

Applications games | Ringing tones | Video clips | MP3 music

Figure 2.15. Example content download

ringing tones to several MB music files. The download times should preferably be low, which puts high requirements on the radio bit rate, especially for the large downloads with several 100 kB.

2.4.4 Multimedia Broadcast Multicast Service, MBMS

A new service introduced in 3GPP Release 6 specifications is Multimedia Broadcast Multicast Service (MBMS). There are two high level modes of operation in MBMS, as given in [13]

1. Broadcast mode, which allows sending audio and video. The already existing Cell Broadcast Service (CBS) is intended for messaging only. The broadcast mode is expected to be a service without charging and there are no specific activation requirements for this mode.
2. Multicast mode allows sending multimedia data for the end users that are part of a multicast subscription group. End users need to monitor service announcements regarding service availability, and then they can join the currently active service. From the network point of view, the same content can be provided in a point-to-point fashion if there are not enough users to justify the high power transmission. A typical example in

3GPP has been the sport results service where, for example, ice hockey results would be available as well as video clips of the key events in different games of the day. Charging is expected to be applied for the multicast mode.

From the radio point of view, MBMS is considered an application independent way to deliver the MBMS User Services, which are intended to deliver to multiple users simultaneously. The MBMS User Services can be classified into three groups as follows [14]:

1. Streaming services, where a basic example is audio and video stream;
2. File download services;
3. Carousel service, which can be considered as a combination of streaming and file download. In this kind of service, an end user may have an application which is provided data repetitively and updates are then broadcast when there are changes in the content.

For MBMS User Services, an operator controls the distribution of the data. Unlike CBS, the end user needs first to join the service and only users that have joined the service can see the content. The charging can then be based on the subscription or based on the keys which enable an end user to access the data. The MBMS content can be created by the operator itself or by a third party and, as such, all the details of what an MBMS service should look like will not be specified by 3GPP, but left for operators and service providers. One possible MBMS high level architecture is shown in Figure 2.16, where the IP multicast network refers here to any server providing MBMS content over the Internet.

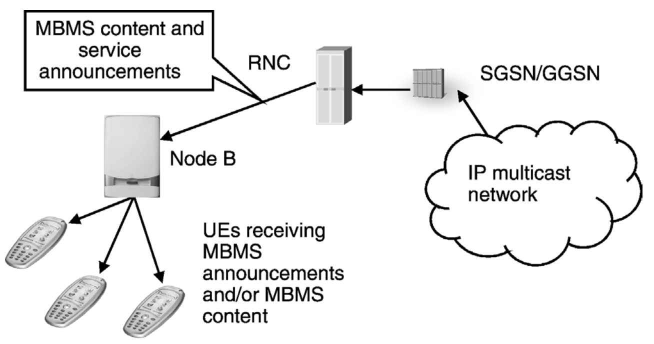


Figure 2.16. Example MBMS high level architecture

The example data rates in [14] range from the 10 kbps text-based information to the 384 kbps video distribution on MBMS. The codecs are expected to be the current ones – such as AMR for voice – to ensure a large interoperability base for different terminals for the services being provided. The MBMS causes changes mostly to Layer 2/3 protocols as described in Chapter 7 in more detail.

2.5 Business Connectivity

Business connectivity considers access to corporate intranet or to Internet services using laptops. We consider shortly two aspects of business connectivity: end-to-end security and the effect of radio latency to the application performance. End-to-end security can be obtained using Virtual Private Networks, VPN, for the encryption of the data. One option is to have a VPN client located on the laptop and the VPN gateway in the corporate premises. Such an approach is often used by large corporates that are able to obtain and maintain required equipment for the remote access service. Another approach uses a VPN connection between the mobile operator core site and the company intranet. The mobile network uses standard UMTS security procedures. In this case the company only needs to subscribe to the operator's VPN service and obtain a VPN gateway. These two approaches are illustrated in Figure 2.17.

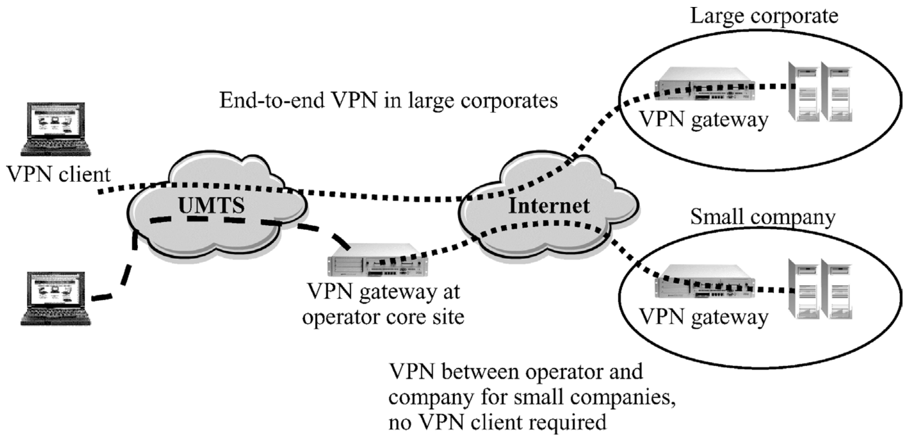


Figure 2.17. Virtual private network architectures

The business connectivity applications can be, for example, web browsing, email access or file download. The application performance should preferably be similar to the performance of DSL or WLAN. The application performance depends on the available bit rate but also on the network latency. The network latency is here measured as the round trip time. The round trip time is the delay of a small IP packet to travel from the mobile to a server and back. The effect of the latency is illustrated using file download over Transmission Control Protocol, TCP, in Figure 2.18. The download process includes TCP connection establishment and file download, including TCP slow start. The end user experienced bit rate is defined here as the download file size divided by the total time. The delay components are illustrated in Figure 2.19.

The user experienced bit rates with round trip times between 0 and 600 ms are shown in Figure 2.20. This figure assumes that a dedicated channel with 384 kbps already exists and no channel allocation is required. The curves show that a low round trip time is beneficial, especially for small file sizes, due to TCP slow start. WCDMA round trip time is analysed in detail in Chapter 10 and it is typically 150–200 ms. Figure 2.21 shows the download time with different round trip times. The download time of a less than 100 kB file is below 3 s as

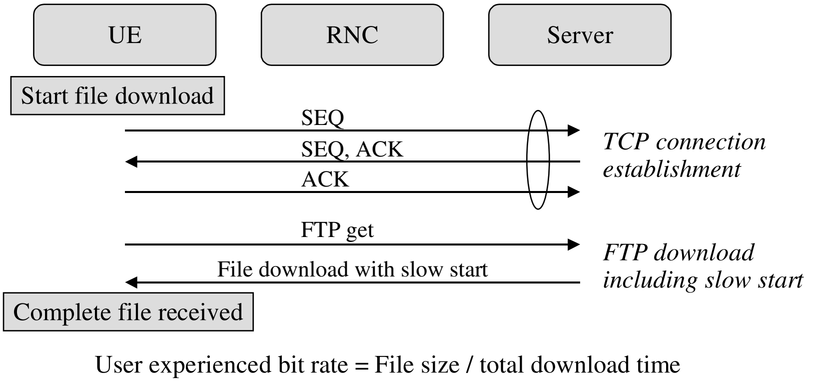


Figure 2.18. Example signalling flow in file download using TCP

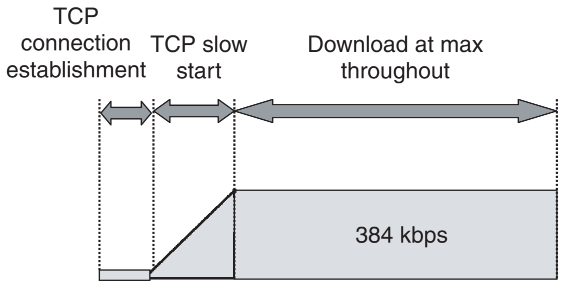


Figure 2.19. Example file download using TCP

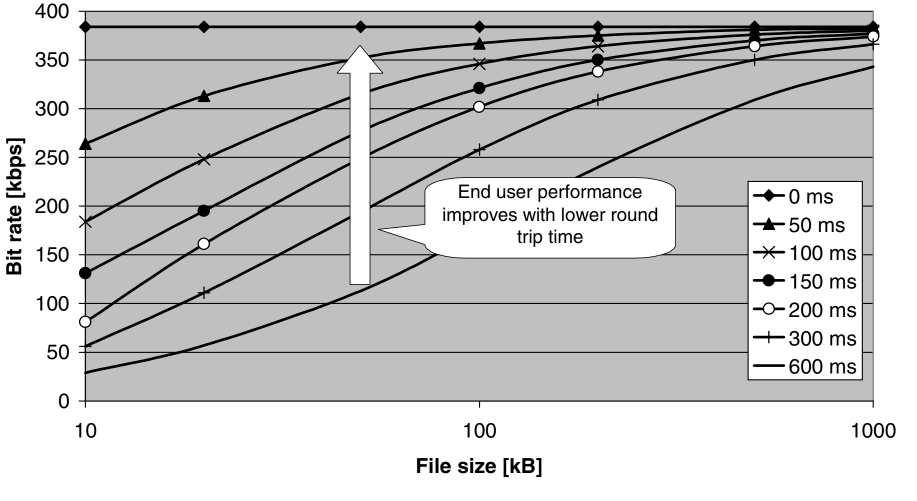


Figure 2.20. Effect of round trip time to the user experienced bit rate with Layer 1 bit rate of 384 kbps

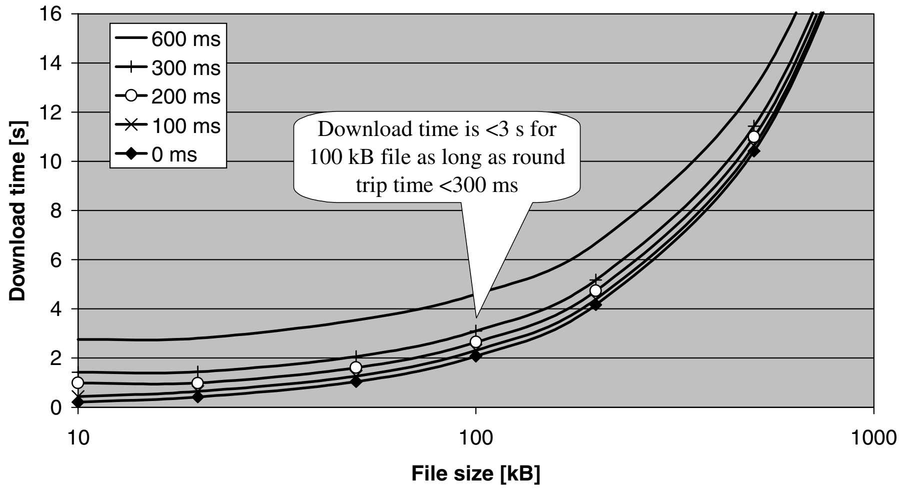


Figure 2.21. Effect of round trip time on the download times with Layer 1 bit rate of 384 kbps

long as the round trip time is below 300 ms. Low round trip time will be more relevant if we need to download several small files using separate TCP sessions.

The application performance is also affected by the application protocol, e.g. HTTP 1.1 vs HTTP 1.0 in web browsing. A web page typically consists of several objects: text and a number of pictures. In the case of HTTP 1.0 each object is downloaded in a separate TCP session, while for HTTP 1.1 all objects from the same server can be downloaded in one TCP connection. The difference is depicted in Figure 2.22. It is beneficial to use HTTP 1.1 to minimise the effect of TCP slow start on the application performance.

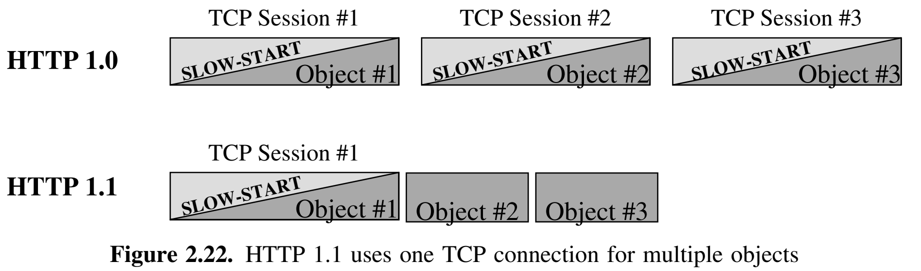


Figure 2.22. HTTP 1.1 uses one TCP connection for multiple objects

2.6 IP Multimedia Sub-system, IMS

IP Multimedia Sub-system, IMS, allows operators to provide their subscribers with multimedia services that are built on Internet applications and protocols [15]. IMS enables IP connectivity between users using the same control and charging mechanisms. The basic session initiation capabilities provided by SIP protocol are utilised to establish peer-to-peer sessions. The IMS concept is shown in Figure 2.23.

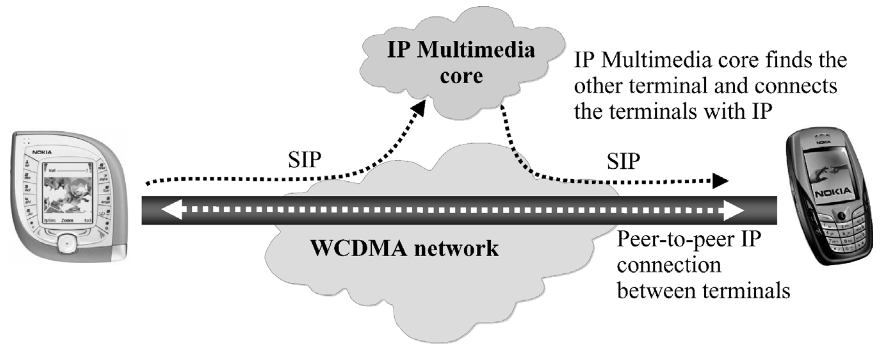


Figure 2.23. Basic principles of the IP Multimedia Sub-system

IMS provides the means for network operators to maintain their role in the value chain by providing new multimedia services and predictable end user performance. The same platform can be used in both real time services, like VoIP, and non-real time services, like content sharing. The IMS network elements are introduced in Chapter 5.

2.7 Quality of Service Differentiation

Chapter 10 covers end-to-end application performance assuming that the system load is reasonably low. When the system load gets higher, it becomes important to prioritise the different services according to their requirements. This prioritisation is called QoS differentiation. 3GPP QoS architecture is designed to provide this differentiation [16]. The terminology is shown in Figure 2.24.

The most relevant parameters of the four UMTS QoS classes are summarised in Table 2.2. The main distinguishing factor between the four traffic classes is how delay-sensitive the traffic

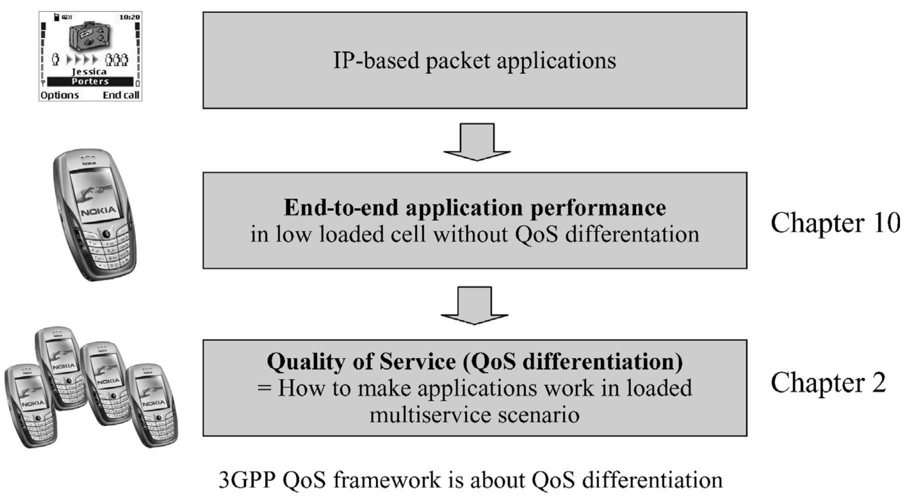


Figure 2.24. Definition of quality of service differentiation

Table 2.2. UMTS QoS classes and their main parameters

	Conversational class	Streaming class	Interactive class	Background class
Transfer delay	80 ms –	250 ms –	—	—
Guaranteed bit rate (kbps)	Up to 2 Mbps	Up to 2 Mbps	—	—
Traffic handling priority	—	—	1,2,3	—
Allocation/retention priority	1,2,3	1,2,3	1,2,3	1,2,3

is: the conversational class is meant for very delay-sensitive traffic, while the background class is the most delay-insensitive. There are, further, three different priority categories, called allocation/retention priority categories, within each QoS class. Interactive has also three traffic handling priorities. Conversational and streaming class parameters also include the guaranteed bit rate and the transfer delay parameters. The guaranteed bit rate defines the minimum bearer bit rate that UTRAN must provide and it can be used in admission control and in resource allocations. The transfer delay defines the required 95th percentile of the delay. It can be used to define the RLC operation mode (acknowledged, non-acknowledged mode) and the number of retransmissions.

The conversational class is characterised by low end-to-end delay and symmetric or nearly symmetric traffic between uplink and downlink in person-to-person communications. The maximum end-to-end delay is given by the human perception of video and audio conversation: subjective evaluations have shown that the end-to-end delay has to be less than 400 ms. The streaming class requires bandwidth to be maintained like conversational class but streaming class tolerates some delay variations that are hidden by dejitter buffer in the receiver. The interactive class is characterised by the request response pattern of the end user. At the message destination there is an entity expecting the message (response) within a certain time. The background class assumes that the destination is not expecting the data within a certain time.

UMTS QoS classes are not mandatory for the introduction of any low delay service. It is possible to support streaming video or conversational Voice over IP from an end-to-end performance point of view by using just background QoS class. QoS differentiation becomes useful for the network efficiency during high load when there are services with different delay requirements. If the radio network has knowledge about the delay requirements of the different services, it will be able to prioritise the services accordingly and improve the efficiency of the network utilisation. The qualitative gain of the QoS differentiation is illustrated in Figure 2.25. Considerable efficiency gains can be obtained in Step 2 just by introducing a few prioritisation classes within interactive or background class by using allocation and retention parameters, ARP. The pure prioritisation in packet scheduling is not alone enough to provide full QoS differentiation gains. Users within the same QoS and ARP class will share the available capacity. If the number of users is simply too high, they will all suffer from bad quality. In that case it would be better to block a few users to guarantee the quality of the existing connections, like streaming videos. That is provided in Step 3 in Figure 2.25 with guaranteed bit rate streaming. The radio network can estimate the available radio capacity and block an incoming user if there is no room to provide the required bandwidth without sacrificing the quality of the existing connections. Finally Step 4 allows further differentiation between guaranteed bit rate services with different delay

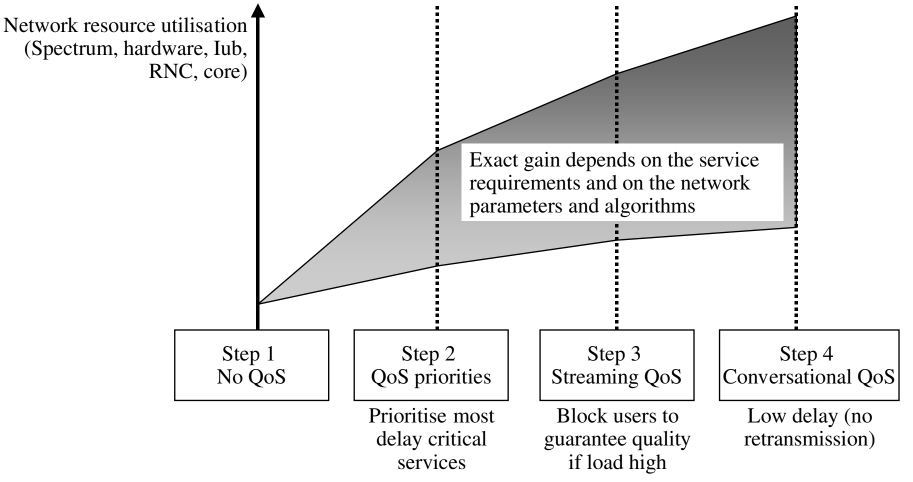


Figure 2.25. Qualitative gain illustration for QoS differentiation

requirements. If the delay requirements are known, the WCDMA RAN can allocate suitable radio parameters – like retransmission parameters – for the new bearer.

An example QoS differentiation scheme is shown in Figure 2.26 with ten different QoS categories: six guaranteed bit rate categories and four non-real time categories. It is assumed in this case that traffic handling priority is equal to allocation and retention priority, and there is no prioritisation within the background class.

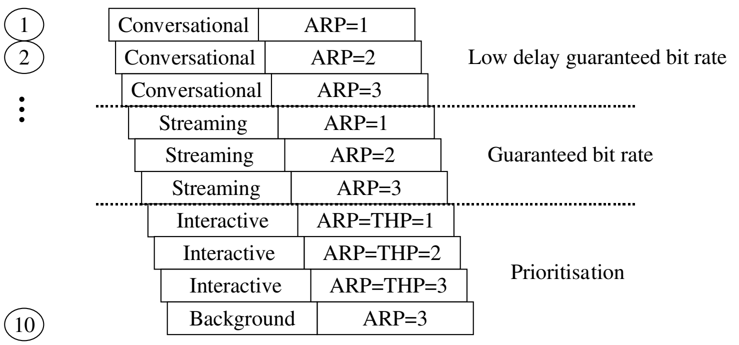


Figure 2.26. Example ten categories by taking a subset of UMTS QoS classes

The next three figures illustrate Steps 1, 2 and 4. Figure 2.27 shows an example where all the services have the same QoS parameters and the same treatment. In this case, all services share the network resources equally: they get the same bit rate and experience the same delay. The network dimensioning must be done so that this bit rate or delay fulfils the most stringent requirements of the services provided in the network. The background type of service, like sending of MMS, will get the same quality, which is unnecessarily good and

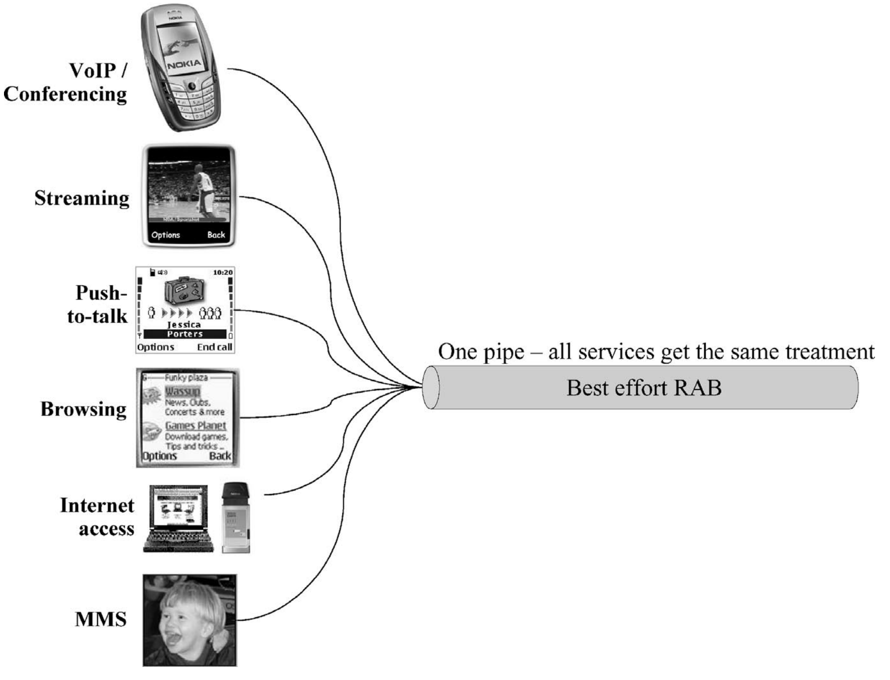


Figure 2.27. No QoS differentiation – all services use the same QoS parameters

wastes network resources. Figure 2.28 shows the case where there are three different pipes with QoS prioritisation in packet scheduling. This approach already provides QoS differentiation and makes the network dimensioning requirements less stringent. Figure 2.29 adds two further pipes with guaranteed bit rates.

The layered architecture of a UMTS bearer service is depicted in Figure 2.30; each bearer service on a specific layer offers its individual services using those provided by the layers below. The QoS parameters are given by the core network to the radio network in radio access bearer set-up.

Figure 2.31 illustrates the mechanisms to define the QoS parameters in radio access bearer set-up.

1. The UE can request QoS parameters. In particular, if the application requires guaranteed bit rate streaming or conversational class, it has to be requested by UE, otherwise, it cannot be given by the network.
2. The access point node, APN, in GGSN can give QoS parameters according to operator settings. Some services may be accessed via certain APNs. That allows the operator to control the QoS parameters for different services and makes it also possible to prioritise operator hosted services compared to accessing other services.
3. The home location register, HLR, may contain subscriber specific limitations for the QoS parameters.
4. The WCDMA radio network must be able to provide the QoS differentiation in packet handling.

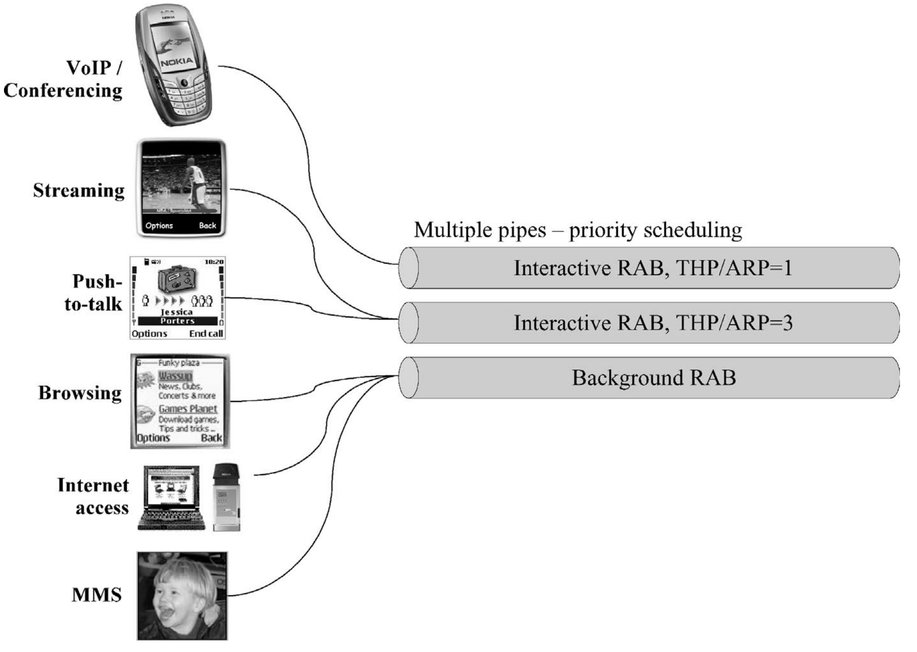


Figure 2.28. QoS prioritisation used with three classes

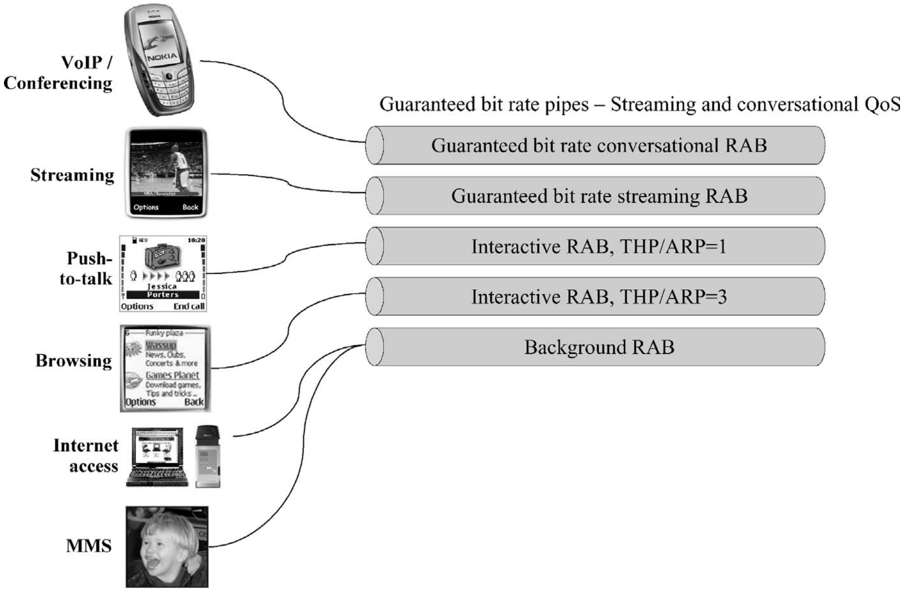


Figure 2.29. QoS differentiation with two guaranteed bit rate classes and three classes for non-real time prioritisation

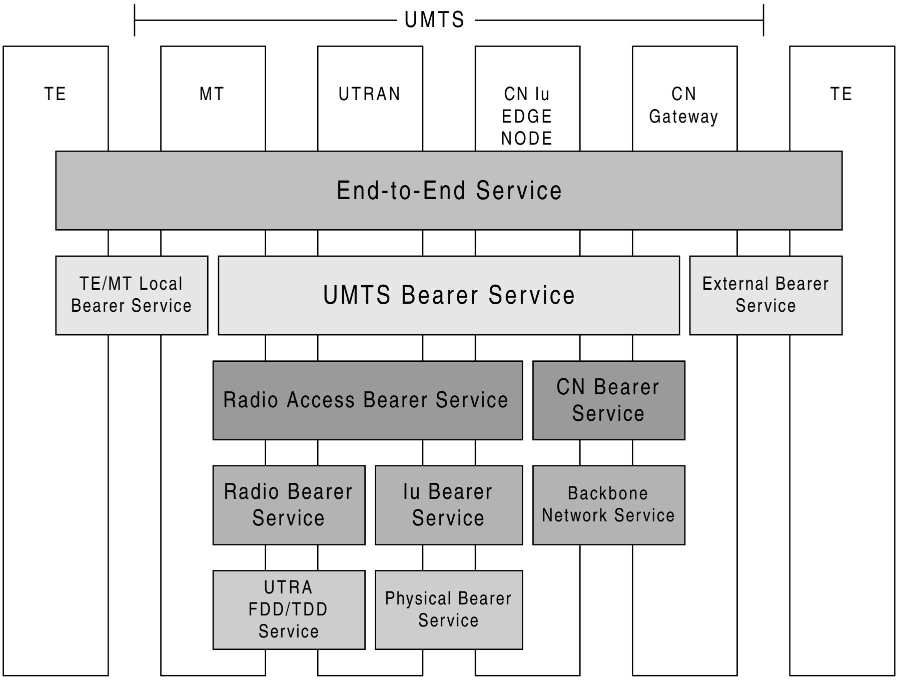
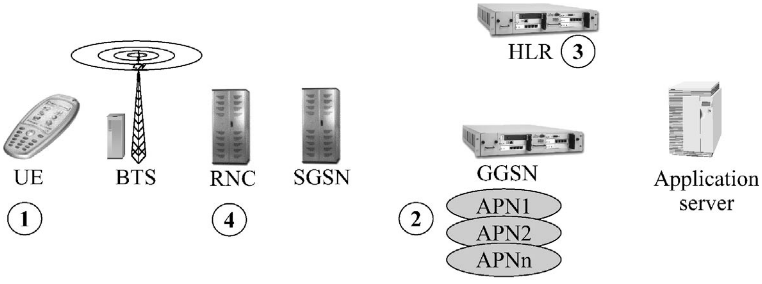


Figure 2.30. Architecture of a UMTS bearer service



1 = UE must request QoS class if it wants streaming or conversational QoS

2 = Different APNs in GGSN can be used to provide different QoS

3 = User specific QoS limitations can be defined in HLR

4 = Radio access network must be able to provide QoS differentiation

Figure 2.31. The role of UE, GGSN and HLR in defining QoS class

2.8 Capacity and Cost of Service Delivery

This section considers the maximum capacity of the radio network in delivering new services and estimates the cost of the service delivery from the UMTS network equipment point of view.

2.8.1 Capacity per Subscriber

The maximum capacity per subscriber for data and for voice traffic is presented below. The data traffic is presented as the maximum amount of downloaded megabytes (MB) of data per average subscriber per month. The voice traffic is presented as the maximum mobile-to-mobile voice minutes per average subscriber per month. The following assumptions are used in the calculations:

- WCDMA carrier capacity is 800 kbps/cell or 80 voice channels/cell, see Chapter 12 for more details;
- High-Speed Downlink Packet Access, HSDPA carrier capacity is 2000 kbps/cell, see Chapter 11 for more details;
- Cell capacity utilisation is 80 % during busy hours;
- Busy hour carries 20 % of daily traffic.
- There are 1000 subscribers per site;
- There are three sectors per site;

The percentage of the traffic carried by the busy hour represents how equally the traffic is distributed during the day. That number is affected by the pricing schemes. Attractive evening or weekend pricing schemes can make the traffic distribution more equal (busy hour carries less than 20 % of the traffic) and more traffic can be carried by the same network. The assumption of 1000 subscribers is a typical average figure for large network operators. More subscribers per site can be found in dense areas. The capacity per subscriber per day can be calculated as follows with a 10 MHz WCDMA 2+2+2 site configuration:

$$\begin{aligned} &\frac{0.8\,\text{Mbps/cell}}{8\,\text{bits/byte}} \cdot 6\,\text{cells} \cdot 3600\,\text{s/hour}\,\frac{80\,\%}{20\,\%}\,\frac{1}{1000\,\text{subs}} \cdot 30\,\text{days/month} \\ &\quad = 260\,\text{MB/sub/month} \end{aligned} \tag{2.1}$$

The maximum data capacity with WCDMA 10 MHz spectrum allocation is 260 MB/sub/month and with HSDPA 650 MB/sub/month. The voice capacity is up to 1725 minutes/sub/month. The data capacities are shown in Figure 2.32 and voice capacities in Figure 2.33. Since data and voice share the same spectrum, the final capacity depends on the traffic share between data and voice traffic. With a 50/50 split, the capacity could be 325 MB + 862 voice minutes per month. These capacity numbers indicate that there is a lot of traffic growth potential with WCDMA on top of today's traffic numbers. Global average voice traffic is currently 150–200 minutes per month.

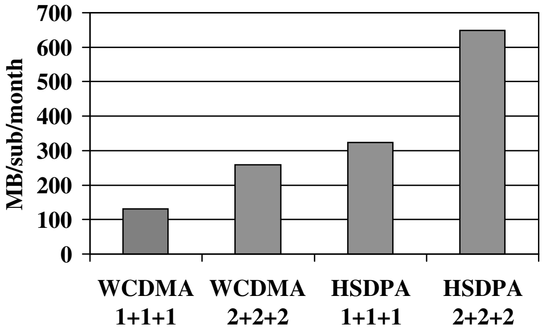


Figure 2.32. Data capacity MB/subscriber/month

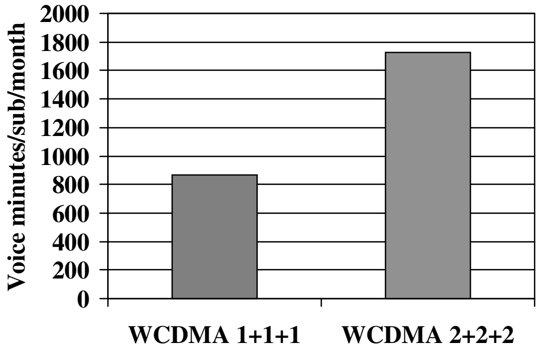


Figure 2.33. Voice capacity minutes/subscriber/month

Most UMTS operators also have GSM900/1800 spectrum that can provide additional capacity for voice and data services. Also, new spectrum for third generation systems will be available at around 2.6 GHz. Chapter 1 gives an overview of the spectrum.

2.8.2 Cost of Capacity Delivery

This section presents cost estimates of delivering megabytes of data or voice minutes over the WCDMA mobile network. The target is to present the calculation methods and to show approximate cost levels. The cost numbers include the depreciation of the radio and the core network capital expenditures (capex) without any implementation costs. The following components are included: base stations, radio transmission, RNC, core network and operations solutions. The price of all this equipment is calculated per transceiver unit (TRX). The assumed per TRX prices are between 15 and 40 k€. The pricing depends on a number of factors and, therefore, a large scale is used. A capex depreciation period of six years is assumed. The mobile network delivery cost per downloaded MB in € can be calculated as follows:

$$\frac{\text{Price per TRX}[€]}{\frac{0.8\,\text{Mbps}}{8\,\text{bits/byte}} \cdot \frac{3600\,\text{s/hour} \cdot 80\,\%}{2\,\%} \cdot 365\,\text{days/year} \cdot 6\,\text{years}} \tag{2.2}$$

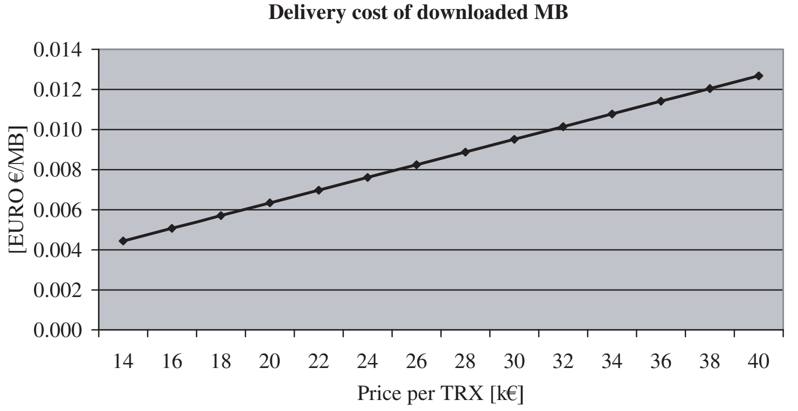


Figure 2.34. Delivery cost of downloaded data MB

The delivery cost for data is shown in Figure 2.34 and for voice in Figure 2.35. The results show that it is possible to push the data delivery capex down to approximately 0.01 €/MB, that means 1 eurocent/MB, and a voice minute below 0.2 eurocent/minute.

The high busy hour utilisation of 80 % can be used for the case when additional capacity is built. If we calculate the delivery cost over the whole network, the busy hour utilisation will be lower on average because parts of the sites are built to provide coverage and they do not collect high traffic volumes.

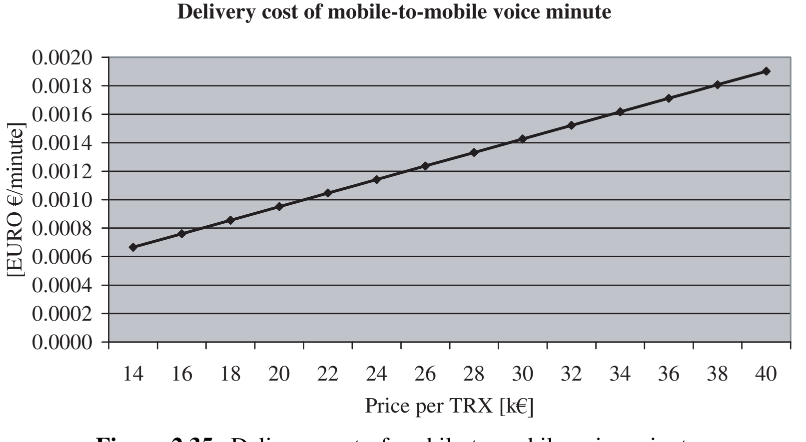


Figure 2.35. Delivery cost of mobile-to-mobile voice minute

The mobile network capex depreciation represents only part of the operator costs. Other costs include, for example, leased line transmission costs, interconnection fees, customer acquisition, advertising and customer care. Therefore, the sales prices cannot be as low as the presented production cost figures which only include capex depreciation. The capacity and the cost calculations above still demonstrate that WCDMA is able to deliver high amounts of traffic with reasonable cost, which are key requirements for enabling new services.

2.9 Service Capabilities with Different Terminal Classes

WCDMA does not use the same principle as GSM with terminal class mark. WCDMA terminals shall tell the network, upon connection set-up, a larger set of parameters indicating the radio access capabilities of the particular terminal. These capabilities determine, for example, the maximum user data rate supported in a particular radio configuration, given independently for the uplink and downlink directions. To provide guidance on which capabilities should be applied together, reference terminal radio access capability combinations have been specified in 3GPP standardisation, see [17]. The following reference combinations have been defined for 3GPP Release '99:

- 32 kbps class. This is intended to provide a basic speech service, including AMR speech as well as some limited data rate capabilities up to 32 kbps.
- 64 kbps class. This is intended to provide a speech and data service, with simultaneous data and AMR speech capability.
- 144 kbps class. This class has the air interface capability to provide, for example, video telephony or various other data services.
- 384 kbps class is being further enhanced from 144 kbps and has, for example, multicode capability, which points toward support of advanced packet data methods provided in WCDMA.
- 768 kbps class has been defined as an intermediate step between 384 kbps and 2 Mbps class.
- 2 Mbps class. This is the state-of-the-art class and has been defined for the downlink direction only.

These classes are defined so that a higher class has all the capabilities covered by a lower class. It should be noted that terminals may deviate from these classes when giving their parameters to the network, thus 2 Mbps is possible for the uplink also, though not covered by any of the classes directly.

3GPP specifications include performance requirements for the bit rates up to 384 kbps, for more details see Section 12.5. Therefore, it is expected that terminals up to 384 kbps will be available in the initial deployment phase.

High-Speed Downlink Packet Access, HSDPA, further enhances the WCDMA bit rate capabilities. HSDPA terminal capabilities are defined in 3GPP Release 5 and extend beyond 10 Mbps. HSDPA is covered in detail in Chapter 11.

2.10 Location Services in WCDMA

2.10.1 Location Services

Location-based services and applications are expected to become one of the new dimensions in UMTS. A location-based service is provided either by a teleoperator or by a third party service provider that utilises available information on the terminal location. The service is either push (e.g. automatic distribution of local information) or pull type (e.g. localisation of emergency calls). Other possible location-based services are discount calls in a certain area,

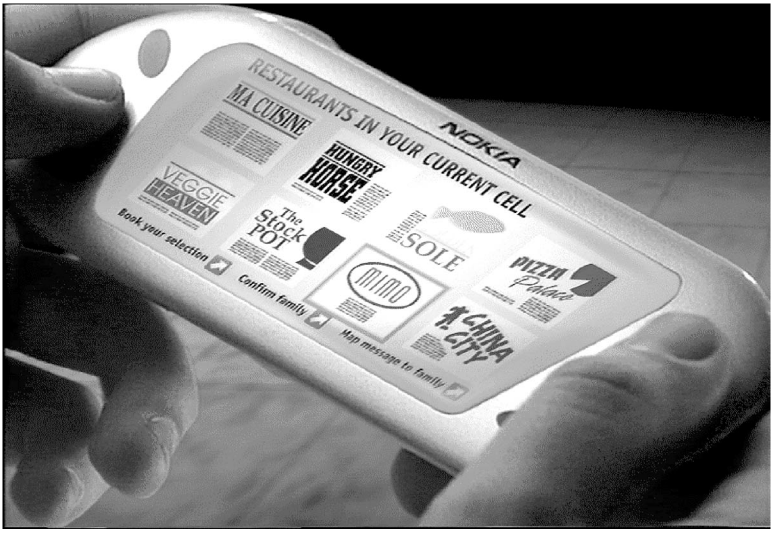


Figure 2.36. 3G concept phone showing location-based service

broadcasting of a service over a limited number of sites (broadcasting video on demand), and retrieval and display of location-based information, such as the location of the nearest gas stations, hotels, restaurants, and so on. Figure 2.36 shows an example. Depending on the service, the data may be retrieved interactively or as background. For instance, before travelling to an unknown city abroad one may request night-time download of certain points of interest from the city. The downloaded information typically contains a map and other data to be displayed on top of the map. By clicking the icon on the map, one gets information from the point. Information to be downloaded background or interactively can be limited by certain criteria and personal interest.

The location information can be input by the user or detected by the network or mobile station. The network architecture of the location services is discussed in Chapter 5. Release '99 of UMTS specifies the following positioning methods:

- the cell coverage based positioning method;
- Observed Time Difference Of Arrival – Idle Period DownLink (OTDOA-IPDL);
- network-assisted GPS methods.

These methods are complementary rather than competing, and are suited for different purposes. These approaches are introduced in the following sections.

2.10.2 Cell Coverage Based Location Calculation

The cell coverage based location method is a network based approach, i.e., it does not require any new functionalities in the mobile. The radio network has the location information with a cell level accuracy when the mobile has been allocated a dedicated channel or when the mobile is in cell_FACH or cell_PCH states. These states are introduced in Chapter 7. If the mobile is in idle state, its location with cell accuracy can be obtained by forcing the mobile to cell_FACH state with a location update as illustrated in Figure 2.37.

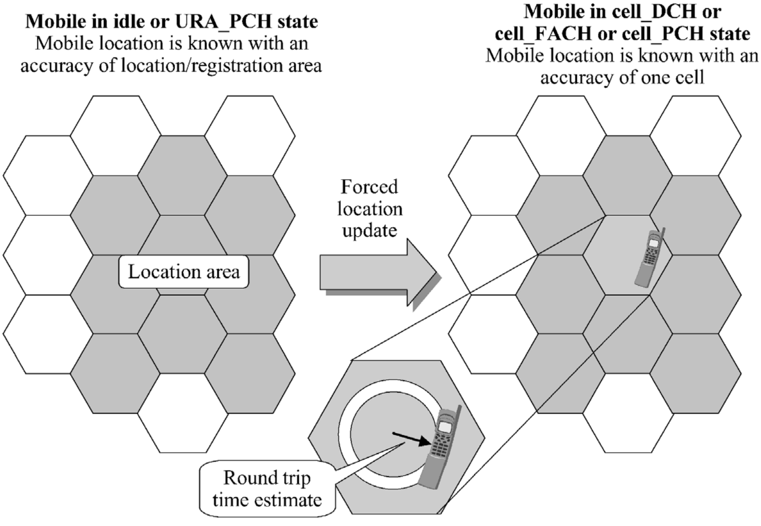


Figure 2.37. Location calculation with cell coverage combined with round trip time

The accuracy of the cell coverage based method depends heavily on the cell size. The typical cell ranges in the urban area are below 1 km and in the dense urban area a few hundred meters providing fairly accurate location information.

The accuracy of the cell coverage based approach can be improved by using the round trip time measurement that can be obtained from the base station. That information is available in cell_DCH state and it gives the distance between the base and the mobile station.

2.10.3 Observed Time Difference Of Arrival, OTDOA

The OTDOA method is based on the mobile measurements of the relative arrival times of the pilot signals from different base stations. At least three base stations must be received by the mobile for the location calculation, as shown in Figure 2.38. A measurement from two base stations defines a hyperbola. With two measurement pairs, i.e. with three base stations, the location can be calculated.

In order to facilitate the OTDOA location measurements and to avoid near–far problems, the WCDMA standard includes idle periods in downlink, IPDL. During those idle periods the mobile is able to receive the pilot signal of the neighbour cells even if the best pilot signal on the same frequency is very strong. A typical frequency of the idle periods is 1 slot every 100 ms, i.e. 0.7 % of the time. The IPDL–OTDOA measurements are shown in Figure 2.39.

The network needs to know the relative transmission times of the pilot signals from different base stations to calculate the mobile location. That relative timing information can be obtained by:

1. OTDOA measurements by the location measurement unit at the base station. The base station measures the relative timing of the adjacent cells. The measurement is similar to the OTDOA measurements by the mobile.

2. The GPS receiver at the base station.

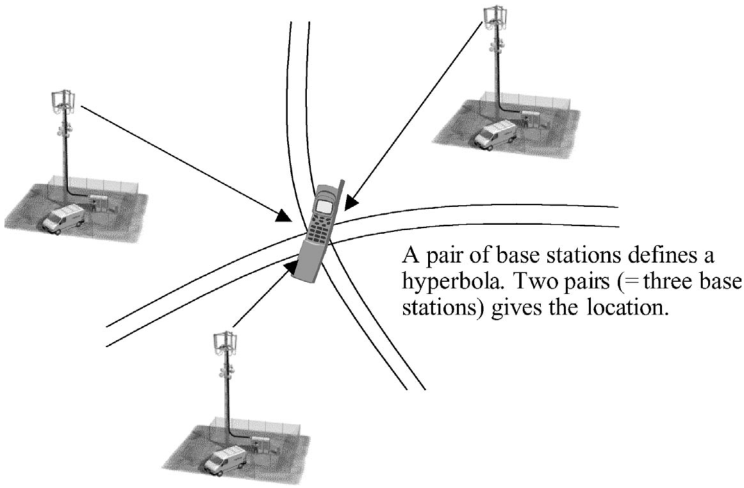


Figure 2.38. Location calculation with three base stations

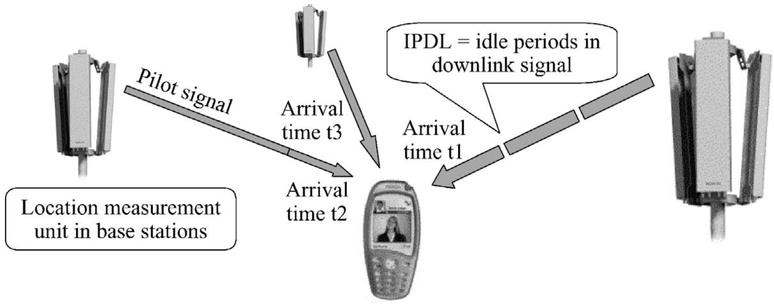


Figure 2.39. IPDL (Idle Period Downlink) – OTDOA (Observed Time Difference of Arrival)

The accuracy of the OTDOA measurements can be in the order of up to tens of meters in very good conditions when several base stations in line-of-sight can be received by the mobile. In practice, such ideal measurement conditions are not typically available in cellular networks. The accuracy depends on the following factors:

1. The number of base stations that the mobile can receive. A minimum of three is required. If more base stations can be received, the accuracy is improved.
2. The relative locations of the base stations. If the base stations are located in different directions from the mobile, the accuracy is improved.
3. The line-of-sight. If there is a line-of-sight between the mobile and the base station, the accuracy is improved.

The requirement of receiving at least three base stations is challenging in the cellular networks. The target of the network planning is to create clear dominance areas of the cells

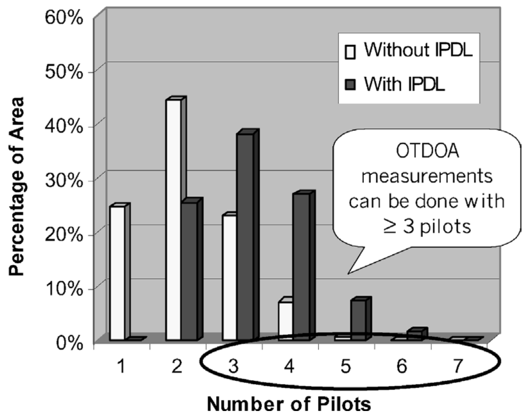


Figure 2.40. Probability of receiving several pilot signals [18]

and to avoid unnecessary overlapping of the cells. That approach maximises the capacity. The clear dominance areas and limited cell overlapping reduce the probability of accurate OTDOA measurements as it is difficult to receive at least three pilot signals. Figure 2.40 shows the probability of a mobile receiving several pilot signals in realistic network scenarios. The probability of receiving at least three pilots is 74 % in Figure 2.40. IPDL allows receipt of the strongest pilot and the second strongest with 100 % probability, but it is challenging to receive at least three pilots with very high probability. The required pilot E_c/I_0 was -18 dB in these simulations and a fully loaded network was assumed. The results show that IPDL greatly improves the performance of OTDOA: without IPDL the probability of receiving at least three pilots would be only 31%. The results also show that it is difficult to obtain a very high probability of OTDOA measurements. The accuracy can be improved by combining OTDOA with the cell coverage based location method.

2.10.4 Assisted GPS

Most accurate location measurements can be obtained with an integrated GPS receiver in the mobile. The network can provide additional information, like visible GPS satellites, reference time and Doppler, to assist the mobile GPS measurements. The assistance data improves the GPS receiver sensitivity for indoor measurements, makes the acquisition times faster and reduces the GPS power consumption. The principle of assisted GPS is shown in Figure 2.41.

A reference GPS receiver in every base station provides the most accurate assistance data and the most accurate GPS measurements by the mobile. The assisted GPS measurements can achieve accuracy of 10 meters outdoors and a few tens of meters indoors. That accuracy also meets the FCC requirements in the USA. If the most stringent measurement probabilities and accuracies are not required, the reference GPS receiver is not needed in every base station, but only a few reference GPS receivers are needed in the radio network. It is also possible to let the mobile GPS make the measurements without any additional assistance data.

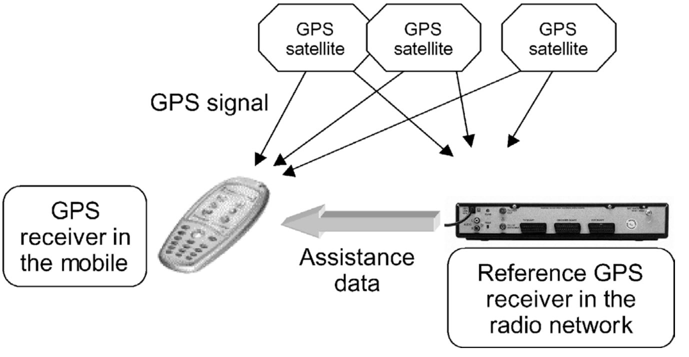


Figure 2.41. Assisted GPS

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