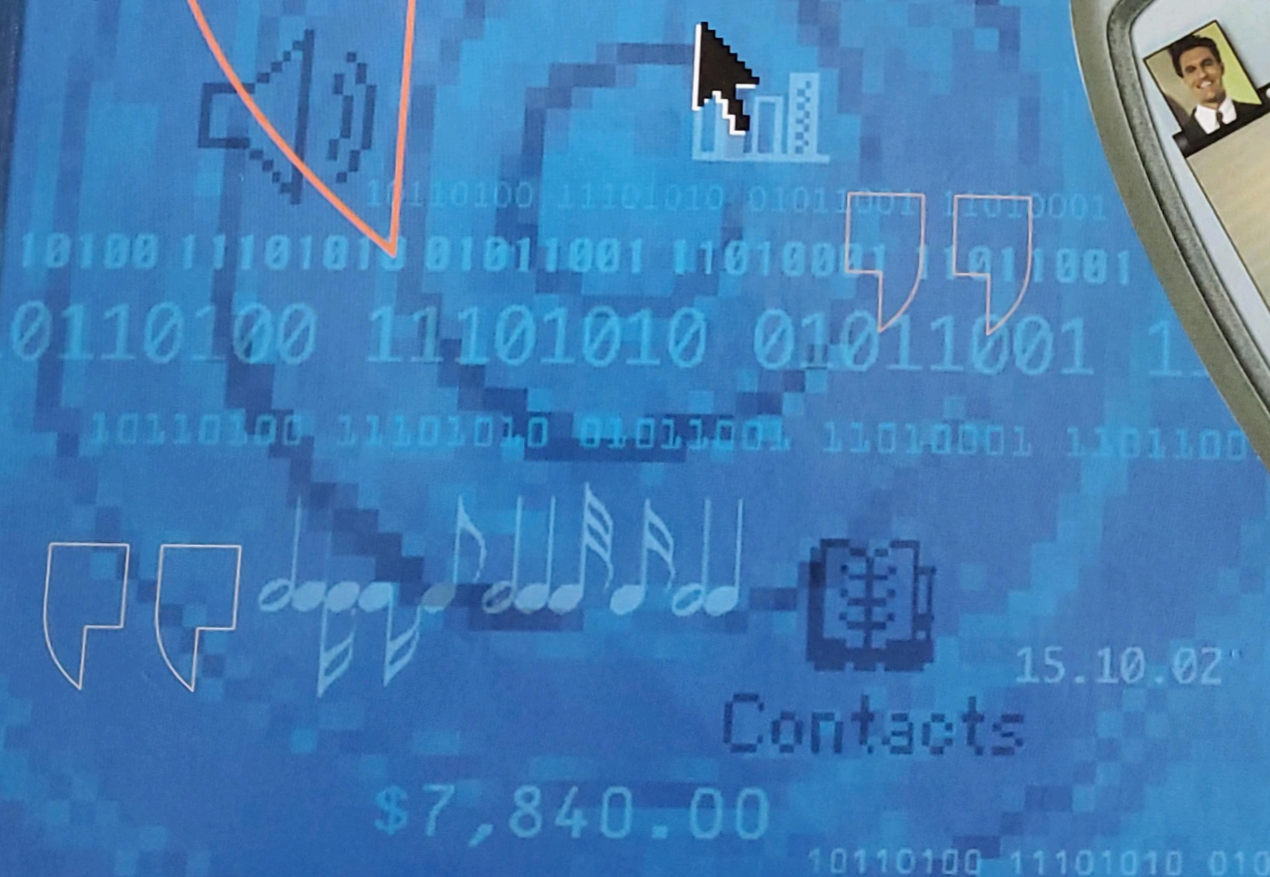


WCDMA FOR UMTS

Radio Access For Third Generation
Mobile Communications



Edited by **Harri Holma**
and **Antti Toskala**

TD/CDMA	Time division CDMA, combined TDMA and CDMA
TDD	Time division duplex
TDMA	Time division multiple access
TE	Terminal equipment
TF	Transport format
TFCI	Transport format combination indicator
TFCS	Transport format combination set
TFI	Transport format indicator
TMSI	Temporary mobile subscriber identity
TPC	Transmission power control
TR	Transparent mode
TS	Technical specification
STD	Time switched transmit diversity
TTA	Telecommunications Association (Korea)
TTT	Telecommunications Technology (Japan)
TSAA	Transmit adaptive antennas
UDP	User datagram protocol
UE	User equipment
UL	Uplink
ULI	Unlicensed mobile communication services
UMTS	Universal mobile communication system
URA	UTRAN registration area
URL	Universal resource locator
U-RNTI	UTRAN RNTI
USCH	Uplink shared channel
USIM	UMTS Subscriber identity module
UTRA	One of the 3 rd generation systems mainly in USA
UTRA	UMTS Terrestrial radio access (GPP)
UTRA	Universal Terrestrial radio access (3GPP)
UTRAN	UMTS Terrestrial radio access network
VAD	Voice activation detection
Vop	Voice over IP
WARC	World administrative radio conference
WCDMA	Wideband CDMA, Code division multiple access
WWW	World wide web
ZZ	Zero forcing

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Introduction

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1.1 WCDMA in Third Generation Systems

Analog cellular systems are commonly referred to as first generation systems. The digital systems currently in use, such as GSM, GPRS, GPRS, GPRS (IS-95) and USTDMA (IS-136), are second generation systems. These systems have enabled voice communications to go wireless in many of the leading markets, and customers are increasingly finding value also in other services such as text messaging and access to data networks, which are starting to grow.

Third generation systems are designed for multimedia communication: with them person-to-person communication can be enhanced with high-quality images and video, and access to information and services on public and private networks will be enhanced by the higher data rates and new flexible communication capabilities of third generation systems. This, together with the continuing evolution of the second generation systems, will create new business opportunities not only for manufacturers and operators, but also for the providers of services.

In the standardisation terms, WCDMA technology has emerged as the most widely adopted third generation air interface. Its specification has been created in 3GPP (the 3rd Generation Partnership Project), which is the joint standardisation project of the standardisation bodies from Europe, Japan, Korea, the USA and China. Within 3GPP, WCDMA is called UTRA (Universal Terrestrial Radio Access) TDD (Frequency Division Duplex) and TDD (Time Division Duplex), the name WCDMA being used to cover both FDD and TDD operation.

In the chapters related to specifications use the 3GPP terms UTRA FDD and TDD, the others using the term WCDMA. This book focuses on the WCDMA FDD technology. The WCDMA TDD mode and its differences from the WCDMA FDD mode are presented in Chapter 12.

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Transmit diversity is included in WCDMA to improve the downlink capacity to support the asymmetric capacity requirements between downlink and uplink. Transmit diversity is not supported by the second generation standards. The major differences between the services and quality requirements in third generation system are: requires advanced radio resource management algorithms to guarantee quality of service and to maximise system throughput. Also, efficient support of non-real-time packet data is important for the new services.

Table 11. Main differences between WCDMA and GSM air interfaces

	WCDMA	GSM
Carrier spacing	5 MHz	200 kHz
Frequency reuse factor	1	1-18
Power control frequency	1500 Hz	2 Hz or lower
Quality control	Radio resource management algorithms	Network planning (frequency planning)
Frequency diversity	5 MHz bandwidth gives multipath diversity with rake receiver	Frequency hopping
Packet data	Load-based packet scheduling	Time slot based scheduling with GPRS
Downlink transmit diversity	Supported for improving downlink capacity	Not supported by the standard, but can be applied

Table 12. Main differences between WCDMA and IS-95 air interfaces

	WCDMA	IS-95
Carrier spacing	5 MHz	1.25 MHz
Chip rate	3.84 Mcps	1.2288 Mcps
Power control frequency	1500 Hz, both uplink and downlink	Uplink: 800 Hz, downlink: slow power control
Base station synchronisation	Not needed	Yes, typically obtained via GPS
Inter-frequency handovers	Yes, measurements with slotted mode	Possible, but measurement method not specified
Efficient radio resource management algorithms	Yes, provides required quality of service	Not needed for speech only networks
Packet data	Load-based packet scheduling	Packet data transmitted as short circuit switched calls
Downlink transmit diversity	Supported for improving downlink capacity	Not supported by the standard

The main differences between WCDMA and IS-95 are discussed below. Both WCDMA and IS-95 utilise direct sequence CDMA. The higher chip rate of 3.84 Mcps in WCDMA gives more multipath diversity than the chip rate of 1.2288 Mcps, especially in small urban

cells. The importance of diversity for system performance is discussed in Sections 9.2.1.2 and 11.2.1.3. Most importantly, increased multipath diversity improves the coverage. The higher chip rate also gives a higher trunking gain, especially for high bit rates, than do narrowband second generation systems.

WCDMA has fast closed-loop power control in both uplink and downlink, while IS-95 uses fast power control only in uplink. The downlink fast power control improves link performance and enhances downlink capacity. It requires new functionalities in the mobile, such as SIR estimation and outer loop power control, that are not needed in IS-95 mobiles.

The IS-95 system was targeted mainly for macro cellular applications. The macro cell base stations are located on masts or rooftops where the GPS signal can be easily received. IS-95 mobiles are not synchronised and this synchronisation is typically obtained via GPS. The need for a GPS receiver in the deployment of the macro and micro cells is problematic, since GPS reception is difficult in urban environments due to the GPS satellites. Therefore, WCDMA is designed to operate with asynchronous base stations where no synchronisation from GPS is needed. The asynchronous base stations make the WCDMA handover slightly different from that of IS-95.

Inter-frequency handovers are considered important in WCDMA, to maximise the use of several carriers per base station. In IS-95, inter-frequency measurements are not specified, making inter-frequency handovers more difficult.

Experience from second generation air interfaces has been important in the development of the third generation interface, but there are many differences, as listed above. In order to make the fullest use of the capabilities of WCDMA, a deep understanding of the WCDMA air interface is needed, from the physical layer to network planning and performance optimisation.

1.5 Core Networks

There are three basic solutions for the core network to which WCDMA radio access networks can be connected. The hosts of the second generation has been either the GSM core network or the IS-41 core network. The third generation offers two options in third generation system. An existing alternative is GPRS with an all-IP-based core network. The most typical connections between the core networks and the air interfaces are illustrated in Figure 1.4. Other connections are also possible and are expected to appear in the standardisation forums in due course.

The market needs will determine which combinations will be used by the operators. It is expected that operators will remain with their second generation core network for voice services and will then add packet data functionalities on top of that. Later, it will be possible to use IP-based core networks for all services.

Because of the different technologies and frequency allocations, global roaming will continue to require specific arrangements between operators, such as multi-mode and multi-band and roaming gateways between the different core networks. To the end user the arrangements must be invisible, and global roaming terminals will probably emerge for those consumers willing to pay for global service.

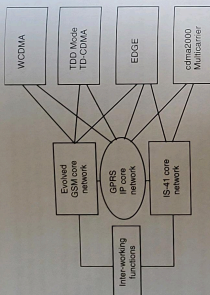


Figure 1.4 Core network relation to the third generation air interface alternatives

References

- [1] Pihkonen, R., Rautava, T. and Penttinen, J., 'TDMA Convergence for Packet Data Services', *IEEE Personal Communications Magazine*, June 1999, Vol. 6, No. 3, pp. 68-73.
- [2] Andremo, P.-G. (ed.), 'UMTS Code Division Testbed (CODIT): Final Review Report', September 1997.
- [3] Nishikawa, T., Dahlman, E., Girard, L. and Klein, A., 'FRAMES Multiple Access for UMTS and IMT-2000', *IEEE Personal Communications Magazine*, April 1998, pp. 16-24.
- [4] Quimper, T., Rikkinen, K., Häkkinen, H., Penttinen, J., Heikinen, A. and Lillsberg, J., 'Design of a 3rd Generation Multirate CDMA System with Multiservice Operations', *IEEE Transactions on Communications*, Vol. 46, No. 1, pp. 1-10, January 1998.
- [5] Pajukoski, K. and Savioja, J., 'Wideband CDMA Test System', *Proc. IEEE Int. Conf. on Personal Indoor and Mobile Radio Communications, PIMRC'97*, Helsinki, Finland, 1-4 September 1997, pp. 669-672.
- [6] Holma, R., Toskala, A. and Laita-aho, M., 'Asynchronous Wideband CDMA for IMT-2000', *3G Telecom Journal*, South Korea, Vol. 8, No. 6, 1998, pp. 1007-1021.

2 UMTS Services and Applications

Jouhi Salonen

2.1 Introduction

The best known new feature of UMTS is higher user bit rates: on circuit-switched connections, 384 Kbps, and on packet-switched connections up to 2 Mbps can be reached. Higher bit rates naturally facilitate some new services, such as video telephony and quick downloading of data. If there is to be a killer application, it is most likely to be quick access to information and the filtering appropriate to the location of a user: see Figure 2.1. Often the requested information is on the Internet, which calls for effective handling of TCP/IP traffic in the UMTS network. At the start of the UMTS era almost all traffic will be voice, but later the share of data will increase. It is, however, difficult to predict the time that transition from voice to data service traffic will move forward. At the same time that transition from voice to data service traffic will move forward, not all of the connections to packet-switched connections. At the start of UMTS service not all of the Quality of Service (QoS) functions will be implemented, and therefore delay-critical applications such as speech and video telephony will be carried on circuit-switched bearers. Later, it will be possible to support delay-critical services as packet data with QoS functions.

Compared to GSM and other existing mobile networks, UMTS provides a new and important feature, namely it allows negotiation of the properties of a radio bearer. This defines the characteristics of the transfer may include throughput, transfer delay and jitter. This feature is important because it allows the network to support a wide range of applications that possess different quality of service (QoS) requirements. It is not possible to predict the nature and usage of many of these applications. Therefore it is neither possible nor sensible to optimise UMTS to only one set of applications. UMTS bearers have to be generic by nature, to allow good support for existing applications and to facilitate the evolution of new applications. Since most of the telecommunications

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The main distinguishing factor between these classes is how delay-sensitive the traffic is: the conversational class is meant for very delay-sensitive traffic, while the background class is the most delay-insensitive. The UMTS QoS classes are summarised in Table 2.1.

Table 2.1. UMTS QoS classes

Traffic class	Conversational class	Streaming class	Interactive class	Background
Fundamental characteristics	Preserve time (variation) between information entities of the stream	Preserve time (variation) between information entities of the stream	Request response pattern Preserve data integrity	Destination is not constrained within a certain time Preserve data integrity
Example of the application	Voice, videotelephony, and low delay video games	Streaming multimedia	Web browsing, network games	Background download of emails

In the initial phase of UMTS the conversational and streaming classes will be transmitted in real-time connections with the WCDMA air interface, while the interactive and background classes will be scheduled in non-real-time packet data. The transmission of non-real-time packet data in WCDMA is described in detail in Chapter 10.

2.3.1 Conversational Class

The best-known application of this class is speech service over circuit-switched boxes. With Internet and multimedia, a number of new applications will require this type, for example voice over IP and video telephony. Real-time conversation is always performed between peers (or groups of five (human) end-users. This is the only type of the four that the required characteristics are strictly imposed by human perception.

Real-time conversation is characterised by the fact that the end-to-end delay is low and the traffic is symmetric or nearly symmetric. The maximum end-to-end delay is given by the processing of video and audio conversation: subjective evaluation of delay is such that the end-to-end delay has to be less than 400 ms. Therefore the limit of acceptable delay is strict, as failure to provide sufficiently low delay will result in unacceptable quality.

2.3.1.1 AMR Speech Service

The speech code in UMTS will employ the Adaptive Multi-rate (AMR) technique. The multi-rate speech code is a single integrated speech code with eight source rates [22] (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 are controlled by the radio access network and do not depend on the speed of the user. To facilitate interoperability with existing cellular networks, some of the modes are identical to those in existing cellular networks. The 12.2 kbps AMR speech code is equal to the GSM EFR code, 7.4 kbps is equal to the US-TDMA speech code, and 6.7 kbps is equal

to the Japanese DPC code. The AMR speech code is capable of switching its bit rate between 12.2 and 0.2 kbps. The AMR speech code has two switching methods: there are two candidates: in-band signalling or dedicated channel.

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 multi-rate coding mode is the so-called Algebraic Code Excited Linear Prediction Code (ACELP). The multi-rate 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 (LP filter coefficients, excitation and codebook indices) and these parameters are sent to the receiver. The receiver reconstructs the speech signal and sends it to the user. The parameters are sent 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 three basic functions to utilise effectively 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 at regular intervals.
- Generation of comfort noise on the RX side during periods when no normal speech frames are received.

DTX has some obvious positive implications: in the user terminal battery life will be prolonged if a smaller battery can 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 mixing the background noise with the speech signal is to prevent the user from recognising the user used to avoid generating possibly annoying sounds as a result of the frame substitution procedure [21, 23]. The AMR speech code can tolerate about a 1% frame error rate (FER) of class A bits without any deterioration of 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^{-3} .

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 in a low-traffic area, the network can increase the AMR bit rate to provide better speech quality. AMR bit rate can be used to extend the cell coverage area. The uplink coverage of the AMR speech code is discussed in Section 11.2.1. With the AMR speech code, it is possible to achieve a trade-off between the network's capacity, coverage and speech quality according to the operator's requirements.