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The UMTS Network and Radio Access Technology

**Air Interface Techniques
for Future Mobile Systems**

JONATHAN P. CASTRO

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Jonathan P. Castro

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Future Mobile Systems

Dr. Jonathan P. Castro

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To:

My family for their endurance, and to my friends and colleagues for their understanding; because while putting together this book I stole too many precious moments from them.

And

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PREFACE

The rapid growth in traffic volume and increase in new services has begun to change the configuration and structure of wireless networks. Thus, future mobile communications systems will be distinguished by high integration of services, flexibility and higher throughput.

To support such features, the efficient use of spectrum and optimum management of radio resources will be essential.

To meet these challenges standardization bodies like ETSI (now expanded to 3GPP), have selected the Wideband Code Division Multiple Access (WCDMA) and the hybrid Time Division – CDMA as the radio techniques for the Universal Mobile Telecommunication Systems (UMTS). Hence, UMTS conceived at the eve of this new millennium will without doubt have a large impact on future wideband mobile networks and serve as the leading platform for wireless multimedia communications.

The specification extracts in this book are intended to provide a concise reference for the many documents related to UMTS systems. After all, the whole UMTS specification set would probably exceed 4000 pages. Thus, it is hoped that the synthesis presented in this book will be useful in some way. In this context, in order to offer a complete source on the UMTS air interface and network issues, this book aims to present a description of the principles, methods and technology used in the standard specification. Different aspects of the UMTS multiple access and network configuration are presented; however, this concise and integrated volume, which embodies the main design elements, goes further. Thus, the content of this book follows structurally the specifications of the 3GPP recommendations to comply entirely with the concept, terminology, approach and style, and not just the technical essence.

In drafting the standards or UMTS technical specifications, the experts have tried to reduce the risk of ambivalent interpretation, and not necessarily ease the understanding for the common reader. The logic, constructive discussions and consensus behind the choices or equivalent solutions have not always been retained in the final specifications. Therefore, in some ways this book tries to present an objective unified view of the key aspects of UMTS. On the other hand, the area of UMTS is vast in content and detail, and not all within the scope of these writings. As a result, the book will allow us to understand the UMTS specifications and get a good grasp of its design, but the technical documents of formerly SMG, and now 3GPP, remain the official specifications with all their appropriate ownership and origin of contribution.

Since this book has been introduced in good faith as a useful reference for UMTS technology, the author does not take responsibility for any misuse or error while dealing

with the information provided. If for some reason some representations have been omitted due to time constraints or the dynamic changing process of the specifications, they will be revised and corrected for later reprints. Thus, the author welcomes any comments and suggestions for improvements or changes that could enhance further this contribution to UMTS.

The chapters in this book cover specific design details of the building blocks in the UMTS air interface, in particular the physical layer. It addresses the technical part of the specification for both the FDD and TDD modes. On the other hand, it also introduces the key criteria for network dimensioning and deployment of 3G systems assuming an evolution from 2G mobile networks from the providers' point of view. To illustrate the progressive steps of UMTS standards such as the evolution towards predominant packet switching oriented communications, this book also introduces the 'All IP Core Network' architecture concept for mobile multimedia.

Chapter 1 deals with concrete requirements for 3G mobile systems after summarizing the rapid growth of wireless communications and the Internet. It also outlines briefly enhancing technologies such as capacity increasing antennas, multi-user detection techniques, and software radio applications.

Chapter 2 presents the fundamentals of system analysis, e.g. multiple access options, which considers narrow-band and wide-band digital channels, as well as the background for the UTRA FDD and TDD modes. It covers signal processing aspects describing the principles of spread spectrum, modulation and spreading, the CDMA performance, PN sequences, power control and handovers. It presents the communications environments envisaged for UMTS operation and deployment. It also describes the channel models used to verify and justify the performance for the selected operating scenarios. It provides a summary of the mathematical formulation for the performance analysis results seen in the forthcoming chapters (e.g. Chapter 7).

Chapter 3 describes the UMTS development platform. It introduces its architecture top down, identifying the core and access network domains. It defines the UTRA identifiers and functions, e.g. system access control, radio channel ciphering and deciphering, mobility functions, and radio resource management and control functions. It also presents mobility management with its signaling connections and impact of mobility handling. Chapter 3 presents the UTRAN synchronization and UTRAN interfaces besides pointing out co-existing 2G/3G network issues. It also introduces the radio interface protocol architecture with its structure in terms of services and function layers. This chapter thus outlines the most relevant elements, which require technical description for design and implementation.

Chapter 4 describes the UTRA physical layer design and configuration, where we introduce all the building blocks in detail with their respective technical description and requirements. It covers dedicated common transport channels, configuration of FDD and TDD physical channels in the uplink and downlink with their spreading and coding characteristics. Spreading and modulation, including scrambling, multiplexing and channel coding are also discussed. The chapter presents the aforementioned characteris-

tics for the FDD and TDD separately for each mode or unified when the case applies to both.

Chapter 5 introduces the UTRA transmission system starting from the spectrum allocation, i.e. the UTRA frequency bands. It presents the radio transmission and reception aspects, describing transmitter and receiver characteristics for the User Equipment (UE) and the Base Station (BS). It describes the maximum output power and output power dynamics, out of synchronization output power handling, transmit on/off power. Details on the output RF spectrum emissions, such as occupied bandwidth and out band emission, spectrum emissions, adjacent channel leakage power ratio, spurious emissions, and transmit modulation and inter-modulation are given. The summary of examples includes a review of simulation scenarios for the co-existence of FDD/FDD when analyzing ACIR with macro-to-macro and macro-to-micro cases. Before presenting results the chapter also reviews propagation models.

Chapter 6 describes the UMTS service components. It covers the UMTS bearer architecture, concepts in quality of service for 3G systems, multimedia transmission and traffic classes in UMTS. The classes include conversational, streaming, interactive, and background types. Sensitivity to IP transmission impairments are also covered here. To provide an overview of potential applications in UMTS this chapter also summarizes service offerings and selected areas of service technology.

Chapter 7 introduces the factors that influence 3G network dimensioning. It discusses coverage and capacity trade-off in the FDD mode pointing out impacts from soft handover, power control and orthogonality deviations. It covers the analysis of parameters for multi-service traffic in PS and CS. It establishes service models starting from capacity projections, and service strategy. Cellular coverage planning issues, i.e. the coverage concept, radio network parameter assumptions for CS and PS, characteristics of CDMA cells (with its theoretical capacity and cell loading effects) constitute the essential parts of this chapter. Link budget principles for the forward and reverse links and their respective formulation are covered. In the latter part, these principles are applied to a field study. For completeness the chapter also describes briefly the dimensioning of the RNC in the UTRAN side. Chapter 7 also presents the dimensioning of the core network and transmission systems. In the last part, results of a field study are provided using hypothetical parameters to illustrate the concepts end-to-end. The illustrations correspond to dimensioning exercises carried out while optimizing 3G networks. However, the input and output values in this chapter do not necessarily reflect actual values that may be used directly while dimensioning a future UMTS network. Finally, to complete the assessment of UMTS network deployment within 2G networks like GSM, this chapter discusses briefly co-location and site sharing, as well as co-location of antenna systems.

Chapter 8 presents issues on resource and network management. It covers radio resource management and signalling, i.e. managing power (fast and low). The conceptual aspects of network management are covered from the network management system point of view. Initial considerations for network optimization are also covered.

Chapter 9 as a prelude to future predominantly PS domain networks, covers the conceptual architecture of UMTS Release 2000 or more specifically Release 4 and 5. It starts

with the evolution of R99 and discusses briefly the long term view and vision of the UMTS architecture. Then it describes the components of R00 with their corresponding interfaces or reference points. This chapter also presents an introduction to and considerations of mobility management, registration aspects, multimedia signalling, service platforms, QoS aspects, and transport issues such as the basic differences of Ipv4 and Ipv6.

ACRONYMS

AAL2	ATM Adaptation Layer Type 2
AAL5	ATM Adaptation Layer Type 5
ATM	Asynchronous Transfer Mode
BHCA	Busy Hour Call Attempt
CAMEL	A version of IN standardised for the mobile environment
CCB	Customer Care and Billing
CDR	Call Detail Record
CPS	Call Processing Server
CSE	CAMEL Service Environment
ACIR	Adjacent Channel Interference Ratio
ACLR	Adjacent Channel Leakage power Ratio
ACS	Adjacent Channel Selectivity
AH	Address Handling
AI	Acquisition Indicator
AICH	Acquisition Indicator Channel
ALCAP	Access Link Control Application Part
AMR	Adaptive MultiRate (speech codec)
AP	Access Preamble
AP-AICH	Access Preamble Acquisition Indicator Channel
API	Access Preamble Indicator
ARQ	Automatic Repeat Request
ASC	Access Service Class
BCCH	Broadcast Control Channel
BCH	Broadcast Channel
BER	Bit Error Rate
BLER	Block Error Ratio
BMC	Broadcast/Multicast Control
BM-IWF	Broadcast Multicast Interworking Function
BS	Base Station
BSS	Base Station Subsystem
C-	Control-
CA	Channel Assignment

CAI	Channel Assignment Indicator
CBC	Cell Broadcast Centre
CBR	Constant Bit Rate
CBS	Cell Broadcast Service
CC	Call Control
CCC	CPCH Control Command
CCCH	Common Control Channel
CCF	Call Control Function
CCH	Control Channel
CCPCH	Common Control Physical Channel
CCTrCH	Coded Composite Transport Channel
CD	Collision Detection
CD/CA-ICH	Collision Detection/Channel Assignment Indicator Channel
CDI	Collision Detection Indicator
CDMA	Code Division Multiple Access
CFN	Connection Frame Number
Chip Rate	Chip rate of W CDMA system, equals to 3,84 M chips per second
CN	Core Network
CPCH	Common Packet channel
CPICH	Common Pilot Channel
CRC	Cyclic Redundancy Check
CS	Circuit Switched
CSCF	Call State Control Function
CSICH	CPCH Status Indicator Channel
CTCH	Common Traffic Channel
CW	Continuous Wave (un-modulated signal)
DC	Dedicated Control (SAP)
DCA	Dynamic channel allocation
DCCH	Dedicated Control Channel
DCH	Dedicated Channel, which is mapped into Dedicated Physical Channel.
DL	Downlink
DPCCH	Dedicated Physical Control Channel
DPCH	Dedicated Physical Channel
DPCH_Ec	Average energy per PN chip for DPCH
DPDCH	Dedicated Physical Data Channel
DRNC	Drift Radio Network Controller
DRNS	Drift RNS
DRX	Discontinuous Reception

DS-CDMA	Direct-Sequence Code Division Multiple Access
DSCH	Downlink Shared Channel
DSMA-CD	Digital Sense Multiple Access - Collision Detection
DTCH	Dedicated Traffic Channel
DTX	Discontinuous Transmission
EDGE	Enhanced Data Rates for GSM Evolution
EFR	Enhanced Full Rate speech codec
E_c/N_0	Received energy per chip divided by the power density in the band
FACH	Forward Link Access Channel
FAUSCH	Fast Uplink Signalling Channel
FBI	Feedback Information
FCS	Frame Check Sequence
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FDR	False transmit format Detection Ratio
FEC	Forward Error Control
FER	Frame Error Rate
FTP	File Transfer Protocol
FSW	Frame Synchronization Word
F _{uw}	Frequency of unwanted signal.
GC	General Control (SAP)
GF	Galois Field
GGSN	Gateway GPRS support node
GMSC	Gateway MSC
GP	Guard Period
GPRS	General Packet Radio System
GSM	Global System for Mobile Communication
GTP	GPRS Tunnelling Protocol
HLR	Home Location register
HO	Handover
HSS	Home Subscriber Server
IC	Interference Cancellation
ICGW	Incoming call gateway
IETF	Internet engineering task force
IMSI	International Mobile Subscriber identity
IN	Intelligent network
I _{oc}	The power spectral density of a band limited white noise source.
I _{or}	The total transmit power spectral density of the DL at the BS antenna connector

IP	Internet Protocol
ISDN	Integrated services digital network
ISI	Inter-symbol interference
ITU	International Telecommunication Union
JD	Joint Detection
kbps	kilo-bits per second
L1	Layer 1 (physical layer)
L2	Layer 2 (data link layer)
L3	Layer 3 (network layer)
LAC	Link Access Control
LAI	Location Area Identity
LCS	Location Services
LLC	Logical Link Control
LAN	Local Area Network
MA	Multiple Access
MAC	Medium Access Control
Mbps	Mega bit per second
Mcps	Mega Chip Per Second
ME	Mobile Equipment
MER	Message Error Ratio
MF	Matched filter
MGCF	Media Gateway Control Function
MGW	Media Gateway Function
MM	Mobility Management
MPEG	Motion picture expert group
MRF	Multimedia Resource Function
MS	Mobile Station
MT	Mobile Terminated
MUD	Multiuser detection
MUI	Mobile User Identifier
NAS	Non Access Stratum
NBAP	Node B Application Protocol
NRT	Non-Real Time
Nt	Notification (SAP)
OVSF	Orthogonal Variable Spreading Factor (codes)
PAD	Padding
PC	Power Control
PPCC	Parallel Concatenated Convolutional Code

PCCH	Paging Control Channel
PCCPCH	Primary Common Control Physical Channel
PCH	Paging Channel
PCPCH	Physical Common Packet Channel
PCS	Personal Communications System
PDCP	Packet Data Convergence Protocol
PDSCH	Physical Downlink Shared Channel
PDSCH	Physical Dedicated Shared Channel
PDP	Packet Data Control
PDU	Protocol Data Unit
PhCH	Physical Channel
PHY	Physical layer
PhyCH	Physical Channels
PI	Paging Indicator
PICH	Paging Indicator Channel
PICH	Page Indicator Channel
PN	Pseudo Noise
PPM	Parts Per Million
PRACH	Physical Random Access Channel
PSC	Primary Synchronisation Code
PSCH	Physical channel
PSTN	Public Switched Telephone Network
PU	Payload Unit
PUSCH	Physical Uplink Shared Channel
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAB	Radio Access Bearer
RACH	Random Access Channel
RAI	Routing Area identity
RANAP	Radio Access Network Application Part
RB	Radio Bearer
RF	Radio Frequency
RL	Radio Link
RLC	Radio Link Control
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RNS	Radio Network Subsystem
RNSAP	Radio Network Subsystem Application Part

RNTI	Radio Network Temporary Identity
RRC	Radio Resource Control
RRM	Radio Resource Management
RSC	Recursive Systematic Convolutional Coder
RSCP	Received Signal Code Power
RSSI	Received Signal Strength Indicator
RT	Real Time
RTP	Real Time Protocol
RU	Resource Unit
RX	Receive
SAB	Service Area Broadcast
SAP	Service Access Point
SCCC	Serial Concatenated Convolutional Code
SCCP	Signalling Connection Control Part
S-CCPCH	Secondary Common Control Physical Channel
SCH	Synchronisation Channel: Primary + Secondary synchronisation channels
SDP	Session Description Protocol
SDU	Service Data Unit
SF	Spreading Factor
SFN	System Frame Number
SHCCH	Shared Channel Control Channel
SI	Status Indicator
SIR	Signal to Interference Ratio
SMS	Short Message Service
SN	Sequence Number
SNR	Signal to Noise Ratio
SPD	Serving Profile Database
SRNC	Serving Radio Network Controller
SRNS	Serving Radio Network Subsystem
SSC	Secondary Synchronisation Code
SSCH	Secondary Synchronisation Channel
SSDT	Site Selection Diversity Transmission
SSDT	Site Selection Diversity TPC
STD	Selective Transmit Diversity
STTD	Space Time Transmit Diversity
TA	Timing Advance
TCH	Traffic Channel
TDD	Time Division Duplex

TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TF	Transport Format
TFC	Transport Format Combination
TFCI	Transport Format Combination Indicator
TFI	Transport Format Indicator
TMSI	Temporary Mobile Subscriber Identity
TPC	Transmit Power Control
TrBk	Transport Block
TrCH	Transport Channel
T-SGW	Transport Signalling Gateway Function
TSTD	Time Switched Transmit Diversity
TTI	Transmission Time Interval
TX	Transmit
TxAA	Transmit Adaptive Antennas
U-	User-
UE	User Equipment
UL	Uplink (Reverse link)
UMTS	Universal Mobile Telecommunications System
URA	UTRAN Registration Area
USCH	Uplink Shared Channel
USIM	UMTS Subscriber Identity Module
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
WCDMA	Wide-band Code Division Multiple Access

1 EVOLVING MOBILE NETWORKS

While the history of mobile communications is long [1–3], and the background of mobile networks thereby is also long, in this chapter we focus on the historic evolution in terms of network architecture and services starting with 2nd generation (2G) mobile systems. In particular we consider the development of the architecture of Global Systems for Mobile Communications (GSM), since it is by far the most widespread mobile system in the world today. This will provide the basis to cover the introduction of Universal Mobile Telecommunication Services (UMTS) in relation to its Core Network (CN) and radio architectures. The latter will in turn serve as the platform to present UMTS Radio Access Technology, which is one the aims of this book.

1.1 THE GROWTH OF MOBILE COMMUNICATIONS

Today wireless voice service is one of the most convenient and flexible means of modern communications. GSM technology has been at the leading edge of this wireless revolution. It is the technology of choice in over 120 countries and for more than 200 operators worldwide. Current estimates are that by the year 2001 there will be around 600 million wireless subscribers (e.g. mobile telephone users), out of which more than 50% will depend on GSM technology.

As the wireless revolution has been unfolding, the Internet has also shown a phenomenal growth simultaneously. The advent of the World Wide Web and web browsers has propelled TCP/IP protocols into the main stream, and the Internet is widespread not

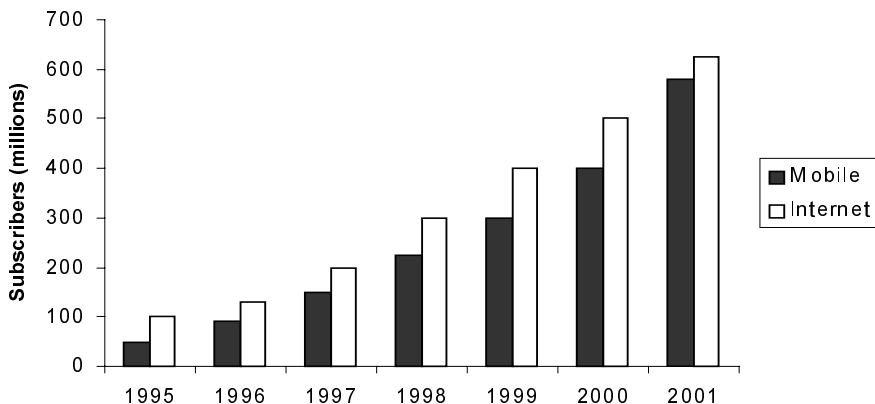


Figure 1.1 The growth of mobile and internet services.

only in the corporate environment but also in households. Large number of consumers have embraced the Internet and use it today to access information online, for interactive business transactions, and e-commerce as well as electronic mail. Figure 1.1 illustrates the growth in mobile and Internet subscribers.

The success of mobile communications, i.e. the ubiquitous presence it has established and the emergence of the Internet point towards a tremendous opportunity to offer integrated services through a wireless network.

One of the main market segments for wireless services besides corporate intranet/internet access is the consumer sector. The availability of intelligent terminals¹ or multipurpose wireless telephones is already ushering a new era of the information age, where subscribers can receive directly through GSM-SMS: news, sport updates, stock quotes, etc. However, the progress of audiovisual techniques and the support for a Web-like interface in a new generation of terminals, will push consumers to a new era of multimedia communications with a focus on services rather than technology.

To support the growth of Internet type services² and future demands for wireless services, ETSI SMG and other standards bodies³ have completed or are now completing specifications to provide a transition platform or evolution path for wireless networks like GSM. Figure 1.2 illustrates the wireless data technology options.

The technology options in Figure 1.2 can be summarized as follows:

- 14.4 kbits/s allows GSM data calls with a rate of 14.4 kbits/s per time slot, resulting in a 50% higher data throughput compared to the current maximum speed of 9.6 kbits/s.
- High Speed Circuit Switched Data (HSCSD) aggregates symmetrically or asymmetrically several circuit channels, e.g. 28.8 kbits/s for two time slots (2 + 2) or 43.2 kbits/s for three time slots (3 + 1).
- General Packet Radio Service (GPRS) enables GSM with Internet access at high spectrum efficiency by sharing time slots between different users. It affords data rates of over 100 kbits/s to a single user while offering direct IP connectivity.
- Enhanced Data Rate for GSM Evolution (EDGE) modifies the radio link modulation scheme from GMSK to 8QPSK. Thereby increasing by three times the GSM throughput using the same bandwidth. EDGE in combination with GPRS (E-GPRS) will deliver single user data rates of over 300 kbits/s.
- UMTS as 3rd generation wireless technology utilizes a Wideband CDMA or TD/CDMA transceiver. Starting with channel bandwidths of 5 MHz it will offer data rates up to 2 Mbits/s. UMTS will use new spectrum and new radio network configurations while using the GSM core infrastructure.

¹ For example WAP terminals.

² Including voice or IP as a new trend.

³ In the USA – T1P1, in Japan – ARIB, in Korea – TTA, and in China – CWTS.

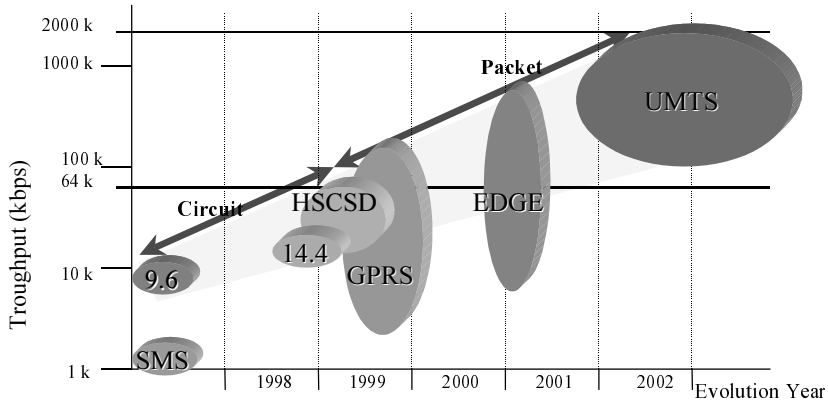


Figure 1.2 Evolution for wireless networks⁴, e.g. GSM.

Although the circuit switched enhancements such as HSCSD will increase transmission rates, it is packet switched enhancements, which will meet the challenges or demands posed on current wireless networks. Thus, GPRS and UMTS with EDGE as an intermediate solution will provide the platform to support integrated services of voice and data including multimedia.

While GPRS and UMTS meet the demands for Internet (IP) features and higher bandwidths in mobile networks, another evolution step is taking place in the network infrastructure. This is the convergence of single networks into a multi-purpose backbone network. The next section covers this step, which will have also impact on the implementation of UMTS radio access technology.

1.1.1 Convergence of Fixed and Mobile Networks

Convergence, i.e. the closer interworking between fixed and mobile telecommunications, although it has long been a buzzword in the telecom market, is now coming into reality. As Ericsson puts it [16], fixed and mobile convergence includes everything from new services to the integration of nodes, networks and operating systems. The user may have, e.g. the same voice mailbox for fixed and mobile telephony, while the operator can also use the large sections of the network in a coordinated manner for different types of access. Thus, convergence is now a new frontier in communications, where UMTS will evolve.

Figure 1.3 illustrate how single service networks will evolve into multi-purpose networks with multi-level access points. With IP becoming more pervasive in the backbone, the challenge of integrating voice and data services in the fixed and mobile environment become more formidable.

⁴ IS-136 has adopted EDGE as its air-interface expansion.

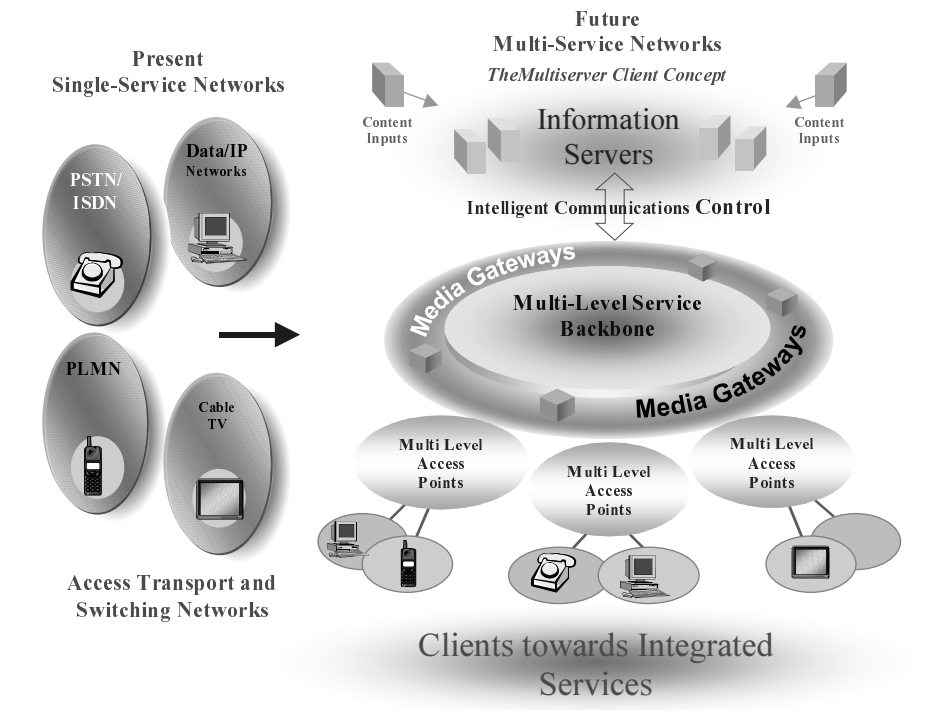


Figure 1.3 Multi-service network.

Table 1.1 The Converging Industry in Telecommunications, Computers and Media

Telecom Industry	Wire-line			Wireless Mobility				New Telecoms
	PSTN	PTN	ISDN	2G	2.5 G	3G mobile	3G Internet	
Computer Industry	Main Frames	Desk top Computing	PC-LAN PC-Servers	PC-WAN	Internet / Intranet WWW Electronic Commerce			Converged Industry
Media Industry					Electronic Publication Interactive Entertainment			New battle ground

It boils down to the transformation of the Telecom, Computer and Media Industry, resulting into the Converged Industry as illustrated in Table 1.1.

Clearly then, UMTS will be part of the convergent Industry with a trend towards multi-services within integrated infrastructures.

1.2 THIRD GENERATION MOBILE SYSTEM REQUIREMENTS

Although third generation (3G) systems involve primarily infrastructure change in the Air-Interface (AI), it also has impact in the service configuration options and the access

to the Core Network (CN). Hence, the 3G, or more specifically UMTS requirements in this section cover three main areas, i.e. services, air-interface, and core network access.

1.2.1 UMTS Services Aspects

The scope of services can be largely focused on different issues like service management, charging and billing, terminals, network management, quality of service, and security. Here, however, we will be looking at services from the principle side in order to establish a framework to present the UMTS air-interface. An extract of the service principles outlined in the ETSI specifications UMTS Services aspects – Service Principles and UMTS Services [4] and Services capabilities [16], can be summarized as follows:

UMTS is the realization of a new generation of mobile communications technology for a world in which personal communications services should allow person-to-person calling, independent of location, the terminal used, the means of transmission (wired or wireless) and the choice of technology.

UMTS shall therefore be in compliance with the following objectives:

- (a) to provide a single integrated system in which the user can access services in an easy to use and uniform way in all environments;
- (b) to allow differentiation between service offerings of various serving networks and home environments;
- (c) to provide a wide range of telecommunications services including those provided by fixed networks and requiring user bit rates of up to 2 Mbits/s as well as services special to mobile communications. These services should be supported in residential, public and office environments and in areas of diverse population densities. These services are provided with a quality comparable with that provided by fixed networks such as ISDN;
- (d) to provide services via hand held, portable, vehicular mounted, movable and fixed terminals (including those which normally operate connected to fixed networks), in all environments (in different service environments – residential, private domestic and different radio environments) provided that the terminal has the necessary capabilities;
- (e) to provide support of roaming users by enabling users to access services provided by their home environment in the same way even when roaming.
- (f) to provide audio, data, video and particularly multimedia services;
- (g) to provide for the flexible introduction of telecommunication services;
- (h) to provide the capability to support Universal Personal Telecommunications (UPT);
- (i) to provide within the residential environment the capability to enable a pedestrian user to access all services normally provided by fixed networks;
- (j) to provide within the office environment the capability to enable a pedestrian user to access all services normally provided by PBXs and LANs;
- (k) to provide a substitute for fixed networks in areas of diverse population densities, under conditions approved by the appropriate national or regional regulatory authority.

(l) to provide support for interfaces which allow the use of terminals normally connected to fixed networks.

In addition UMTS aims:

- to enable users to access a wide range of telecommunications services, including many that are today undefined as well as multi-media and high data rates.
- to facilitate the provision of small, easy to use, low cost terminals with long talk time and long standby operation;
- to provide an efficient means of using network resources (particularly radio spectrum).

Based on the above objectives, specific requirements related to services are outlined in the ETSI Specifications [15–17]. These requirements are primarily concerned with items such as Quality of Service, Security and Charging, Service Management, etc.

1.2.2 UMTS Terrestrial Radio Access Aspects

The UMTS Terrestrial Radio Access (UTRA) system requirements are based on the service requirements. The latter sets the demands, which UTRA specification aims to meet. Table 1.2 summarizes key (selected) requirements identified for the UTRA concept from [18]:

Table 1.2 UTRA High Level Requirements

Key requirements	Description
	Bearer capabilities
Maximum user bit rates	• <i>Rural Outdoor</i>⁵: at least 144 kbit/s (goal to achieve 384 kbit/s), maximum speed: 500 km/h • <i>Suburban Outdoor</i>⁶: at least 384 kbps (goal to achieve 512 kbit/s), maximum speed: 120 km/h • <i>Indoor/Low range outdoor</i>⁷: at least 2 Mbps, maximum speed: 10 km/h • The UTRA definition should allow evolution towards higher bit rates
Flexibility	• Negotiation of bearer service attributes (bearer type, bit rate, delay, BER, etc.) • Parallel bearer services (service mix), real-/non-real-time communication modes, etc. • Circuit and packet oriented bearers • Supports scheduling (and pre-emption) of bearers (including control bearers) within priority • Adaptability of link to quality, traffic and network load, as well as radio conditions. • Wide range of bit rates should be supported with sufficient granularity

⁵ The specified bit rate will be available throughout the operator's service area, with the possibility of large cells.

⁶ The specified bit rate will be available with complete coverage of a suburban or urban area, using microcells or smaller macrocells.

⁷ The specified bit rate will be available indoors and localised coverage outdoors.

	• Variable bit rate real time capabilities should be provided • Bearer services appropriate for speech shall be provided
Handover	• Provide seamless (to user) handover between cells of one operator • The UTRA should not prevent seamless HO between different operators or access networks • Efficient handover between UMTS and 2nd generation systems, e.g. GSM, should be possible
	Operational requirements
Compatibility with services provided by present core transport networks	• ATM bearer services • GSM services • IP (internet protocol) based services • B/N-ISDN services
Radio access network planning	• If radio resource planning is required, automatic planning shall be supported
Public network operators	• It shall be possible to guarantee pre-determined levels of QoS and quality to public UMTS ops
Private and residential operators	• The radio access scheme should be suitable for low cost applications where range, mobility and user speed may be limited • Multiple unsynchronized systems should be able to successfully co-exist in the same environment • It should be possible to install base stations without co-ordination • Frequency planning should not be needed
	Efficient spectrum usage
Spectrum efficiency	• High spectrum efficiency for typical mixtures of different bearer services • Spectrum efficiency at least as good as GSM for low bit rate speech
Variable asymmetry of total band usage	• Variable division of radio resource between up-link and down-link resources from a common pool (NB: this division could be in either frequency, time, or code domains)
Spectrum utilization	• Allow multiple operators to use the band allocated to UMTS without co-ordination⁸ • It should be possible to operate the UTRA in any suitable frequency band that becomes available such as first and second generation system's bands
Coverage/capacity	• The system should be flexible to support a variety of initial coverage/capacity configurations and facilitate coverage/capacity evolution • Flexible use of various cell types and relations between cells (e.g. indoor cells, hierarchical cells) within a geographical area without undue waste of radio resources • Ability to support cost effective coverage in rural areas
Mobile terminal viability	• Hand-portable and PCM-CIA card sized UMTS terminals should be viable in terms of size, weight, operating time, range, effective radiated power and cost
Network complexity and cost	• The development and equipment cost should be kept at a reasonable level, taking into account cell site cost, cross-connect, signalling load and traffic overhead (e.g. due to handovers)

⁸ The feasibility of spectrum sharing requires further study.

Mobile station types	It should be possible to provide a variety of mobile station types of varying complexity, cost and capabilities in order to satisfy the needs of different types of users
	Requirements from bodies outside SMG
Alignment with IMT 2000	UTRA shall meet at least the technical requirements for submission as a candidate technology for IMT 2000 (FPLMTS)
Minimum bandwidth allocation	It should be possible to deploy and operate a network in a limited bandwidth (e.g. 5 MHz)
Electro-magnetic compatibility (EMC)	The peak and average power and envelope variations have to be such that the degree of interference caused to other equipment is not higher than in today's systems
RF radiation effects	UMTS shall be operative at RF emission power levels, which are in line with the recommendations related to electromagnetic radiation
Security	The UMTS radio interface should be able to accommodate at least the same level of protection as the GSM radio interface does
Co-existence with other systems	The UMTS Terrestrial Radio Access should be capable of co-existing with other systems within the same or neighbouring band depending on systems and regulations
	Multi-mode implementation capabilities
	It should be possible to implement dual mode UMTS/GSM terminals cost effectively

By looking at the bearer capabilities in Table 1.2, we can see that evolution towards higher rates will initially apply mainly to indoor rates. In this environment convergence will also have higher impact. In addition, UTRA will not only prevent seamless HO between different operators or access networks, but also support HO between 2G and 3G systems, e.g. GSM and UMTS.

UTRA will support key technologies, like ATM, IP, BISDN, as well as GSM, when it comes down to core network (CN) transports. This will consolidate the trend of 2G CN towards integrated circuit switched and packet switched services.

1.3 ENHANCING TECHNOLOGIES

1.3.1 Capacity Increasing Antennas

By increasing the number of BS antennas we can resolve the uplink limitation of WCDMA. However, this approach does not allow a single step solution because many factors intervene before completing the process. These factors include: propagation environment, BS configuration, environmental issues as a result of power levels, and network integration in terms of the RNC. However, here we consider first the BS configuration by looking at the antenna design. We need low correlation between the antennas achievable by adequate separation between the antennas. The beam forming technique may exploit a uniform linear array, where the inter-antenna spacing falls near 1/2 of a carrier wavelength. Then sectors using narrow beams will have an increased antenna gain when compared to typical sector antenna.

While pico and micro environments have higher angular diversity, the macro environment has lower angular diversity, but higher multi-path diversity. Thus, the macro environment can benefit from beam forming techniques, because the latter applies more to lower angular diversity conditions. The optimum number of branches will depend on the accuracy of the channel estimation, Godara [19,20] presents more beam forming options related to mobile applications.

1.3.2 Multi-user Detection Techniques

Multi-user Detection (MUD) techniques may apply to both the UL and DL. However, initially due to processing power constraints in the MS, MUD may be exploited first in the BS. Thus, here we look at performance enhancement primarily in the UL while implementing MUD in the BS. The two UTRA modes, i.e. FDD and TDD can benefit from MUD techniques. In fact, the joint detection algorithm is already an inherent part of the TDD mode.

Capacity within interference-limited WCDMA can improve through the use of efficient receivers. This implies that the structured multiple access interference can be dealt with at the receiver through multi-user detectors [21]. MUD techniques have been covered at length in Refs. [22,23]. Here we aim to point out some of the promising techniques, which can apply to future releases of the WCDMA mode.

Studies in MUD techniques for WCDMA BS receivers [24–26], indicate that a multi-stage parallel interference cancellation (PIC) may suite well WCDMA systems with a single spreading factor (SF). The parallel interference cancellation implies that interference gets cancelled from all users concurrently. MUD techniques for multi-service WCDMA with a variable spreading factor has been studied in Ref. [27], where a group-wise serial interference cancellation (GSIC) receiver [28–30] appears to be the most promising of the present receiver designs. In this technique, users with a given SF are also detected concurrently, after which the MAI⁹ originated by them gets suppressed by the users having different SF.

1.3.3 Software Radio Applications

Although 3G wireless communications concepts, e.g. IMT-2000 family of networks, aim towards global standardization to break away with multiple standards deployed in particular geographical areas, there is a need for multi-frequency transceivers operating in common hardware platforms for practical solutions in the medium and long-term.

This solution appears more realistic today through Software Radio (SR), the application of flexible and programmable transceivers. Thus, SR sets itself as a key technology to drive the realization of global standards in 3G systems. The evolution of GSM to UMTS alone will benefit multi-band multi-mode (GSM 900, 1800, 1900, GPRS, UMTS (FDD and TDD) terminals. On the other hand, SR not only applies to terminals or Mo-

⁹ Multiple Access Interference.

mobile Stations (MS) but also the to the Base Stations (BS). In the sequel we cover SR as part of the enabling techniques in the MS and BS.

The main limitation of the feasibility of MUD in real commercial systems has been the disproportionate processing speeds afforded by current DSP¹⁰ technology and the requirement of the detection and estimation algorithms. Although overall performance of DSPs has increased and keeps increasing, 3G systems also are pushing the signal processing capabilities higher and higher. Tasks such as high-data-rate signal acquisition, more accurate channel estimation for highly selective fading environments, fast signal quality estimation algorithms involved in power control, and optimum combining of signals for diversity gains in space and time, demand all the power a processor can produce. These demands can be realized more rapidly through Software Defined Radio (SDR).

Thus, while compatibility between standards remains attractive, SDRs will shape into software and hardware reconfigurable radios in the RF, intermediate frequency (IF), as well as base-band processing stages [31–34].

1.3.4 Implementation and Integration Aspects

Research studies aiming to improve the overall performance of multiple access techniques such as WCDMA or TDCDMA have provided interesting and applicable methods. However, these results may not necessarily be part of the first UTRA commercial systems in the next 2 years. Thus, it will be some time before techniques such as Software Radio, Adaptive Antennas, and Multi-user Detection enhance capacity, coverage and increase system stability.

Implementation and integration appear as key limitations to bring these advanced techniques into operating systems or near future¹¹ exploitable networks. Processing power demands for example, do not allow rapid implementation of the above methods. Furthermore, integrating such techniques into smaller components is a great challenge. This means, that while less optimum supporting techniques like system on a chip, maximizing power consumption, or operating at very low power come into place; the aforementioned improvements will remain academic.

At present, while UMTS frequency licensing becomes big business for governments, operators seem to have fall into the spin of supremacy and consolidation for market share and have somehow forgotten the timeliness of technology. Manufacturers are finding themselves in a race to supply plain vanilla solutions and are incapable of implementing true breakthroughs in multiple access or radio-access techniques.

Thus, it seems reasonable to think that it may be to the benefit of industry as a whole and governments themselves to concentrate on putting more resources into the realization of new communications technologies than just coping with spectrum allocation and acquisition to offer services with higher transmission rates. Such an approach will make

¹⁰ Digital Signal Processor.

¹¹ Recent evaluation on end-to-end industrial solutions do not yet show these techniques as part of a product.

UMTS a clear platform for advanced technology from the start and not just one more alternative to provide new mobile applications.

1.4 CONCLUSIONS

Chapter 1 has presented a window to perceive the environment into which UMTS¹² will develop. It has set the background to introduce *UMTS Radio Access Technology*, the aim of this book. From the impressive growth of GSM and the Internet, as well as the UMTS air-interface specification requirements, UMTS Terrestrial Radio Access (UTRA) is well positioned to play the key role in the convergence of telecommunications towards integrated services. Therefore, the contents of future chapters describe in more detail some of key elements shown generically in this chapter.

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2 SYSTEM ANALYSIS FUNDAMENTALS

2.1 FUNDAMENTALS OF SYSTEM ANALYSIS

Third generation systems focus on providing a universal platform to afford multifarious communications options at all levels, i.e. the radio as well as the core network sides. This implies the application of optimum techniques in multiple access and inter-working protocols for the physical and upper layers, respectively. This chapter discusses the background of the multiple access or radio part of the UMTS specification. Several sources [5–9] have already covered all types of fundamentals related to the air-interface. Thus, we focus only on the communications environment to access the radio link performance for coverage analysis and network dimensioning in forthcoming chapters.

2.1.1 Multiple Access Options

The access technologies utilized in UTRA are unique because of the type of implementation and not because they are new. The combination of CDMA and TDMA techniques in one fully compatible platform, make UTRA special. The WCDMA and hybrid TDMA/CDMA form the FDD and TDD modes to co-exist seamlessly to meet the UMTS services and performance requirements. In the sequel we cover the fundamental characteristics for each access technique which serves as a building block for the UTRA modes.

2.1.1.1 *Narrow-band Digital Channel Systems*

The two basic narrow-band techniques include FDMA (using frequencies) and TDMA (using time slots). In the first case, frequencies are assigned to users while guard bands maintain between adjacent signal spectra to minimize interference between channels. In the second case, data from each user takes place in time intervals called slots. The advantages of FDMA lie on efficient use of codes and simple technology requirements. But the drawbacks of operating at a reduced signal/interference ratio and the inhibiting flexibility¹ of bit rate capabilities outweigh the benefits. TDMA allows flexible rates in multiples of basic single channels and sub-multiples for low-bit rate broadcast transmission. It offers frame-by-frame signal management with efficient guard band arrangements to control signal events. However, it requires substantial amounts of signal processing resources to cope with matched filtering and synchronization needs.

¹ The maximum bit per channel remains fixed and low.

2.1.1.2 Wide-band Digital Channel Systems

Some of the drawbacks and limitations in the narrow-band channel systems made room for wide-band channel system designs. In wide-band systems the entire bandwidth remains available to each user, even if it is many times larger than the bandwidth required to convey the information. These systems include primarily Spread Spectrum (SS) systems, e.g. Direct Sequence Spread Spectrum (DSSS) and Frequency Hopping Spread Spectrum (FHSS). In DSSS, emphasized in this book, the transmission bandwidth exceeds the coherent bandwidth, i.e. the received signal after de-spreading resolves into multiple time-varying delay signals that a RAKE receiver can exploit to provide an inherent time diversity receiver in a fading environment. In addition, DSSS has greater resistance to interference effects when compared to FDMA and TDMA. The latter greatly simplifies frequency band assignment and adjacent cell interference. In addition, capacity improvements with DSSS or more commonly referred to as DS-CDMA², resulting from the voice activity factor, which we cannot apply effectively to FDMA or TDMA. With DS-CDMA, e.g. adjacent micro-cells share the same frequencies, whereas interference in FDMA and TDMA does not allow this. Other benefits and features can be found in [10–12]. Here we focus on the WCDMA or FDD mode and TDMA/CDMA or TDD mode of the UTRA solution.

2.1.1.3 The UTRA FDD Mode: WCDMA

Figure 2.1 illustrates some of the UTRA Frequency Division Duplexing (FDD) characteristics. This mode uses Wide-band Direct-Sequence Code Division Multiple Access (DS-CDMA), denoted WCDMA. To support bit rates up to 2 Mbps, it utilizes a variable spreading factor and multi-code links. It supports highly variable user data rates through the allocation of 10 ms frames, during which the user data rate remains constant, although the latter may change from frame to frame depending on the network control. It realizes a chip rate of 3.84 Mcps within 5 MHz carrier bandwidth, although the actual carrier spacing can be selected on a 200 kHz grid between approximately 4.4 and 5 MHz, depending on the interference situation between the carriers.

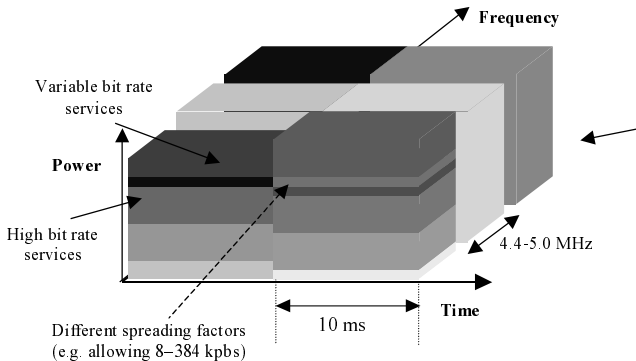


Figure 2.1 The UTRA WCDMA or FDD mode characteristics.

² Direct Sequence Code Division Multiple Access.

The FDD has a self timing point of reference through the operation of asynchronous BSs, and it uses coherent detection in the up- and downlink based on the use of pilot reference symbols. Its architecture allows the introduction of advanced capacity and coverage enhancing CDMA receiver techniques, e.g. multi-user detection and smart adaptive antennas. In addition, it will seamlessly co-exist with GSM networks through its inter-system handover functions of WCDMA.

2.1.1.4 The UTRA TDD Mode: TD/CDMA

The 2nd UTRA mode results from the combination of TDMA–FDMA and exploits spreading as part of its CDMA component. It operates in Time Division Duplexing using the same frequency channel.

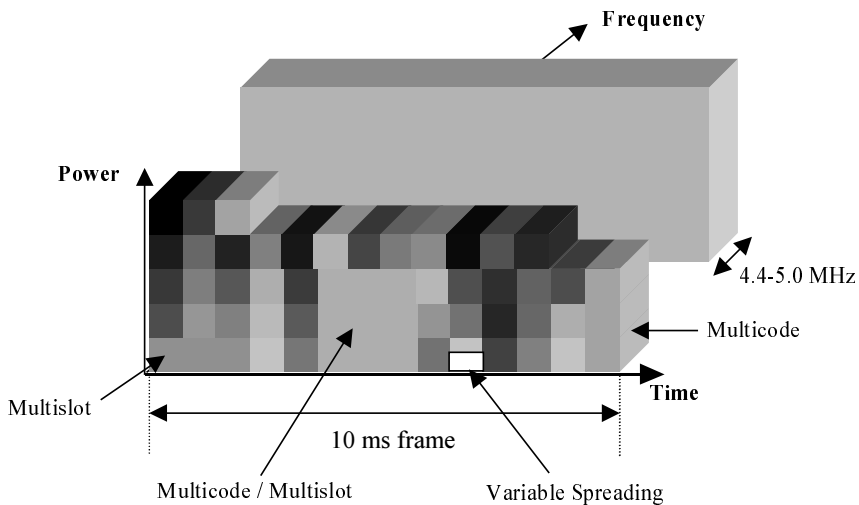


Figure 2.2 UTRA TDD mode characteristics.

In this mode, the MSs can only access a Frequency Division Multiplexing (FDM) channel at specific times and only for a specific period of time. Thus, if a mobile gets one or more Time Slots (TS) allocated, it can periodically access this set of TSs throughout the duration of the frame. Spreading codes described in Chapter 4 separate user signals within one or more slots. Hence, in the TDD mode we define a physical channel by a code, one TS, and one frequency, where each TS can be assigned to either the uplink or the downlink depending on the demand. Users may obtain flexible transmission rates by occupying several TSs of a frame as illustrated in Figure 2.2, without additional processing resources from the transceiver hardware. On the other hand, when more than one frequency channel gets occupied, utilization of transceiver resources will increase if the wide-band transmission cannot prevent it. We achieve variable data rates through either multi-code transmission with fixed spreading or through single code with variable spreading. In the 1st case, a single user or users may get multiple spreading codes within the same TS; while in the 2nd case, the physical channel spreading factor may vary according to the data rate.

2.1.2 Signal Processing Aspects

In the following, we review Signal Processing characteristics for the WCDMA as well as TD/CDMA as a base to describe key functions of the UTRA FDD and TDD modes. These include spreading aspects and modulation and coding.

2.1.2.1 The Spread Spectrum Concept

Digital designs of communications systems aim to maximise capacity utilization. We can for example increase channel capacity by increasing channel bandwidth, and/or transmitted power. In this context, CDMA operates at much lower S/N ratios as a result of the extra channel bandwidth used to achieve good performance at low signal-to-noise ratio. From Shannon's channel capacity principle [22] expressed as:

$$C = B \log_2 \left[1 + \frac{S}{N} \right], \quad (2.1)$$

where B is the bandwidth (Hz), C is the channel capacity (bits/s), S is the signal power, and N is the noise power; we can find a simple definition of the bandwidth as:

$$B = \frac{C}{1.44} \times \frac{N}{S}. \quad (2.2)$$

Thus, for a particular S/N ratio, we can achieve a low information error rate by increasing the bandwidth used to transfer information. To expand the bandwidth here, we add the information to the spreading spectrum code before modulation. This approach applies for example to the FDD mode, which uses a code sequence to determine RF bandwidth. The FDD mode has robustness to interference due to higher system processing gain³ G_p . The latter quantifies the degree of interference rejection and can be defined as the ratio of RF bandwidth to the information rate:

$$G_p = \frac{B}{R}. \quad (2.3)$$

From Ref. [23] in a spread-spectrum system, thermal noise and interference determine the noise level. Hence, for a given user, the interference is processed as noise. Then, the input and output S/N ratios can relate as:

$$\left(\frac{S}{N} \right)_o = G_p \left(\frac{S}{N} \right)_i. \quad (2.4)$$

Relating the S/N ratio to the E_b/N_o ratio⁴, where E_b is the energy per bit and N_o is the noise power spectral density, we get:

$$\left(\frac{S}{N} \right)_i = \frac{E_b \times R}{N_o \times B} = \frac{E_b}{N_o} \times \frac{1}{G_p}. \quad (2.5)$$

From the preceding equations we can express E_b/N_o in terms of the S/N input and output ratios as follows:

³ Reference processing gains for spread spectrum systems have been established between 20 and 50 dBs.

⁴ Unless otherwise specified, here we assume that N_o includes thermal and interference noise.

$$\frac{E_b}{N_o} = G_p \times \left(\frac{S}{N} \right)_i = \left(\frac{S}{N} \right)_o. \quad (2.6)$$

2.1.2.2 Modulation and Spreading Principles

In wide-band spread-spectrum systems like the FDD mode, the entire bandwidth of the system remains available to each user. To such systems, the following principles apply: first, the spreading signal has a bandwidth much larger than the minimum bandwidth required to transfer desired information or base-band data. Second, data spreading occurs by means of a code spreading signal, where the code signal is independent of the data and is of a much higher rate than the data signal. Lastly, at the receiver, de-spreading takes place by the cross-correlation of the received spread signal with a synchronized replica of the same signal used to spread the data [23].

2.1.2.2.1 Modulation

If we view Quadrature Phase-Shift Keying (QPSK) as two independent Binary Phase-Shift Keying (BPSK) modulations, then we can assume the net data rate doubles. We now provide the background for QPSK to serve as background to the applications in UTRA presented in Chapter 4.

For all practical purposes we start with M-PSK, where $M = 2^b$, and $b = 1, 2$ or 3 (i.e. 2-PSK or BPSK, 4-PSK or QPSK and 8-PSK). In the case of QPSK modulation the phase of the carrier can take on one of four values 45° , 135° , 225° , or 315° as we shall see later. The QPSK power spectral density (V^2/Hz) could be then defined as

$$S(f) = A^2 T_s \left\{ \frac{\sin[\pi T_s (f - f_c)]}{[\pi T_s (f - f_c)]} \right\}^2, \quad (2.7)$$

where f_c is the unmodulated carrier frequency, A is the carrier amplitude, and T_s is the symbol interval. When T_b is the input binary bit interval, T_s may be expressed as

$$T_s = T_b \log_2 M. \quad (2.8)$$

The power spectral density of an unfiltered M-PSK signal occupies a bandwidth which is a function of the symbol rate $r_s = (1/T_s)$. Thus, for a given transmitter symbol, the power spectrum for any M-PSK signal remains the same regardless of the number M of symbol levels used. This implies that BPSK, QPSK and 8-PSK signals each have the same spectral shape if T_s remains the same in each case.

Spectral Efficiency

For a M-ary PSK scheme each transmitted symbol represents $\log_2 M$ bits. Hence, at a fixed input bit rate, as the value of M increases, the transmitter symbol rate decreases; which means that there is an increase in spectral efficiency for larger M .

Thus, if for any digital modulation the spectral efficiency η_s , (i.e. the ratio of the input data rate r_b and the allocated channel bandwidth B) is given by:

$$\eta_s = \frac{r_b}{B}, \quad (\text{bit/s Hz}), \quad (2.9)$$

the 8-PSK spectral efficiency will be three times as great as that for BPSK. However, this will be achieved at the expense of the error probability.

Now allocating the RF bandwidth of a M-PSK signal we should remember that its spectrum rolls off relatively slowly. Therefore, it is necessary to filter the M-PSK signal so that its spectrum is limited to a finite bandpass channel region avoiding adjacent channel interference. Using *Nyquist filtering* or *raised cosine filtering* prevents the *adjacent channel interference*, as well as the *intersymbol interference* (ISI) due to filtering. The raised-cosine spectra are characterized by a factor α_B , known as the *excess bandwidth factor*. This factor lies in the range 0–1, and specifies the excess bandwidth of the spectrum compared to that of an ideal bandpass spectrum ($\alpha_B = 0$) for which the bandwidth would be $B = r_s$. Typical values of α_B used in practice are 0.3–0.5 [3].

Thus, for M-PSK transmission using the Nyquist filtering with roll-off α_B the required bandwidth will be given by

$$B = r_s (1 + \alpha_B). \quad (2.10)$$

Then the maximum bit rate in terms of the transmission bandwidth B , and the roll-off factor α_B can be defined as

$$r_b = \frac{B \log_2 M}{1 + \alpha_B}. \quad (2.11)$$

However, if we assume an M-PSK with an ideal Nyquist filtering (i.e. $\alpha_B = 0$) the signal spectrum is centred on f_c , it is constant over the bandwidth $B = 1/T_s$, and it is zero outside that band. Then the transmitted bandwidth for the M-PSK signal, and the respective spectral efficiency are given by

$$B = \frac{1}{T_b \log_2 M}, \quad \text{and} \quad \eta_s = \frac{r_b}{B} = \log_2 M. \quad (2.12)$$

Bit Error Rate (BER) Performance

In M-PSK modulation, the input binary information stream is first divided into b bit blocks, and then each block is transmitted as one of M possible symbols; where each symbol is a carrier frequency sinusoid having one of M possible phase values [3]. Among the M-PSK schemes, BPSK and QPSK are the most widely used. Nevertheless, here we review only the QPSK scheme. In QPSK each transmitted symbol (Figure 2.3) represents two input bits as follows:

<i>Input bits</i>	<i>Transmitted symbols</i>
00	$A \cos(\omega_c t + 45^\circ)$
01	$A \cos(\omega_c t + 135^\circ)$
11	$A \cos(\omega_c t + 235^\circ)$
10	$A \cos(\omega_c t + 315^\circ)$

The conversion from binary symbol to phase angles is done using Gray coding. This coding permits only one binary number to change in the assignment of binary symbols to adjacent phase angles, thereby minimizing the demodulation errors, which in a digital receiver result from incorrectly selecting a symbol adjacent to a correct one.

Figure 2.3 illustrates a block diagram frequently used for any form of M-PSK modulation. For QPSK, the multiplexer basically converts the binary input stream into two parallel, half rate signal $v_I(t)$ and $v_Q(t)$ (i.e. the in-phase and quadrature signals). These signals taking values $+A/\sqrt{2}$ or $-A/\sqrt{2}$ in any symbol interval, are fed to two balanced modulators with input carriers or relative phase 0° and 90° , respectively. Then the QPSK signal could be given by

$$s(t) = v_I(t) \cos w_c t + v_Q(t) \sin w_c t. \quad (2.13)$$

If we assume T_s is the time interval and $v_I = +A/\sqrt{2}$ and $v_Q = -A/\sqrt{2}$, it can be shown that the output $s(t)$ is

$$s(t) = A \cos \left(w_c t - \frac{\pi}{4} \right). \quad (2.14)$$

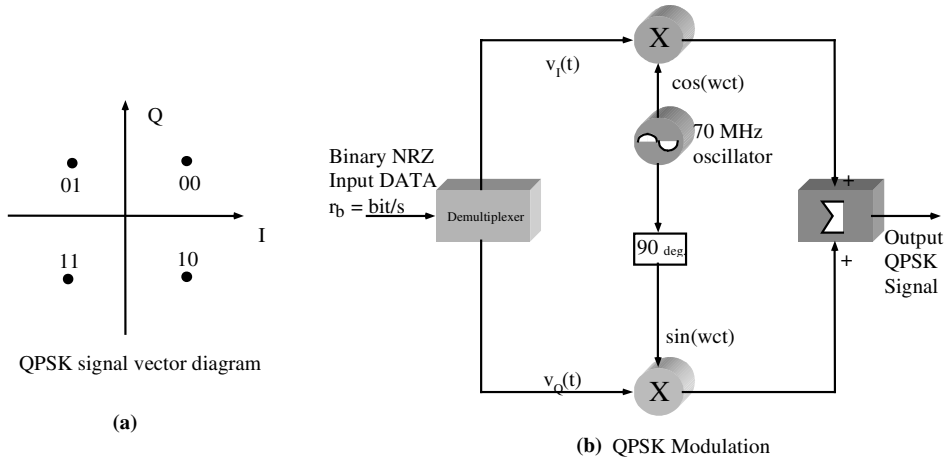


Figure 2.3 QPSK configuration, after [3].

Assuming a *coherent demodulator*, the latter includes a *quadrature detector* consisting of two balanced multipliers with carrier inputs in phase quadrature, followed by root-Nyquist filter in the output I and Q arms. Then, the resultant I and Q signals are sampled at the centre of each symbol to produce the demodulator output I and Q signals, which in turn are delivered to the decoder [3].

Generally, an M-PSK modulator produces symbols with one of M phase values spaced $2\pi/M$ apart. Then each signal is demodulated correctly at the receiver when the phase is within π/M radians of the correct phase at the demodulator sampling instant. If noise is present, evaluation of the probability of error requires a calculation of the probability

that the received phase lies outside the angular segment within π/M radians of the transmitted symbol at the sampling instant.

Therefore, the probability that a demodulator error occurs can be referred to as the *symbol error probability* P_s . In the context of the M -ary modulation scheme with $M = 2^b$ bits, each symbol represents b bits. The most probable symbol errors are then those that choose an incorrect symbol adjacent to the correct one. When using Gray coding, only one bit error results from a symbol error. Thereupon, the bit error probability P_b is related to the symbol error probability by

$$P_b = \frac{P_s}{m}. \quad (2.15)$$

In the case of QPSK, symbol errors occur when the noise pushes the received phasor into the wrong quadrant as illustrated in Figure 2.4. In this figure it is assumed that the transmitted symbol has a phase of $\pi/4$ rad, corresponding to the demodulator I and Q values of $v_I = V$ and $v_Q = V$ volts (i.e. noise-free case). Thus, if we consider that the noise phasors (n_1 and n_2) are pointing in directions that are most likely to cause errors, then a symbol error will occur if either n_1 or n_2 exceeds V .

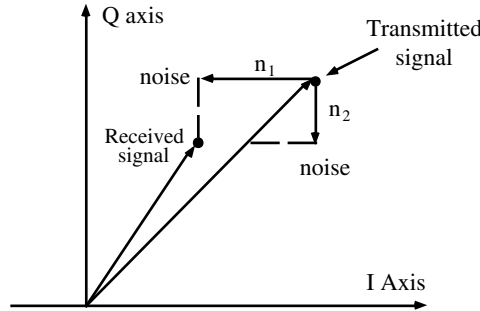


Figure 2.4 Transmitted and received signal vectors [3].

Now, if for simplicity we also assume that a QPSK signal is transmitted without Nyquist filtering and demodulated with hard-decisions, the probability of a correctly demodulate symbol value is equal to the product of the probabilities that each demodulator low-pass filter output lies in the correct quadrant. Then the probability that the demodulated symbol value is correct is given by

$$P_c = (1 - P_{e1})(1 - P_{e2}), \quad (2.16)$$

where P_{e1} and P_{e2} are the probabilities that the two filter output sample values are in the wrong quadrant. When showing that the low-pass filters are equivalent to integrators, which is the optimum choice if Nyquist filtering is not used, P_{e1} and P_{e2} can be expressed as

$$P_{e1} = P_{e2} = Q\left(\sqrt{\frac{E_s}{N_o}}\right), \quad (2.17)$$

where $E_s = A^2 T_s / 2$ is the energy per symbol, $N_o/2$ is the two-sided noise power spectral density (in V^2/Hz) at the demodulator input, and the function $Q(x)$ is the complementary integral Gaussian function. The error function $\text{erf}(x)$ given by

$$\text{erf}(x) = \frac{2}{\sqrt{\pi}} \int_0^x \exp(-y^2) dy, \quad (2.18)$$

and complementary error function $\text{erfc}(x)$ expressed as

$$\text{erfc}(x) = 1 - \text{erf}(x) \quad (2.19)$$

are not fully identical to the integral Gaussian function $G(x)$, and the complementary integral $G_c(x)$ or $Q(x)$ in our case. Now if we assume $G_c(x) = Q(x)$, we can use the following

$$Q(x) = \frac{1}{2} \text{erfc}\left(\frac{x}{\sqrt{2}}\right). \quad (2.20)$$

function to evaluate our error probabilities. Then since $P_{e1} = P_{e2}$, the *symbol error probability* could be written as

$$P_s = 1 - P_c = 2P_{e1} - P_{e1}^2, \quad (2.21)$$

which at $P_{e1} \leq 1$ becomes

$$P_s \approx 2P_{e1}. \quad (2.22)$$

Substituting P_{e1} from equation (2.17) into equation (2.22), the QPSK *symbol error probability* can be given by

$$P_s \approx 2Q\left(\sqrt{\frac{E_s}{N_o}}\right). \quad (2.23)$$

Now, for QPSK $E_s = 2E_b$, where E_b is the energy per bit; then making use of equation (2.C.3) we get the bit error rate probability P_{BER} for the QPSK system as follows:

$$P_{\text{BER}} = Q\left(\sqrt{\frac{2E_b}{N_o}}\right). \quad (2.24)$$

Here we found the P_{BER} assuming that no Nyquist filtering was present. However according to Ref. [3], this P_{BER} also holds when root-Nyquist filters are used at the transmitter and receiver under the assumption that the demodulator input energy E_b and the noise power density N_o are the same for both cases.

2.1.2.3 CDMA System Performance

As noted earlier, CDMA systems tolerate more interference than typical TDMA or FDMA systems. This implies that each additional active radio user coming into the

network increases the overall level of interference to the cell site receivers receiving CDMA signals from mobile station transmitters. This depends on its received power level at the cell site, its timing synchronization relative to other signals at the cell site, and its specific cross-correlation with other CDMA signals. Consequently, the number of CDMA channels in the network will depend on the level of total interference that the system can tolerate. As a result, the FDD mode behaves as an interference limited system, where technical design will play a key role in the overall quality and capacity performance. Thus, despite advanced techniques such as multi-user detection and adaptive antennas, a robust system will still need a good bit error probability with a higher level of interference.

When we consider that at the cell site all users receive the same signal level assuming Gaussian noise as interference, the modulation method has a relationship that defines the bit error rate as a function of the E_b/N_o ratio. Therefore, if we know the performance of the signal processing methods and tolerance of the digitized information to errors, we can define the minimum E_b/N_o ratio for a balanced system operation. Then, if we maintain operation at this minimum E_b/N_o , we can obtain the optimum performance of the system. From Ref. [23] we can define the relationship between the number of mobile users M , the processing gain G_p , and the E_b/N_o ratio as follows:

$$M \approx \frac{G_p}{(E_b/N_o)}. \quad (2.25)$$

On the other hand, the E_b/N_o performance can be seen better in relation with Shannon's limit in AWGN⁵, which simplified can be presented as:

$$\frac{C}{B} < \frac{1}{\log_e 2} \left(\frac{S}{N} \right) \quad \text{and} \quad \frac{C}{B} < \frac{1}{\log_e 2} \left(\frac{E_b}{N_o} \right) \left(\frac{C}{B} \right), \quad (2.26)$$

then

$$\frac{E_b}{B} \geq \log_e 2 = 0.69 = -1.59 \text{ dB} \quad (2.27)$$

provides error-free communications. Then for Shannon's limit the number of users can be projected from:

$$M = \frac{G_p}{0.69} = 1.45 G_p. \quad (2.28)$$

Shannon's theoretical limit implies that a WCDMA system can support more users per cell than classical narrow-band systems limited by the number of dimensions. On the other hand, this limit in practice has $E_b/N_o = 6 \text{ dB}$ as a typical value. However, due to practical limitations, accommodating as many users in a single cell as indicated by Shannon's limit is not possible in a CDMA system, and this applies also to the UTRA FDD. Thus, cell capacity depends upon many factors, (e.g. receiver-modulation per-

⁵ Additive White Gaussian Noise.

formance, power control accuracy, intersystem interference), and the upper-bound theoretical capacity of an ideal noise-free CDMA channel has also limitations by the processing gain G_p [23].

Multiple transmissions in neighbouring CDMA cells using the same carrier frequency cause interference, denoted by β factor. This event will cause reduction of the number of users in a cell, because the interference from users in other cells has to be added to the interference generated by the other mobiles in the user's cell. β may range from 0.4 to 0.55. In addition to the interference factor, we also introduce the imperfect power control or power control accuracy factor α , which ranges from 0.5 to 0.9. Interference can be reduced by the voice activity factor v ranging from 0.45 to 1.0. If we use directional antennas at the base station, the sectorized cell will have a sectors, the antennas used at the cell each will radiate into a sector of $360/a$ degrees, resulting in an interference improvement factor λ . Average values for β , α , v and λ (3 sector cell) are 0.5, 0.85, 0.6 and 2.55, respectively [23]. Then incorporating all the preceding factors the user capacity equation becomes:

$$M \approx \frac{G_p}{E_b/N_o} \times \frac{1}{1+\beta} \times \alpha \times \frac{1}{v} \times \lambda. \quad (2.29)$$

In the forthcoming section we also review pseudorandom sequences as part of the signal processing aspects relevant for the operation of the UTRA modes.

2.1.2.4 Pseudorandom Sequences

Pseudorandom noise (PN), i.e. deterministic periodic sequences in WCDMA perform the following tasks: bandwidth spreading of the modulated signal to wider transfer bandwidths, signal discrimination among users transmitting in the same bandwidth of multiple access methods.

The characteristics of these sequences are: 1/2 relative frequencies of zero and one; for zeros or ones half of all run lengths are of length 1; one-quarter are of length 2, one-eighth are of length 3; etc. When a PN sequence shifts by any non-zero number of elements, the resulting sequence will have an equal number of agreements and disagreements with respect to the original sequence.

We generate PN sequences by combining feedback shift register outputs. This register consists of consecutive two-state memory or storage stages and feedback logic. Binary sequences shift through the shift register in response to clock pulses. We logically combine the contents of the stages to produce the input to the first stage. The initial contents of the stages and feedback logic determine the successive contents of the stages. We call a feedback shift register and its output linear when the feedback logic consists entirely of modulo-2 adders.

The output sequences get classified as either maximal length or non-maximal length. The first ones are the longest sequences that can be generated by a given shift register of a given length, while all other sequences besides maximal length sequences are non-maximal length sequences. In the binary shift register sequence generators, the maximal length sequence has $2^n - 1$ chips, where n is the number of stages in the shift registers. A

property of the maximal length sequences implies that for an n – stage linear feedback shift register, the sequence repetition period in clock pulses is $T_0 = 2^n - 1$. When a linear feedback shift register generates a maximal sequence, then all its non-zero output sequences result in maximal sequences, regardless of the initial stage. A maximal sequence contains $2^{n-1} - 1$ zeros and 2^{n-1} ones per period.

Other characteristics of PN sequences, e.g. properties of maximal length PN sequences, auto-correlation, cross-correlation, and orthogonal functions are described in Ref. [23]. In the following we review additional WCDMA characteristics, such as power control and soft handover.

2.1.2.5 Power Control Characteristics

Accurate and *fast power* control becomes imperative in WCDMA. It increases network stability and prevents near-far effect (UL) or cell blocking by overpowered MSs. Open loop or slow power control would not cope with the highly non-correlated fast fading between UL and DL as a consequence of the large frequency separation. Chapter 4 describes the technical details of fast power control. The latter applies to both the UL and DL. In the 1st case, the BS balances the MS's power after comparing the received Signal-to-Interference Ratio (SIR) to a SIR_{target} . In the 2nd case, we aim to provide sufficient additional power to MSs at the cell edges in order to minimise other-cell interference.

The *outer loop* or slow power control adjusts the BS's reference SIR_{target} based on the needs of a single or independent radio link. It aims to maintain constant quality established by the network through a target BER or FER for example. The RNC handles the command steps to lower or increase the reference SIR_{target} .

2.1.2.6 Soft Handovers Characteristics

While there is hard handover for carrier change or hierarchical cell transition, and inter-system hand over to pass from FDD to TDD or GSM, in WCDMA two types of soft handovers characterise the cell transition process. These include *Softer* and *Soft* handovers. In the 1st case, a MS finds itself in the overlapped cell coverage area of two adjacent sectors of a BS. The MS communicates simultaneously with BS through two channels (2 DL codes) corresponding one to each sector. The MS's rake receives and processes the two signals, where its fingers generate the necessary de-spreading codes for each sector. The UL process occurs in the BS, where the BS receives the MS's channel in each sector and routes them to the same rake receiver for the typical maximal ratio combining process under one active power control loop per connection.

In the 2nd case, i.e. soft handover, a MS finds itself in the overlapping cell coverage area of two sectors corresponding to different BSs. Communications between MS and BS occur simultaneously through two channels, one from each BS. In the DL, the MS receives both signals for maximal ratio combining. In the UL, the MS code channel arrives from both BS, and is routed to the RNC for combining, in order to allow the same frame reliability indicator provided for outer loop power control when selecting the best frame. Two active power control loops participate in soft handover, i.e. one for each BS.

While softer handover may occur only in about 10 % of links, soft handover may occur in about 30 % of the links. Thus, for the latter provision in terms of extra power, Rake processing, RNC transmission lines will be essential.

2.2 THE 3G COMMUNICATIONS ENVIRONMENT

This section provides dedicated reference models for the test environments cited in the forthcoming chapters, in particular the deployment contents presented in Chapter 7. These test environments aim to cover the range of UMTS operating environments. Thus, the necessary parameters to identify the reference models include the test propagation environments, traffic conditions and user information rate for reference voice and data services. It also presents some performance objectives and criteria for each operating environment.

The test operating environments are direct extracts from the recommendations considered for the evaluation process of the Radio Transmission Technologies (RTTs) submitted to ETSI and ITU as UTRA candidate solutions. Thus, the contents bring together or are based entirely on the specifications outlined in Refs. [1–4].

2.2.1 Mapping High Level Requirements onto Test Environments

This section maps high level service requirements summarized in Chapter 1 onto test environments described in the next sections. The mapping identifies the maximum user bit rate in each test environment, together with the maximum speed, expected range and associated wide-band channel model. Table 2.1 illustrates the suggested reference values.

Table 2.1 Radio Transmission Test Environments [2]

High level description	Maximal bit rate	Maximal speed	Test environment channel models	Cell coverage
Rural outdoor	144 kbit/s	500 km/h	Vehicular channel A & B	Macrocell
Suburban outdoor	384 kbit/s	120 km/h	Outdoor to indoor and pedestrian channel A & B	Microcell
			Vehicular channel A	Macrocell
Indoor/ Low range out-door	2048 kbit/s	10 km/h	Indoor channel A & B	Picocell
			Outdoor to indoor and pedestrian channel A	Microcell

2.2.1.1 Reference Services

The UMTS minimum set of services to appropriately characterize bearers include: ranges of supported data rates, BER requirements, one way delay requirements, activity factor and traffic models. The forthcoming section covers traffic models and Table 2.2 provides example values for access reference services, such as: Speech, Low Delay Data (LDD), Long delay Circuit Switched Data (LCD), Unrestricted Delay Data (UDD). The latter corresponds to connectionless data for packet services; 12.2 kbps corresponds to AMR rates not necessarily part of the early test recommendations [2].

Table 2.2 Reference Data Rates

Test environments	Indoor office	Outdoor to indoor and pedestrian	Vehicular 120 km/h	Vehicular 500 km/h
Speech (kbps)	8, 12, 2	8, 12, 2	8, 12, 2	8, 12, 2
BER	$\leq 10^{-3}$	$\leq 10^{-3}$	$\leq 10^{-3}$	$\leq 10^{-3}$
Delay (ms)	20	20 ms	20	20
Activity (%)	50	50%	50	50
LDD data (kbps)	144, 384, 2048	64, 144, 384	32, 144, 384	32, 144
BER	$\leq 10^{-6}$	$\leq 10^{-6}$	$\leq 10^{-6}$	$\leq 10^{-6}$
Delay (ms)	50	50	50	50
Activity (%)	100	100	100	100
UDD data (packet) connection-less information types	See Section 2.2.8.1 and Table 2.7	See Section 2.2.8.1 and Table 2.7	See Section 2.2.8.1 and Table 2.7	See Section 2.2.8.1 and Table 2.7
LCD data (kbps)	144, 384, 2048	64, 144, 384	32, 144, 384	32, 144
BER	$\leq 10^{-6}$	$\leq 10^{-6}$	$\leq 10^{-6}$	$\leq 10^{-6}$
Delay (ms)	300	300	300	300
Activity (%)	100	100	100	100

2.2.2 Channel Types

As a global standard, UMTS aims for a broad range of environment characteristics, e.g. large and small cities, tropical, rural, and desert areas. Reference parameters describing the propagation models for these areas include:

1. time delay-spread with its structure and its statistical variability (e.g. probability distribution of time delay spread);
2. geometrical path loss rule (e.g. R^{-4}) and excess path loss;
3. shadow fading and multi-path fading characteristics (e.g. Doppler spectrum, Rician vs. Rayleigh) for the envelope of channels; and
4. operating radio frequency.

Characterization of rapid fading variation occurs by the channel impulse response, where response modelling takes place using a tapped delay line implementation. The Doppler spectrum characterizes the tap variability. These environments are represented in terms of propagation from [2] by: indoor office, outdoor to indoor and pedestrian, vehicular, and mixed.

2.2.3 Indoor Office

This environment has small cells and low transmit powers, where both BSs and pedestrian users remain indoors, with path loss rule varying due to scatter and attenuation by walls, floors, and metallic structures, e.g. partitions and filing cabinets, all producing some type of shadowing effects. These effects include: log-normal shadow fading with standard deviation of 12 dB, and fading ranges from Rician to Rayleigh, with Doppler frequency offsets set by walking speeds.

The indoor office path loss is based on the COST⁶ 231 model; this low increase of path loss versus distance is a worst case from the interference point of view and is defined as follows:

⁶ COST 231 Final Report (e.g. propagation environments) , Commission of the European Communities.

$$L = L_{FS} + L_c + \sum k_{wi} L_{wi} + n^{((n+2)/(n+1)-b)} * L_f, \quad (2.30)$$

where L_{FS} is the free space between transmitter and receiver, L_c is the constant loss, k_{wi} is the number of penetrated walls of type i , n is the number of penetrated floors, L_{wi} is the loss of wall type i , L_f is the loss between adjacent floors, and b is the empirical parameter. L_c normally is set to 37 dB. $n = 4$ is the average for an indoor office environment. For capacity calculations in moderately pessimistic environments, the model can be modified to $n = 3$.

Table 2.3 Weighted Average for Loss Categories Indoor Environment [2]

Loss category	Description	Factor (dB)
L_f	Typical floor structures (i.e. offices) - hollow pot tiles - reinforced concrete - thickness typ. < 30 cm	18.3
L_{w1}	Light internal walls - plasterboard - walls with large numbers of holes (e.g. windows)	3.4
L_{w2}	Internal walls - concrete, brick - minimum number of holes	6.9

Under the simplifying assumptions of the office environment the indoor path loss model has the following form:

$$L = 37 + 30 \log_{10}(R) + 18.3 n^{((n+2)/(n+1)-0.46)}, \quad (2.31)$$

where R is the transmitter-receiver separation given in metres and n is the number of floors in the path. L shall in no circumstances be less than free space loss. A log-normal shadow fading standard deviation of 12 dB can be expected.

Physical deployment

The specific assumptions about the indoor physical deployment environment can be summarized from [2] as: area per floor = 5000 m², number of floors = 3, room = 20 × 10 × 3 (m), corridor = 100 × 5 × 3 (m), log-normal standard deviation = 12 (dB), MS velocity = 3 (km/h). Figure 2.5 illustrates a default deployment scheme, where base stations use omnidirectional antennas. For spectrum efficiency evaluation, quality statistics should only be collected in the middle floor. See the mobility model in [2].

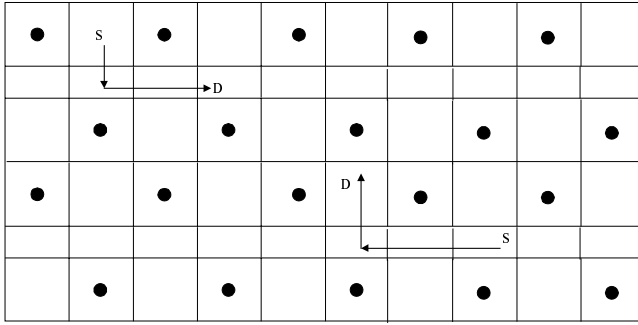


Figure 2.5 Indoor office model and deployment scheme.

2.2.4 Outdoor to Indoor and Pedestrian

We also characterize this environment by small cells and low transmit power, where BS with low antenna heights stay outdoors while pedestrian users may remain in the streets and/or inside buildings and residences. A geometrical path loss rule of R^{-4} may satisfy; however, a wider range would serve better. When the path has a line of sight on a canyon-like street, the path loss follows a R^{-2} rule with Fresnel zone clearance. In the absence of Fresnel zone clearance, a path loss rule of R^{-4} will apply, but a range up to R^{-6} may occur due to trees and other obstructions along the path. Log-normal shadow fading with a standard deviation of 10 dB applies to outdoors and 12 dB to indoors. Building penetration loss averages 12 dB with a standard deviation of 8 dB. Walking speeds set Rayleigh and/or Rician fading rates, but not faster fading due to reflections from moving vehicles.

Generally, the total transmission loss L (dB) between isotropic antennas for outdoor transmission loss equals the sum of:

- free space loss, L_{fs} ,
- the diffraction loss from rooftop to the street, L_{rts} , and
- the reduction due to multiple screen diffraction past rows of buildings, L_{msd}

where L_{fs} and L_{rts} are independent of the BS antenna height, while L_{msd} depends on whether the base station antenna is at, below or above building heights. Then L is given as:

$$L(d) = L_{fs} + L_{rts} + L_{msd} . \quad (2.32)$$

Given a MS-to-BS separation R , the free space loss L_{fs} between them is given by:

$$L_{fs} = -10 \log_{10} \left(\frac{\lambda}{4\pi d} \right)^2 . \quad (2.33)$$

The diffraction from the rooftop down to the street level gives the excess loss to the mobile station:

$$L_{\text{rts}} = -10 \log_{10} \left[\frac{\lambda}{2\pi^2 r} \left(\frac{1}{\theta} - \frac{1}{2\pi + \theta} \right)^2 \right], \quad (2.34)$$

where

$$\theta = \tan^{-1} \left(\frac{|\Delta h_m|}{x} \right) \quad \text{and} \quad r = \sqrt{(\Delta h_m)^2 + x^2}. \quad (2.35)$$

Δh_m is the difference between the mean building height and the mobile antenna height; x is the horizontal distance between the mobile and the diffracting edges [2].

For the general model, the multiple screen diffraction loss from the base antennas due to propagation past rows of buildings is:

$$L_{\text{msd}} = -10 \log_{10} (Q_M^2), \quad (2.36)$$

where Q_M is a factor dependent on the relative height of the base station antenna as being either at, below or above the mean building heights [13,14].

In this case the base-station antenna height is near mean rooftop level, then:

$$Q_M = \frac{d}{R}. \quad (2.37)$$

The total transmission loss for the near rooftop case then becomes:

$$L = -10 \log_{10} \left(\frac{\lambda}{2\sqrt{2}\pi R} \right)^2 - 10 \log_{10} \left[\frac{\lambda}{2\pi^2 r} \left(\frac{1}{\theta} - \frac{1}{2\pi + \theta} \right)^2 \right] - 10 \log_{10} \left(\frac{d}{R} \right)^2. \quad (2.38)$$

When $\Delta h_b = -5$ m, $\Delta h_m = 10.5$ m, $x = 15$ m, and $b = 80$ m, as typical in an urban and suburban environment, the above path loss expression reduces to a simple function of the transmitter to receiver distance R (km) and frequency f (MHz),

$$L = 40 \log_{10}(R) + 30 \log_{10}(f) + 49, \quad (2.39)$$

where R is the base station–mobile station separation in kilometres; f is the carrier frequency of 2000 in MHz for UMTS band application. L shall in no circumstances be less than free space loss. This model applies to the non-line-of-sight (NLOS) case only and describes worse case propagation assuming log-normal shadow fading with a standard deviation of 10 dB for outdoor users and 12 dB for indoor users. The average building penetration loss is 12 dB with a standard deviation of 8 dB [2].

A more detailed model uses a recursive approach [15] that calculates the path loss as a sum of LOS and NLOS segments. The shortest path along streets between the BS and the MS has to be found within the Manhattan environment and the path loss (dB) is given by:

$$L = 20 \log_{10} \frac{4\pi d_n}{\lambda}, \quad (2.40)$$

where d_n is the illusory distance, λ is the wavelength, n is the number of straight street segments between BS and MS (along the shortest path).

The illusory distance = sum of the street segments obtained recursively using the expressions $k_n = k_{n-1} + d_{n-1}c$ and $d_n = k_n s_{n-1} + d_{n-1}$ where c is a function of the street crossing, e.g. for a 90° street crossing $c = 0.5$. Furthermore, s_{n-1} = length in meters of the last (straight path) segment. We set the initial values as: $k_0 = 1$ and $d_0 = 0$, and we get the illusory distance as the final d_n when the last segment has been added. When extending the model to cover the micro cell dual slope behaviour, we express L as:

$$L = 20 \log_{10} \left(\frac{4\pi d_n}{\lambda} D \left(\sum_{j=1}^n s_{j-1} \right) \right) \quad \text{where } D(x) = \begin{cases} x / x_{br}, & x > x_{br} \\ 1, & x \leq x_{br} \end{cases} \quad (2.41)$$

Before the break point x_{br} the slope is 2, after the break point it increases to 4. The break point x_{br} is set to 300 m; x is the distance from the transmitter to the receiver.

When taking into account propagation effects going above roof-tops, path loss calculation occurs according to the shortest geographical distance using the COST Walfish-Ikegami Model and with antennas below roof-tops, i.e.:

$$L = 24 + 45 \log(d + 20), \quad (2.42)$$

where d is the shortest physical geographical distance from the transmitter to the receiver in metres. The final path loss results from the minimum between the street path loss value and the path loss based on the shortest geographical distance, i.e.

Pathloss = min(manhattan pathloss, macro path loss)

This path loss model applies only to microcell coverage with antenna located below the roof-top. When the urban structure has macrocell coverage, the first path loss case applies [2].

Physical Deployment

The same test service requirements of the preceding section (indoor) apply for the outdoor-to-indoor pedestrian environment. The specific assumptions about the physical deployment environment include:

- Indoor: building penetration loss = 12 dB, standard deviation = 8, log-normal standard deviation = 10 and MS velocity = 3 km.
- Outdoor: building penetration loss = NA, log-normal standard deviation = 10, MS velocity = 3 km.

A Manhattan-like structure defined for the outdoor to indoor and pedestrian environment can be applied with following assumptions: 6.5 km^2 , $200 \times 200 \text{ m}$ block, 30 m street width, and 10 m BS height. Figure 2.6 illustrates the default deployment scheme with BS using omni-directional antennas, where expected quality statistics would only arise from among cells marked with a T. See the mobility model in Ref. [2].

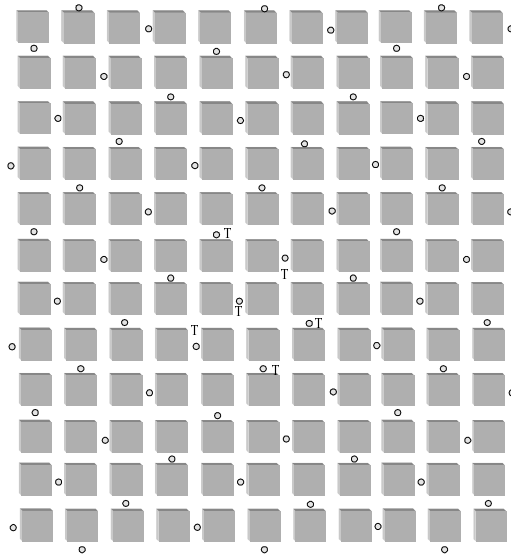


Figure 2.6 Manhattan-like urban model example.

2.2.5 Vehicular

We characterize the vehicular environment by larger cells and higher transmit power. The recommendations imply a geometrical path loss rule of R^{-4} and log-normal shadow fading with 10 dB standard deviation in urban and suburban areas. Rural areas with flat terrain will have lower path loss than that of urban and suburban areas. In mountainous areas we can apply a path loss rule closer to R^{-2} assuming that BS locations do not suffer from blocking. Vehicle speeds set Rayleigh fading rates.

The vehicular environment applies to scenarios in urban and suburban areas outside the high rise core where the buildings have nearly uniform height. In this model, the BS has antenna height above rooftop level with Q_M as the factor depending on the relative height of the BS antenna

$$Q_M = 2.35 \left(\frac{\Delta h_b}{d} \sqrt{\frac{b}{\lambda}} \right)^{0.9}, \quad (2.43)$$

where Δh_b is the height difference between BS antenna and the mean building rooftop height, and b is the average separation between rows of buildings. Then the total transmission loss for the above rooftop case becomes:

$$L = -10 \log_{10} \left[\left(\frac{\lambda}{4\pi R} \right)^2 \right] - 10 \log_{10} \left[\frac{\lambda}{2\pi^2 r} \left(\frac{1}{\theta} - \frac{1}{2\pi + \theta} \right)^2 \right] - 10 \log_{10} \left[(2.35)^2 \left(\frac{\Delta h_b}{R} \sqrt{\frac{d}{\lambda}} \right)^{1.8} \right]. \quad (2.44)$$

In the building environment, measurements [16] showed that the path loss slope behaves as a linear function of the base station antenna height relative to the average rooftop Δh_b . Then, the above path loss equation can be defined as:

$$L = -10 \log_{10} \left[\left(\frac{\lambda}{4\pi R} \right)^2 \right] - 10 \log_{10} \left[\frac{\lambda}{2\pi^2 r} \left(\frac{1}{\theta} - \frac{1}{2\pi + \theta} \right)^2 \right] - 10 \log_{10} \left[(2.35)^2 \frac{(\Delta h_b \sqrt{d/\lambda})^{1.8}}{R^{2(1-4 \times 10^{-3} \Delta h_b)}} \right], \quad (2.45)$$

$$\text{where } \theta = \tan^{-1} \left(\frac{|\Delta h_m|}{x} \right), \quad r = \sqrt{(\Delta h_m)^2 + x^2}. \quad (2.46)$$

Δh_m is the difference between the mean building height and the mobile antenna height; and x is the horizontal distance between the mobile and the diffracting edges.

If $\Delta h_m = 10.5$ m, $x = 15$ m, and $b = 80$ m, typical urban and suburban environment values with average four story building heights, then the above path loss expression L reduces to a simple function of the transmitter to receiver distance R (km). We measure the BS antenna height from the average rooftop Δh_b in metres, and frequency f in MHz.

$$L = [40(1 - 4 \times 10^{-3} \Delta h_b)] \log_{10}(R) - 18 \log_{10}(\Delta h_b) + 21 \log_{10}(f) + 80 \quad (\text{dB}). \quad (2.47)$$

When we assume a fixed BS antenna height of 15 m above the average rooftop, i.e. $\Delta h_b = 15$ m, and a carrier frequency of 2000 MHz, the vehicular path loss L becomes:

$$L = 128.1 + 37.6 \log_{10}(R). \quad (2.48)$$

L shall in no circumstances be less than the free space loss. This model applies to the Non-Line of Sight (NLOS) case only and describes the worse case propagation. Log-normal shadow fading with 10 dB standard deviation are assumed in both urban and suburban areas [2]. The path loss model is valid for a range of Δh_b from 0 to 50 m.

Physical Deployment

When assuming a cell radius of 2000 m, services up to 144 kbit/s apply; and with a cell radius of 500, services above 144 kbps (e.g. 384 kbps) apply. If the BS antenna height remains above the average roof top height of 15 m, a hexagonal cell lay out with distances between base stations equal to 6 km can serve as reference. Figure 2.7 illustrates this type of tri-sector cells using the GSM based antenna pattern shown in Figure 2.8.

Mobility Model

The vehicular reference mobility model uses a pseudorandom mobility model with semi-directed trajectories, the mobile's position gets updated according to the decorrelation length and direction can change at each position update following a given probability within a sector. For a reference example, we can assume constant mobile speeds of 120 km/h, with a direction probability change at position update of 0.2, and a maximal angle for direction update of 45° . Mobiles get uniformly distributed on the map and their direction randomly chosen at initialization [2].

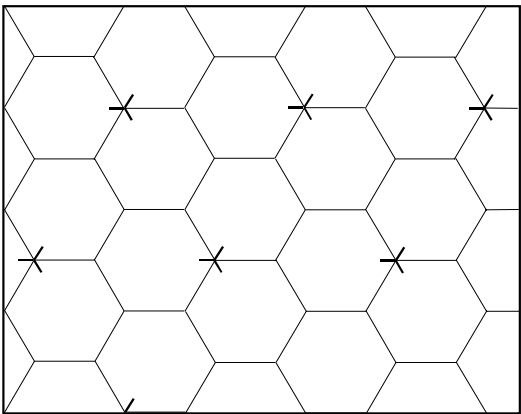


Figure 2.7 Vehicular deployment model.

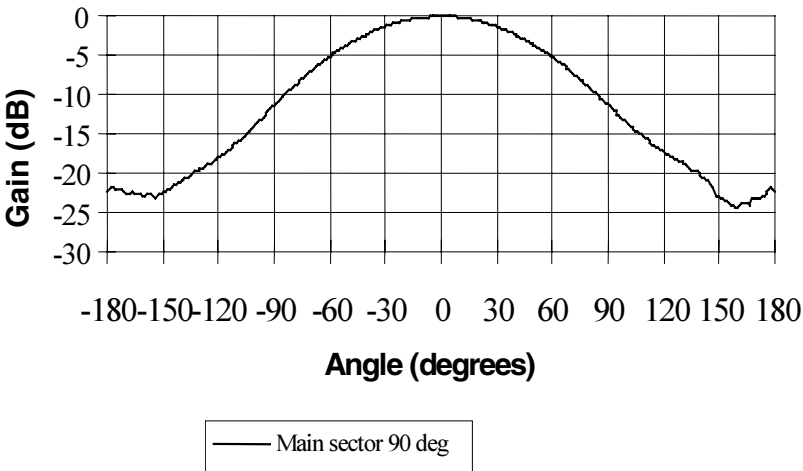


Figure 2.8 Horizontal antenna pattern example based on GSM three sectored antenna [2].

2.2.6 Mixed

Here we illustrate a mix environment by a vehicular (macrocells) and an outdoor to indoor (microcells) environment taking place in the same geographical area. In this area, fast moving terminals (e.g. vehicles, trains) will most likely connect to the macrocells to reduce the hand-off rate (number of hand-offs per minute) and slow moving terminals (pedestrians, boats on a shore) will probably connect to the microcells to achieve high capacity. The reference assumptions [2] about combined outdoor and vehicular physical deployment environments can be as follows: the log-normal standard deviations = 10 dB for both outdoor and vehicular environments, mobile speeds are 3

km/h and 80–120 km/h for outdoor and vehicular, respectively. The proportions of users are 60% and 40% for outdoor and vehicular, respectively.

Mobility Model

The mobility model will follow outdoor and vehicular patterns allowing appropriate handover between macrocells and microcells for all users.

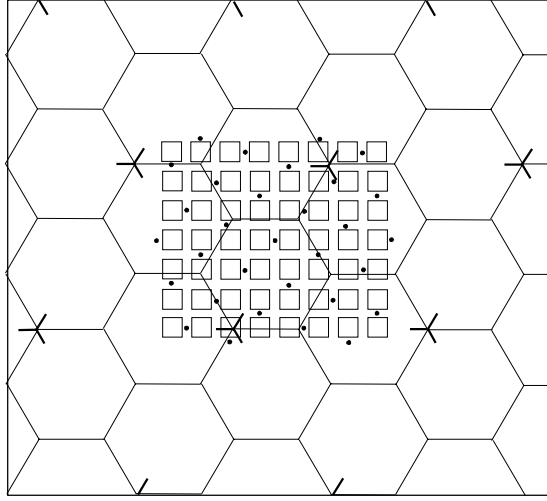


Figure 2.9 Mixed physical environment and proposed deployment model.

2.2.6.1 Long-term Fading Decorrelation Length

We characterize the log-normal fading in the logarithmic scale around the mean path loss L (dB) by a Gaussian distribution with zero mean and standard deviation. In this context, due to the slow fading process versus distance Δx , adjacent fading values correlate. Then, the normalized auto-correlation function $R(\Delta x)$ can be described with sufficient accuracy by an exponential function [17].

$$R(\Delta x) = \exp\left(-\frac{|\Delta x|}{d_{\text{cor}}} \ln 2\right), \quad (2.49)$$

where the decorrelation length d_{cor} , depends on the environment. From this principle, we may assume 20 m for the decorrelation in the vehicular, and 5 m for the outdoor to indoor pedestrian environment. For the latter the evaluation of decorrelation length may not be fully valid.

2.2.7 Channel Impulse Response

The environments described in the preceding sections have a channel impulse response based on a tapped-delay line model. The number of taps, the time delay relative to the

first tap, the average power relative to the strongest tap, and the Doppler spectrum of each tap characterize the model. Most of the time delay spreads are relatively small, but occasionally, there are worst case multi-path characteristics that lead to larger delay spreads. Here we consider the worst case. Two multi-path channels capture this delay spread better than a single tapped delay line. The reference simulation channel model can use a discrete wide sense stationary uncorrelated scattering (WSSUS) channel model, where the sum of delay replicas represents the received signal of the input signal weighted by independent zero-mean complex Gaussian time variant processes. Hence, if $z(t)$ and $w(t)$ denote the complex low pass representations of the channel input and output, respectively, then [2]

$$w(t) = \sum_{n=1}^N \sqrt{p_n} g_n(t) z(t - \tau_n), \quad (2.50)$$

where p_n is the strength of the n th weight, and $g_n(t)$ is the complex Gaussian process weighting the n th replica.

The Doppler spectrum of the n th path or the power spectrum of $g_n(t)$ controls the fading rate due to the n th path. Therefore, to define this channel model we can specify only the Doppler spectra of the tap weights $\{P_n(\nu); n = 1, \dots, N\}$, the tap delays $\{\tau_n; n = 1, \dots, N\}$, and the tap weight strengths $\{p_n(\nu); n = 1, \dots, N\}$. Interpreting the process $g_n(t)$ as the superposition of unresolved multi-path components arriving from different angles and in the vicinity of the delay interval, we have:

$$\left(\tau_n - \frac{1}{2W} < \tau < \tau_n + \frac{1}{2W} \right), \quad (2.51)$$

where W is the bandwidth of the transmitted signal.

Generally, each ray has a different Doppler shift corresponding to a different value of the cosine of the angle between the ray direction and the velocity vector. Which means that we can assume: first, very large number of receive-rays arrive uniformly distributed in azimuth at the MS and at zero elevation for each delay interval for outdoor channels⁷. At the BS in general the received rays arrive in a limited range in azimuth. Second, for indoor channels⁸ a very large number of receive-rays arrive uniformly distributed in elevation and azimuth for each delay interval at the BS.

The first assumption matches the ones made in by Clarke [18] and Jakes [19] in narrow band channel modelling. Thus the same Doppler spectrum will result, i.e.

$$P_n(\nu) = P(\nu) = \frac{1}{\pi} \frac{1}{\sqrt{(V/\lambda)^2 - \nu^2}}, \quad |\nu| < \frac{V}{\lambda}, \quad (2.52)$$

⁷ Also, the antenna pattern is assumed to be uniform in the azimuthal direction.

⁸ Assuming the antenna is either a short or half-wave vertical dipole.

where V is the velocity of the mobile and λ is the wavelength at the carrier frequency. The term **CLASSIC** is used to identify this Doppler spectrum [2].

The second assumption results in a Doppler spectrum that is nearly flat, and the choice of a flat spectrum has been made, i.e.

$$P_n(v) = P(v) = \frac{\lambda}{2V}, \quad |v| < \frac{V}{\lambda}. \quad (2.53)$$

Hence, this Doppler spectrum is referred to as **FLAT** [2].

Table 2.4, Table 2.5 and 2.6 describe the tapped-delay-line parameters for each of the environments introduced in the preceding sections. Three parameters characterize each tap of the channels, i.e. the time delay relative to the first tap, the average power relative to the strongest tap, and the Doppler spectrum of each tap. A $\pm 3\%$ variation in the relative time delay allows channel sampling rate matching some in simulations [2].

Table 2.4 Indoor Office Reference Environment Tapped-Delay-Line Parameters [2]

Tap	Channel A		Channel B		Doppler spectrum
	Relative delay (ns)	Average power (dB)	Relative delay (ns)	Average power (dB)	
1	0	0	0	0	FLAT
2	50	-3.0	100	-3.6	FLAT
3	110	-10.0	200	-7.2	FLAT
4	170	-18.0	300	-10.8	FLAT
5	290	-26.0	500	-18.0	FLAT
6	310	-32.0	700	-25.2	FLAT

Table 2.5 Outdoor to Indoor and Pedestrian Reference Environment Tapped-Delay-Line Parameters [2]

Tap	Channel A		Channel B		Doppler spectrum
	Relative delay (ns)	Average power (dB)	Relative delay (ns)	Average power (dB)	
1	0	0	0	0	CLASSIC
2	110	-9.7	200	-0.9	CLASSIC
3	190	-19.2	800	-4.9	CLASSIC
4	410	-22.8	1200	-8.0	CLASSIC
5	—	—	2300	-7.8	CLASSIC
6	—	—	3700	-23.9	CLASSIC

Table 2.6 Vehicular Reference Environment, High Antenna, Tapped-Delay-Line Parameters [2]

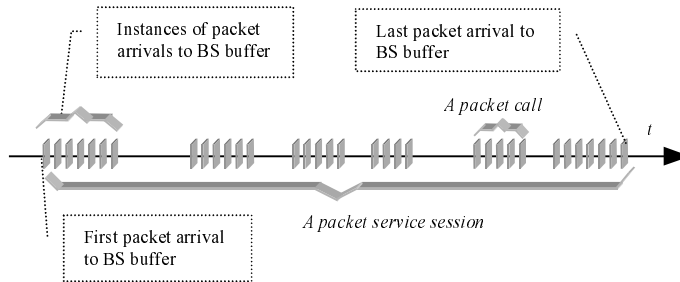
Tap	Channel A		Channel B		Doppler spectrum
	Relative delay (ns)	Average power (dB)	Relative delay (ns)	Average power (dB)	
1	0	0.0	0	-2.5	CLASSIC
2	310	-1.0	300	0	CLASSIC
3	710	-9.0	8900	-12.8	CLASSIC
4	1090	-10.0	12900	-10.0	CLASSIC
5	1730	-15.0	17100	-25.2	CLASSIC
6	2510	-20.0	20000	-16.0	CLASSIC

2.2.8 Traffic Types and Propagation Models

We can represent real time services (e.g. speech and CS data services) by generating calls according to a Poisson process assuming a mean call duration of 120 s. The speech would be an on-off model, with activity and silent periods being generated by an exponential distribution. The mean value for active and silence periods is 3 s and independent of the up and downlink, and both are exponentially distributed. For circuit switched data services, we can assume a traffic model with constant bit rate model and 100% activity.

2.2.8.1 Packet or Non-real Time Services

We can represent non-real time services by a WWW browsing session consisting of a sequence of packet calls. We only consider the packets from a source, which may be at either end of the link but not simultaneously. A subscriber may initiate a packet call when requesting an information entity. During this call several packets may be generated, which means that the packet call constitutes a sequence of packets bursts [20,21]. Figure 2.10 illustrates the bursts during the packet call typically seen in fixed network packet transmissions.

**Figure 2.10** Packet service session characteristics.

The behaviour of Figure 2.10 can be modelled through the following parameters:

- Session arrival process $\rightarrow$ Poisson process
- Number of packet calls per session $\rightarrow N_{pc} \in \text{Geom}(\mu_{N_{pc}})$
- Reading time between packet calls $\rightarrow D_{pc} \in \text{Geom}(\mu_{D_{pc}})$
- Number of datagrams within a packet call $\rightarrow N_d \in \text{Geom}(\mu_{N_d})$

- Inter arrival time between datagrams (within a packet call) $\rightarrow D_d \in \text{Geom}(\mu_{Dd})$
- Size of a datagram, $S_d \rightarrow$ Pareto distribution with cut-off:

$$\left\{ \begin{array}{l} f_x(x) = \frac{\alpha k^\alpha}{x^{\alpha+1}}, \quad x \geq k \\ F_x(x) = 1 - \left(\frac{k}{x}\right)^\alpha, \quad x \geq k \\ \mu = \frac{k\alpha}{\alpha-1}, \alpha > 1 \\ \sigma^2 = \frac{k^2\alpha}{(\alpha-2)(\alpha-1)^2}, \quad \alpha > 2 \end{array} \right. \quad (2.54)$$

Packet_Size is defined by the following formula:

$$\text{Packet\_Size} = \min(P, m), \quad (2.55)$$

where P is the normal Pareto distributed random variable ($\alpha = 1.1$, $k = 81.5$ bytes) and m is the maximum allowed packet size, $m = 66666$ bytes. The PDF of the Packet_Size becomes:

$$f_n(x) = \begin{cases} \frac{\alpha k^\alpha}{x^{\alpha+1}}, & k \leq x < m \\ \beta, & x = m \end{cases} \quad (2.56)$$

where β is the probability that $x > m$. It can easily be calculated as:

$$\beta = \int_m^\infty f_x(x) dx = \left(\frac{k}{m}\right)^\alpha, \quad \alpha > 1. \quad (2.57)$$

Then it can be calculated as:

$$\mu_n = \int_{-\infty}^\infty x f_n(x) dx = \int_k^m x \frac{\alpha k^\alpha}{x^{\alpha+1}} dx + m \left(\frac{k}{m}\right)^\alpha = \dots \text{calculating} \dots = \frac{\alpha k - m(k/m)^\alpha}{\alpha - 1} \quad (2.58)$$

with the parameters above the average size: $\mu_n = 480$ bytes [2,20,21] indicates that according to the values for α and k in the Pareto distribution, the average packet size μ is 480 bytes. Average requested file size is $\mu_{Nd} \times \mu = 25 \times 480$ bytes ≈ 12 kbytes. The inter-arrival time is adjusted in order to get different average bit rates at the source level. Table 2.7 illustrates characteristics of connectionless information rates for WWW from [20].

Table 2.7 Characteristics of Connection-Less Information Types [20]

Packet based information rates, e.g. Internet services	Avg. no. of packet calls in a session	Avg. reading time between packet calls (s)	Avg. amount of packets within a packet call	Avg. inter-arrival time between packets (s) ⁹	Parameters for packet size distribution
WWW surfing UDD 8 kbit/s	5	412	25	0.5	$k = 81.5$ (= 1.1)
WWW surfing UDD 32 kbit/s	5	412	25	0.125	$k = 81.5$ (= 1.1)
WWW surfing UDD 64 kbit/s	5	412	25	0.0625	$k = 81.5$ (= 1.1)
WWW surfing UDD 144 kbit/s	5	412	25	0.0277	$k = 81.5$ (= 1.1)
WWW surfing UDD 384 kbit/s	5	412	25	0.0104	$k = 81.5$ (= 1.1)
WWW surfing UDD 2048 kbit/s	5	412	25	0.00195	$k = 81.5$ $\alpha = 1.1$

2.3 CONCLUDING REMARKS

This chapter summarizes the essential background to investigate the UTRA physical layer and its impact on its architecture. It has provided the reference models to represent the communication environments and the signal processing issues. It was not the aim of the author to cover the different topics in depth but to set them as review points for further study when required.

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⁹ The different interarrival times correspond to average bit rates of 8, 32, 64, 144, 384 and 2048 kbit/s.

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3 THE UMTS DEVELOPMENT PLATFORM

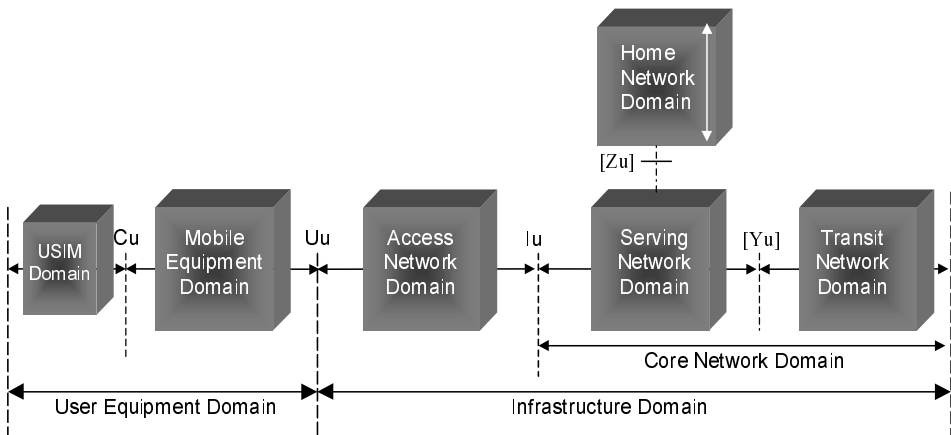
3.1 ARCHITECTURE AND DEPLOYMENT SCENARIOS

The architecture at the domain and functional levels, as well as the deployment scenarios are presented based on the 3GPP (ETSI) specifications noted in [1,2]. The terminology and basic principles are kept for consistency with a simplified approach in some cases, and for a pragmatic representation of the subject in others.

3.1.1 The UMTS High Level System Architecture

3.1.1.1 The UMTS Domains

Figure 3.1 illustrates the different UMTS domains. The identified domains imply the evolution of current or existing network infrastructures, but do not exclude new ones. The Core Network (CN) domain can evolve for example from the GSM, N-ISDN, B-ISDN, and PDN infrastructures.



Cu = Reference point between USIM and ME

Iu = Reference point between Access and Serving Network domains

Uu = Reference point between User Equipment and Infrastructure domains, UMTS radio interface

[Yu] = Reference point between Serving and Transit Network domains

[Zu] = Reference point between Serving and Home Network domains

Figure 3.1 UMTS architecture domains and reference points.

The generic architecture incorporates two main domains, i.e. the *user equipment domain* and the *infrastructure domain*. The first concerns the equipment used by the user to access UMTS services having a radio interface to the infrastructure. The second consists of the physical nodes, which perform the various functions required to terminate the radio interface and to support the telecommunication services requirements of the users. The rest of the sub-domains are defined in Table 3.1.

Figure 3.16 in Appendix A illustrates the four (Application, Home, Serving, and Transport) strata. It also shows the integrated UMTS functional flow, i.e. the interactions between the USIM, MT/ME, Access Network, Serving Network and Home Network domains, including interactions between TE, MT, Access Network, Serving Network, Transit Network domains and the Remote Party.

Table 3.1 The UMTS Architecture Domains

USER EQUIPMENT DOMAINS: dual mode and multi-mode handsets, removable smart cards, etc.	
Mobile Equipment (ME) domain	Consists of: • The Mobile Termination (MT) entity performing the radio transmission and related functions, and • the Terminal Equipment (TE) entity containing the end-to-end application, (e.g. a laptop connected to a handset).
USIM domain	The User Services Identity Modulo (USIM) domain contains data and procedures to unambiguously and securely identify itself, (e.g. smart card)
INFRASTRUCTURE DOMAINS	
Access Network (AN) domain	Consists of the physical entities managing the access network resources and provides the users with mechanisms to access the core network
Core Network (CN) domain	Consists of the physical entities providing support for the network features and telecommunication services; e.g. management of user location information, control of network features and services, switching and transmission mechanisms for signalling and for user generated information. It includes: 1. Serving Network (SN) domain representing the core network functions local to the user's access point and thus their location changes when the user moves. 2. Home Network (HN) domain representing the core functions conducted at a permanent location regardless of the user's access point. The USIM is related by subscription to the HN. 3. Transit Network (TN) domain, which is the CN part between the SN and the remote party.

3.1.1.2 The IMT 2000 Family

The UMTS high level architecture integrates the physical aspects through the domain concept and functional aspects through the strata concept. The separation according to [1] allows a UMTS network to fit within the context of the IMT 2000 family of networks as illustrated in Figure 3.2.

Basically there are two CN options for the air interface of the IMT 2000 family of networks, i.e. GSM and IS-41 networks. The first one, which also includes the IP packet

network, will serve the UTRA modes and the UWC-136 (packet) evolving based on EDGE. While GPRS may become an IP core network on its own, where UMTS and other air interfaces will directly connect to it, today it is part of the GSM infrastructure. IS-41 will serve primarily USA regions in the evolution of IS-136 in TDMA and IS-95 in CDMA.

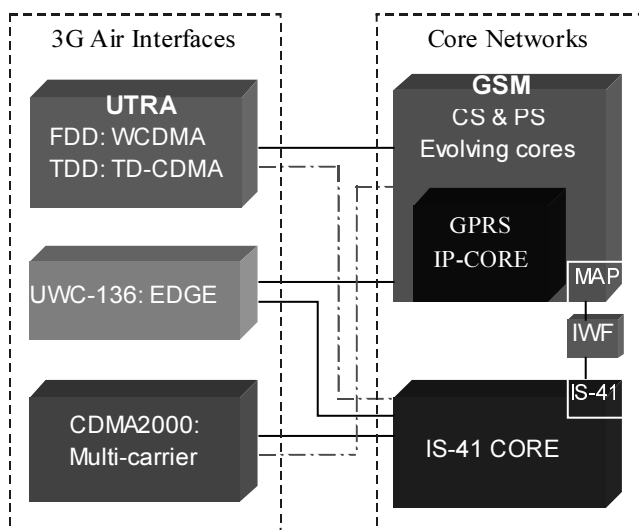


Figure 3.2 The IMT 2000 family of networks.

3.1.2 Coexistence of Present and Future Networks

While UMTS will bring new services and allow new access options, its deployment and introduction will be in several phases. The first no doubt will evolve within a mixed environment where coexistence with 2nd generation systems like GSM (including GPRS) will be predominant. Figure 3.3 illustrates the main network elements of a typical GSM network incorporating the Circuit Switched (CS) segment and the GPRS entities as part of the Packet Switched (PS) segment. It also includes the future UMTS elements on the radio interface side. Hence while some operators or service providers will deploy completely new network infrastructures, others will use GSM architecture as the basis for UMTS or 3G systems. This means that *UMTS will complement the existing GSM system in some cases, not replace it.*

Clearly for all the elements to coexist as illustrated in Figure 3.3 they must all contain the necessary HW/SW (including protocols) enabling features for inter-working. Today, for example, the SMG (ETSI) and 3GPP organization have the task of making 2nd and 3rd generation elements inter-work seamlessly through specification and recommendations. The technical specifications for practical reasons are issued in releases, e.g. the contents of the 1st edition of this book will be based on Release 1999 covering the evolution of GSM and the introduction of UMTS.

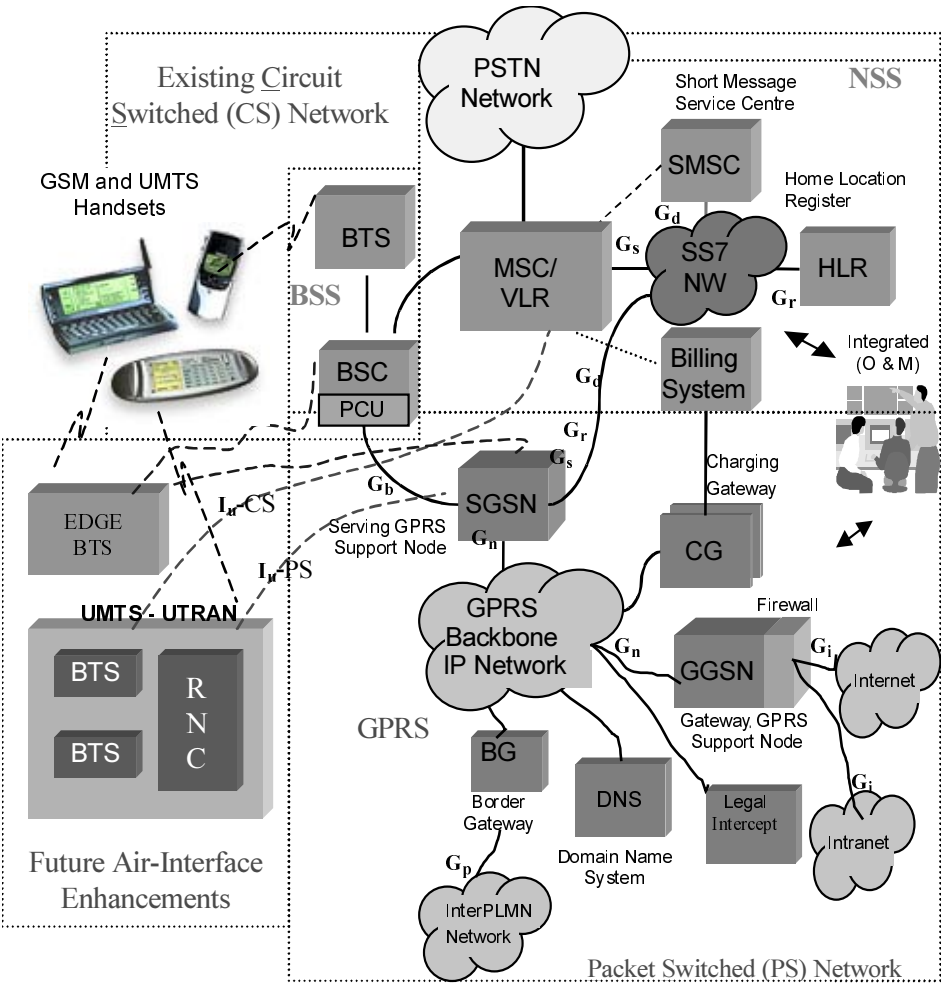


Figure 3.3 Coexistence of 2nd and 3rd generation mobile network elements.

3.2 THE CORE NETWORK DOMAIN

3.2.1 Network Evolution Towards UMTS

Evolution here implies seamless and dynamic interoperability of 2G (2.5G) and 3G technologies in the Core Network (CN) and Radio Network (RN) sides. We will thus cover these evolution implications next taking into account the integrated network elements illustrated in Figure 3.3, i.e. the PS and CS building blocks in the CN side and UTRAN and EDGE in the RN side.

To structure the presentation following the domain concept, we first cover the core network domain. Forthcoming chapters address the access network and mobile equipment

domains. Furthermore, for completeness we also define the basic functions of the CN building blocks.

The UMTS platform as illustrated in Figure 3.3, will incorporate a number of 2G/3G¹ functional elements joined by standard interfaces. Together these network elements will route multifarious information traffic and provide:

- resource allocation;
- mobility management;
- radio link management;
- call processing;
- billing record generation;
- operational and maintenance functions; and
- collection of performance statistics.

The CN comprises of circuit and packet switching systems, trunk transmission, signalling systems, the access network and service platforms.

Figure 3.4 highlights the 3G side represented in layers to point out some of the new elements incorporated to the legacy GSM network. Each layer contains a distinct network element based on the CN infrastructure evolution. However, it is not restricted to the CN layers, it includes, e.g. the radio layer and others as follows:

- the radio network layer illustrating new WCDMA base stations (BSs) and the RNC to be described in Chapter 4;
- the mobile switching layer, which regroups the 3G SGSN, 3G MSCs with their upgraded associated components, such as HLR, VLR, AuC, EIR, and their new CPS unit enabling IP telephony;
- the transit-IP layer, which not only serves as the backbone layer for transiting traffic between nodes, but also incorporates the IP bypass mediation device for signalling and user data between CS and PS. The GGSN may be also part of this layer.
- the signalling layer, comprising mainly of STPs connected to the other elements;
- the management layer composed of the integrated network management systems and network mediation systems as illustrated in Figure 3.3;
- the service layer will comprise all value-added service platforms, such as SMC, VMS, intelligent network platform and customer care centres, ISP, billing platform, etc;

¹ GSM 1800 MHz evolving elements and new UMTS or 3G specific elements co-existing seamlessly.

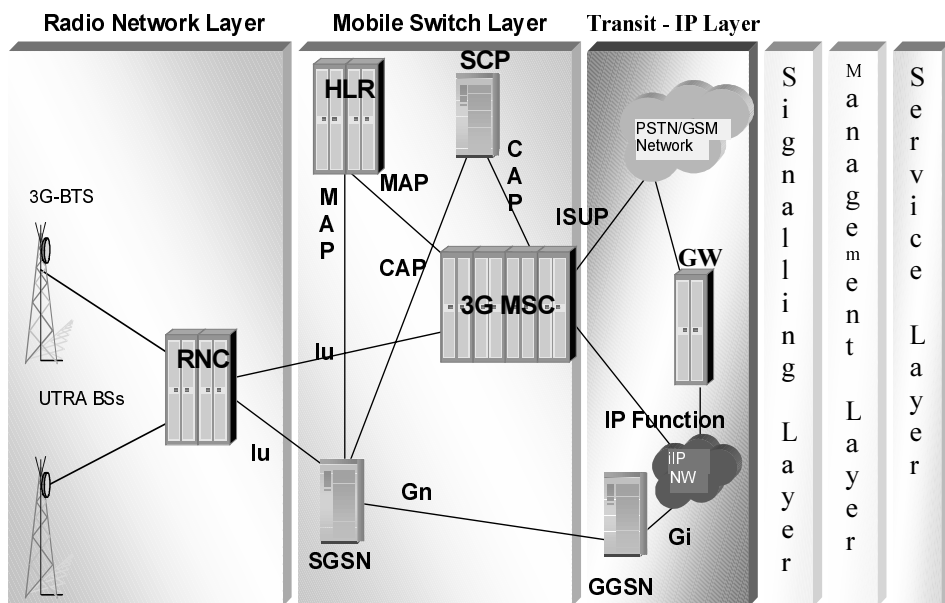


Figure 3.4 Representation of the CN and radio access layers.

For background completeness on the CN side, the main GPRS elements and terminal connections to a GPRS network are described next from the functional level.

3.2.1.1 Main Packet Switched Network Elements

The *Serving GPRS Support Node* (SGSN) performs the following key tasks:

- authentication and mobility management;
- protocol conversion between the IP backbone and the protocols used in the BSS and MS;
- collection of charging data and traffic statistics;
- routing data to the relevant GGSN when connection to an external network is required (all intra-network MS to MS connections must also be made via a GGSN).

The *Gateway GPRS Support Node* (GGSN) acts as the interface between the GPRS network and external networks; it is simply a router to a sub-network. When the GGSN receives data addressed to a specific user, it checks if the address is active. If it is, the GGSN forwards the data to the SGSN serving the mobile; if the address is inactive the data is discarded. The GGSN also routes mobile originated packets to the correct external network.

3.2.1.1.1 Terminal Attachment to the GPRS Network

The connection between a GPRS terminal and the network has two parts:

1. *Connection to the GSM network (GPRS Attach)* – When the GPRS terminal is switched on, it sends an ‘attach’ message to the network. The SGSN collects the user data from the HLR and authenticates the user before attaching the terminal.

2. *Connection to the IP network (PDP context)* – Once the GPRS terminal is attached, it can request an IP address (e.g. 172.19.52.91) from the network. This address is used to route data to the terminal. It can be static (the user always has the same IP address), or dynamic (the network allocates the user a different IP address for each connection).

Dedicated standard (ETSI specified) interfaces assuring the interconnection between the key network elements and enabling multi-vendor configurations include:

- Gb-interface (SGSN-BSS);
- Gn-interface (GSN-GSN);
- Gp-interface (inter-PLMN interface);
- Gi (GGSN-external IP networks);
- Gr (SGSN-HLR);
- Gs (SGSN-MSC/VLR);
- Gd (SGSN to SMS-GMSC/SMS-IW MSC).

Other GPRS elements illustrated in Figure 3.3 are:

1. *Domain Name Servers* – These are standard IP devices that convert IP names into IP addresses e.g. vms.orange.ch → 172.19.52.92
2. *Firewalls* – These protect the IP network against external attack (e.g. from hackers). The firewall might reject all packets that are not part of a GPRS subscriber initiated connection.
3. *Border Gateway* – This is a router providing, e.g. a direct GPRS tunnel between different operators’ GPRS networks via an inter-PLMN data network, instead via the public Internet.
4. *Charging Gateway* – GPRS charging data are collected by all the SGSNs and GGSNs in the network. The charging gateway collects all this data together, processes it and passes it to the billing system.

3.2.1.2 Open Interfaces

To practically visualize the inter-operating environment we will take as reference a GSM Network in transition towards a 3G system.

Evolving CN elements, e.g. will concurrently support interfaces (thereby signalling) for both 2G and 3G radio networks, i.e. existing elements will be enable through field upgrades with Iu interface towards high-capacity 3G.

The 3G RAN, e.g. will connect to a GSM CN via the Iu interface. This interface provides a logical separation between CS and PS signalling giving the possibility to physically separate the interfaces, i.e.

- Iu-CS interface for circuit-switched traffic, based on the ATM transport protocol, and
- Iu-PS interface for packet-switched traffic, based most likely on IP over ATM.

The Iu interfaces above assume that: the MSC can also multiplex the Iu-PS interface to the SGSN with only one physical interface from RNC to the core network, and that the MSC will get an ATM module to interact with the ATM based RAN.

A second new interface besides the Iu in the CN concerns IP links. It is foreseen that by the time UMTS is deployed, MCSs will support IP connections. Thus the solution can be envisaged as follows:

- a new feature in the MSC, will be the integrated IP function protocol between two MSCs signalling and user data between CS and PS;
- the integrated IP function will introduce a new type of trunk signalling to the MSC switching system, i.e. SS7 over the IP network;
- the transmission over the IP network will be done using the User Datagram Protocol (UDP) from the TCP/IP stack; both signalling transmission and media transmission will use the protocol;
- data, fax and compressed speech will be packetized to IP packets and transmitted to the other switch using the Real-Time Transport Protocol (RTP) on the UDP.

Other key interfaces in the evolution to 3G include:

- A-Interface MSC to GSM BSS will continue as needed for applications like Radio Resource Management (RRM), Mobility Management (MM), and Link Management (LM);
- MAP performing signalling between the MSC and other NSS elements and performing critical operations between switching and database elements to support roaming;
- CCS7 – Common Channel Signalling system (7) links the MSC to a PSTN or to an ISDN using a single channel to carry the signalling of multiple speech circuits; the digital Channel Associated Signalling (CAS) used between exchanges will also continue as needed;
- in the short term, the File Transfer Access and Management (X.25 FTAM) interface will continue to communicate with billing systems as IP links to new billing centres develop;
- standard V.24 interfaces connecting O&M terminals to the MSC will probably continue, while more sophisticated WWW type interfaces will be implemented with evolving MSC operating systems.

In conclusion, we can say that the two critical interfaces that UMTS introduces to the CN are primarily the Iu and IP. These interfaces add new dimension to the existing GSM infrastructures besides enriching the type of links a CN may have.

In the following we cover the essential transition steps in terms of the 3G architecture requirements.

3.2.2 Key Release 99 Architectural Requirements

The general working assumptions for Release 99 (R99), which cover the phase 1 UMTS/Release '99 GSM standards and reflecting in part the elements illustrated in Figure 3.3, can be summarized from [3] as follows:

- a Core Network based on an evolved 2G MSC and an evolved SGSN
- an optionally evolved Gs interface
- Mobile IPv4 with Foreign Agent (FA) care-of addresses to end-users over the UMTS/GPRS network, where the FA is located in the GGSN.
- class A GSM mobiles.
- Transcoder location shall be according to the “Evolution of the GSM platform towards UMTS” outlined in 3G TS 23.930
- UMTS/IMT 2000 Phase1 (R99) network architecture and standards shall allow the operator to choose between Integrated and Separated CNs for transmission (including L2)
- The UMTS standard shall allow for both separated and combined MSC/VLR and SGSN configurations.
- The UE shall be able to handle separated or combined MSCs and SGSNs.
- There can be several user planes to these CN nodes.

The following general concepts should be followed:

- Separate the layer 3 control signalling from the layer 2 transport discussion (do not optimize layer 3 for one layer 2 technology).
- MSC-MSC layer 3 call control is out of scope of standardization in SMG.
- As future evolution may lead to the migration of some services from the CS domain to the PS domain without changes to the associated higher-layer protocols or functions. UMTS release 99 shall provide the flexibility to do this in a way that is backwards compatible with release 99 UEs, provided this does not introduce significant new complexity or requirements in the system.

3.2.3 The R99 Core Network Synthesis

In the preceding section we have already covered the evolution of the CN. However, here we are summarizing them again in the light of the R99 to provide a concise background for R00 introduced in Chapter 9. The synthesis here considers the classical 2G (GSM) type CN architecture evolving to 3G CN. However, some suppliers may already

use the layered architecture as illustrated in Chapter 7, to introduce R99 and directly evolve into R00 with minimum or no structural changes.

The UMTS R99 CN will start with a hybrid GSM network. Thus real-time services such as voice and video will continue the usage of CS paths through the Mobile Switching Centre (MSC). Non real-time services, e.g. Internet type (email, ftp, information services, etc.), will also continue passing through the GPRS network.

While IP services may appeal to both, public and private, the latter may exploit more IP services such as IP telephony, person-to-person multimedia conferencing, mobile Internet, etc; CS voice telephony may initially remain the best solution for the mass market. On the other hand, if end-to-end mobile IP telephony consolidates its flexibility with minimized cost and high quality and IP native terminals are widespread, the mass market may embrace IP services very rapidly.

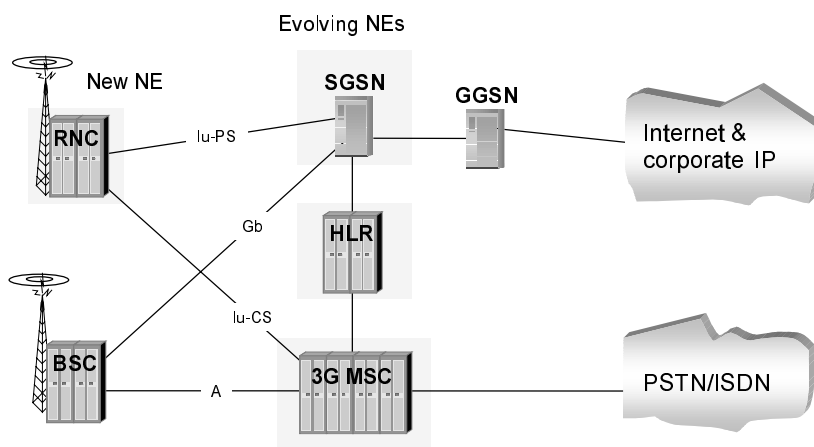


Figure 3.4b 2G Elements evolving for UMTS R99.

Figure 3.4b illustrates the evolving CN elements to meet the R99 specifications. Notice that the evolution affects primarily the SGSN, MSC, and HLR. These elements will either have new architecture platforms or follow SW upgrade with selected HW additions. The process will vary from supplier to supplier. The new element, i.e. RNC, corresponds to the radio network. For completeness we next review the main functions and transition steps of the CS and PS network elements. The Value Added Services (VAS) platforms (e.g. voice mail, SMSC, IN, pre-paid, etc.), which also reside in the CN, will evolve at their own pace.

3.2.4 Circuit Switched (CS) Network Elements (NE)

3.2.4.1 The 3G Mobile Switching Centre (3GMSC)

The 3G MSC will become the main element of the R99 CS network just as it is in GSM. In general depending on the manufacturers, a 3G MSC will include a VLR and a SSP to serve both GSM BSS and 3G RAN concurrently by incorporating both A and Iu inter-

faces. The Iu interface will be realized through a HW upgrade, e.g. a plug-in board to the MSC in some cases, or an ATM module using a new HW platform connecting to the MSC in other cases. Thus, the same 3GMSC will control services and charging for CS 2G/3G services in seamless co-existence. Furthermore, some 3GMSCs will interconnect through IP interfaces and multiplex Iu traffic from the RNC.

3.2.4.1.1 ATM Functionality

For all practical purposes we assume here that the ATM functionality takes place in a dedicated element, which we will call the ATM unit. This unit will have as a primary function to provide inter-working of the 3GMSC with the UMTS Radio Access Network (UTRAN). In most cases this unit will also provide transcoding functions for 3G CS services. The ATM unit will then support:

- Iu interface for ATM transmission in the UTRAN side and TDM transmission in the MSC side
- speech transcoding and user plane adaptation for CS data services
- 3G to 2G protocol conversion (e.g. WCDMA RANAP and GSM BSSAP).

3.2.4.2 The 3G Home Location Register

In most cases the 3GHLR will evolve from the 2GHLR through SW upgrade, i.e. the same HLR (including e.g. AuC and EIR) will serve 2G/3G dual-mode users with centralized subscriber information. Thus, service providers will use only one centre to activate 2G subscribers (e.g. GSM voice or GPRS) and UMTS subscribers with all their different traffic profiles. The 3GHLR then stores:

- subscriber profile data for 2G and 3G services
 - subscriber identity
 - semi-dynamic information, e.g. current subscriber profile with activated services, etc.
 - dynamic data, e.g. mobility management data.
- authentication information
- equipment identity info specifying listed mobile equipment identities.

3.2.4.3 New Services

While expected CS services may have rates up to 384 kbps, initial new services may start first with 64 and 144 kbps. Among these services sets we will have, e.g.:

- Virtual Home Environment (VHE): comprehensive set of services, features and tools, which have the same *look and feel at home or abroad*
- new bearer service defined by QoS parameters and a toolbox for implementing operator specific services

- roaming between 2G/3G network, i.e. UMTS and GSM will be supported as well as handovers from one network to another
- enhanced location services, i.e. new multimedia applications will be possible with the higher rate transmissions (e.g. real time video displays, dynamic broadcasting and easy banking).

When comparing to GSM, e.g. in practice the new evolved NEs will then be capable of:

- providing multiple service components in one stream to a single user terminal simultaneously for multimedia, thus offering transparency for video communications
- offering high bit rates for both circuit- and packet-switched bearer services
- offering multiple connections concurrently to single user MS to serve speech + packet data for example.

In addition, during the transition to full IP CN e.g. only the CS part will be able to support services requiring constant bit rate with small delay variations through the protocol/rate adaptation conversion carried out by the ATM unit. Thus, the 3G NEs will support 64² to 384 kbps transparent or real-time CS data services easily.

3.2.5 Packet Switched (PS) Network Elements (NE)

3.2.5.1 3G Serving GPRS Support Node (3G SGSN)

As illustrated in Figure 3.4b, for UMTS R99 the SGSN will require an evolution to 3G SGSN to successfully support 3G PS services. This evolution becomes imperative to acts as a link between the 3G Radio Access Network (RAN) and the packet core, in order to perform both control and traffic handling (gateway) functions for the PS domain part of the 3G system. An evolved SGSN or a 3G SGSN will then provide:

- an interface between the Radio Network Controller (Iu) and the 3G Core Network (Gn, Gp). The physical Iu interface will generally use an ATM STM-1 optical interface, while the Gn and Gp physical interfaces can use Ethernet or ATM technology.
- SS7 network interface to communicate with the HLR (Gr), EIR (Gf) and the SMSC (Gd). Some may use X.25 or IP to connect the SMSC. The physical SS7 interfaces for Gf, Gr and Gd will generally use E1-PCM or T1-PCM connections.
- interface with the charging gateway through the Ga interface.

Other functions of the 3G-SGSN (some may depend on the manufacturers road map) include: IPv6, mobility management, session management, QoS, multiple PDP context per IP address, tunnelling, charging, IPSec, lawful interception, SMS, pre-paid.

² With UDI ISDN H.324 inter-working for example.

3.2.5.1.1 Functional Characteristics GPRS and the 3G SGSN

As noted above, the 3G SGSN and the 2G SGSN do have some capital differences when it comes to functionality. If we pair them with the RNC and BSC, respectively, we can see the different functions in Table 3.1b.

Table 3.1b The 2G- and 3G-SGSN functional characteristics

	3G SGSN	RNC	2G SGSN	BSC
Mobility management	X	X	X	
Interaction with HLR, MSC/HLR	X	3G MSC only	X	
Charging and statistics	X		X	
High capacity routing	X			
IP telephony	X	X		
Real time multimedia	X	X		
Authentication	X		X	
Radio protocol to IP conversion		X	X	
GTP tunnelling to GGSN	X		X	
Ciphering		X	X	
Compression		X	X	

While there remains similarities between the 2G/3G SGSNs, major differences also exist, e.g. ciphering and compression no longer takes place at the 3G SGSN but at the RNC. Mobility management takes place in the 3G SGSN and the RNC. Before, the BSC did not perform any of these functions. 3G SGSN will support delay sensitive applications, e.g. IP telephony and real time multimedia requiring high capacity routing; these functions did not take place at the 2G SGSN. Thus, migration to the 3G SGSN has added some functions with demand for higher processing capacity.

3.2.5.2 The Gateway GPRS Support Node (GGSN)

A 2G GGSN in most cases will not need structural changes besides probably a SW upgrade to support 3G. In any case, it acts as an interface between the GPRS/3G network and external networks. It connects the GPRS and 3G core to the Internet world, ISPs and private or corporate Intranets, thereby allowing 2G (GPRS) and 3G mobile users to have data services. On the other hand, because we see the GGSN as a router from external networks, major changes will occur only with additional unique functions. Other common functions, depending also on the manufacturer's road map as in the SGSN, can be listed as follows:

- integrated support for GPRS and 3G
- support for multiple access points per GGSN and multiple PDP contexts per IP address
- Ipv6 and QoS
- support standard routing protocols, e.g. RIPv1-v2, OSPF, IGRP, BGP4, DVMPR, including static routing

- IPSec and lawful interception
- prepaid, hot billing, fixed rate charges, CDR consolidation³ and duplicate CDR removal.

3.2.5.3 *Charging, Border and Lawful Interception Gateways*

No changes in the intrinsic functions of the CG, BG, and LIG occurred for 3G. Thus, charging, e.g. takes place as in GPRS, i.e. charging data gets collected by all the SGSNs and GGSNs in the network.

For the Border Gateway (BG) we will continue using the Public Land Mobile Network (PLMN) solutions based on roaming agreements between operators. Hence, main tasks such as providing a secure connection between PLMNs over the Inter-PLMN backbone network will continue.

The Lawful Interception Gateway (LIG) will also remain essential for network functionality in the 3G infrastructure, allowing authorities to monitor and intercept 3G mobile data calls. Therefore, providers around the globe will be able to meet their local authority requirements before starting commercial 3G services.

3.2.5.4 *DNS, DHCP and the IP Backbone*

As in the preceding elements, the DNS, DHCP and IP backbone does not need unique functions for 3G.

For example, with 3G, DNS will continue allowing the SGSNs to translate logical names into physical addresses of the GSNs. The 3G SGSN will thus use the DNS server to determine the IP address of the GGSN when activating a PDP context and to find the address of the SGSN when doing an inter-SGSN routing area update. The principle of two DNS servers, one primary and one secondary (backup DNS server) will also continue.

The Dynamic Host Control Protocol (DHCP) in GPRS and 3G also has the same function; i.e. it automates IP address management. In GPRS as well as 3G static IP address allocation will take place in the HLR and dynamic IP address allocation will come either from RADIUS/DHCP servers within the private/corporate network or from the GGSN's internal addresses pool.

Finally, the IP backbone will continue, as in GPRS, with the only difference that higher capacity LAN switches may be required.

3.2.5.5 *Network Connectivity in the 3G Packet Core*

We do not expect major change in network connectivity for 3G PS either. For example, an *access point* (configured in the GGSN) will continue to be the logical connection point that the GGSN will provide to allow MS to attach to external networks⁴. Sub-

³ Putting together CDRs from various services.

⁴ A single access point always refers to a certain external network, e.g. a company Intranet or an ISP.

scribers, e.g., will continue to use one access point in order to access their own ISP provider's services, and a 2nd access point to connect to the private/corporate Intranet. In general, the configuration parameters for an access point will include:

- static IP address range or dynamic IP address pool for subscribers
- set number of PDP contexts for the access point
- dedicated RADIUS server addresses and passwords (e.g. authentication and accounting)
- DHCP server address and IP address allocation method (e.g. DHCP, GGSN, RADIUS)
- user authentication method (e.g. RADIUS) and default router.

To select the desired network access, the subscriber uses the Access Point Name (APN). The SGSN uses the APN to resolve the IP address of the correct GGSN.

One GGSN can provide more than one access point on the same physical interface (e.g. Ethernet port) when using tunnelling to separate data traffic routed to different networks. Likewise, we can link one access point to several GGSNs for redundancy or resilient using a round robin approach.

3.2.5.6 Implementing Quality of Service (QoS)

QoS managed on a link by link basis, is also an inherent feature in 3G. Once the traffic has been classified dedicated and integrated for example, where the first would require a certain level of quality and the 2nd a certain bandwidth guaranteed, the process of implementation will follow typical steps. The MS and the network negotiate and agree QoS level during the PDP-context activation. However, based on the availability of radio resources, it may also be re-negotiated later during the session. We can illustrate these steps as follows:

- The MS sends an *active PDP* context message to the SGSN containing, e.g. the access point name and a number of other parameters including a QoS profile.
- The SGSN verifies the request with the HLR based on the user's subscription profile. If there is contradiction, the QoS request may be rejected. Otherwise, the SGSN will issue a DNS query to identify the IP address of the GGSN associated with the requested access point.
- The SGSN then issues a *create PDP context* request to the GGSN with the re-negotiated QoS profile. The GGSN may accept or decline this request.
- The GGSN replies with its acceptance of the PDP context request to the SGSN and the SGSN in turn to the MS.

In the IP backbone, *dedicated* traffic for example, will make use of the Type of Service (ToS) field in the IPv4 or Ipv6 header to classify traffic. Then the GGSN and SGSN map the QoS profile requested in the PDP context activation message to *dedicated* code points.

We can also map *dedicated* classes to ATM Permanent Virtual Circuits (PVCs) to provide guaranteed QoS. However, this is still undergoing consolidation in the standardization bodies, e.g. IETF.

We may also map *dedicated* classes to MPLS labels, where the latter defines a protocol for encapsulating IP packets. MPLS, which has awareness of routing devices, uses a four-byte label added to the original packets. Thus, MPLS aims to simplify routing decisions by allowing packets to go along specific routes based on QoS parameters among others.

3.2.5.7 Implementing Ipv6

The introduction of IP terminals in 3G will dramatically increase the need for new IP addresses. Thus, we turn to IPv6 to enable new services and solve the problems inherent with IPv4 networks. IPv6 offers an enlarged address space, e.g. in IPv4 the address length is only 32 bits whereas IPv6 addresses have a length of 128 bits. These longer addresses enable IPv6 to offer a total of 2^{128} IP addresses.

The IPv6 protocol offers the following to 3G mobile networks, besides the enhanced address range:

- same IP address wherever you roam, i.e. global reachability
- multifarious 3G services to the mass market
- enhanced security through standardized and mandated security features
- availability of IP addresses for billions of terminals
- built-in QoS enhancing performance.

On the other hand, supporting IPv6 in 3G mobiles implies changes in various elements of the 3G network, and this may not come from all equipment manufacturers at the same time, or may not happen in R99. These changes include:

- the HLR and SGSN will need SW upgrades to support IPv6 parameters
- the GGSN will need to support IPv6 protocol stack in its external interfaces (i.e. Gi)
- the MSs will need the IPv6 protocol stack.

Since the subscriber's IPv6 traffic would be carried over the 3G backbone within the GTP tunnels, it is not expedient that all the backbone routers, or even the SGSN protocol stacks, migrate from the start to IPv6. On the other hand, even if IPv6 capable MSs could still use IPv4 backbones because of the GTP Tunneling Protocol that separates the backbone IP layer from the subscriber IP packet payload; IPv6 will still benefit the backbone layer, e.g. from the build in security features.

3.2.6 Coexistence Interoperability Issues

Although it seems that only one new interface, i.e. Iu, appears when incorporating the UMTS radio network to the 2G or 2.5G CN, inter-networking impacts spread to all the

integrated network elements as shown in Figure 3.3. In particular, mobility management and call control bring in new interoperability requirements. These requirements are summarized next before we concentrate on describing the different UMTS building blocks of the radio network in forthcoming chapters.

3.2.6.1 Iu Interface Inter-Working Characteristics

The Iu principles presented in [5] apply to PS and CS networks. In this context, UTRAN supports two logically independent signalling flows via the Iu interface to combined or separated network nodes of different types like MSC and SGSN [3].

Thus, UTRAN contains *domain distribution function* routing application independent UE control signalling to a corresponding CN domain. The UE indicates the addressed application type through a protocol discriminator for example. Then UTRAN maps this onto a correct Iu instance to forward signalling. UTRAN services, including radio access bearers, are CN domain independent, e.g. we can get speech bearer either through the PS or CS core network. The Iu includes control and user planes.

Because only a RNC can identify the actual packet volume successfully transferred to a UE, it indicates the volume of all not transferred downlink data to the 3G-SGSN so this latter can correct its counter.

3.2.6.1.1 Iu Control Plane

Both PS and circuit CS domains use the Signalling Connection Control Part (SCCP) protocol to transport Radio Access Network Application Part (RANAP) messages over the Iu interface. Likewise, both SCCP and RANAP protocols comply with ITU-T recommendations. In R99, SCCP messages in CS domain use a broadband SS7 stack comprising MTP3b on top of SAAL-NNI. In the PS domain UMTS specs allow operators to chose one out of two standardized protocol suites, i.e. broadband SS7 stack comprising MTP3b on top of SAAL-NNI or IETF/Sigtran CTP protocol suite for MTP3 users with adaptation to SCCP. Figure 3.5 illustrates the different RANAP stack options.

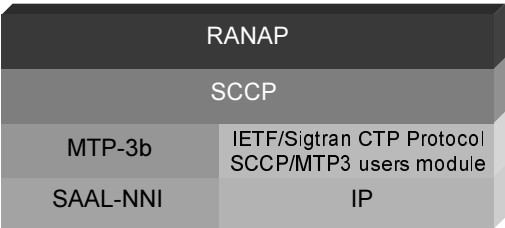


Figure 3.5 Stack options in the RANAP protocol.

3.2.6.1.2 Iu User Plane

The user plane towards the IP domain works based on an evolved Gn interface, where we achieve tunnelling of user data packets over the Iu interface through the user plane part of GTP over UDP/IP. The tunnelling protocol corresponds to an evolution of the user plane part of the GTP protocol used in GPRS stacked on top of UDP/IP. When

transport data uses ATM PVCs, the Iu IP layer provides Iu network layer services such as routing, addressing, load sharing and redundancy. This leads to an IP network configured to transfer Iu data units between RNSs and 3G-SGSNs.

We can access common layer 2 resources between UTRAN and the IP domain of a CN through one or several AAL5/ATM permanent VCs. More than one permanent AAL5/ATM VCs, for example allows load sharing and redundancy.

The UMTS user data plane in the network consists of two tunnels, i.e. a first IP/UDP/GTP tunnel between RNC and 3G SGSN on the Iu interface and a second IP/UDP/GTP tunnel between GGSN and 3G SGSN on the Gn interface.

The double tunnel architecture provides hierarchical mobility, allows direct RNC connection to the IP domain backbone, ensures traffic routing through 3G-SGSN to perform appropriate charging and legal interface functions, and it also makes room for future exploitation of Iu and Gn interfaces. The protocol stack is shown in Figure 3.6.

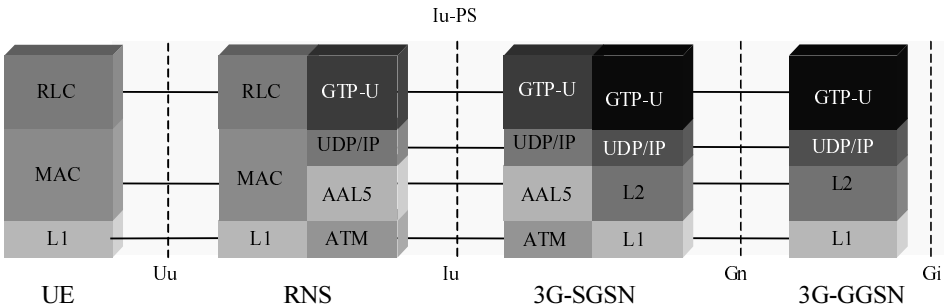


Figure 3.6 Protocol architecture for the IP domain user plane.

Specifications in [3] outline user data retrieval principles in UMTS and at GSM-UTMS handover for the PS domain. In the following we cover the Radio Access Domain, which in part will cover UTMS Mobility Management (UMM) and UMTS call control to complete the context of interoperability between 2G and 3G systems, i.e. GSM and UMTS.

3.3 THE ACCESS NETWORK DOMAIN

While the access network domain incorporates elements starting the physical layer and radio protocols, here we concentrate on the UTRAN as part of the radio access network.

3.3.1 UTRAN Architecture

The UTRAN as illustrated in Figure 3.7, contains Radio Network Subsystems (RNSs) communicating with the Core Network (CN) through the Iu interface. In turn a RNS contains a Radio Network Controller (RNC) and one or more Node B⁵. A Node B con-

⁵ Throughout this book we use 'Node B' as noted by the 3GPP specifications. However, we should know that a Node B is just a 3G-BTS performing more functions than a 2G BTS in GSM for example.

nects to the RNC through the Iub interface, and it can support either a FDD or TDD or combined dual mode operation.

The RNC takes care of handover decisions requiring signalling to the UE; it comprises a combining/splitting function to support macro diversity between different Node Bs. RNCs can interconnect each other through the Iur logical interface. The latter can be conveyed over a direct physical connection between RNCs or through any appropriate transport network.

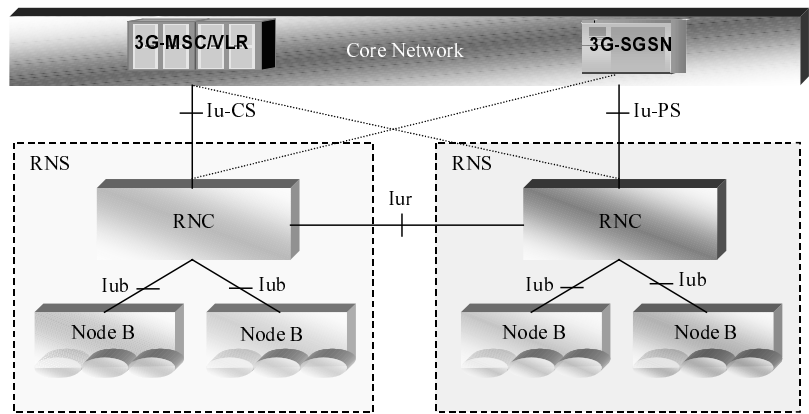


Figure 3.7 UTRAN architecture example.

All UE connections between UTRAN have a serving RNS. When service relocation demands it, a drift RNSs support the serving RNS by providing new radio resources. Figure 3.8 illustrates the role of a RNS (serving or drift) on a per connection basis between a UE and UTRAN.

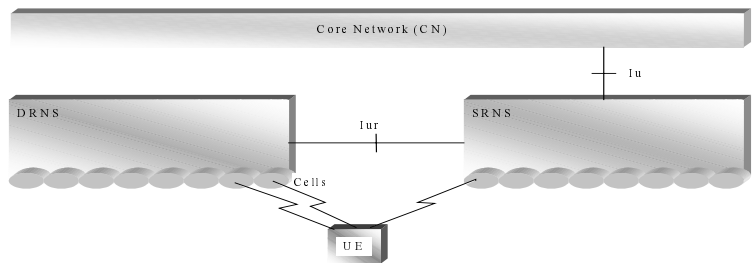


Figure 3.8 Serving and drift RNS.

3.4 UTRAN IDENTIFIERS AND FUNCTIONS

3.4.1 Identifiers

The unique RNC-Ids are defined by the operator, and set in the RNC via O&M.

Element	Identifiers
PLMN	The PLMN-Id is made of Mobile Country Code (MCC) and Mobile Network Code (MNC): $\text{PLMN-Id} = \text{MCC} + \text{MNC}$
CN Domain	Identifies a CN domain edge node for relocation tasks. It is made up of the PLMN-Id and of the LAC or RAC of the first accessed cell in the target RNS. The two CN domain identifiers are: – CN CS Domain-Id = $\text{PLMN-Id} + \text{LAC}$ – CN PS Domain-Id = $\text{PLMN-Id} + \text{LAC} + \text{RAC}$
RNC	RNC-Id together with the PLMN identifier is used to globally identify the RNC. RNC-Id or the RNC-Id together with the PLMN-Id is used as RNC identifier in UTRAN Iub, Iur and Iu interfaces. SRNC-Id is the RNC-Id of the Serving RNC. C-RNC-Id is the RNC-Id of the controlling RNC. D-RNC-Id is the RNC Id of the Drift RNC. $\text{Global RNC-Id} = \text{PLMN-Id} + \text{RNC-Id}$
Service area	Used to uniquely identify an area consisting of one or more cells belonging to the same location area Such an area is called a service area and can be used for indicating the location of a UE to the CN. The Service Area Code (SAC) together with the PLMN-Id and the LAC will constitute the service area identifier. $\text{SAI} = \text{PLMN-Id} + \text{LAC} + \text{SAC}$
Cell	Used to uniquely identify a cell within an RNS The Cell-Id together with the identifier of the controlling RNC (CRNC-Id) constitutes the UTRAN Cell Identity (UC-Id) UC-Id or C-Id is used to identify a cell in UTRAN Iub, Iur and Iu interfaces. $\text{UC-Id} = \text{RNC-Id} + \text{C-Id}$
Local Cell	Used to uniquely identify the set of resources within a Node B required to support a cell (as identified by a C-Id). Also used for the initial configuration of a Node B when no C-Id is defined.
UE (Radio Network Temporary Identities (RNTI) as UE identifiers	Used as in UTRAN and in signalling messages between UE and UTRAN. They include: Serving RNC RNTI (s-RNTI), Drift RNC RNTI (d-RNTI), Cell RNTI (c-RNTI), UTRAN RNTI (u-RNTI) See their use in [6]
Resource identifiers (see [6])	Radio network control plane identifiers, Transport network control plane identifiers and binding identifier.

3.4.2 System Access Control

Through the system access 3G subscribers connect to the UMTS network to use services and/or facilities. Subscriber system access may be initiated from either the mobile side, e.g. a mobile originated call, or the network side, e.g. a mobile terminated call. In the following we summarize key system access control functions; specifications in [6] describe additional details.

3.4.2.1 Admission and Congestion Control

Admission control admits or denies new users, new radio access bearers or new radio links resulting from network tasks, e.g. handover events. It aims to avoid overload situa-

tions and bases its decisions on interference and resource measurements. It also serves during initial UE access, RAB assignment/reconfiguration and handover depending on the required events. Finally, its functions depend on UL interference and DL power information located in the controlling RNC. The serving RNC performs admission control towards the Iu interface.

Congestion control monitors, detects and handles situations when the system reaches near overload or an overload situation while users remain connected. Thus, when somewhere in the network, limited resources degrade service quality, congestion control brings the system back and restores stability seamlessly.

3.4.2.2 System Information Broadcasting

This function provides the mobile station with the access stratum and non-access stratum information used by the UE for its operation within the network.

3.4.3 Radio Channel Ciphering and Deciphering

This computation function protects radio-transmitted data against unauthorized third parties. Ciphering and deciphering usage may depend on a session key, derived through signalling and/or session dependent information.

3.4.4 Mobility Functions

3.4.4.1 Handover

Handover manages radio interface mobility based on radio measurements in order to maintain CN quality of service. It may be directed to/from another system (e.g. UMTS to GSM handover). Control for this function may originate in the network, or may come independently from the UE. Hence, it may be located in the SRNC, the UE, or both.

3.4.4.2 SRNS Relocation

This function coordinates events when a SRNS role passes to another RNS. It manages the Iu interface connection mobility from one RNS to another. The SRNC initiates the *SRNS relocation*, which finds a home in the RNC and CN as illustrated in Figure 3.9.

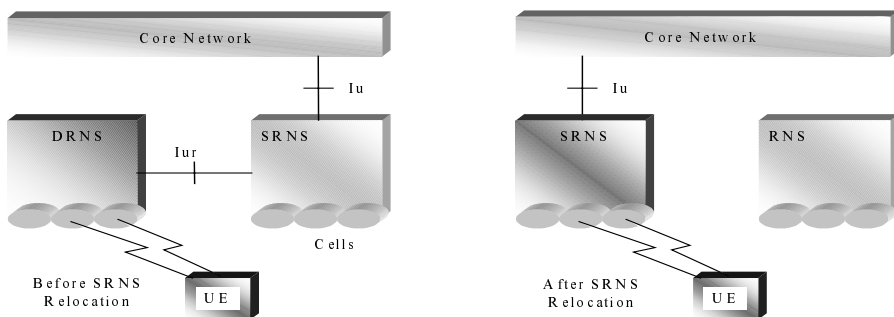


Figure 3.9 A serving RNS relocation example.

3.4.5 Radio Resource Management and Control Functions

Radio resource management concerns the allocation and maintenance of radio communication resources. In UMTS CS and PS services share these resources.

Not all functions apply to both FDD and TDD modes. For example, macro-diversity applies only to FDD while dynamic channel allocations applies only to TDD.

3.4.5.1 Radio Resource Configuration

This function configures the radio network resources, i.e. cells and common transport channels (e.g. BCH, RACH, FACH, PCH), and takes the resources into or out of operation.

3.4.5.2 Radio Environment Survey

The radio environment survey performs quality estimates and measurements on radio channels from current and surrounding cells; as in [6] these functions, located in the UE and UTRAN, include:

- received signal strengths (current and surrounding cells);
- estimated bit error ratios, (current and surrounding cells);
- estimation of propagation environments (e.g. high-speed, low-speed, satellite, etc.);
- transmission range (e.g. through timing information);
- Doppler shift;
- synchronization status;
- received interference level;
- total DL transmission power per cell.

3.4.5.3 Macro-diversity Control – FDD

In FDD, *macro-diversity control* manages duplication/replication of information streams to receive/transmit the same information through multiple physical channels (or different cells) from/towards a single mobile terminal.

This function also controls combining of information streams generated by a single source (diversity link), but conveyed via several parallel physical channels (diversity sub-links). Macro-diversity control interacts with *channel coding control* to reduce bit error ratio when combining different information streams. Depending on the physical network configuration, combining/splitting may occur at the SRNC, DRNC or Node B level.

3.4.5.4 TDD – Dynamic Channel Allocation (DCA)

The TDD mode uses fast or slow DCA. Fast DCA implies assigning resources to the radio bearers in relation to the admission control. Slow DCA implies assigning radio

resources, including time slots, to different TDD cells depending on the varying cell load.

3.4.5.5 Allocation/De-allocation and Control of Radio Bearers

The allocation/de-allocation function located in the CRNC and SRNC, translates the connection element set up requests into physical radio channel allocation according to the QoS of the radio access bearer. It gets activated, e.g. during a call when user service request varies or during macro-diversity.

Radio bearer control located both in the UE and in the RNC, manages connection element set up and release in the radio access sub-network. It participates in the processing of the end-to-end connection set up and release, as well as the managing and maintenance of the end-to-end connection, which is located in the radio access sub network.

3.4.5.6 Radio Protocols Function

This function provides user data and signalling transfer capability across the UMTS radio interface by adapting the services (according to the QoS of the radio access bearer) to the radio transmission. This function includes:

- multiplexing of services and multiplexing of UEs on radio bearers;
- segmentation and reassembly;
- acknowledged/unacknowledged delivery according to the radio access bearer QoS.

3.4.5.7 RF Power Control

Power control manages the transmitted power level in order to minimize interference and keep connection quality. Table 3.2 illustrates the different functions.

Table 3.2 RF Power Control Functions

Power control	Link	Function description
Outer loop	UL	Located in the SRNC sets the target quality value for the UL inner loop power control, which is located in Node B for FDD and is located in the UE for TDD. It receives input from quality estimates of the transport channel. The UL outer loop power control is mainly used for long-term quality control of the radio channel. In FDD, if the connection involves both a SRNS and a DRNS the function UL outer loop power control (located in the SRNC) sets the target quality for the UL inner loop power control function (located in Node B).
	DL	Sets the target quality value for the DL inner loop power control. It receives input from quality estimates of the transport channel measured in the UE. It is used for a long-term quality control of the radio channel. Located mainly in the UE with some control parameters are set by the UTRAN, where the SRNC under an algorithms control sends the target down link power range based on the measurement report from UE.

Inner Loop	UL	Sets the power of the uplink dedicated physical channels. <i>In FDD</i> , it is a closed loop process receiving quality target from UL outer loop power control and quality estimates of the uplink dedicated physical control channel. The UE gets power control commands on the downlink dedicated physical control channel. Resides in both the UTRAN and the UE. <i>In TDD</i> located in the UE, it is a open loop process receiving quality target from the UL outer loop power control and uses the quality target and quality estimates of downlink channels to set the transmit power.
	DL	This function located in both UTRAN and UE, sets the power of the downlink dedicated physical channels while receiving the quality target from DL outer loop power control and quality estimates of the downlink dedicated physical control channel. Power control commands are sent on the uplink dedicated physical control channel to the UTRAN.
Open Loop	UL	This function located in both UTRAN and UE sets the initial power of the UE at random access. It uses UE measurements and broadcasted cell/system parameters as input
	DL	Function located in both UTRAN and UE, sets the initial power of downlink channels. It receives downlink measurement reports from the UE

3.4.5.8 Radio Channel Coding and Control

This function located in both UE and UTRAN brings redundancy into the data source flow, thereby increasing its rate by adding information calculated from the data source. This allows detection or correction of signal errors introduced by the transmission medium. Channel coding algorithm(s) and redundancy level may vary in the different types of logical channels and different types of data.

Channel coding control residing in both UE and UTRAN, generates control information required by the channel coding/ decoding execution functions, e.g. channel coding type, code rate, etc.

3.4.5.9 Radio Channel Decoding

Channel decoding aims to reconstruct the information source using the added redundancy by the channel coding function to detect or correct possible errors in the received data flow. This function may also employ a priori error likelihood information generated by the demodulation function to increase the efficiency of the decoding operation. The channel decoding function, located in both the UE and UTRAN, complements the channel coding function.

3.4.5.10 Initial Random Access

This function, located in the UTRAN, detects initial MS access attempts and responds accordingly. Handling this initial access may incorporate procedures to resolve colliding events. Successful attempts will obtain the right to resource allocation request.

3.4.5.11 NAS Core Network Distribution Functions

Non-Access Stratum Messages (NAS) messages in the RRC protocol have transparent transfer within the access stratum through a direct transfer procedure. A UE/SRNC distribution function handles a CN domain indicator, service descriptor, and flow ID being part of the AS message to direct messages to the corresponding NAS entity, i.e. the appropriate mobility management instance in the UE domain and the corresponding CN domain.

In the downlink the SRNC provides the UE with the necessary information on the originating CN domain for the individual NAS message.

In the uplink, the UE distribution function inserts the appropriate CN domain values, domain indicator, service descriptor, and flow ID IEs in the AS message. The SRNC evaluates the CN domain indicator, service descriptor, and flow ID contained in the AS message and distributes the NAS message to the corresponding RANAP instance for transfer over the Iu interface.

3.4.5.12 Timing Advance in TDD

This function aligns uplink radio signals from the UE to the UTRAN. It is based on uplink burst timing measurements performed by the Node B L1, and on timing advance commands sent downlink to the UE.

3.4.5.13 NAS Service Specific Function

A UE or SRNC service specific function provides a SAP for particular services (e.g. priority levels). In the downlink direction, the SRNC may base the routing on this SAP.

3.5 MOBILITY MANAGEMENT

3.5.1 Signalling Connection

The UE may or may not have a signalling connection, and in the radio interface dedicated or common channels can be used [7].

When an established signalling connection exists over the Dedicated Control Service Access Point (DC-SAP) from the access stratum, the CN reaches the UE by a dedicated connection SAP on the CN side with a context between UTRAN and UE for the given connection. This context disappears when the connection is released and a *dedicated connection* can be initiated only from the UE.

When a dedicated connection does not exist, the CN reaches the UE through the notification SAP, where the CN message may request the UE to establish a dedicated connection. The UE is addressed with a user/terminal identity and a geographical area.

The location of the UE is known either at cell level (higher activity) or in a larger area consisting of several cells (lower activity). Knowing the location minimizes the number of location update messages for moving UEs with low activity and removes paging needs for UEs with high activity.

3.5.2 Impacts of Mobility Handling

In the presence of a dedicated connection to the UE, the UTRAN handles the UE radio interface mobility, such as soft handover, and procedures for handling mobility in the RACH/PCH substrate. The radio network cell structure should not necessarily be known outside the UTRAN.

In the absence of a dedicated connection to the UE, mobility handling occurs directly between the UE and CN outside the access stratum, e.g. through registration procedures. While paging the UE, the CN indicates a geographical area which becomes the actual paged cell in UTRAN. Within a cell structure we may use location area identities or other means to identify a geographical area independently.

While a dedicated connection lasts, the UE suppresses its registrations to the CN and re-registers if required. Thus, the UTRAN does not contain any permanent location registers for the UE, but only temporary contexts for the duration of the dedicated connection. This context may typically contain location information (e.g. current cell(s) of the UE) and information about allocated radio resources and related connection references [6].

3.6 UTRAN SYNCHRONIZATION AND O&M REQUIREMENTS

3.6.1 Synchronization Model

The main synchronization issues in UTRAN include: network, node, transport channel, radio interface, and time alignment synchronization. Figure 3.10 illustrates the nodes involved in these issues (with exception of network and node synchronization).

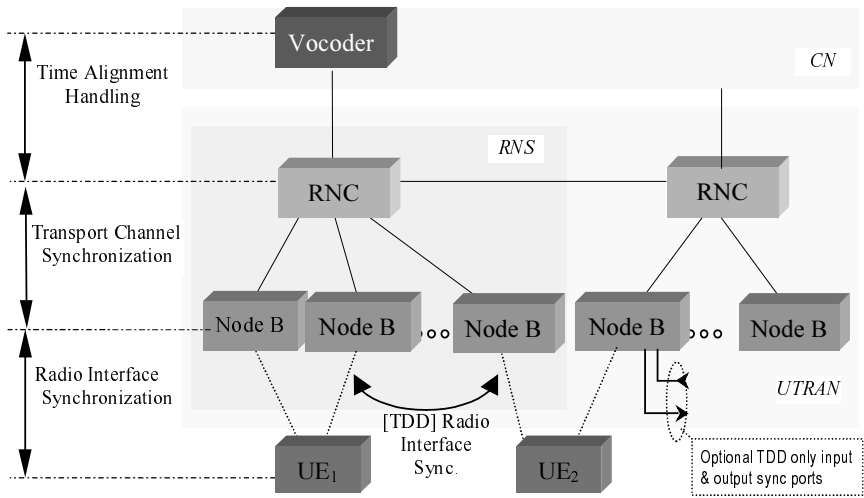


Figure 3.10 Synchronization issues model.

3.6.2 Node B O&M

Figure 3.11 illustrates the two Node B O&M types, i.e. the *implementation specific O&M* linked to the actual implementation of Node B, and the *logical O&M* having impacts on the traffic carrying resources in Node B controlled from the RNC.

3.6.2.1 Implementation Specific O&M

Implementation specific O&M functions depend on both HW and SW management components of Node B, and their transport from Node B to the management system occurs via the RNC. The implementation specific O&M interface shares the same physical bearer with the Iub interface, where [8] specifies the routing function and the transport bearer.

Routing across the RNC in the UTRAN is optional, but signalling between co-located equipment and its management system is required, this may be carried over the same bearer as the implementation specific O&M.

3.6.2.2 Logical O&M

The logical O&M represents the signalling associated with the control of logical resources owned by the RNC but physically implemented in Node B (e.g. channels, cells, etc.). The RNC controls these logical resources. A number of O&M procedures physically implemented in Node B impact on the logical resources requiring an information exchange between RNC and Node B. All messages needed to support this information exchange are classified as logical O&M forming an integral part of NBAP [6].

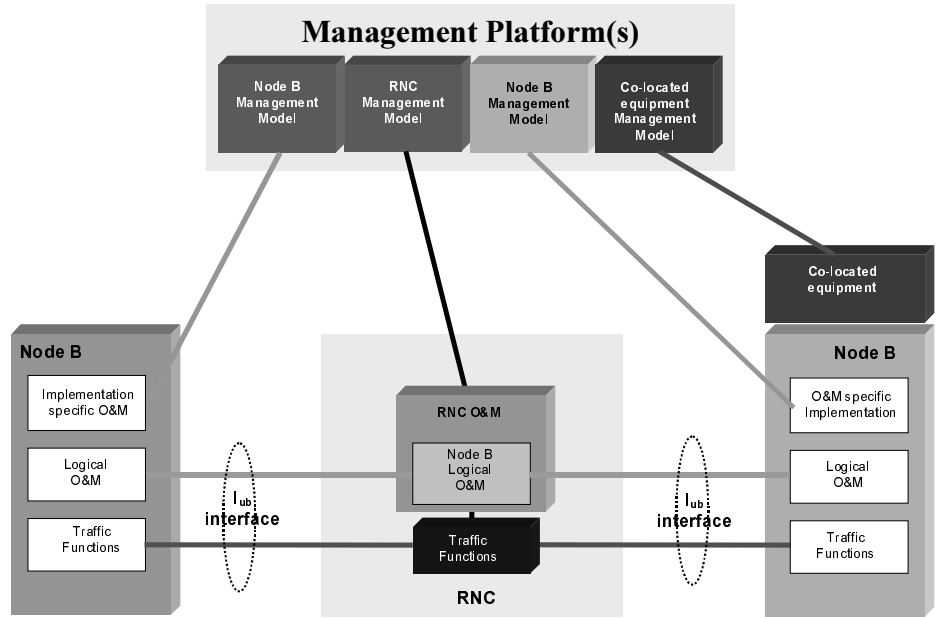


Figure 3.11 RNS architecture with O&M interfaces [6].

Figure 3.11, with representative logical connections, shows the concept of an interface from the RNC to the management system and O&M functions within the management system for clarity only. The specifications of these functional connections do not correspond to the scope of the radio network specification.

3.7 UTRAN INTERFACES

3.7.1 General Protocol Model for UTRAN Interfaces

Figure 3.12 presents the general UTRAN interfaces protocol model. The structure assumes that layers and planes are logically independent of each other, and if needed, the whole structure may evolve later with standardization work.

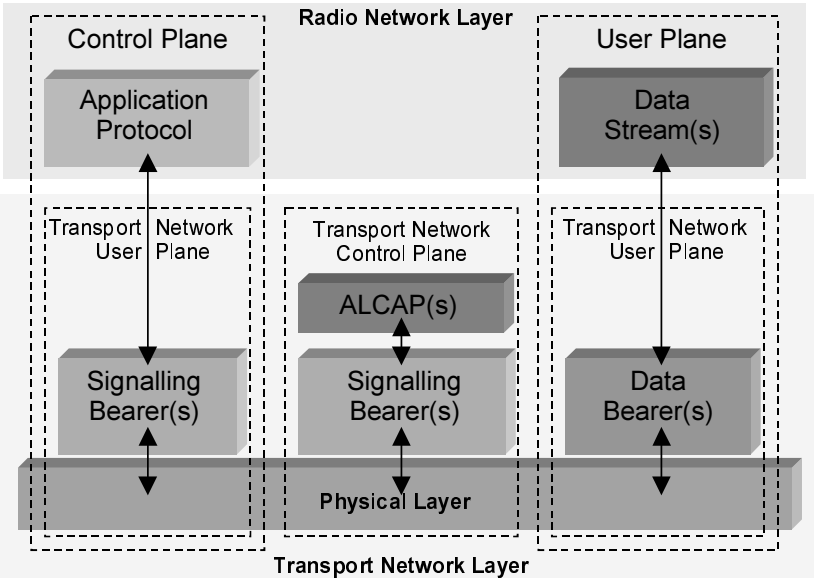


Figure 3.12 General protocol model for UTRAN Interfaces.

3.7.2 Horizontal Layers

The radio network and transport network layers constitute the main components of the protocol structure. The first layer contains all visible UTRAN related issues, and the second layer represents standard UTRAN transport technology for selection without any specific requirements.

3.7.3 Vertical Planes

3.7.3.1 Control Plane

The control plane includes the application protocol, i.e. RANAP, RNSAP or NBAP, and the signalling bearer for transporting the application protocol messages.

Among other things, the application protocol is used for setting up bearers for (i.e. radio access bearer or radio link) in the radio network layer. In the three plane structure the bearer parameters in the application protocol are not directly tied to the user plane technology, but are rather general bearer parameters.

The signalling bearer for the application protocol may or may not be of the same type as the signalling protocol for the ALCAP. The signalling bearer is always set up by O&M actions.

3.7.3.2 User Plane

The user plane includes the data stream(s) and the data bearer(s) for the data stream(s). The data stream(s) is/are characterized by one or more frame protocols specified for that interface.

3.7.3.3 Transport Network Control Plane

The transport network control plane does not include any radio network layer information, and is completely in the transport layer. It includes the ALCAP protocol(s) that is/are needed to set up the transport bearers (data bearer) for the user plane. It also includes the appropriate signalling bearer(s) needed for the ALCAP protocol(s).

The transport network control plane is a plane that acts between the control plane and the user plane. The introduction of transport network control plane makes it possible for the application protocol in the radio network control plane to be completely independent of the technology selected for data bearer in the user plane.

When the transport network control plane is used, the transport bearers for the data bearer in the user plane are set up in the following fashion. First there is a signalling transaction by the application protocol in the control plane, which triggers the set up of the data bearer by the ALCAP protocol that is specific for the user plane technology.

The independence of control plane and user plane assumes that ALCAP signalling transaction takes place. It should be noted that ALCAP might not be used for all types data bearers. If there is no ALCAP signalling transaction, the transport network control plane is not needed at all. This is the case when pre-configured data bearers are used.

It should also be noted that the ALCAP protocol(s) in the transport network control plane is/are not used for setting up the signalling bearer for the application protocol or for the ALCAP during real time operation.

The signalling bearer for the ALCAP may or may not be of the same type as the signalling bearer for the application protocol. The signalling bearer for ALCAP is always set up by O&M actions.

3.7.3.4 Transport Network User Plane

The data bearer(s) in the user plane, and the signalling bearer(s) for application protocol, belong also to the transport network user plane. As described in the previous section, the data bearers in the transport network user plane are directly controlled by the

transport network control plane during real time operation, but the control actions required for setting up the signalling bearer(s) for application protocol are considered O&M actions.

The following section is an informative section which aims to provide an overall picture of how the MAC layer is distributed over Uu, Iub and Iur for the RACH, FACH and DCH [7].

3.8 RADIO INTERFACE PROTOCOL ARCHITECTURE

This section covers essential aspects on the radio interface protocols based on [9]. For completeness and to remain close to the technical specifications we use the same terminology and keep the approach of the proposed architecture.

3.8.1 Protocol Structure

Radio interface protocols establish, adapt, and free radio bearer services in the UTRA platform. They have functions in Layers 1–3, i.e. physical (L1), link (L2) and network (L3) layers in the OSI terminology. L2 has in turn the following sub-layers: Medium Access Control (MAC), Radio Link Control (RLC), Packet Data Convergence Protocol (PDCP) and Broadcast/Multicast Control (BMC). L3 and RLC consist of Control (C) and User (U) planes. The PDCP and BMC sub-layers exist only in the U plane.

Layer 3 has sub-layers in the C-plane. The lowest one, the Radio Resource Control (RRC), interfaces with L2 and terminates in the UTRAN. The next sub-layer provides duplication avoidance functionality [10] and terminates in the CN. It remains part of the access stratum to provide access stratum services to higher layers. However, we assume that higher layer signalling such as Mobility Management (MM) and Call Control (CC) does not belong to the non-access stratum⁶.

In the architecture representation shown by Figure 3.13, each block indicates an instance of the respective protocol. At the interface between sub-layers, we mark with ovals Service Access Points (SAP) for peer-to-peer communication. The SAP between MAC and the physical layer provides the transport channels, and SAPs between RLC and the MAC sub-layer provide the logical channels. In the C-plane, the General Control (GC) defines through Notification (Nt) and Dedicated Control (DC) SAPs the interface between duplication avoidance and higher L3 sub-layers (CC, MM).

Figure 3.13 also illustrates connections between RRC and MAC as well as RRC and L1 affording local inter-layer control services. We have as well, an equivalent interface control between RRC and the RLC sub-layer, between RRC and the PDCP sub-layers and between RRC and the BMC sub-layer. These interfaces enable the RRC to control the configuration of the lower layers. Thus, separate control SAPs defined between RRC and each lower layer (PDCP, RLC, MAC, and L1) exist.

⁶ Higher level signalling is not in the scope of 3GPP TSG RAN. On the other hand, the UTRA radio interface protocol architecture has similarities to the current ITU-R protocol architecture, ITU-R M.1035.

The RLC sub-layer provides ARQ functionality in conjunction with the applied radio transmission technique. In this case, we do not see a difference between RLC instances in C and U planes. When the Iu connection-point remains unchanged, the CN may request the UTRAN full data protection. However, when the Iu connection point changes (e.g. SRNS relocation, streamlining, etc.), the UTRAN may not guarantee full data protection, but rely on duplication avoidance functions in the CN.

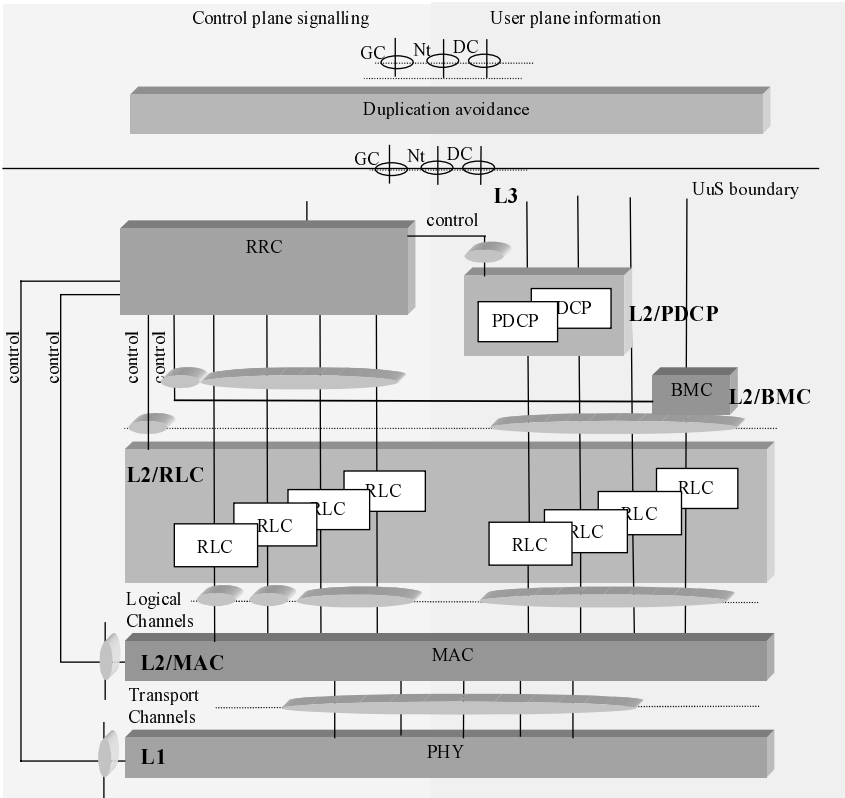


Figure 3.13 Radio interface protocol architecture (ovals are service access points) (after [9]).

3.8.1.1 Service Access Points and Service Primitives

At SAPs each layer provides services where a set of primitives or operations defines every service that a layer provides to the upper layer(s). There exists control services at Control SAPs (C-SAP) enabling the RRC layer to control lower layers locally (i.e. not requiring peer-to-peer communication)⁷. See primitives in [9].

⁷ C-SAP primitives can bypass one or more sub-layers.

3.8.2 Services and Functions in Layer 1

L1 or the physical layer provides information transfer services to MAC and upper layers. We characterize these services by *how* and with *what* features information gets transferred over the air interface, and we denote them *transport channels*⁸.

3.8.2.1 Services

3.8.2.1.1 Transport channels

In principle we classify transport channels in two groups, i.e. common and dedicated channels. The 1st group has in-band identification of UEs when addressing particular UEs. The 2nd group has identification of UEs through the physical channel, i.e. code and frequency for FDD and code, time slot and frequency for TDD.

Table 3.3 Transport Channel Summary [8]

Transport channels	description
<i>Common</i>	
Random Access Channel (RACH)	A contention based uplink channel used for transmission of relatively small amounts of data, e.g. for initial access or non-real-time dedicated control or traffic data.
Common Packet Channel (CPCH)	A contention based channel used for transmission of bursty data traffic. This channel only exists in FDD mode and only in the uplink direction. The common packet channel is shared by the UEs in a cell and therefore, it is a common resource. The CPCH is fast power controlled.
Forward Access Channel (FACH)	Common downlink channel without closed-loop power control used for transmission of relatively small amount of data.
Downlink Shared Channel (DSCH)	A downlink channel shared by several UEs carrying dedicated control or traffic data.
Uplink Shared Channel (USCH)	An uplink channel shared by several UEs carrying dedicated control or traffic data, used in TDD mode only.
Broadcast Channel (BCH)	A downlink channel used for broadcast of system information into an entire cell.
Paging Channel (PCH)	A downlink channel used for broadcast of control information into an entire cell allowing efficient UE sleep mode procedures. Currently identified information types are paging and notification. Another use could be UTRAN notification of change of BCCH information.
<i>Dedicated</i>	
Dedicated Channel (DCH)	A channel dedicated to one UE used in uplink or downlink
Fast Uplink Signalling Channel (FAUSCH)	An uplink channel used to allocate dedicated channels in conjunction with FACH

Each transport channel, excluding the FAUSCH⁹, gets an associated *transport format* when having a fixed or slow changing rate, or an associated *transport format set* when having a fast changing rate. We define the transport format as a combination of encoding, interleaving, bit rate and mapping onto physical channels [11]. We define the trans-

⁸ They transport signal and traffic information.

⁹ It only conveys a reservation request.

port format set as a group of transport formats. In the context of the latter, e.g. variable rate DCH has a transport format set, i.e. one transport format for each rate, whereas a fixed rate DCH has a single transport format [9].

3.8.2.2 L1 Functions

Chapter 4 describes the main functions of L1; here we list a summary from [9] to complete the services and functions introduction.

- error detection on transport channels and indication to higher layers;
- FEC encoding/decoding and interleaving/de-interleaving of transport channels;
- multiplexing of transport channels and de-multiplexing of coded composite transport channels;
- rate matching;
- modulation and spreading/demodulation and de-spreading of physical channels;
- macro-diversity distribution/combining and soft handover execution;
- mapping of coded composite transport channels on physical channels;
- power weighting and combining of physical channels;
- frequency and time (chip, bit, slot, frame) synchronization;
- measurements and indication to higher layers (e.g. FER, SIR, interference power, transmit power, etc.);
- closed-loop or fast power control;
- RF processing;
- support of uplink synchronization (TDD only);
- support of timing advance on uplink channels (TDD only).

3.8.3 Services and Functions in Layer 2

3.8.3.1 Services and Functions in The MAC Sub-layer

Specification in [12] provide the details of the MAC protocol; here we simply summarize the main services and functions.

3.8.3.1.1 Services to upper layers

- *Data transfer* → provides unacknowledged transfer of MAC SDUs between peer MAC entities without segmentation.
- *Reallocation of radio resources and MAC parameters* → performs on request of RRC execution of radio resource reallocation and change of MAC parameters. In TDD mode, in addition, the MAC handles resource allocation autonomously.
- *Reporting of measurements* → reports local measurements, e.g. traffic volume and quality indication to the RRC.

3.8.3.1.2 Logical channels

The MAC layer provides data transfer services on logical channels. We classify these channels in two groups, i.e. *control channels* for control-plane information transfer, and *traffic channels* for user-plane information transfer (see Table 3.4).

Table 3.4 Summary of Logical Channels

Logical channels	Description
<u>Control Channels (CCH)</u> Transfer of control plane information only	<u>Broadcast Control Channel (BCCH)</u> A downlink channel for broadcasting system control information
	<u>Paging Control Channel (PCCH)</u> A downlink channel transferring paging information.
	<u>Dedicated Control Channel (DCCH)</u> A point-to-point bi-directional channel that transmits dedicated control information between a UE and the network
	<u>Common Control Channel (CCCH)</u> Bi-directional channel for transmitting control information between network and UEs
	<u>Shared Channel Control Channel (SHCCH)</u> Bi-directional channel that transmits control information for uplink and downlink shared channels between network and UEs
<u>Traffic Channel (TCH)</u>	<u>Dedicated Traffic Channel (DTCH)</u> A DTCH is a point-to-point channel, dedicated to one UE, for the transfer of user information. A DTCH can exist in both uplink and downlink.
	<u>Common Traffic Channel (CTCH)</u> A point-to-multipoint unidirectional channel for transfer of dedicated user information for all or a group of specified UEs.

3.8.3.1.3 Mapping between logical channels and transport channels

Table 3.5 illustrates connections between logical and transport channels:

Table 3.5 Connections Between Logical and Transport Channels

Channel	Connected to
BCCh	BCH, may also FACH
CCCH	PCH
CCCH	RACH and FACH
SHCCH	RACH, USCH/FACH and DSCH
DTCH	Either RACH and FACH; RACH and DSCH to DCH and DCSCH, DCH AND DSCH, DCH, CPCH (FDD only) or USCH (TDD only) ⁹
CTCH	FACH
DCCH	Either RACH and FACH, RACH and DSCH, DCH and DSCH, DCH, CPCH (FDD only), FAUSCH, USCH (TDD only)

Figures 3.14 and 3.15 illustrate the mappings as seen from the UE and UTRAN sides including both the FDD and TDD modes.

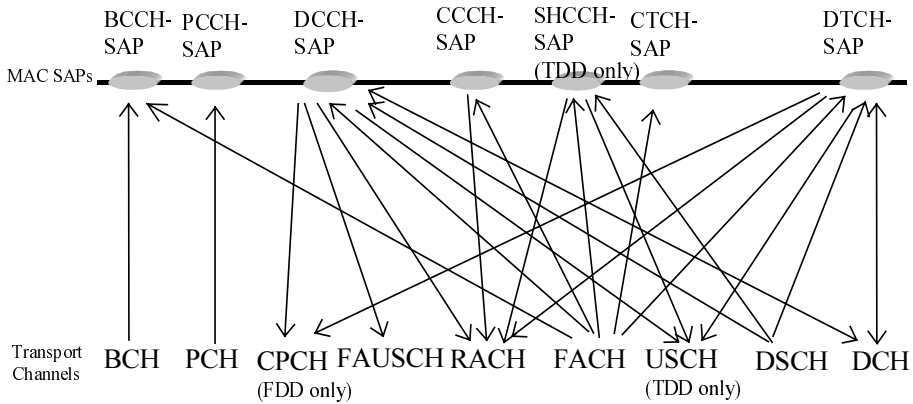


Figure 3.14 Logical channels mapped onto transport channels, seen from the UE side [9].

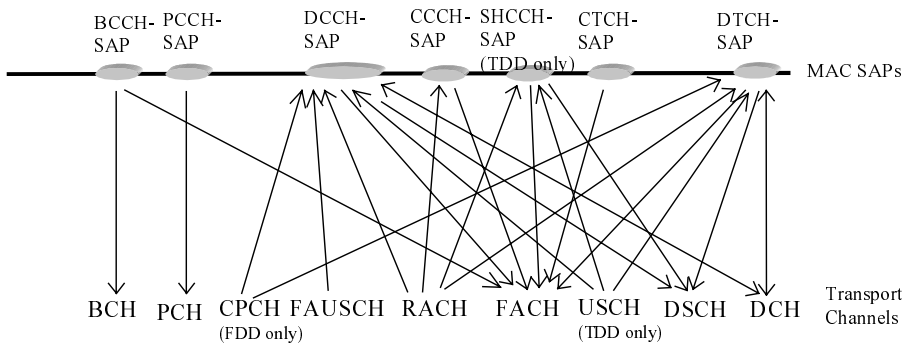


Figure 3.15 Logical channels mapped onto transport channels, seen from the UTRAN side [9].

3.8.3.2 MAC Functions

The functions of MAC include:

- *Mapping between logical channels and transport channels.*
- *Selection of appropriate transport format for each transport channel depending on instantaneous source rate.*
- *Priority handling between data flows of one UE.* “Priorities are e.g. given by attributes of Radio Bearer services and RLC buffer status. The priority handling is achieved by selecting a Transport Format Combination for which high priority data is mapped onto L1 with a ‘high bit rate’ Transport Format, at the same time letting lower priority data be mapped with a ‘low bit rate’ (could be zero bit rate) Transport Format. Transport format selection may also take into account transmit power indication from Layer 1” [9].

- *Priority handling between UEs by means of dynamic scheduling* → aiming for a dynamic scheduling for efficient spectrum utilization¹⁰. The MAC realizes priority handling on common and shared transport channels.
- *Identification of UEs on common transport channels*. When addressing a particular UE on a common downlink channel, or when a UE uses the RACH, we need in-band identification of the UE.
- *Multiplexing/de-multiplexing of higher layer PDUs into/from transport blocks delivered to/from the physical layer on common transport channels*. The MAC supports multiplexing for common transport channels as complement to the physical layer.
- *Multiplexing/demultiplexing of higher layer PDUs into/from transport block sets delivered to/from the physical layer on dedicated transport channels*. The MAC allows service multiplexing for dedicated transport channels.
- *Traffic volume monitoring*. Based on the MAC's measurement of traffic volume on logical channels and reported to RRC, the latter performs transport channel switching decisions.
- *Dynamic Transport Channel type switching* → switching execution between common and dedicated transport channels takes place based on a switching decision derived by the RRC.
- *Ciphering*. This function prevents unauthorized acquisition of data. Ciphering occurs in the MAC layer for transparent RLC mode.
- *Access Service Class selection for RACH transmission*. The RACH resources (i.e. access slots and preamble signatures for FDD, timeslot and channelization code for TDD) may be divided between different Access Service Classes (ASC) in order to provide different priorities of RACH usage. More than one ASC or all ASCs can get assigned to the same access slot/signature space. Each ASC will also have a set of back-off parameters associated with it, some or all of which may be broadcasted by the network. The MAC function applies the appropriate back-off and indicates to the PHY layer the RACH partition associated with a given MAC PDU transfer [9].

3.8.3.3 RLC Services and Functions

3.8.3.3.1 Services

- *RLC connection establishment/release*.
- *Transparent data transfer* → transmits higher layer PDUs without adding any protocol information, but may include segmentation/re-assembly functionality.
- *Unacknowledged data transfer* → transmits higher layer PDUs without guaranteeing delivery to the peer entity. The unacknowledged data transfer mode has the following characteristics:

¹⁰ In the TDD we represent transportable data in terms resource units sets.

- *Detection of erroneous data*: delivering only correct SDUs to the receiving higher layer by using the sequence-number check function.
- *Unique delivery*: delivering SDUs only once to the receiving upper layer using the duplication detection function.
- *Immediate delivery*: delivering SDUs to the higher layer receiving entity as soon as it arrives at the receiver.
- *Acknowledged data transfer*. Transmits higher layer PDUs and guarantees delivery to the peer entity. It has the following characteristics:
 - *Error-free delivery*: ensured by means of retransmission.
 - *Unique delivery*: delivering each SDU only once to the receiving upper layer using duplication detection function.
 - *In-sequence delivery*: supports for in-order delivery of SDUs, i.e. delivering SDUs to the receiving higher layer entity in the same order as the transmitting higher layer entity submits them to the RLC sub-layer.
 - *Out-of-sequence delivery*: it shall also be possible to allow the receiving RLC entity to deliver SDUs to a higher layer in a different order than submitted to RLC sub-layer at the transmitting side.
- *QoS setting*. Configurable by Layer 3 to provide different levels of QoS.
- *Notification of unrecoverable errors*. Notifying the upper layer of errors that cannot be resolved by RLC¹¹ itself by normal exception handling procedures, e.g. by adjusting the maximum number of retransmissions according to delay requirements.

3.8.3.4 RLC Functions

- *Segmentation and reassembly*. This function performs segmentation/reassembly of variable-length higher layer PDUs into/from smaller RLC Payload Units (PUs). The RLC PDU size is adjustable to the actual set of transport formats.
- *Concatenation*
- *Padding*. In the absence of concatenation and non-filled RLC PDUs of given size, the remainder of the data field gets filled with padding bits.
- *Transfer of user data*. Conveyance of data between users of RLC services.
- *Error correction*. Error correction by retransmission (e.g. Selective Repeat, Go Back N, or a Stop-and-Wait ARQ) in acknowledged data transfer mode.
- *In-sequence delivery of higher layer PDUs*. Preserves the order of higher layer PDUs when submitted for transfer by RLC using the acknowledged data transfer service.

¹¹ There is a single RLC connection per radio bearer.

- *Duplicate detection.* Detects duplicated received RLC PDUs and ensures that the resultant higher layer PDU get delivered only once to the upper layer.
- *Flow control.* Allows an RLC receiver to control the rate at which the peer RLC transmitting entity may send information.
- *Sequence number check (unacknowledged data transfer mode).* Guarantees the integrity of reassembled PDUs and provides a mechanism for the detection of corrupted RLC SDUs through checking the sequence number in RLC PDUs when they are reassembled into a RLC SDU.
- *Protocol error detection and recovery.* Detects and recovers from errors in the operation of the RLC protocol.
- *Ciphering.* Prevents unauthorized acquisition of data. Ciphering occurs in the RLC layer for non-transparent RLC mode.
- *Suspend/resume function.* Suspension and resumption of data transfer as in e.g. LAPDm.

3.8.4 PDCP Services and Function

The Packet Data Convergence Protocol (PDCP) service provides transmission and reception of network PDUs in acknowledged/unacknowledged and transparent RLC mode. As part of its function, first it maps network PDUs from one network protocol to one RLC entity. Second it compresses in the transmitting entity and decompresses in the receiving entity redundant network PDU control information (header compression/decompression), including TCP/IP header compression and decompression when necessary. See more service and function details in [19].

3.8.5 Broadcast/Multicast Control – Services and Functions

The BMC provides broadcast/multicast transmission service in the user plane on the radio interface for common user data in transparent or unacknowledged mode. Its essential functions include from [19]:

- *Storage of cell broadcast messages* → stores messages received over the CBC-RNC interface for scheduled transmission.
- *Traffic volume monitoring and radio resource request for CBS* → at the UTRAN side, it calculates the required transmission rate for cell broadcast service based on the messages received over the CBC-RNC interface, and requests appropriate CTCH/FACH resources from RRC.
- *Scheduling of BMC messages* → The BMC receives scheduling information along with each cell broadcast message over the CBC-RNC-interface. Based on this UTRAN scheduling information, it generates schedule messages and schedules BMC message sequences correspondingly. At the UE side, it evaluates the scheduled messages and indicates scheduling parameters to RRC, which are used by the RRC to configure lower layers for CBS discontinuous reception.

- *Transmission of BMC messages to UE* → transmits BMC messages (scheduling and cell broadcast messages) based on a schedule.
- *Delivery of cell broadcast messages to upper layer (NAS)* → delivers correctly received cell broadcast messages to upper layer (NAS) in the UE, neglecting corrupted ones.

Specifications are given in [19], “Data flows through Layer 2”.

3.8.6 Uu Stratum Services and Functions in Layer 3

Here we provide an overview on Layer 3 services and functions based on the Uu Stratum. Further detailed description of the RRC protocol and structured procedures involving RRC can be found in [20–22].

The main Uu stratum services include: general control, notification and dedicated control. The first provides a common information broadcast service to all Ues in a certain geographical area. The second provides paging and notification broadcast services to a specific UE(s) in a certain geographical area. The third provides services for establishment/release of a connection and transfer of messages using this connection. It should also be possible to transfer a message during the establishment phase.

3.8.7 The Radio Resource Control (RRC) Functions

The RRC layer handles the control plane signalling of Layer 3 between the UEs and UTRAN. Its main functions include:

- *Broadcast of information provided by the access and non-access stratum (core network).* It performs information broadcasting from the network to all UEs. The system information is normally repeated on a regular basis.
- *Establishment, re-establishment, maintenance and release of an RRC connection between the UE and UTRAN.* Higher layers request the UE side to establish the first signalling connection for the UE. The establishment of an RRC connection includes an optional cell re-selection, an admission control, and a Layer 2 signalling link establishment.
- *Establishment, reconfiguration and release of radio bearers.* Can, on request from higher layers, perform the establishment, reconfiguration and release of radio bearers in the user plane.
- *Assignment, reconfiguration and release of radio resources for the RRC connection.* It handles the assignment of radio resources (e.g. codes, CPCH channels) needed for the RRC connection including needs from both the control and user plane.
- *RRC connection mobility functions.* Performs evaluation, decision and execution related to RRC connection mobility during an established RRC connection, e.g. handover, inter-system handover preparation, cell re-selection and cell/paging area update procedures, based on e.g. measurements done by the UE.

- *Paging/notification.* May broadcast paging information from the network to selected Ues, upon request from higher layers on the network side when necessary, or can also initiate paging during an established RRC connection.
- *Routing of higher layer PDUs.* Performs at the UE side routing of higher layer PDUs to the correct higher layer entity, at the UTRAN side to the correct RANAP entity.
- *Control of requested QoS.* It ensures that the QoS requested for the radio bearers can be met, e.g. allocation of a sufficient number of radio resources.
- *UE measurement reporting and control of the reporting.* The RRC layer controls the measurements performed by the UE in terms of what to measure, when to measure and how to report, including both UMTS air interface and other systems. It also performs the reporting of the measurements from the UE to the network.
- *Outer loop power control.* The RRC layer controls setting of the target of the closed loop power control.
- *Control of ciphering.* Provides procedures for setting of ciphering (on/off) between the UE and UTRAN.
- *Slow DCA.* It applies only to the TDD mode and involves allocating preferred radio resources dynamically based on long-term decision criteria.
- *Arbitration of radio resources on uplink DCH.* Controls rapid radio resource allocations on uplink DCH using a broadcast channel to send control information to all involved users.
- *Initial cell selection and re-selection in idle mode.* Selection of the appropriate cell based on idle mode measurements and cell selection criteria.
- *Integrity protection.* Adds a Message Authentication Code (MAC-I) to sensitive and/or containing sensitive information RRC messages.
- *Initial configuration for CBS.* Performs the initial configuration of the BMC sub-layer.
- *Allocation of radio resources for CBS.* Allocates radio resources for CBS based on traffic volume requirements indicated by BMC.
- *Configuration for CBS discontinuous reception.* Configures the lower layers (L1, L2) of the UE when the latter listen to the resources allocated for CBS.
- *Timing advance control.* Controls the operation of timing advance, which is applicable only to the TDD mode.

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APPENDIX A: UMTS FUNCTIONAL DOMAINS

Figure 3.16 illustrates the four (application, home, serving, and transport) strata. It also shows the integrated UMTS functional flow, i.e. the interactions between the USIM, MT/ME, access network, serving network and home network domains, including interactions between TE, MT, access network, serving network, transit network domains and the remote party.

The direct flows between non-contiguous domains (i.e. non-directly interconnected domains) are transparently transported across all the domains and interfaces located on the communication path between these end domains. The protocols may or may not be UMTS specific, as long as they can inter-work seamlessly to facilitate roaming.

When looking at the lower part of Figure 3.16, the home network domain becomes the transit network domain in the upper part. Thus, the integrated UMTS functional flow illustrated Figure 3.16, includes the representation or notation of the remote party indicating the remote-end entity (e.g. user or machine). It shows the end-to-end character of the communication. However, the specification of the remote party is outside the scope of the UMTS specification [1].

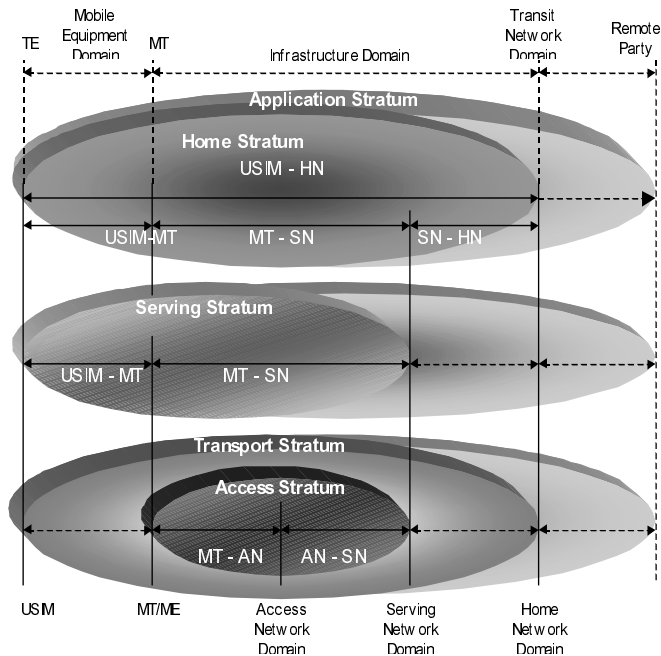


Figure 3.16 UMTS Architecture functional flow.

Because of the incorporation of the remote party in Figure 3.16, the home (through the application stratum, the serving and transport strata are extended all the way to the remote party line in the representation. Hence, two diversion paths reflect the flows exchanged between serving and home domains on one side and between serving and transit on the other side. Starting the first layer the stratum levels are defined in Table 3.6.

Table 3.6 Definition of the Strata Layers

Stratum levels	
Transport stratum	Supports the transport of user data and network control signalling from other strata through UMTS. It includes: • considerations of the physical transmission formats used for transmission; • mechanisms for error correction and recovery; • mechanisms to encrypt data over the radio interface and in the infrastructure part if required; • mechanisms for adaptation of data to use the supported physical format (if required); and • mechanisms to transcode data to make efficient use of, e.g. the radio interface (if required); • may include resource allocation and routing local to the different interfaces (if required); • the <i>access stratum</i>, which is specific to UMTS, as the part of the transport stratum.
Access stratum	• Consists of User Equipment (UE) and infrastructure parts, as well as access-technique specific protocols between these parts (i.e. specific physical media formats between the UE and the infrastructure used to carry information). • It provides services related to the transmission of data over the radio interface and the management of the radio interface to the other parts of UMTS. The access stratum includes the following protocols: • <i>Mobile Termination – Access Network (MT–AN)</i> protocol supporting transfer of detailed radio-related information to coordinate the use of radio resources between the MT and AN. • <i>Access Network – Serving Network (AN–SN)</i> protocol supporting the access from the SN to the resources provided by the AN. It is independent of the specific radio structure of the AN.
Serving stratum	Consists of protocols and functions to route and transmit user or network generated data/information from source to destination. The source and destination may be within the same or different networks. It contains functions related to telecommunication services, and includes: • <i>USIM – Mobile Termination (USIM–MT)</i> protocol supporting access to subscriber-specific information to allow functions in the user equipment domain. • <i>Mobile Termination – Serving Network (MT–SN)</i> protocol supporting access from the MT to the services provided by the serving network domain. • <i>Terminal Equipment – Mobile Termination (TE–MT)</i> protocol supporting exchange of control information between the TE and the MT.
Home stratum	• Consists of protocols and functions related to the handling and storage of subscription data and possibly home network specific services. • It also includes functions to allow domains other than the home network domain to act on behalf of the home network. • It contains functions related to subscription data management and customer care, as well as billing and charging, mobility management and authentication. The home stratum includes the following protocols:

	• <i>USIM – Home Network (USIM–HN)</i> protocol supporting co-ordination of subscriber-specific information between USIM & HN. • <i>USIM – Mobile Termination (USIM–MT)</i> protocol providing the MT with access to user specific data and resources necessary to perform actions on behalf of the home network. • <i>Mobile Termination – Serving Network (MT–SN)</i> protocol supporting user specific data exchanges between the MT and the SN. • <i>Serving Network – Home Network (SN–HN)</i> protocol providing the SN with access to HN data and resources necessary to perform its actions on behalf of the HN, e.g. to support the users communications, services and features (including VHE).
Application stratum	• It represents the application process itself, provided to the end-user. • It includes end-to-end protocols and functions making use of services provided by the <i>home</i>, <i>serving</i> and <i>transport</i> strata and necessary infrastructure supporting services and/or value added services. • The functions and protocols within the application stratum may adhere to GSM/UMTS standards or may be outside the scope of the UMTS standards. • End-to-end functions are applications consumed by users at the edge of/outside the overall network. • Authenticated and authorised users may access the applications by using any variety of available user equipment.

The data transport services offered to higher layers by L1 occurs through the use of transport channels via the MAC sub-layer. Table 4.1 illustrates some of the L1 or physical layer services. Through inter-working (e.g. a UE) provision of compatible bearers is assured.

Based on the types of physical channels L1 has two multiple access techniques:

- a Direct-Sequence Code Division Multiple Access (DS-CDMA) with the information spread within 5 MHz bandwidth, also referred to as Wide-band CDMA (WCDMA); and
- a Time Division Multiple Access (TDMA) + CDMA often denoted as TDMA/CDMA or TD/CDMA resulting from the extra slotted feature.

Table 4.1 Main Functions of the UTRA Physical Layer

1. Macro-diversity distribution/combining and soft handover execution	2. Power weighting and combining of physical channels
3. Error detection on transport channels and indication to higher layers	4. Modulation and spreading/ demodulation and de-spreading of physical channels
5. FEC encoding/decoding of transport channels	6. Frequency and time (chip, bit, slot, frame) synchronization
7. Multiplexing of transport channels and demultiplexing of coded composite transport channels	8. Radio characteristics measurements including FER, SIR, interference power, etc., and indication to higher layers
9. Rate matching (data multiplexed on DCH)	10. Inner-loop power control
11. Mapping of coded composite transport channels on physical channels	12. RF processing

The two access schemes afford UTRA two transmission modes, i.e. Frequency Division Duplex (FDD) corresponding to WCDMA operating with pair bands, and Time Division Duplex (TDD) corresponding to TD/CDMA operating with unpaired bands. The flexibility to operate in either FDD or TDD mode allows efficient spectrum utilization within the frequency allocation in different regions, e.g. Europe, Asia, etc.

The FDD mode or WCDMA is thus a duplex method where uplink and downlink transmissions use two different radio frequencies separated, e.g. by 190 MHz. The TDD mode is a duplex method where uplink and downlink transmissions occur over the same radio frequency by using synchronized time intervals. In the TDD, time slots in a physical channel are divided into transmission and reception parts. Information on uplink and downlink are transmitted reciprocally. The UTRA has QPSK as modulation scheme. In the WCDMA or FDD mode the spreading (and scrambling) process is closely associated with modulation. The different UTRA families of codes are:

- channelization codes derived with a code tree structure to separate channels from same the source, and codes to separate different cells;

Table 4.2 illustrates the harmonized parameters of the two UTRA modes.

A 10 ms radio frame divided into 15 slots (2560 chip/slot at the chip rate 3.84 Mcps) applies to two modes. A physical channel is therefore defined as a code (or number of codes) and additionally in TDD mode the sequence of time slots completes the definition of a physical channel. The information rate of the channel varies with the symbol rate being derived from the 3.84 Mcps chip rate and the spreading factor.

We derive the symbol rate from the 3.84 Mcps chip rate and the spreading factor to obtain a variable rate in the channel. The information rate of the channel, e.g. varies with

spreading factors from 256 to 4 for FDD uplink, from 512 to 4 for FDD downlink; and from 16 to 1 for TDD uplink and downlink. Consequently, modulation symbol rates vary from 960 k symbols/s to 15 k symbols/s (7.5 k symbols/s) for FDD uplink (downlink) respectively, and for TDD the momentary modulation symbol rates varies from 3.84 M symbols/s to 240 k symbols/s.

The UTRA has QPSK as modulation scheme. In the WCDMA or FDD mode the spreading (and scrambling) process is closely associated with modulation. The different UTRA families of codes are:

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Table 4.2 UTRA FDD and TDD Harmonized Parameters

Parameters	UTRA TDD	UTRA FDD
Multiple access	TDMA, CDMA (inherent FDMA)	CDMA (inherent FDMA)
Duplex method	TDD	FDD
Channel spacing and carrier chip rate	5 MHz (nominal) and 3.84 Mcps	
Time slot and frame length	15 slots/frame and 10 ms	
Spreading factor	1,2,4,8,16	4...512
Channel allocation	Slow and fast DCA supported	No DCA required
Types of burst	Traffic bursts, random access and synchronization burst	DTX time mask defined, burst not applicable
Multi-rate concept	Multi-code, multi-slot and orthogonal variable spreading	Multi-code and orthogonal variable spreading
Forward error correction (FEC) codes	Convolutional coding $R=1/2$ or $1/3$ constraint length $K=9$, turbo coding (8-state PCCC $R=1/3$) or service specific coding	
Interleaving	Inter-frame interleaving (10, 20, 40 and 80 ms)	
Modulation	QPSK	
Detection	Coherent, based on midamble	Coherent, based on pilot symbols
Dedicated channel power control	UL: open loop; 100 or 200 Hz DL: closed loop; rate ≤ 800 Hz	Fast closed loop; rate = 1500 Hz
Intra-frequency handover	Hard handover	Soft and softer handovers
Inter-frequency handover	Hard handover	
Intra-cell interference cancellation	Support for joint detection	Support for advanced receivers at base station

- gold codes with 10 ms period (38400 chips at 3.84 Mcps) used in the FDD mode, with the actual code itself length $2^{18}-1$ chips, and scrambling codes of length 16 used in the TDD mode;
- User Equipment (UE) separating codes: gold codes with 10 ms period, or alternatively S(2) codes 256 chip period for FDD mode, and codes with period of 16 chips and midamble sequences of different length depending on the environment for the TDD mode.

The key physical layer procedures involved with UTRA operation are:

- power control, with both inner loop and slow quality loop for FDD mode, and for TDD mode open loop in uplink and inner loop in downlink;
- cell search operation.

Measurements reported to higher layers and network containing radio characteristics like FER, SIR, interference power, etc. are:

- handover measurements within UTRA, e.g. determination of relative strength of a cell. In the FDD mode, identification of timing relation between cells to support asynchronous soft handover;
- other measurement procedures are: preparation for HO to GSM900/1800/1900; UE procedures before random access process; and procedures for Dynamic Channel Allocation (DCA) in the TDD mode.

4.2 DEDICATED AND COMMON TRANSPORT CHANNELS

Transport channels are defined by how and with what features data is transferred over the air interface. The generic classification of transport channels includes two groups, i.e. *dedicated* and *common* channels. The first group uses inherent UE addressing, while the second uses explicit UE addressing when addressing is required.

4.2.1 Dedicated Transport Channels

There is primarily one transport Dedicated Channel (DCH) for up- or downlink in the FDD and TDD modes, which is used to carry user or control information between the UTRAN and a UE. The DCH is transmitted over the entire cell or over only a part of the cell using, e.g. beam-forming antennas.

4.2.2 Common Transport Channels

While the intrinsic function of each common transport channel may not necessarily be identical in the FDD and TDD modes, both sets have basically the same function and acronym. Table 4.3 summarizes the essential definitions for the two modes.

Table 4.3 Summary of Common Transport Channels

FDD mode	TDD mode
<i>BCH – Broadcast Channel</i>	<i>BCH – Broadcast Channel</i>
Downlink transport channel that is used to broadcast system- and cell-specific information. The BCH is always transmitted over the entire cell and has a single transport format.	
<i>FACH – Forward Access Channel</i>	<i>FACH – Forward Access Channel(s)</i>
Downlink transport channel used to carry control information to a mobile station when the system knows the cell location of the mobile station. In the FDD, it can be transmitted over the entire cell or over only a part of the cell using, e.g. beam-forming antennas, and it can also be transmitted using slow power control. In the TDD may carry short user packets.	
<i>PCH – Paging Channel</i>	<i>PCH – Paging Channel</i>
Downlink transport channel transmitted always over the entire cell, used to carry control information to a mobile station when the system does not know the location cell of the mobile station. In the FDD mode transmission of the PCH is associated with the transmission of physical-layer generated paging indicators, to support efficient sleep-mode procedures.	
<i>RACH – Random Access Channel</i>	<i>RACH – Random Access Channel</i>
Uplink transport channel, always received from the entire cell, used to carry control information from the mobile station. In FDD, the RACH is characterized by a collision risk and by using open loop power control for transmission. In TDD it may also carry short user packets.	

<i>CPCH – Common Packet Channel</i>	<i>USCH – Uplink Shared Channel</i>
Uplink transport channel associated with a dedicated channel on the downlink, which provides power control and CPCH control commands (e.g. emergency stop). It is characterized by initial collision risk and by using inner loop power control for transmission.	Uplink transport channel shared by several UEs carrying dedicated control or traffic data.
<i>DSCH – Downlink Shared Channel</i>	<i>DSCH - Downlink Shared Channel</i>
Downlink transport channel shared by several UEs carrying dedicated control or traffic data. In FDD it is associated with one or several downlink DCH(s). It may be transmitted over the entire cell or over only a part of the cell using e.g. beam-forming antennas.	

Both FDD and TDD have a similar number of transport channels; however, the FDD mode does not have an Uplink Shared Channel (USCH) and the TDD mode does not have a Common Packet Channel (CPCH).

The CPCH transport channel in FDD performs essential power control commands, which may not be required in TDD. Likewise, the USCH transport channel performs essential commands in TDD, which may not be required in FDD.

4.3 CONFIGURATION OF FDD PHYSICAL CHANNELS

Physical channels in FDD inherit primarily a layered structure of radio frames and time slots. A radio frame is a processing unit consisting of 15 slots with a length of 38 400 chips, and slot is a unit consisting of fields containing bits with a length of 2560 chips. The slot configuration varies depending on the channel bit rate of the physical channel; thus, the number of bits per slot may be different for different physical channels and may, in some cases, vary with time. The basic physical resource is the code/frequency plane, and on the uplink, different information streams may be transmitted on the I and Q branches. Thus, a physical channel corresponds to a specific carrier frequency, code, and on the uplink there is in addition a relative phase (0 or $\pi/2$) element.

4.3.1 Uplink and Downlink Modulation

The uplink modulation uses a chip rate of 3.84 Mcps, where the complex-valued chip sequence generated by the spreading process has QPSK modulation as seen in Figure 4.2. The pulse-shaping characteristics are described in [3].

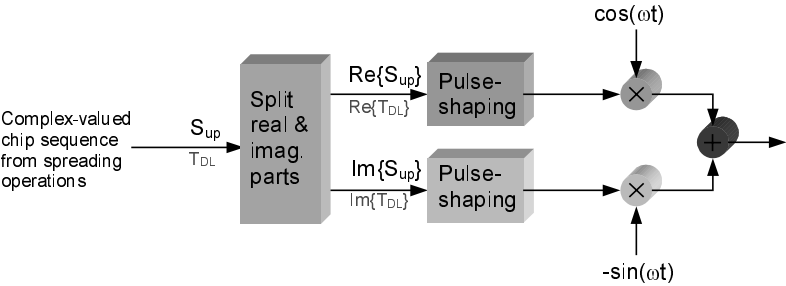


Figure 4.2 Uplink/downlink modulation process.

The downlink modulation also has a chip rate of 3.84 Mcps, with a QPSK modulated complex-valued chip sequence generated by the spreading process. Figure 4.2 does also represent the downlink modulation process. However, the DL pulse-shaping characteristics are described in [4].

4.3.2 Dedicated Uplink Physical Channels

The two types of uplink dedicated physical channels, i.e. Dedicated Physical Data Channel (DPDCH) and Dedicated Physical Control Channel (DPCCH) are I/Q code multiplexed within each radio frame. The uplink DPDCH carries the DCH transport channel, while the uplink DPCCH carries L1 control information such as: known pilot bits to support channel estimation for coherent detection, Transmit Power Control (TPC) commands, Feedback Information (FBI), and an optional Transport Format Combination Indicator (TFCI).

The TFCI informs the receiver about the instantaneous transport format combination of the transport channels mapped to the uplink DPDCH transmitted simultaneously. There is one and only one uplink DPCCH on each radio link; however, there may be zero, one, or several uplink DPDCHs on each radio link. Figure 4.3 illustrates the frame structure of the uplink dedicated physical channels, where each frame has 10 ms length split into 15 slots (T_{slot}) of 2560 chips length, corresponding to one power control period.

Parameter k in Figure 4.3 determines the number of bits per uplink DPDCH slot. It is related to the spreading factor defined as $SF = 256/2^k$, which may range from 256 down to 4. The SF in the uplink DPCCH is always equal to 256 corresponding to 10 bits per uplink DPCCH slot. Table 4.4 illustrates the exact number of bits in the uplink DPDCH, while Table 4.5 shows the different uplink DPCCH fields (i.e. N_{pilot} , N_{TFCI} , N_{FBI} , and N_{TPC}). The pilot patterns are given Table 4.6 and the TPC bit pattern is given in Table 4.8. Upper layers configure the slot format. The channel symbol rate and SF for all cases in Table 4.5 are 15 and 256, respectively. Channel bit and symbol rates illustrated in Tables 4.4 and Table 4.5 reflect rates before spreading.

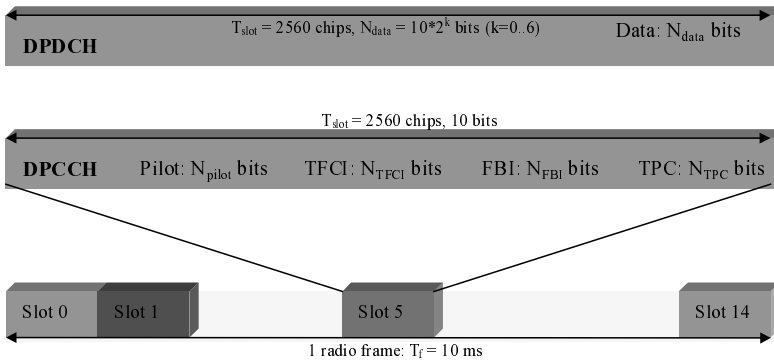


Figure 4.3 Uplink frame structure DPDCH/DPCCH.

The FBI bits (S field and D field) support the techniques requiring feedback from the UE to the UTRAN access point, including closed loop mode transmit diversity and Site Selection Diversity Transmission' (SSDT). The open SSDT signalling uses the S field and the closed loop mode transmit diversity signalling uses the D field. The S field consists of 0, 1 or 2 bits while the D field consists of 0 or 1 bit. Table 4.5 shows the total FBI field size, i.e. the N_{FBI} . Simultaneous use of SSDT power control and closed loop mode transmit diversity requires that the S field consists of 1 bit. The use of the FBI fields is described in detail in [5].

Table 4.4 DPDCH Fields

Slot format #i	Channel bit rate (kbps)	Channel symbol rate (ksps)	SF	Bits/frame	Bits/slot	N_{data}
0	15	15	256	150	10	10
1	30	30	128	300	20	20
2	60	60	64	600	40	40
3	120	120	32	1200	80	80
4	240	240	16	2400	160	160
5	480	480	8	4800	320	320
6	960	960	4	9600	640	640

There are two types of uplink dedicated physical channels; those that include TFCI (e.g. for several simultaneous services) and those that do not include TFCI (e.g. for fixed-rate services). These types are reflected by the duplicated rows of Table 4.5. It is the UTRAN that determines if a TFCI should be transmitted and it is mandatory for all UEs to support the use of TFCI in the uplink. The mapping of TFCI bits onto slots is described in [3]. In compressed mode, DPCCH slot formats with TFCI fields are changed. There are two possible compressed slot formats for each normal slot format. They are labelled A and B and the selection between them is dependent on the number of slots that are transmitted in each frame in compressed mode.

Table 4.5 DPCCH Fields

Slot format #i	Channel bit rate (kbps)	Bits/frame	Bits/slot	N_{pilot}	N_{TPC}	N_{TFCI}	N_{FBI}	Slots/frame
0	15	150	10	6	2	2	0	15
0A	15	150	10	5	2	3	0	10–14
0B	15	150	10	4	2	4	0	8–9
1	15	150	10	8	2	0	0	8–15
2	15	150	10	5	2	2	1	15
2A	15	150	10	4	2	3	1	10–14
2B	15	150	10	3	2	4	1	8–9
3	15	150	10	7	2	0	1	8–15
4	15	150	10	6	2	0	2	8–15
5	15	150	10	5	1	2	2	15
5A	15	150	10	4	1	3	2	10–14
5B	15	150	10	3	1	4	2	8–9

The pilot bit patterns are described in Tables 4.6 and 4.8. The shadowed column part of pilot bit pattern is defined as FSW, which can be used to confirm frame synchronization. (The value of the pilot bit pattern other than FSWs shall be '1'.)

Table 4.6 Pilot Bit Patterns for Uplink DPCCH with $N_{\text{pilot}} = 3, 4, 5$ and 6

Slot	$N_{\text{pilot}} = 3$			$N_{\text{pilot}} = 4$				$N_{\text{pilot}} = 5$					$N_{\text{pilot}} = 6$					
Bit #	0	1	2	0	1	2	3	0	1	2	3	4	0	1	2	3	4	5
0	1	1	1	1	1	1	1	1	1	1	1	0	1	1	1	1	1	0
1	0	0	1	1	0	0	1	0	0	1	1	0	1	0	0	1	1	0
2	0	1	1	1	0	1	1	0	1	1	0	1	1	0	1	1	0	1
3	0	0	1	1	0	0	1	0	0	1	0	0	1	0	0	1	0	0
4	1	0	1	1	1	0	1	1	0	1	0	1	1	1	0	1	0	1
5	1	1	1	1	1	1	1	1	1	1	1	0	1	1	1	1	1	0
6	1	1	1	1	1	1	1	1	1	1	0	0	1	1	1	1	0	0
7	1	0	1	1	1	0	1	1	0	1	0	0	1	1	0	1	0	0
8	0	1	1	1	0	1	1	0	1	1	1	0	1	0	1	1	1	0
9	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
10	0	1	1	1	0	1	1	0	1	1	0	1	1	0	1	1	0	1
11	1	0	1	1	1	0	1	1	0	1	1	1	1	1	0	1	1	1
12	1	0	1	1	1	0	1	1	0	1	0	0	1	1	0	1	0	0
13	0	0	1	1	0	0	1	0	0	1	1	1	1	0	0	1	1	1
14	0	0	1	1	0	0	1	0	0	1	1	1	1	0	0	1	1	1

Table 4.7 presents the relationship between the TPC bit pattern and transmitter power control command.

Table 4.7 TPC Bit Pattern

TPC bit pattern		Transmitter power control command
$N_{\text{TPC}} = 1$	$N_{\text{TPC}} = 2$	
1	11	1
0	00	0

While there is only DPCCH per radio link, several parallel DPDCHs using different channelization codes [4] can be transmitted for the multi-code operation in the uplink dedicated physical channels.

Table 4.8 Pilot Bit Patterns for Uplink DPCCH with $N_{\text{pilot}} = 7$ and 8

Slot #	$N_{\text{pilot}} = 7$							$N_{\text{pilot}} = 8$							
Bit #	0	1	2	3	4	5	6	0	1	2	3	4	5	6	7
0	1	1	1	1	1	0	1	1	1	1	1	1	1	1	0
1	1	0	0	1	1	0	1	1	0	1	0	1	1	1	0
2	1	0	1	1	0	1	1	1	0	1	1	1	0	1	1
3	1	0	0	1	0	0	1	1	0	1	0	1	0	1	0
4	1	1	0	1	0	1	1	1	1	1	0	1	0	1	1
5	1	1	1	1	1	0	1	1	1	1	1	1	1	1	0
6	1	1	1	1	0	0	1	1	1	1	1	1	0	1	0
7	1	1	0	1	0	0	1	1	1	1	0	1	0	1	0

8	1	0	1	1	1	0	1	1	0	1	1	1	1	1	0
9	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
10	1	0	1	1	0	1	1	1	0	1	1	1	0	1	1
11	1	1	0	1	1	1	1	1	1	1	0	1	1	1	1
12	1	1	0	1	0	0	1	1	1	1	0	1	0	1	0
13	1	0	0	1	1	1	1	1	0	1	0	1	1	1	1
14	1	0	0	1	1	1	1	1	0	1	0	1	1	1	1

4.3.2.1 Spreading DPCCH/DPDCH

In the uplink spreading principle of DPCCH and DPDCHs real-valued sequences of +1 and -1 represent the binary values '0' and '1', respectively. We spread the DPCCH to the chip rate by the channelization code c_c , and the n th DPDCH (or DPDCH_n) to the chip rate by the channelization code $c_{d,n}$. As illustrated in Figure 4.4, we can transmit one DPCCH and up to six parallel DPDCHs simultaneously, i.e. $1 \leq n \leq 6$ [8].

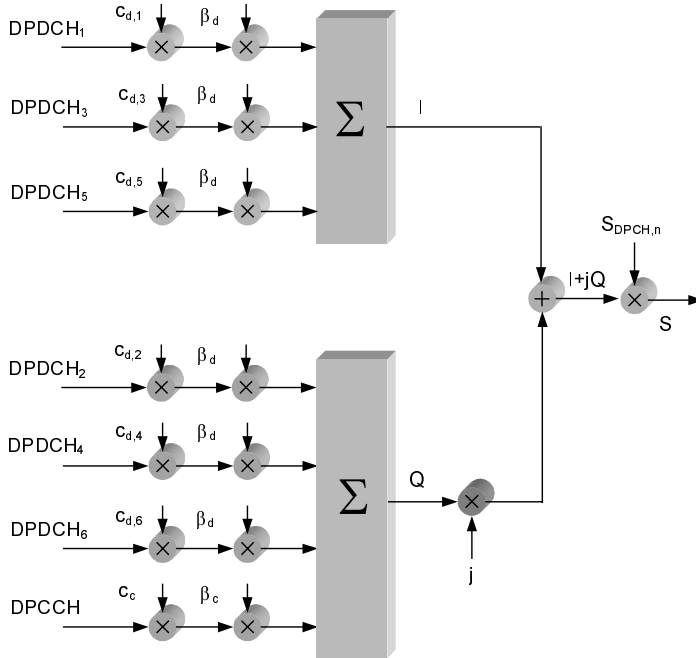


Figure 4.4 Spreading for uplink DPCCH and DPDCHs.

After channelization, gain factors β_c for DPCCH and β_d for all DPDCHs weight the real-valued spread signals, where at every instant in time, at least one of the values β_c and β_d have the amplitude 1.0. Likewise after the weighting, we sum the stream of real-valued chips on the I- and Q-branches and then treat them as a complex-valued stream of chips. After we scramble these streams by the complex-valued scrambling code $S_{\text{dpch},n}$, the scrambling code application aligns with the radio frames, i.e. the first scrambling chip corresponds to the beginning of a radio frame.

Table 4.9 illustrates quantization steps of the β -values quantized into 4 bit words. After the weighting, we sum the stream of real-valued chips on the I- and Q-branches and then treat them as a complex-valued stream of chips. After we scramble these streams by the complex-valued scrambling code $S_{\text{dpcch},n}$. The scrambling code application aligns with the radio frames, i.e. the first scrambling chip corresponds to the beginning of a radio frame.

Table 4.9 The Quantization of the Gain Parameters

Signalling values for β_c and β_d	Quantized amplitude ratios β_c and β_d
15	1.0
14	0.9333
13	0.8666
12	0.8000
11	0.7333
10	0.6667
9	0.6000
8	0.5333
7	0.4667
6	0.4000
5	0.3333
4	0.2667
3	0.2000
2	0.1333
1	0.0667
0	Switch off

4.3.3 Common Uplink Physical Channels

4.3.3.1 Physical Random Access Channel - PRACH

The PRACH carries the Random Access Channel (RACH).

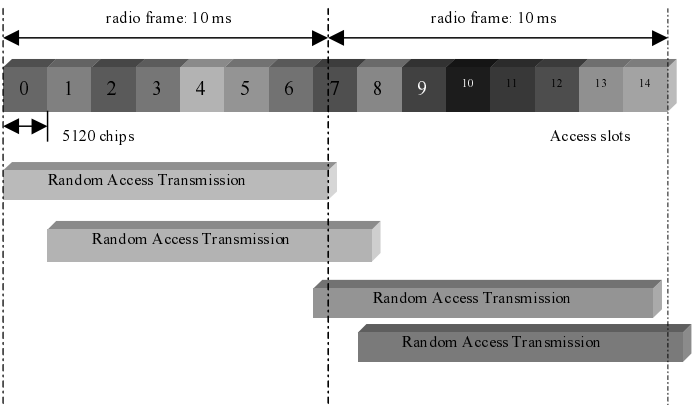


Figure 4.5 RACH access slot numbers and spacing.

4.3.3.1.1 The Random-access Transmission Structure

The random-access transmission uses a slotted ALOHA technique with fast acquisition indication. The UE can start the random-access transmission at the beginning of a number of well-defined time intervals, denoted *access slots* as illustrated in Figure 4.5. There are 15 access slots per two frames and they are spaced 5120 chips apart. The information about the type of access slots available for random-access transmission comes from the upper layers.

Figure 4.6 illustrates the random-access transmission structure, where the transmission consists of one or several *preambles* of length 4096 chips and a *message* of length 10 ms or 20 ms. Each preamble has 256 repetitions of 16 chips signature. Thus, there is a maximum of 16 available signatures, see [4] for more details.

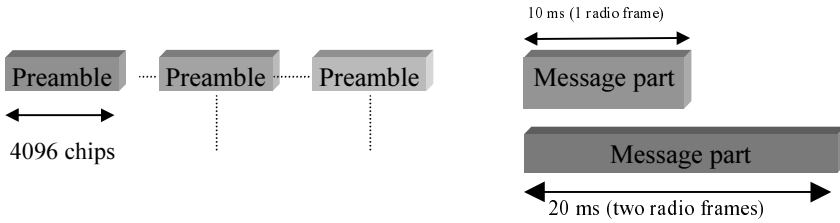


Figure 4.6 Structure of the random-access transmission.

4.3.3.1.2 The RACH Message Part

Figure 4.7 illustrates the random-access message part radio frame structure, where the 10 ms message part radio frame is split into 15 slots, each having a length $T_{\text{slot}} = 2560$ chips. Furthermore, each slot consists of two parts, i.e. a data part to which the RACH transport channel is mapped and a control part that carries Layer 1 control information; they are transmitted in parallel.

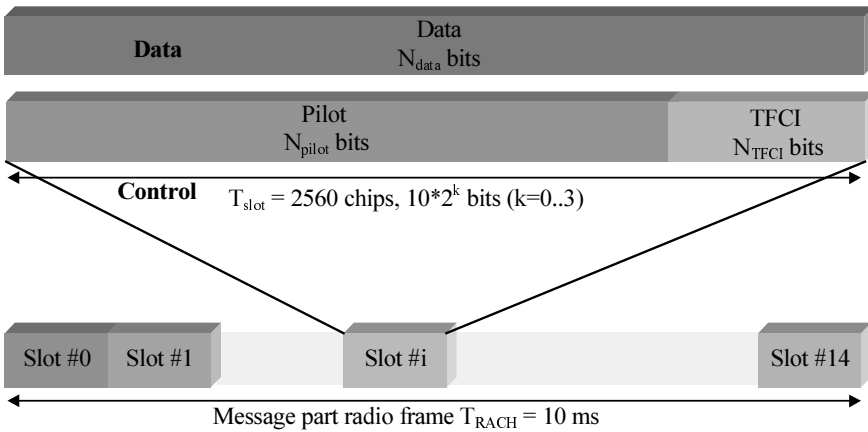


Figure 4.7 Random-access message part radio frame structure.

A 10 ms message part consists of one message part radio frame, while a 20 ms message part consists of two consecutive 10 ms message part radio frames. The message part length can be determined from the used signature and/or access slot, as configured by higher layers. Table 4.10 illustrates data and control fields of the random access message.

Table 4.10 Random Access Message Data and Control Fields

Slot format # <i>i</i>	Channel bit rate (kbps)	Channel symbol rate (ksps)	SF	Bits/frame	Bits/slot	N_{pilot}	N_{data}
0	15	15	256	150	10		10
1	30	30	128	300	20		20
2	60	60	64	600	40		40
3	120	120	32	1200	80		80
Control fields							N_{TFCI}
0	15	15	256	150	10	8	2

The data part consists of 10×2^k bits, where $k = 0, 1, 2, 3$. This corresponds to a spreading factor of 256, 128, 64, and 32 for the message data part, respectively.

The control part consists of 8 known pilot bits to support channel estimation for coherent detection and 2 TFCI bits. This corresponds to a spreading factor of 256 for the message control part. The pilot bit pattern is described in Table 4.11. The total number of TFCI bits in the random-access message is $15 \times 2 = 30$.

The TFCI of a radio frame indicates the transport format of the RACH transport channel mapped to the simultaneously transmitted message part radio frame. In the case of a 20 ms PRACH message part, the TFCI is repeated in the second radio frame.

Table 4.11 Pilot Bit Patterns for RACH Message Part with $N_{\text{pilot}} = 8$

Slot #	$N_{\text{pilot}} = 8$							
Bit #	0	1	2	3	4	5	6	7
0	1	1	1	1	1	1	1	0
1	1	0	1	0	1	1	1	0
2	1	0	1	1	1	0	1	1
3	1	0	1	0	1	0	1	0
4	1	1	1	0	1	0	1	1
5	1	1	1	1	1	1	1	0
6	1	1	1	1	1	0	1	0
7	1	1	1	0	1	0	1	0
8	1	0	1	1	1	1	1	0
9	1	1	1	1	1	1	1	1
10	1	0	1	1	1	0	1	1
11	1	1	1	0	1	1	1	1
12	1	1	1	0	1	0	1	0
13	1	0	1	0	1	1	1	1
14	1	0	1	0	1	1	1	1

4.3.3.2 Physical Common Packet Channel (PCPCH)

The Physical Common Packet Channel (PCPCH) carries the CPCH. The CPCH transmission is based on the Digital Sense Multiple Access – Collision Detection (DSMA-CD) technique with fast acquisition indication. The UE can start transmission at the beginning of a number of well-defined time-intervals, relative to the frame boundary of the received BCH of the current cell. The access slot timing and structure are identical to those defined for the RACH. Figure 4.8 illustrates the structure of the CPCH access transmission. The PCPCH access transmission consists of one or several Access Preambles [A-P] of length 4096 chips, one Collision Detection Preamble (CD-P) of length 4096 chips, a DPCCH Power Control Preamble (PC-P) which is either 0 slots or 8 slots in length, and a message of variable length $N \times 10$ ms.

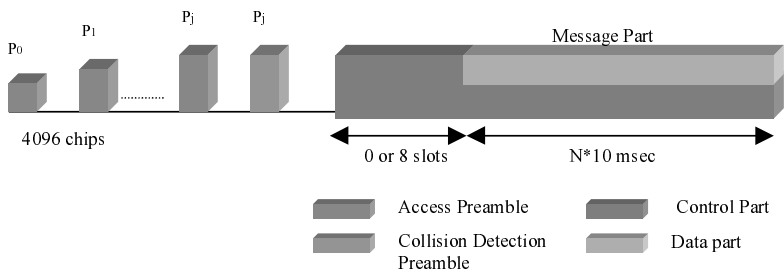


Figure 4.8 Structure of the CPCH access transmission.

4.3.3.2.1 CPCH Access –Power Control and Detection Preamble Parts

- Like in the RACH, the access CPCH preamble uses signature sequences, but the number of sequences can be lower. The scrambling codes may differ from the gold codes segment used in the RACH or could be the same scrambling code.
- Table 4.12 defines the DPCCH fields form the CPCH PC-P part. The power control preamble length parameter takes the values 0 or 8 slots, as set by the higher layers. When the power control preamble length is set to 8 slots, pilot bit patterns from slot #0 to slot #7 defined in Table 4.8 shall be used for CPCH PC-P.

Table 4.12 DPCCH Fields for CPCH Power Control Preamble Segment

Slot format #i	Chan- nel bit rate (kbps)	Channel symp. rate (ksps)	SF	Bits/ Fram e	Bits/ Slot	N_{pilot}	N_{TPC}	N_{TFCI}	N_{FBI}
0	15	15	256	150	10	6	2	2	0
1	15	15	256	150	10	8	2	0	0
2	15	15	256	150	10	5	2	2	1
3	15	15	256	150	10	7	2	0	1
4	15	15	256	150	10	6	2	0	2
5	15	15	256	150	10	5	1	2	2

- Also like in the RACH, the detection CPCH preamble uses signature sequences. However, the scrambling code set differs from the gold code segment used to form the RACH scrambling code.

4.3.3.2.2 CPCH Message Part

With similar message part structure of the RACH, each CPCH message part consists of up to $N\_Max\_frames^1$ 10 ms frames, with a 10 ms frame split into 15 slots, each having $T_{slot} = 2560$ chips length. In addition, every slot consists of a data part that carries higher layer information and a control part that carries Layer 1 control information. The data and control parts are transmitted in parallel.

The DPDCH field entries defined in Table 4.4 apply also to the data part of the CPCH message part. The control part of the CPCH message part has a spreading factor of 256, and it uses the same slot format as the control part of the CPCH PC-P. The pilot bit patterns defined in Tables 4.6 and 4.8 apply also to the pilot bit patterns of the CPCH message part.

Figure 4.9 illustrates the uplink common packet physical channel frame structure. Each frame of length 10 ms is split into 15 slots having $T_{slot} = 2560$ chips length corresponding to one power-control period.

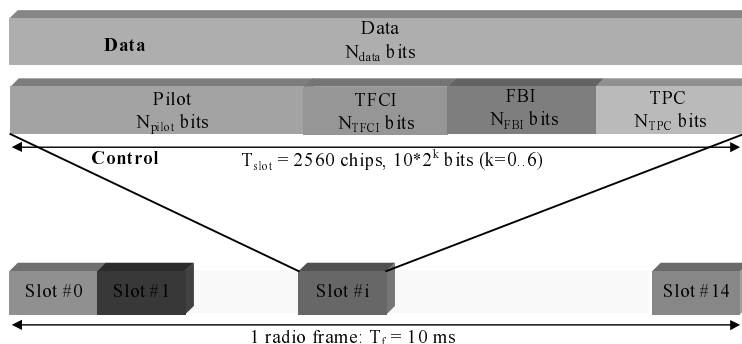


Figure 4.9 Frame structure for uplink data and control parts associated with PCPCH.

The data part consists of 10×2^k bits, where $k = 0, 1, 2, 3, 4, 5, 6$, corresponding to spreading factors of 256, 128, 64, 32, 16, 8, 4, respectively.

4.3.3.3 Spreading Common Uplink Physical Channels

4.3.3.3.1 PRACH

The PRACH preamble part consists of a complex-valued code and the message part includes the data and control parts, Figure 4.10 illustrates its spreading principle. In the message part, real-value sequences represent the binary control and data parts i.e. the binary value '0' maps to the real value +1, while the binary value '1' maps to the real

¹ N_Max_frames is a higher layer parameter.

value -1 . The channelization code c_c spreads the control part, while channelization code c_d spreads the data part.

After channelization, gain factor β_c for the control part and β_d for the data part weight the real-valued spread signals, where at least every instant in time one of the value β_c and β_d have the amplitude 1.0. Table 4.9 illustrates quantization steps of the β -values quantized into 4 bit words.

Once the weighting takes place, we treat the stream of real-valued chips on the I- and Q-branches as a complex-valued stream of chips. Then the complex-valued scrambling code $S_{r\text{-msg},n}$ scrambles this complex-valued signal. The 10 ms scrambling code application aligns with the 10 ms message part radio frames, i.e. the first scrambling chip corresponds to the beginning of a message part radio frame [8].

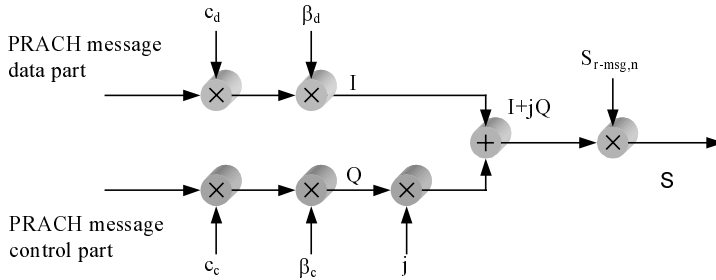


Figure 4.10 Spreading of PRACH message part.

4.3.3.3.2 PCPCH

As in the PRACH, the PCPCH preamble part consists of a complex-valued code, and the PCPCH message part includes data and control parts, Figure 4.11 illustrates its spreading principle.

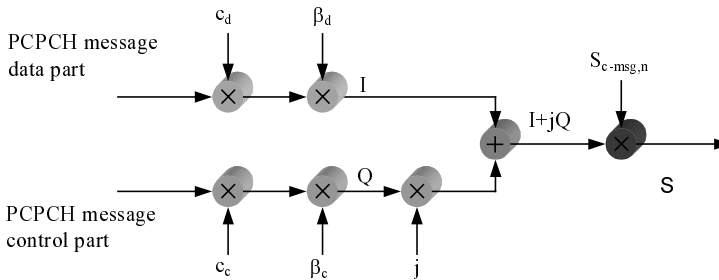


Figure 4.11 Spreading of PCPCH message part.

In the message part, real-value sequences represent the binary control and data parts, i.e. the binary value '0' maps to the real value $+1$, while the binary value '1' maps to the

real value -1 . The channelization code c_c spreads the control part, while channelization code c_d spreads the data part. Channelization and weighting follows the same pattern as in the PRACH.

4.3.4 Uplink Channelization Codes

The Orthogonal Variable Spreading Factor (OVSF) channelization codes preserve orthogonality between a user's different physical channels. A tree illustrated in Figure 4.12 defines these codes.

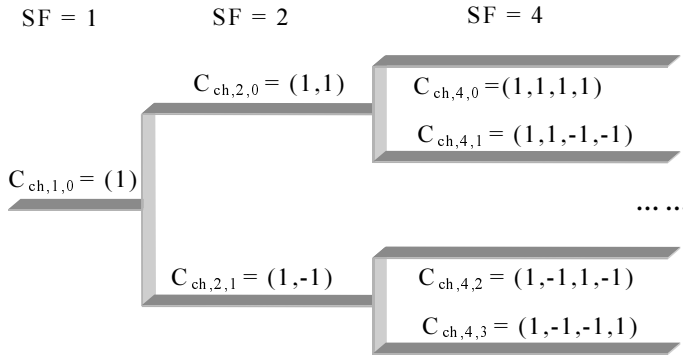


Figure 4.12 Orthogonal Variable Spreading Factor (OVSF) code-tree generation.

The channelization codes in the OVSF tree have a unique description as $C_{ch,SF,k}$, where SF is the spreading factor of the code and k is the code number, $0 \leq k \leq SF - 1$. Each level in the code tree defines channelization codes of length SF, corresponding to a spreading factor of SF. From [8] the generation method for the channelization code is defined as:

$$C_{ch,1,0} = 1, \quad (4.1)$$

$$\begin{bmatrix} C_{ch,2,0} \\ C_{ch,2,1} \end{bmatrix} = \begin{bmatrix} C_{ch,1,0} & C_{ch,1,0} \\ C_{ch,1,0} & -C_{ch,1,0} \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix}, \quad (4.2)$$

$$\begin{bmatrix} C_{ch,2^{(n+1)},0} \\ C_{ch,2^{(n+1)},1} \\ C_{ch,2^{(n+1)},2} \\ C_{ch,2^{(n+1)},3} \\ \vdots \\ C_{ch,2^{(n+1)},2^{(n+1)}-2} \\ C_{ch,2^{(n+1)},2^{(n+1)}-1} \end{bmatrix} = \begin{bmatrix} C_{ch,2^n,0} & C_{ch,2^n,0} \\ C_{ch,2^n,0} & -C_{ch,2^n,0} \\ C_{ch,2^n,1} & C_{ch,2^n,1} \\ C_{ch,2^n,1} & -C_{ch,2^n,1} \\ \vdots & \vdots \\ C_{ch,2^n,2^n-2} & C_{ch,2^n,2^n-2} \\ C_{ch,2^n,2^n-1} & -C_{ch,2^n,2^n-1} \end{bmatrix}. \quad (4.3)$$

The leftmost value in each channelization code word corresponds to the chip transmitted first in time.

4.3.4.1 DPCCH/DPDCH Code Allocation

According to [8] for the DPCCH and DPDCHs the following applies: the DPCCH is always a code $c_c = C_{ch,256,0}$ as spread; and when we transmit only one DPDCH, the DPDCH₁ has code $c_{d,1} = C_{ch,SF,k}$ as spread, where SF is the spreading factor of DPDCH₁ and $k = SF/4$. However, when we transmit more than one DPDCH, all DPDCHs have spreading factors equal to 4. The DPDCH_{*n*} is spread by the code $c_{d,n} = C_{ch,4,k}$, where $k = 1$ if $n \in \{1, 2\}$, $k = 3$ if $n \in \{3, 4\}$, and $k = 2$ if $n \in \{5, 6\}$.

4.3.4.2 PRACH Message Part Code Allocation

The preamble signature s , $0 \leq s \leq 15$, points to one of the 16 nodes in the code tree that corresponds to channelization codes of length 16. To spread the message part we use the sub-tree below a specified node, while to spread the control part we use the channelization code c_c with SF = 256 in the lowest branch of the sub-tree, i.e. $c_c = C_{ch,256,m}$ where $m = 16 \times s + 15$. The data part uses any of the channelization codes from spreading factor 32 to 256 in the upper-most branch of the sub-tree. More exactly, we spread the data part by channelization code $c_d = C_{ch,SF,m}$, SF is the data part spreading factor and $m = SF \times s/16$ [8].

4.3.4.3 PCPCH Message Part Code Allocation

For the control part and data part the following applies: the control part has always code $c_c = C_{ch,256,0}$ as spread; and the data part has code $c_d = C_{ch,SF,k}$ as spread, where SF is the spreading factor of the data part and $k = SF/4$. The data part may use the code from spreading factor 4 to 256, and a UE can increase SF during a message transmission on frame by frame basis [8].

Finally, the same channelization code of the message control part applies to the PCPCH power control preamble.

4.3.5 Uplink Scrambling Codes

All uplink physical channels use a complex-valued scrambling code. While either long or short scrambling codes apply to the DPCCH/DPDCH, to the PRACH and PCPCH message parts only long scrambling codes apply. Higher layers assign the 2^{24} long and 2^{24} short uplink scrambling codes.

4.3.5.1 Long Scrambling Sequence

The long scrambling sequences $c_{long,1,n}$ and $c_{long,2,n}$ result from the position wise modulo 2 sum of 38 400 chip segments and two binary m sequences generated by means of two generator polynomials of degree 25. The 1st m sequences, i.e. x comes from the primitive (over GF (2)) polynomial $X^{25} + X^3 + 1$; while the 2nd m sequences, i.e. y comes from the polynomial $X^{25} + X^3 + X^2 + X + 1$. The resulting sequences constitute a seg-

ment set of gold sequences, where the sequence $c_{\text{long},2,n}$ is a 16 777 232 chip shifted version of the sequence $c_{\text{long},1,n}$ [8]. Figure 4.13 illustrates a configuration of long uplink scrambling sequence generator.

For completeness in the following we include an extract of the long scrambling sequence definition from [8]. Where $n_{23} \dots n_0 = 24$ bit binary representation of the scrambling sequence number n with n_0 as the least significant bit, x sequence which depends on the chosen scrambling sequence number n is denoted x_n , $x_n(i)$ and $y(i)$ denote the i th symbol of the sequence x_n and y , respectively. Then m sequences x_n and y can be defined as:

$$x_n(0) = n_0, \quad x_n(1) = n_1, \dots, x_n(22) = n_{22}, \quad x_n(23) = n_{23}, \quad x_n(24) = 1, \quad (4.4)$$

$$y(0) = y(1) = \dots = y(23) = y(24) = 1, \quad (4.5)$$

where $x_n(0)$ and $y(0)$ are the initial conditions.

The recursive definition of subsequent symbols include:

$$x_n(i+25) = x_n(i+3) + x_n(i) \bmod 2, \quad i = 0, \dots, 2^{25} - 27, \quad (4.6)$$

$$y(i+25) = y(i+3) + y(i+2) + y(i+1) + y(i) \bmod 2, \quad i = 0, \dots, 2^{25} - 27. \quad (4.7)$$

The binary gold sequence z_n can be defined as:

$$z_n(i) = x_n(i) + y(i) \bmod 2, \quad i = 0, 1, 2, \dots, 2^{25} - 2, \quad (4.8)$$

then the real valued gold sequence Z_n is defined by:

$$Z_n(i) = \begin{cases} +1 & \text{if } z_n(i) = 0 \\ -1 & \text{if } z_n(i) = 1 \end{cases} \quad \text{for } i = 0, 1, \dots, 2^{25} - 2. \quad (4.9)$$

Now, the real-valued long scrambling sequences $c_{\text{long},1,n}$ and $c_{\text{long},2,n}$ are defined as:

$$c_{\text{long},1,n}(i) = Z_n(i), \quad i = 0, 1, 2, \dots, 2^{25} - 2 \quad (4.10)$$

and

$$c_{\text{long},2,n}(i) = Z_n((i + 16777232) \bmod (2^{25} - 1)), \quad i = 0, 1, 2, \dots, 2^{25} - 2. \quad (4.11)$$

Finally, we define the complex-valued long scrambling sequence $C_{\text{long},n}$, as

$$C_{\text{long},n}(i) = c_{\text{long},1,n}(i) \left(1 + j(-1)^i c_{\text{long},2,n} \left(2 \lfloor i/2 \rfloor \right) \right), \quad (4.12)$$

where $i = 0, 1, \dots, 2^{25} - 2$ and $\lfloor \cdot \rfloor$ denotes rounding to the nearest lower integer.

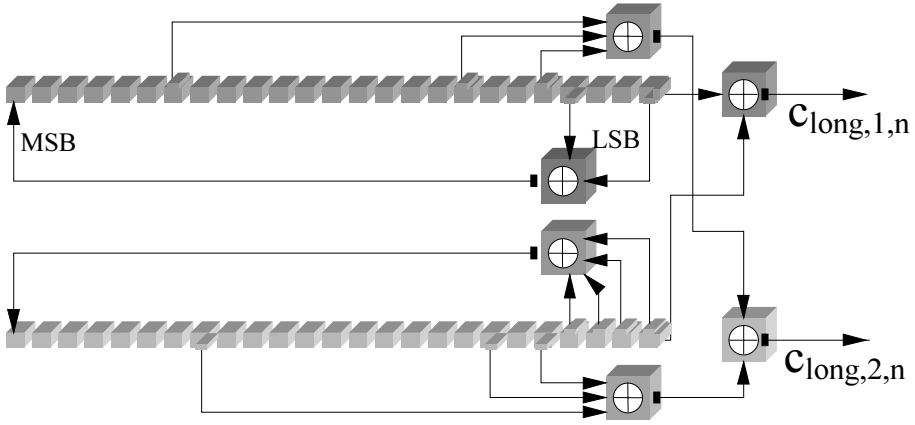


Figure 4.13 Configuration of the uplink long scrambling sequence generator.

4.3.5.2 Short Scrambling Sequence

The short scrambling sequences $c_{\text{short},1,n}(i)$ and $c_{\text{short},2,n}(i)$ originate from a family sequence of periodically extended S(2) codes, where $n_{23}n_{22}\dots n_0 = 24$ bit binary representation of the code number n . We obtain the n th quaternary S(2) sequence $z_n(i)$, $0 \leq n \leq 1677721$ by modulo 4 addition of three sequences, a quaternary sequence $a(i)$ and two binary sequences $b(i)$ and $d(i)$, where the initial loading of the three sequences comes from the code number n . The sequence $z_n(i)$ of length 255 results from the following relation:

$$z_n(i) = a(i) + 2b(i) + 2d(i) \bmod 4, \quad i = 0, 1, \dots, 254, \quad (4.13)$$

where we obtain the quaternary sequence $a(i)$ recursively through the polynomial $g_0(x) = x^8 + x^5 + 3x^3 + x^2 + 2x + 1$ as

$$a(0) = 2n_0 + 1 \bmod 4, \quad (4.14)$$

$$a(i) = 2n_i \bmod 4, \quad i = 1, 2, \dots, 7, \quad (4.15)$$

$$a(i) = 3a(i-3) + a(i-5) + 3a(i-6) + 2a(i-7) + 3a(i-8) \bmod 4, \quad i = 8, 9, \dots, 254, \quad (4.16)$$

and the binary sequence $b(i)$ comes also recursively from the polynomial $g_1(x) = x^8 + x^7 + x^5 + x + 1$ as

$$b(i) = n_{8+i} \bmod 2, \quad i = 0, 1, \dots, 7, \quad (4.17)$$

$$b(i) = b(i-1) + b(i-3) + b(i-7) + b(i-8) \bmod 2, \quad i = 8, 9, \dots, 254, \quad (4.18)$$

and the binary sequence $d(i)$ is again generated recursively by the polynomial $g_2(x) = x^8 + x^7 + x^5 + x^4 + 1$ as

$$d(i) = n_{16+i} \bmod 2, \quad i = 0, 1, \dots, 7, \quad (4.19)$$

$$d(i) = d(i-1) + d(i-3) + d(i-4) + d(i-8) \bmod 2, \quad i = 8, 9, \dots, 254. \quad (4.20)$$

We extend the sequence $z_n(i)$ to length 256 chips by setting $z_n(255) = z_n(0)$.

Table 4.13 defines the mapping from $z_n(i)$ to the real-valued binary sequences $c_{\text{short},1,n}(i)$ and $c_{\text{short},2,n}(i)$, $i = 0, 1, \dots, 255$.

Table 4.13 Mapping from $z_n(i)$ to $c_{\text{short},1,n}(i)$ and $c_{\text{short},2,n}(i)$, $i = 0, 1, \dots, 255$

$z_n(i)$	$c_{\text{short},1,n}(i)$	$c_{\text{short},2,n}(i)$
0	+1	+1
1	-1	+1
2	-1	-1
3	+1	-1

Finally, we define the complex-valued short scrambling sequence $C_{\text{short},n}$, as:

$$C_{\text{short},n}(i) = c_{\text{short},1,n}(i \bmod 256) \left(1 + j(-1)^i c_{\text{short},2,n} \left(2 \lfloor (i \bmod 256) / 2 \rfloor \right) \right). \quad (4.21)$$

Figure 4.14 illustrates an implementation of the short scrambling sequence generator for the 255 chip sequence extension by one chip.

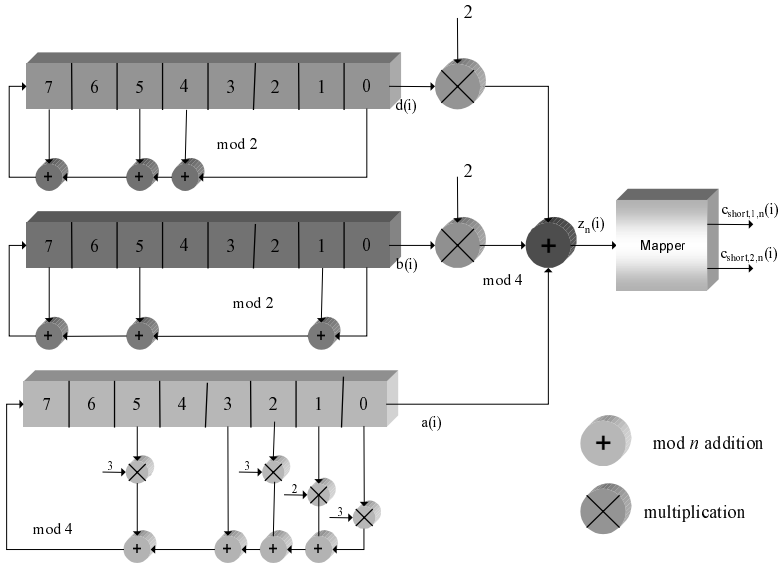


Figure 4.14 255 chip sequence uplink short scrambling sequence generator.

4.3.5.3 Scrambling Codes in Uplink Dedicated Physical Channels

The uplink DPCCCH/DPDCH may use either long or short scrambling codes with different constituent codes in each case.

From [8], when using long scrambling codes we define the n th uplink DPCCCH/DPDCH scrambling code denoted $S_{\text{dpch},n}$, as

$$S_{\text{dpch},n}(i) = C_{\text{long},n}(i), \quad i = 0, 1, \dots, 38399, \quad (4.22)$$

where the lowest index corresponds to the chip transmitted first in time and Section 4.3.5.1 defines $C_{\text{long},n}$. Likewise, when using short scrambling codes we define the n th uplink DPCCCH/DPDCH scrambling code denoted $S_{\text{dpch},n}$, as

$$S_{\text{dpch},n}(i) = C_{\text{short},n}(i), \quad i = 0, 1, \dots, 38399, \quad (4.23)$$

where the lowest index corresponds to the chip transmitted first in time and Section 4.3.5.2 defines $C_{\text{short},n}$.

4.3.5.4 PRACH and PCPCH Message Part Scrambling Code

The PRACH message part uses 10 ms long scrambling code, and there are 8192 possible PRACH scrambling codes. From [8] we define the n th PRACH message part scrambling code, denoted $S_{\text{r-msg},n}$, where $n = 0, 1, \dots, 8191$, based on the long scrambling sequence as

$$S_{\text{r-msg},n}(i) = C_{\text{long},n}(i + 4096), \quad i = 0, 1, \dots, 38399, \quad (4.24)$$

where the lowest index corresponds to the chip transmitted first in time and Section 4.3.5.1 defines $C_{\text{long},n}$.

The message part scrambling code has a one-to-one correspondence to the scrambling code utilized in the preamble part. For one PRACH, we use the same code number in both scrambling codes, i.e. if the PRACH preamble scrambling code uses $S_{\text{r-pre},m}$ then the PRACH message part scrambling code uses $S_{\text{r-msg},m}$, where the number m is the same for both codes [8].

As in PRACH, PCPCH uses 10 ms long scrambling codes in the message part. They are cell-specific and each scrambling code has a one-to-one correspondence to the signature sequence and the access sub-channel utilized by the access preamble part. Both long and short scrambling codes may scramble the PCPCH message part. We define up to 64 uplink-scrambling codes per cell and up to 32768 different PCPCH scrambling codes in the system. For the long scrambling sequence we define the n th PCPCH message part scrambling code ($S_{\text{c-msg},n}$, $n = 8192, 8193, \dots, 40959$) as:

$$S_{\text{c-msg},n}(i) = C_{\text{long},n}(i), \quad i = 0, 1, \dots, 38399, \quad (4.25)$$

where the lowest index corresponds to the chip transmitted first in time and Section 4.3.5.1 defines $C_{\text{long},n}$. For the short scrambling codes we have

$$S_{\text{c-msg},n}(i) = C_{\text{short},n}(i), \quad i = 0, 1, \dots, 38399. \quad (4.26)$$

A total of 512 groups each containing 64 codes comprise the 32768 PCPCH scrambling codes. The group of PCPCH preamble scrambling codes in a cell and the primary

scrambling code used in the downlink of the cell match one-to-one. $S_{c\text{-msg},n}$ as defined in the preceding paragraphs with $n = 64 \times m + k + 8176$, is the k th PCPCH scrambling code within the cell with downlink primary scrambling code m , where $k = 16, 17, \dots, 79$ and $m = 0, 1, 2, \dots, 511$ [8].

4.3.5.5 Scrambling Code in the PCPCH Power Control Preamble

The PCPCH power control preamble uses the same scrambling code as the PCPCH message part (Section 4.3.2.1), where the phase of the scrambling code is such that the end of the code aligns with the frame boundary at the end of the power control preamble.

4.3.5.6 PRACH Preamble Codes

Complex valued sequence constitutes the random access preamble code $C_{\text{pre},n}$. It originates from a preamble scrambling code $S_{\text{r-pre},n}$ and a preamble signature $C_{\text{sig},s}$ as:

$$C_{\text{pre},n,s}(k) = S_{\text{r-pre},n}(k) \times C_{\text{sig},s}(k) \times \exp \left[j \left(\frac{\pi}{4} + \frac{\pi}{2} k \right) \right], \quad k = 0, 1, 2, 3, \dots, 4095, \quad (4.27)$$

where $k = 0$ corresponds to the chip transmitted first in time and we define $S_{\text{r-pre},n}$ and $C_{\text{sig},s}$ next. A total of 8192 PRACH preamble part scrambling codes result from the long scrambling sequences. We define the n th preamble scrambling code, $n = 0, 1, \dots, 8191$, as:

$$S_{\text{r-pre},n}(i) = c_{\text{long},l,n}(i), \quad i = 0, 1, \dots, 4095, \quad (4.28)$$

where Section 4.3.5.1 defines the sequence $c_{\text{long},l,n}$.

As for the PCPCH, we divide the 8192 PRACH preamble scrambling codes in 512 groups with 16 codes in each. And again as in the earlier scrambling codes, we have one-to-one correspondence between the group of PRACH preamble scrambling codes in a cell and the primary scrambling code used in the downlink of the cell. $S_{\text{r-pre},n}(i)$ as defined in equation (4.28) with $n = 16 \times m + k$, represents the k th PRACH preamble scrambling code within the cell with downlink primary scrambling code m , $k = 0, 1, 2, \dots, 15$ and $m = 0, 1, 2, \dots, 511$.

The preamble signature s has 256 repetitions of the signature $P_s(n)$ from the set of 16 Hadamard codes of length 16 (Table 4.14), where $n = 0, \dots, 15$. The specifications in [8] define it as:

$$C_{\text{sig},s}(i) = P_s(i \bmod 16), \quad i = 0, 1, \dots, 4095. \quad (4.29)$$

Table 4.14 Preamble Signatures

Preamble signature	Value of n															
	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
$P_0(n)$	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1
$P_1(n)$	1	-1	1	-1	1	-1	1	-1	1	-1	1	-1	1	-1	1	-1
$P_2(n)$	1	1	-1	-1	1	1	-1	-1	1	1	-1	-1	1	1	-1	-1
$P_3(n)$	1	-1	-1	1	1	-1	-1	1	1	-1	-1	1	1	-1	-1	1
$P_4(n)$	1	1	1	1	-1	-1	-1	-1	1	1	1	1	-1	-1	-1	-1
$P_5(n)$	1	-1	1	-1	-1	1	-1	1	1	-1	1	-1	-1	1	-1	1
$P_6(n)$	1	1	-1	-1	-1	-1	1	1	1	1	-1	-1	-1	-1	1	1
$P_7(n)$	1	-1	-1	1	-1	1	1	-1	1	-1	-1	1	-1	1	1	-1
$P_8(n)$	1	1	1	1	1	1	1	1	-1	-1	-1	-1	-1	-1	-1	-1
$P_9(n)$	1	-1	1	-1	1	-1	1	-1	-1	1	-1	1	-1	1	-1	1
$P_{10}(n)$	1	1	-1	-1	1	1	-1	-1	-1	-1	1	1	-1	-1	1	1
$P_{11}(n)$	1	-1	-1	1	1	-1	-1	1	-1	1	1	-1	-1	1	1	-1
$P_{12}(n)$	1	1	1	1	-1	-1	-1	-1	-1	-1	-1	-1	1	1	1	1
$P_{13}(n)$	1	-1	1	-1	-1	1	-1	1	-1	1	-1	1	1	-1	1	-1
$P_{14}(n)$	1	1	-1	-1	-1	-1	1	1	-1	-1	1	1	1	1	-1	-1
$P_{15}(n)$	1	-1	-1	1	-1	1	1	-1	-1	1	1	-1	1	-1	-1	1

4.3.5.7 PCPCH Preamble Codes

Like in PRACH, PCPCH access preamble codes $C_{c\text{-acc},n,s}$, have complex value sequences. We define them from the preamble scrambling codes $S_{c\text{-acc},n}$ and a preamble signature $C_{\text{sig},s}$ as:

$$C_{c\text{-acc},n,s}(k) = S_{c\text{-acc},n}(k) \times C_{\text{sig},s}(k) \times \exp \left[j \left(\frac{\pi}{4} + \frac{\pi}{2} k \right) \right], \quad k = 0, 1, 2, 3, \dots, 4095, \quad (4.30)$$

where $S_{c\text{-acc},n}$ and $C_{\text{sig},s}$ are defined in the sequel.

Code generation takes place as in PRACH, resulting in 32768 PCPCH scrambling codes in total. We define n th PCPCH access preamble scrambling code, where $n = 8192, 8193, \dots, 40959$, as:

$$S_{c\text{-acc},n}(i) = c_{\text{long},1,n}(i), \quad i = 0, 1, \dots, 4095, \quad (4.31)$$

where the sequence Section 4.3.5.1 defines $c_{\text{long},1,n}$.

When PRACH and PCPCH share access resources, the scrambling codes applied in PRACH preamble apply also to PCPCH preamble; and as in the PRACH part we divide the 32768 PCPCH preamble scrambling codes into 512 groups with 64 codes in each group. There exists a one-to-one correspondence between the group of PCPCH access preamble scrambling codes in a cell and the primary scrambling code used in the downlink of the cell. The k th PCPCH scrambling code within the cell with downlink primary scrambling code m , $k = 16, 17, \dots, 79$ and $m = 0, 1, 2, \dots, 511$, corresponds to $S_{c\text{-acc},n}$ as defined in Section 4.3.5.7 with $n = 64 \times m + k + 8176$.

When PCPCH and PRACH share scrambling code resources and the index k is less than 16, the corresponding PRACH formulae apply. Otherwise, if the index k is greater than or equal to 16, the formula in this section applies. The CPCH-access burst preamble part carries one of the 16 different orthogonal complex signatures identical to the ones used by the preamble part of the random-access burst [8].

4.3.5.7.1 Collision Detection(CD) Preamble

As in PRACH, the PCPCH CD preamble codes $C_{c-cd,n,s}$ have complex valued sequences. We define these preamble codes from the preamble scrambling codes $S_{c-cd,n}$ and a preamble signature $C_{sig,s}$ as:

$$C_{c-cd,n,s}(k) = S_{c-cd,n}(k) \times C_{sig,s}(k) \times \exp \left[j \left(\frac{\pi}{4} + \frac{\pi}{2} k \right) \right], \quad k = 0, 1, 2, 3, \dots, 4095, \quad (4.32)$$

where we define $S_{c-cd,n}$ in the sequel and $C_{sig,s}$ in Section 4.3.5.6.

The 32768 PCPCH CD preamble-scrambling code originates from the same scrambling code utilized in the CPCH access preamble. We define the n th PCPCH CD access preamble scrambling code, where $n = 8192, 8193, \dots, 40959$, as:

$$S_{c-cd,n}(i) = c_{long,1,n}(i), \quad i = 0, 1, \dots, 4095, \quad (4.33)$$

where Section 4.3.5.1 defines the sequence $c_{long,1,n}$.

When RACH and CPCH share scrambling code resources, RACH preamble scrambling codes will also apply to the CPCH CD preamble. As in the cases above, we divide the 32768 PCPCH scrambling codes into 512 groups with 64 codes each. There exists also a one-to-one correspondence between the group of PCPCH CD preamble scrambling codes in a cell and the primary scrambling code used in the downlink of the cell. The k th PCPCH scrambling code within the cell with downlink primary scrambling code m , $k = 16, 17, \dots, 79$ and $m = 0, 1, 2, \dots, 511$, corresponds to $S_{c-cd,n}$ as defined in equation (4.33) with $n = 64 \times m + k + 8176$.

When PCPCH and PRACH share scrambling code resources, and the index k is less than 16 the corresponding PRACH formula applies. Otherwise, when the index k is greater than or equal to 16, the preceding formulae apply. The CD preamble part of the CPCH access burst carries one of 16 different orthogonal complex signatures identical to the ones utilized by the preamble part of the random access burst [8].

4.3.6 Uplink Power Control Procedure

The FDD mode has unique procedures compared to the TDD. These include fast power control and soft handover procedures. Other procedures are synchronization and random access.

4.3.6.1 PRACH and DPCCH/DPDCH Power Control

The uplink PRACH message part applies gain factors to manage the control/data part relative power similar to the uplink dedicated physical channels. Thus, power control steps in the dedicated physical channels apply also to the RACH message part, with the differences that [11]:

- β_c is the gain factor for the control part (similar to DPCCH);
- β_d is the gain factor for the data part (similar to DPDCH);
- no inner or fast loop power control is performed, but open loop power control.

Before the uplink power control procedure simultaneously controls the power of a DPCCH and its corresponding DPDCHs when present, high layers set the initial uplink DPCCH transmit power. The network determines this relative transmit power offset between DPCCH and DPDCHs using the gain factors signalled to the UE. The inner or fast power control loop operation adjusts the power of the DPCCH and DPDCHs in steps of 1 dB or multiples of one and smaller steps through emulation at 1500 Hz command rate. The DPCCH uplink transmit power takes place immediately before the start of its pilot field. This change occurs with respect to its previous value derived by the UE, i.e. Δ_{DPCCH} (in dB). The previous DPCCH power value corresponds to the one used in the previous slot, except in the event of an interruption in transmission due to the use of compressed mode. In the latter case, the previous value corresponds to the one used in the last slot before the transmission gap. While in power control, the UE transmit power will not exceed a maximum allowed value, i.e. the lowest out of the terminal maximum output power and the one set by higher layer signalling. If the UE transmit power falls below the required minimum output power and the derived value of $\Delta_{\text{DPCCH}} < 0$, the UE may reduce the Δ_{DPCCH} magnitude [11].

4.3.6.1.1 The transmit power control function

The uplink inner-loop power control adjusts the UE transmit power to keep the received uplink signal-to-interference ratio (SIR) at a given target, i.e. $\text{SIR}_{\text{target}}$. The serving cells in the active set estimate signal-to-interference ratio (SIR_{est}) of the received uplink DPCH. Then they generate TPC commands and transmit them once per slot according to the following rules: if $\text{SIR}_{\text{est}} > \text{SIR}_{\text{target}}$ then the TPC command enables transmission of '0', otherwise if $\text{SIR}_{\text{est}} < \text{SIR}_{\text{target}}$ then the TPC command enables transmission of '1'. Upon receipt of one or more TPC commands in a slot, the UE derives a single TPC command, TPC_cmd , for each slot, i.e. it combines multiple TPC commands if more than one is received in a slot. The UTRAN uses two algorithms supported by the UE² to realize a TPC_cmd .

Algorithm 1

1. UE not in soft handover, $\rightarrow$ each slots receives only one TPC command, then:
 - If the received $\text{TPC\_cmd} = 0$ then TPC_cmd for that slot = -1.

² The step size Δ_{TPC} is a UE specific parameter, under UTRAN control, which can have values 1 dB or 2 dB.

- If the received $\text{TPC\_cmd} = 1$, then TPC_cmd for that slot = 1.
2. UE is in soft handover, $\rightarrow$ each slot may receive multiple TPC commands from different cells in the active set. In receiver diversity (i.e. softer handover), the UTRAN transmits the same command in all the serving cells the UE is in softer handover with, and the TPC commands known to be the same get combined into one TPC command (see more details in [11]).

Algorithm 2³

1. UE is not in soft handover, $\rightarrow$ each slot receives only one TPC command and the UE processes received TPC commands on a 5-slot cycle. The non-overlapping sets of 5 slots align to the frame boundaries. The TPC_cmd logic is as follows:
 - The first 4 slots of a set have $\text{TPC\_cmd} = 0$.
 - In the fifth slot of a set, the UE uses hard decisions on each of the 5 received TPC commands as follows [11]:
 - If all 5 hard decisions within a set are 1 then $\text{TPC\_cmd} = 1$ in the 5th slot.
 - If all 5 hard decisions within a set are 0 then $\text{TPC\_cmd} = -1$ in the 5th slot.
 - Otherwise, $\text{TPC\_cmd} = 0$ in the 5th slot.
2. UE is in soft handover, $\rightarrow$ each slot receives multiple TPC commands from different cells in the active set. UE is in soft handover, $\rightarrow$ each slot may receive multiple TPC commands from different cells in the active set. In receiver diversity (i.e. softer handover), the UTRAN transmits the same command in all the serving cells the UE is in softer handover with, and the TPC commands known to be the same get combined into one TPC command (see more details in [11]).

After deriving the combined TPC command TPC_cmd using one of the two supported algorithms, the UE adjusts the transmit power of the uplink DPCH with a step of Δ_{DPCH} (in dB), which is given by $\Delta_{\text{DPCH}} = \Delta_{\text{TPC}} \times \text{TPC\_cmd}$.

Out of sync handling

The UE shuts its transmitter off when the UE estimates the DPCH quality over the last 200 ms period to be worse than a threshold Q_{out} . This criterion never occurs during the first 200 ms of the dedicated channel's existence. The UE can turn its transmitter on when the UE estimates the DPCH quality over the last 200 ms period to be better than a threshold Q_{in} . This criterion always occurs during the first 200 ms of the dedicated channel's existence. At the transmission resumption the power of the DPCH shall remain the same as when the UE transmitter went off [12].

³ Allows emulation of smaller step sizes than the minimum power control step or to turn off uplink power control.

4.3.6.2 Compressed Mode Power Control

The compressed mode, which has compressed frames containing transmission gaps, uses the same transmit power control function outlined in the preceding section, but with additional features aiming for fastest recovery of the signal-to-interference ratio (SIR) close to the target SIR after each transmission gap. In this mode, compressed frames may exist either in the uplink or the downlink or both. In the first case, the DPDCH(s) and DPCCH uplink transmissions stop during the gaps. In the 2nd case, if the gaps cause the absence of downlink TPC commands, the corresponding TPC_cmd derived by the UE goes to zero.

A transmit power change of the uplink DPCCH compensates the variation in the total pilot energy in both compressed and non-compressed frames in the uplink DPCCH due the different number of pilot bits per slot. Thus, at the start of each slot the UE derives a power offset Δ_{PILOT} value. The compensation uses the value in the most recently transmitted slot; Δ_{PILOT} (in dB) follows:

$$\Delta_{\text{PILOT}} = 10 \log_{10} (N_{\text{pilot,prev}} / N_{\text{pilot,curr}}), \quad (4.34)$$

where $N_{\text{pilot,prev}}$ is the number of pilot bits in the most recently transmitted slot, and $N_{\text{pilot,curr}}$ is the number of pilot bits in the current slot. If no compensation takes place during transmission gaps in the downlink, $\Delta_{\text{PILOT}} = \text{zero}$. Furthermore, during compressed mode the UE will adjust the transmit power of the uplink DPCCH with a step of Δ_{DPCCH} (in dB) as follows:

$$\Delta_{\text{DPCCH}} = \Delta_{\text{TPC}} \times \text{TPC\_cmd} + \Delta_{\text{PILOT}}. \quad (4.35)$$

The latter may not occur if otherwise specified. After an uplink transmission gap, the UE applies a change in the transmit power of the uplink DPCCH by an amount Δ_{DPCCH} (in dB) at the beginning the 1st slot, with respect to the uplink DPCCH power in the most recently transmitted uplink slot, where:

$$\Delta_{\text{DPCCH}} = \Delta_{\text{RESUME}} + \Delta_{\text{PILOT}}. \quad (4.36)$$

The UE determines the Δ_{RESUME} value (in dB) according to the Initial Transmit Power (ITP) mode, which is a UE specific parameter signalled by the network with other compressed mode parameters. Table 4.15 summarizes the ITP mode. If a downlink TPC command is transmitted in the first slot of a downlink transmission gap, then $\delta_{\text{last}} = \delta_i$ computed in the first slot of the downlink transmission gap. Otherwise $\delta_{\text{last}} = \delta_i$ computed in the last slot before the downlink transmission gap. δ_i will be updated according to the following recursive relation:

$$\begin{aligned} \delta_i &= 0.9375\delta_{i-1} - 0.96875\text{TPC\_cmd}_i\Delta_{\text{TPC}}, \\ \delta_{i-1} &= \delta_i. \end{aligned} \quad (4.37)$$

δ_{i-1} is the value of δ_i computed for the previous slot. $\delta_{i-1} = 0$ when we activate the uplink DPCCH, and also at the end of the first slot after each downlink transmission gap. This relation gets executed in all slots with simultaneous uplink and downlink DPCCH

transmission, and in the first slot of a downlink transmission gap if a downlink TPC command is transmitted in that slot. TPC_cmd_i is the most recent power control command derived by the UE [11].

After a transmission gap in either the uplink or the downlink, there exists a recovery period following resumption of simultaneous uplink and downlink DPCCCH transmission. This period has a Recovery Period Length (RPL) and is expressed as a number of slots. The RPL is equal to the minimum value out of the transmission gap length and 7 slots. Table 4.15 illustrates the two recovery period modes for the power control algorithm, where the Recovery Period Power (RPP) control mode is signalled with the other compressed mode parameters [13]. For RPP mode 0, the step size does not change during the recovery period and ordinary transmit power control applies, i.e. the algorithm for processing TPC commands determined by the value of PCA. For RPP mode 1, during RPL slots after each transmission gap, power control algorithm 1 applies with a step size $\Delta_{\text{RP-TPC}}$ instead of Δ_{TPC} , regardless of the value of PCA. We define the uplink DPCCCH transmit power change, which does not apply to the first slot after the transmission gap, as:

$$\Delta_{\text{DPCCCH}} = \Delta_{\text{RP-TPC}} \times \text{TPC\_cmd} + \Delta_{\text{PILOT}}, \quad (4.38)$$

where $\Delta_{\text{RP-TPC}}$ denotes the recovery power control step size and is expressed in dB. When the PCA has the value 1, ..., $\Delta_{\text{RP-TPC}}$, it is equal to the minimum value of 3 dB and $2\Delta_{\text{TPC}}$, and when the PCA has the value 2, $\Delta_{\text{RP-TPC}}$, it is equal to 1 dB. After the recovery period, normal transmit power control function resumes using the algorithm specified by the value of PCA and with step size Δ_{TPC} . When the PCA has 2 as value, slot sets over which the TPC commands are processed remain aligned to the frame boundaries in the compressed frame. For both RPP mode 0 and RPP mode 1, if the transmission gap or the recovery period results in any incomplete sets of TPC commands, $\text{TPC\_cmd} = \text{zero}$ for those incomplete slots sets [11].

Table 4.15 Initial Transmit Power (ITP) and Recovery Period Power (RPP) Control Modes

ITP	DESCRIPTION
0	$\Delta_{\text{RESUME}} = \Delta_{\text{TPC}} \times \text{TPC\_cmd}_{\text{gap}}$
1	$\Delta_{\text{RESUME}} = \delta_{\text{last}}$
RPP	
0	Transmit power control applies using the algorithm determined by the value of PCA with step size Δ_{TPC}
1	Transmit power control applies using algorithm 1 with step size $\Delta_{\text{RP-TPC}}$ during RPL slots after each transmission gap

4.3.6.3 DPCCCH Power Control Preamble

DCHs can use power control preamble for initialization, and both UL and DL DPCCCHs get transmitted during the uplink power control preamble. However, the UL DPCCCH does not start before the end of the power control preamble. The network signals the power control preamble length as a UE-specific parameter, where values can take 0 slots or 8 slots. When the preamble length > zero, power control details used during

the power control preamble differ from the ordinary power control used afterwards. The uplink DPCCCH change transmit power after the first slot of the power control preamble can be defined as:

$$\Delta_{\text{DPCCCH}} = \Delta_{\text{TPC-init}} \times \text{TPC\_cmd}. \quad (4.39)$$

When $\text{PCA} = 1$, then $\Delta_{\text{TPC-init}}$ is equal to the minimum value out of 3 dB and $2\Delta_{\text{TPC}}$; and when $\text{PCA} = 2$, then $\Delta_{\text{TPC-init}} = 2\text{dB}$. TPC_cmd is derived according to algorithm 1 regardless of the value of PCA. Normal transmit power control with the power control algorithm determined by the value of PCA and step size Δ_{TPC} , applies as soon as the sign of TPC_cmd reverses for the first time, or at the end of the power control preamble if the power control preamble ends first. The specifications in [11] describe the setting of the uplink DPCCCH/DPDCH power difference.

4.3.6.4 Power Control in the PCPCH Message Part and Preamble

Message part. The uplink inner or fast loop power control adjusts the UE transmit power to keep the received uplink signal-to-interference ratio (SIR) at a given $\text{SIR}_{\text{TARGET}}$ set by upper layer outer loop.

The network estimates the SIR_{EST} of the received PCPCH, then it generates TPC commands and transmits the commands once per slot according to the following rule:

- if $\text{SIR}_{\text{EST}} > \text{SIR}_{\text{TARGET}}$ then the TPC command to transmit = '0', or
- if $\text{SIR}_{\text{est}} < \text{SIR}_{\text{target}}$ then the TPC command to transmit = '1'.

The UE derives a TPC_cmd for each slot. The UE will support the UTRAN controlled algorithms 1 and 2 to derive a TPC_cmd . These come in step size Δ_{TPC} and can have values of 1 dB or 2 dB. After deriving the TPC command TPC_cmd using one of the two supported algorithms, the UE adjusts the transmit power of the uplink PCPCH in steps of Δ_{TPC} dB according to the TPC command. If $\text{TPC\_cmd} = 1$, then the transmit power of the uplink PCPCH increases by Δ_{TPC} dB. If $\text{TPC\_cmd} = -1$, then the transmit power of the uplink PCPCH decreases by Δ_{TPC} dB. If $\text{TPC\_cmd} = 0$, then the transmit power of the uplink PCPCH remains unchanged. Any power increase or decrease takes place immediately before the start of the pilot field on the PCPCH control channel [11].

4.3.6.5 Power Control in the Power Control Preamble Part

The UE begins the power control preamble using the same power level applied for the CD preamble. The initial power control step size utilized in the power control preamble differs from the one applied in the message part as follows:

- When inner loop power control algorithm 1 applies to the message part, then the initial step size in the power control preamble = $\Delta_{\text{TPC-init}}$, where $\Delta_{\text{TPC-init}}$ is the minimum value out of 3 dB and $2\Delta_{\text{TPC}}$, where Δ_{TPC} is the power control step size used for the message part.

- When inner loop power control algorithm 2 applies to the message part, then we use initially the inner loop power control algorithm 1 in the power control preamble, with a step size of 2dB.
- In either one of the cases, the power control algorithm and step size revert to ones used for the message part as soon as the sign of the TPC commands reverses for the first time [11].

4.3.7 Downlink Physical Channels

4.3.7.1 Downlink Transmit Diversity

Table 4.16 outlines possible applications of open and closed loop transmit diversity modes on different types of downlink physical channel. Simultaneous use of STTD and closed loop modes on the same physical channel is not possible. Furthermore, when Tx diversity applies to any of the downlink physical channels, it also apply to the P-CCPCH and SCH.

In addition, the PDSCH and the DPCH associated with this PDSCH shall use the same transmit diversity mode. A transmit diversity mode (open loop or closed loop) on the associated DPCH may not change while the duration of the PDSCH frame, and within the slot prior to the PDSCH frame. Nevertheless, changing from closed loop mode 1 to mode 2 or vice versa, is possible.

Table 4.16 Application of Tx Diversity Modes on Downlink Physical Channel Types

Physical channel type	Open loop mode		Closed loop
	TSTD	STTD	Mode
P-CCPCH	×	√	×
SCH	√	×	×
S-CCPCH	×	√	×
DPCH	×	√	√
PICH	×	√	×
PDSCH	×	√	√
AICH	×	√	×
CSICH	×	√	×

√, may apply; ×, does not apply.

4.3.7.2 Open Loop Transmit Diversity

- *Space Time Block Coding Based Transmit Antenna Diversity (STTD)* employs a space time block coding based transmit diversity. It is optional in UTRAN but mandatory at the UE. STTD encoding works on blocks of four consecutive channel bits. Figure 4.15 illustrates a block diagram of a generic STTD encoder for channel bits b_0, b_1, b_2, b_3 . Channel coding, rate matching and interleaving occurs as in the non-diversity mode. The bit b_i has real valued $\{0\}$ for DTX bits and $\{1, -1\}$ for all other channel bits.

- *Time Switched Transmit Diversity for SCH (TSTD)* can apply to the SCH; like STTD, it is optional in UTRAN but mandatory in the UE.

4.3.7.3 Closed Loop Transmit Diversity

Closed loop transmit diversity is described in [5].

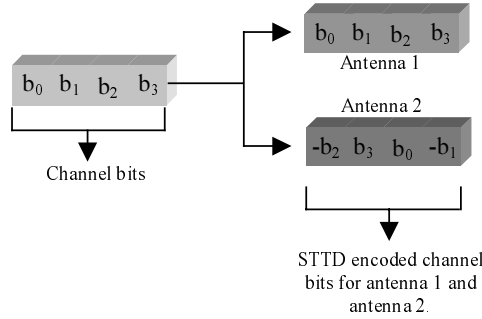


Figure 4.15 The STTD encoder – block diagram example.

4.3.8 Dedicated Downlink Physical Channels

The only Downlink Dedicated Physical Channel (downlink DPCH) transmits dedicated data generated at Layer 2 and above, i.e. the dedicated transport channel (DCH), in time-multiplex with control information generated at Layer 1 (known pilot bits, TPC commands, and an optional TFCI). The downlink DPCH is therefore a time multiplex of a downlink DPDCH and a downlink DPCCH.

Figure 4.16 illustrates the downlink DPCH frame structure, where each frame has 10 ms length split into 15 slots with $T_{\text{slot}} = 2560$ chips length, corresponding to one power-control period.

As in earlier frame structures, the parameter k in Figure 4.16 determines the total number of bits per downlink DPCH slot. It is related to the spreading factor SF of the physical channel as $\text{SF} = 512/2^k$. The spreading factor may thus range from 512 down to 4.

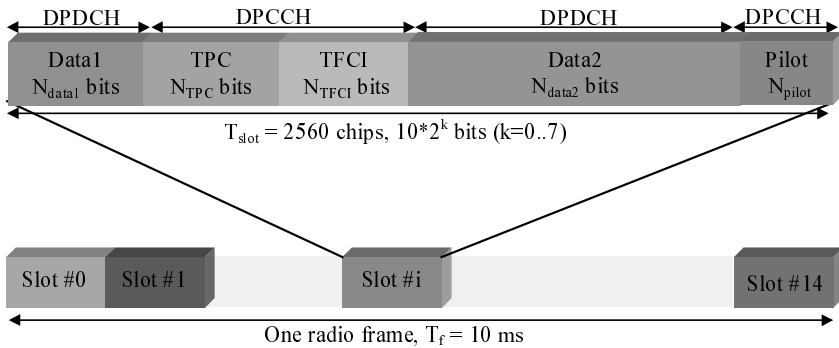


Figure 4.16 Downlink DPCH frame structure.

Appendix-A defines the exact number of bits of the different downlink DPCH fields, i.e. N_{pilot} , N_{TPC} , N_{TFCI} , N_{data1} and N_{data2} . Upper layers configure and reconfigure the slot format usage.

The two basic types of downlink DPCH reflected in Appendix A, are the ones that contain TFCI (e.g. various simultaneous services) and those that do not contain TFCI (e.g. fixed-rate services). While support of downlink TFCI inclusion in the network may be optional, it is mandatory in all UEs. The UTRAN determines if a TFCI should be transmitted or not.

4.3.8.1 The Compressed Mode

There are two compressed slot formats, i.e. A and B. Format B is possible by the spreading factor reduction and format A by all other transmission time reduction methods.

Table 4.17 shows the DPCH pilot bit patterns (order left to right), where shadowed columns define the Frame Synchronization Word (FSW) part, which can be used to confirm frame synchronization. All other non-FSW pilot bit pattern columns contain ‘11’ for each slot.

The downlink compressed-mode through ‘spreading factor reduction’ has double the number of bits in the TPC and pilot fields, where symbol repetition fills up the fields when necessary. In the normal mode we denote the bits in one of these fields by $x_1, x_2, x_3, \dots, x_X$, while in the compressed mode we denote the corresponding field as $x_1, x_2, x_1, x_2, x_3, x_4, x_3, x_4, \dots, x_X$.

Table 4.17 Pilot Bit Patterns for Downlink DPCH with $N_{\text{pilot}} = 2, 4, 8$ and 16

Slot	$N_{\text{pilot}}=2$	$N_{\text{pilot}}=4$		$N_{\text{pilot}}=8^b$				$N_{\text{pilot}}=16^c$							
Sym _b	0	0	1	0	1	2	3	0	1	2	3	4	5	6	7
0	11	11	11	11	11	11	10	11	11	11	10	11	11	11	10
1	00	11	00	11	00	11	10	11	00	11	10	11	11	11	00
2	01	11	01	11	01	11	01	11	01	11	01	11	10	11	00
3	00	11	00	11	00	11	00	11	00	11	00	11	01	11	10
4	10	11	10	11	10	11	01	11	10	11	01	11	11	11	11
5	11	11	11	11	11	11	10	11	11	11	10	11	01	11	01
6	11	11	11	11	11	11	00	11	11	11	00	11	10	11	11
7	10	11	10	11	10	11	00	11	10	11	00	11	10	11	00
8	01	11	01	11	01	11	10	11	01	11	10	11	00	11	11
9	11	11	11	11	11	11	11	11	11	11	11	11	00	11	11
10	01	11	01	11	01	11	01	11	01	11	01	11	11	11	10
11	10	11	10	11	10	11	11	11	10	11	11	11	00	11	10
12	10	11	10	11	10	11	00	11	10	11	00	11	01	11	01
13	00	11	00	11	00	11	11	11	00	11	11	11	00	11	00
14	00	11	00	11	00	11	11	11	00	11	11	11	10	11	01

^a Pattern does not apply to slot formats 2B and 3B.

^b Pattern does not apply to slot formats 0B, 1B, 4B, 5B, 8B, and 9B.

^c Pattern does not apply to slot formats 6B, 7B, 10B, 11B, 12B, and 13B.

- For the other slot formats, symbol repetition shall be applied to the pilot bit pattern with the half size.
- In compressed mode through spreading factor reduction, symbol repetition is applied to the symbol patterns described in Table 4.12.

Table 4.18 illustrates the relationship between the TPC symbol and the transmitter power control command.

Table 4.18 TPC Bit Pattern

TPC Bit Pattern			Transmitter power control command
$N_{\text{TPC}} = 2$	$N_{\text{TPC}} = 4$	$N_{\text{TPC}} = 8$	
11	1111	11111111	1
00	0000	00000000	0

4.3.8.2 Multi-Code Transmission

Multi-code transmission in the downlink, i.e. the Coded Composite Transport Channel (CCTrCH) is mapped onto several parallel downlink DPCHs using the same spreading factor. As illustrated in Figure 4.17, the Layer 1 control information is transmitted only on the first downlink DPCH. DTX bits are transmitted during the corresponding time period for the additional downlink DPCHs.

When there are several CCTrCHs mapped to different DPCHs transmitted to the same UE, different spreading factors can be used on the DPCHs. However, even in this case, Layer 1 control information is transmitted only on the first DPCH, while DTX bits are transmitted during the corresponding time period on the additional DPCHs.

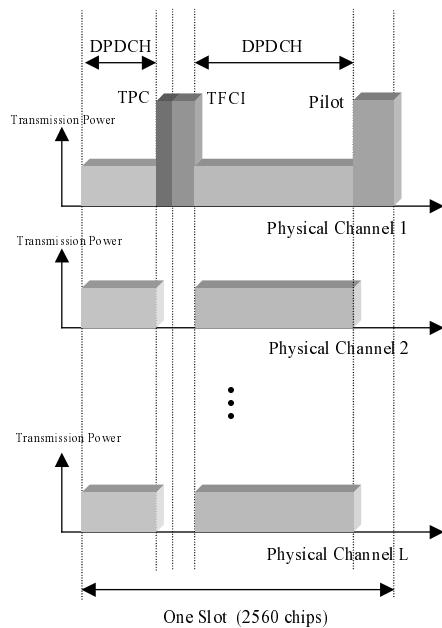


Figure 4.17 Downlink slot format in case of multi-code transmission.

4.3.8.3 STTD in the DPCH

In the following we describe how antenna diversity occurs using the pilot bit pattern for the DPCH channel transmitted on antenna 2 illustrated in Table 4.19:

- $N_{\text{pilot}} = 2$ diversity antenna pilot pattern results from STTD encoding the two pilot bits defined in Table 4.17 with the last two bits (data or DTX) of the second data field (data 2) of the slot. Hence, for $N_{\text{pilot}} = 2$ the last two bits of the second data field (data 2) after STTD encoding, follow the diversity antenna pilot bits defined in Table 4.19.
- $N_{\text{pilot}} = 4$ diversity antenna pilot bit pattern results from STTD encoding both the shadowed and non-shadowed pilot bits in Table 4.17.
- $N_{\text{pilot}} = 8$, 16 diversity antenna pilot bit pattern shown in Table 4.19 results from STTD encoding the corresponding (shadowed) bits in Table 4.17. The non-shadowed pilot bit pattern is orthogonal to the corresponding (non-shadowed) pilot bit pattern in Table 4.17.
- Pilot bit patterns $N_{\text{pilot}} > 4$ in compressed mode with the ‘spreading factor reduction’ method will get symbol repetition for the pilot bit patterns of Table 4.19.

Table 4.19 Pilot Bit Patterns of Downlink DPCH for Antenna 2 using STTD

Slot 0	$N_{\text{pilot}}=2^a$	$N_{\text{pilot}}=4^b$		$N_{\text{pilot}}=8^c$				$N_{\text{pilot}}=16^d$							
Sym	0	0	1	0	1	2	3	0	1	2	3	4	5	6	7
0	01	01	10	11	00	00	10	11	00	00	10	11	00	00	10
1	10	10	10	11	00	00	01	11	00	00	01	11	10	00	10
2	11	11	10	11	11	00	00	11	11	00	00	11	10	00	11
3	10	10	10	11	10	00	01	11	10	00	01	11	00	00	00
4	00	00	10	11	11	00	11	11	11	00	11	11	01	00	10
5	01	01	10	11	00	00	10	11	00	00	10	11	11	00	00
6	01	01	10	11	10	00	10	11	10	00	10	11	01	00	11
7	00	00	10	11	10	00	11	11	10	00	11	11	10	00	11
8	11	11	10	11	00	00	00	11	00	00	00	11	01	00	01
9	01	01	10	11	01	00	10	11	01	00	10	11	01	00	01
10	11	11	10	11	11	00	00	11	11	00	00	11	00	00	10
11	00	00	10	11	01	00	11	11	01	00	11	11	00	00	01
12	00	00	10	11	10	00	11	11	10	00	11	11	11	00	00
13	10	10	10	11	01	00	01	11	01	00	01	11	10	00	01
14	10	10	10	11	01	00	01	11	01	00	01	11	11	00	11

^a The pilot bits precede the last two bits of the data 2 field.

^b Bit pattern does not apply to slot formats 2B and 3B.

^c Bit pattern does not apply to slot formats 0B, 1B, 4B, 5B, 8B, and 9B.

^d Bit pattern does not apply to slot formats 6B, 7B, 10B, 11B, 12B, and 13B.

^e Bit pattern applies also to slot formats 2B and 3B.

¹ In other slot formats we apply symbol repetition to the pilot bit pattern with the half size.

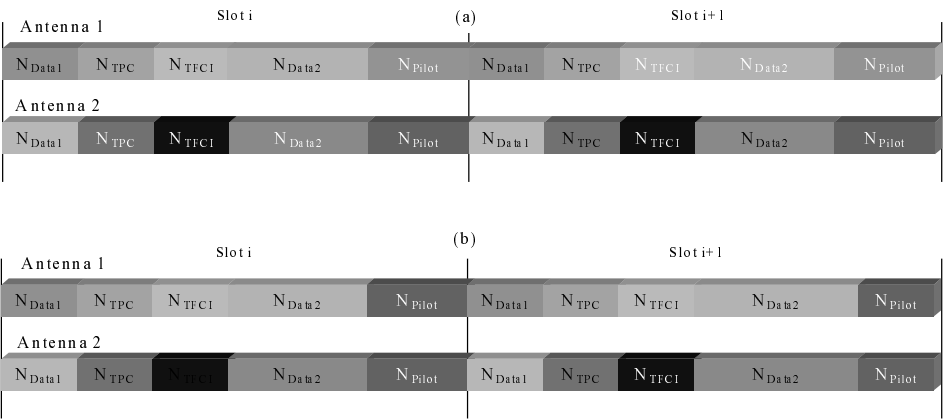
² Appendix B illustrates the bit pattern for compressed mode with spread reduction and $N_{\text{pilot}} = 4$.

STTD encoding for the DPDCH, TPC, and TFCI fields follows the definition in Section 4.3.7.2. For DPCH with SF = 512 the first two bits in each slot, i.e. TPC bits, are not STTD encoded. These bits are transmitted with equal power from the two antennas; however, the remaining four bits are STTD encoded.

4.3.8.4 Closed Loop Mode Transmit Diversity and Dedicated Channel Pilots

Closed loop mode 1 uses orthogonal pilot patterns between the transmit antennas. Table 4.17 defines the pilot patterns used on antenna 1 and Table 4.19 defines pilot patterns used on antenna 2. Figure 4.18 illustrates the two antenna slot structures with the pilot pattern bits shaded in grey.

In closed loop mode 2, the same pilot pattern is used on both antennas with bit pattern defined in Table 4.17.



- ¹ Closed-loop mode 1 uses structure (a),
- ² Closed-loop mode 2 uses structure (b),
- ³ The grey shading indicates the bit pattern orthogonality.

Figure 4.18 Slot structures for downlink dedicated physical channel diversity transmission.

4.3.8.5 The Downlink DPCCCH for CPCH

Downlink DPCCCH for CPCH is a special case of downlink dedicated physical channel of ‘slot format # 0’ illustrated in the table of Appendix A, where the spreading factor for the DL-DPCCCH is 512. Figure 4.19 illustrates the CPCH downlink DPCCCH frame structure.

The CPCH downlink DPCCCH incorporates known pilot bits, TFCI, TPC commands and CPCH Control Commands (CCC). CPCH control commands support CPCH signalling. These commands include Layer 1 control command such as start of message indicator, and higher layer control command such as emergency stop command. The exact number of bits of DL DPCCCH fields (N_{pilot} , N_{TFCI} , N_{CCC} and N_{TPC}) is determined in Table 4.20. Table 4.17 defines the pilot bit pattern for $N_{pilot}=4$ used for DPCCCH for CPCH.

Table 4.20 DPCCCH Fields for CPCH Message Transmission

Slot format #i	Channel bit rate (kbps)	Channel symbol rate (ksps)	SF	Bits/slot	DPCCCH bits/slot				Transmitted slots/frame N_{Tr}
					N_{TPC}	N_{TFCI}	N_{CCC}	N_{Pi-}	
0	15	7.5	512	10	2	0	4	4	15

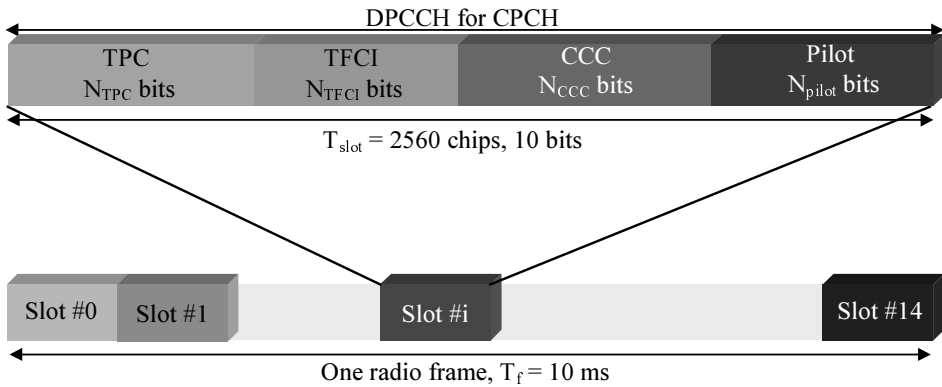


Figure 4.19 Frame structure for downlink DPCCH for CPCH.

The transmission of the CPCH control command uses the CCC field in Figure 4.19 upon request from higher layers, where a given pattern is mapped onto the CCC field. If no requests exist, nothing is transmitted in the CCC field; thus, there is one to one mapping between the CPCH control command and the pattern. ‘CPCH emergency stop transmission’ maps the [1111] pattern onto the CCC field. This stop command cannot be transmitted during the first $N_{\text{Start\_Message}}$ frames of DL DPCCH after power control preamble. The CCC field gets the [1010] pattern for the start of message indicator during the first $N_{\text{Start\_Message}}$ frames of DL DPCCH after power control preamble.

4.3.9 Common Downlink Physical Channels

4.3.9.1 Common Pilot Channel (CPICH)

Figure 4.20 illustrates the frame structure of the CPICH, which is a fixed rate (30 kbps, SF = 256) downlink physical channel that carries a pre-defined bit/symbol sequence.

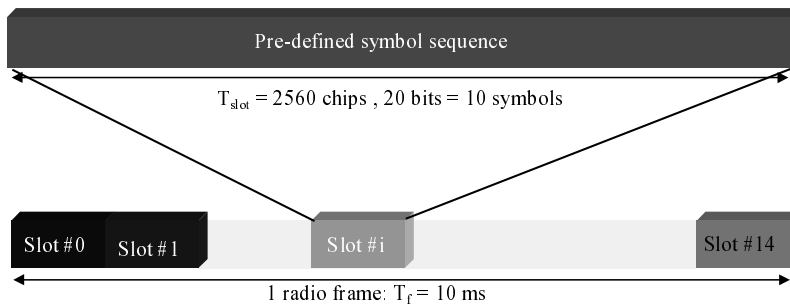


Figure 4.20 Frame structure for the common pilot channel.

During *transmit diversity* on any downlink cell channel, either with open or closed loop power control, the CPICH shall be transmitted from both antennas using the same channelization and scrambling code (in this case, as illustrated in Figure 4.21).

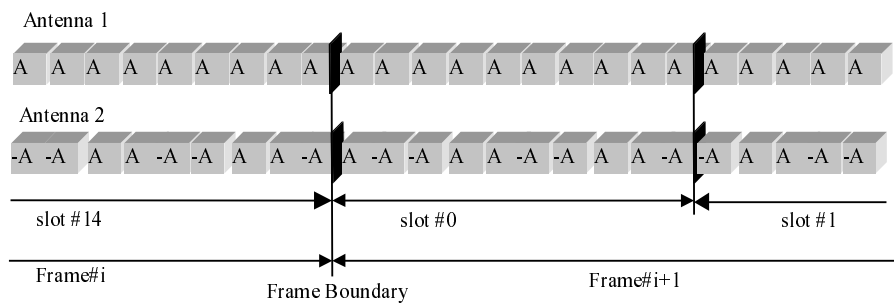


Figure 4.21 Common pilot channel (with $A = 1 + j$) modulation pattern.

Antenna 1 and Antenna 2 do have different CPICH pre-defined symbol sequences. In the absence of *transmit diversity*, the symbol sequence of Antenna 1 applies.

In the sequel we describe the two types of common pilot channels, i.e. primary and secondary CPICHs, which differ in their use and the limitations placed on their physical features.

4.3.9.1.1 Primary and Secondary Common Pilot Channels (P-CPICH)

Table 4.21 illustrates the main characteristics of the P-CPICHs. The coding and spreading section present more details on scrambling issues.

Table 4.21 Characteristics of Primary and Secondary P-CPICHs

Primary P-CPICH characteristics	Secondary P-CPICH characteristics
The P-CPICH uses always the same channelization code	The S-CPICH uses an arbitrary channelization code of SF = 256
The primary scrambling code scrambles the P-CPICH	Either the primary or a secondary scrambling code scrambles a S-CPICH
Each cell has only one P-CPICH	A cell may contain zero, one, or several S-CPICHs
The P-CPICH is broadcast over the entire cell	A S-CPICH may be transmitted over the entire cell or only over a part of the cell
P-CPICH serves as phase reference for DL: SCH, primary CCPCH, AICH, and PICH	A secondary CPICH may be the reference for secondary CCPCH and DL- DPCH.
It is also the <i>default</i> phase reference for all other downlink physical channels	Upper layers inform the UE when a secondary CPICH is used as reference...

4.3.9.2 Primary Common Control Physical Channel (P-CCPCH)

The primary CCPCH, as a downlink physical channel with a fixed rate (30 kbps, SF = 256), carries the BCH transport channel. Figure 4.22 shows the frame structure of the primary CCPCH, which differs from the downlink DPCH in that it does not transmit TPC commands, or TFCI, or pilot bits. In addition, it is not transmitted during the first 256 chips of each slot. The technical specification in [1] describes the primary CCPCH structure with STTD encoding.

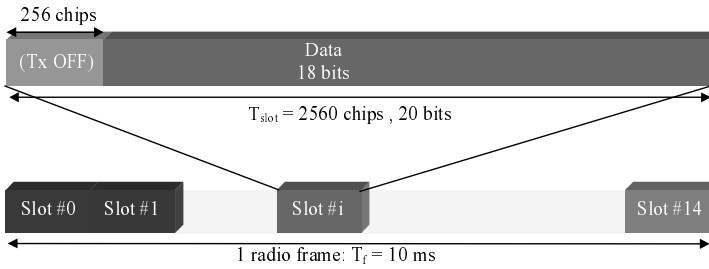


Figure 4.22 Primary CCPCH frame structure.

4.3.9.3 Secondary Common Control Physical Channel (S-CCPCH)

The two CCPCH types, i.e. those that include TFCI and those that do not, carry the FACH and PCH. Since it is the UTRAN which determines when a TFCI shall be transmitted, it is mandatory that all UEs support the use of TFCI. Possible secondary CCPCH rates are the same as for the downlink DPCH. Figure 4.23 illustrates a secondary CCPCH frame structure, where the parameter k determines the total number of bits per downlink secondary CCPCH slot. It is related to the spreading factor $SF (= 256/2^k)$ of the physical channel with spreading range of 256 down to 4.

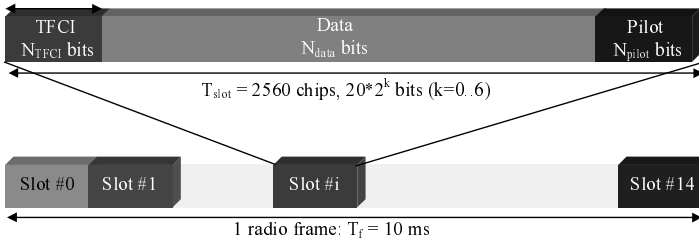


Figure 4.23 Frame structure for the secondary common control physical channel.

Table 4.22 presents the number of bits per field, as well as the channel bit, and symbol rates before spreading for the secondary CCPCH. Table 4.23 illustrates the pilot patterns.

We can map the FACH and PCH to the same or to separate secondary CCPCHs. If the first case occurs, one frame can serve both. Key characteristics and differences are:

- A CCPCH does not have inner-loop power control as a downlink dedicated physical channel does.
- While a transport channel mapped to the primary CCPCH (BCH) can support only a fixed pre-defined transport format combination, a secondary CCPCH can support multiple transport format combinations using TFCI.
- We can transmit a primary CCPCH over the entire cell.
- A secondary CCPCH may be transmitted in a narrow lobe in the same way as a dedicated physical channel, only when carrying the FACH.

Table 4.22 Secondary CCPCH Fields

Slot format #i	Channel bit rate (kbps)	Channel symbol rate (ksps)	SF	Bits/ frame	Bits/ slot	N _{data}	N _{pilot}	N _{TFCI}
0	30	15	256	300	20	20	0	0
1	30	15	256	300	20	12	8	0
2	30	15	256	300	20	18	0	2
3	30	15	256	300	20	10	8	2
4	60	30	128	600	40	40	0	0
5	60	30	128	600	40	32	8	0
6	60	30	128	600	40	38	0	2
7	60	30	128	600	40	30	8	2
8	120	60	64	1200	80	72	0	8*
9	120	60	64	1200	80	64	8	8*
10	240	120	32	2400	160	152	0	8*
11	240	120	32	2400	160	144	8	8*
12	480	240	16	4800	320	312	0	8*
13	480	240	16	4800	320	296	16	8*
14	960	480	8	9600	640	632	0	8*
15	960	480	8	9600	640	616	16	8*
16	1920	960	4	19200	1280	1272	0	8*
17	1920	960	4	19200	1280	1256	16	8*

* If TFCI bits are not used, then DTX shall be used in TFCI field.

We can use the shadowed part of the pilot symbol pattern illustrated in Table 4.23 as frame synchronization words. Symbol patterns other than the frame synchronization word contain ‘11’. The transmission order of the two-bit pair representing an I/Q pair of QPSK modulation illustrated in Table 4.23 goes from left to right.

Table 4.23 Pilot Symbol Pattern

Slot #	N _{pilot} = 8				N _{pilot} = 16							
Symb. #	0	1	2	3	0	1	2	3	4	5	6	7
0	11	11	11	10	11	11	11	10	11	11	11	10
1	11	00	11	10	11	00	11	10	11	11	11	00
2	11	01	11	01	11	01	11	01	11	10	11	00
3	11	00	11	00	11	00	11	00	11	01	11	10
4	11	10	11	01	11	10	11	01	11	11	11	11
5	11	11	11	10	11	11	11	10	11	01	11	01
6	11	11	11	00	11	11	11	00	11	10	11	11
7	11	10	11	00	11	10	11	00	11	10	11	00
8	11	01	11	10	11	01	11	10	11	00	11	11
9	11	11	11	11	11	11	11	11	11	00	11	11
10	11	01	11	01	11	01	11	01	11	11	11	10
11	11	10	11	11	11	10	11	11	11	00	11	10
12	11	10	11	00	11	10	11	00	11	01	11	01
13	11	00	11	11	11	00	11	11	11	00	11	00
14	11	00	11	11	11	00	11	11	11	10	11	01

When a slot format uses TFCI bits, its value in each radio frame corresponds to a certain transport format combination of the FACHs and/or PCHs in actual use, which is (re)negotiated at each FACH/PCH addition/removal.

4.3.9.3.1 STTD Encoding the S-CCPCH structure

If the UTRAN perceives antenna diversity and we transmit S-CCPCH using open loop transmit diversity, STTD encoding applies to the S-CCPCH data symbols. Table 4.24 shows the pilot symbol pattern for antenna 2.

Table 4.24 Antenna 2 Pilot Symbol Pattern for STTD Encoding in the S-CCPCH

Slot #	$N_{\text{pilot}} = 8$				$N_{\text{pilot}} = 16$							
Symbol #	0	1	2	3	0	1	2	3	4	5	6	7
0	11	00	00	10	11	00	00	10	11	00	00	10
1	11	00	00	01	11	00	00	01	11	10	00	10
2	11	11	00	00	11	11	00	00	11	10	00	11
3	11	10	00	01	11	10	00	01	11	00	00	00
4	11	11	00	11	11	11	00	11	11	01	00	10
5	11	00	00	10	11	00	00	10	11	11	00	00
6	11	10	00	10	11	10	00	10	11	01	00	11
7	11	10	00	11	11	10	00	11	11	10	00	11
8	11	00	00	00	11	00	00	00	11	01	00	01
9	11	01	00	10	11	01	00	10	11	01	00	01
10	11	11	00	00	11	11	00	00	11	00	00	10
11	11	01	00	11	11	01	00	11	11	00	00	01
12	11	10	00	11	11	10	00	11	11	11	00	00
13	11	01	00	01	11	01	00	01	11	10	00	01
14	11	01	00	01	11	01	00	01	11	11	00	11

4.3.9.4 Synchronization Channel (SCH)

The Synchronization Channel (SCH) as a downlink signal for cell search consists of two sub-channels, i.e. the primary and secondary SCH. These channels shown in Figure 4.24 have 10 ms radio frames divided into 15 slots, each having 2560 chips length.

The primary SCH consists of a modulated code of length 256 chips, the Primary Synchronization Code (PSC) denoted c_p in Figure 4.24, transmitted once every slot. The PSC is the same for every cell in the system.

The secondary SCH consists of repeatedly transmitting a length 15 sequence of modulated codes of length 256 chips, the Secondary Synchronization Codes (SSC), transmitted in parallel with the primary SCH. The SSC is denoted $c_s^{i,k}$ in Figure 4.24, where $i = 0, 1, \dots, 63$ is the number of the scrambling code group, and $k = 0, 1, \dots, 14$ is the slot number. Each SSC is chosen from a set of 16 different codes of length 256. This sequence on the secondary SCH indicates which of the code groups the cell's downlink scrambling code belongs to.

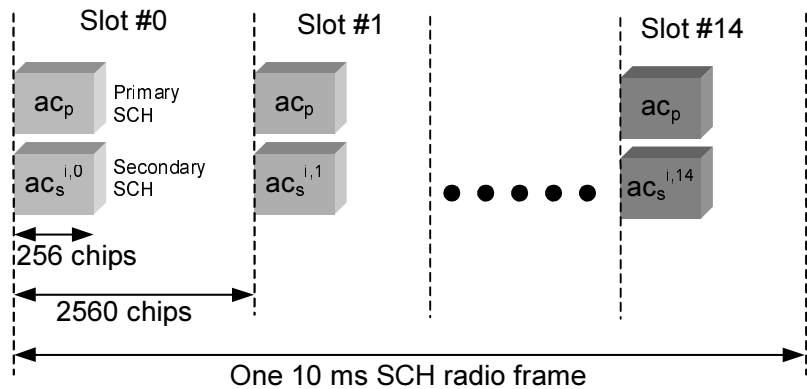


Figure 4.24 Structure of synchronization channel (SCH).

Symbol a modulates the primary and secondary synchronization codes as illustrated in Figure 4.24. Table 4.25 shows the presence/absence of STTD encoding on the P-CCPCH.

Table 4.25 STTD Encoding on the P-CCPCH

P-CCPCH STTD encoded	$a = +1$
P-CCPCH not STTD encoded	$a = -1$

4.3.9.4.1 SCH transmitted by TSTD

Figure 4.25 shows the SCH's structure transmitted by the TSTD scheme, where we transmit on antenna 1 both PSC and SSC in even numbered slots, and on antenna 2 both PSC and SSC in odd numbered slots.

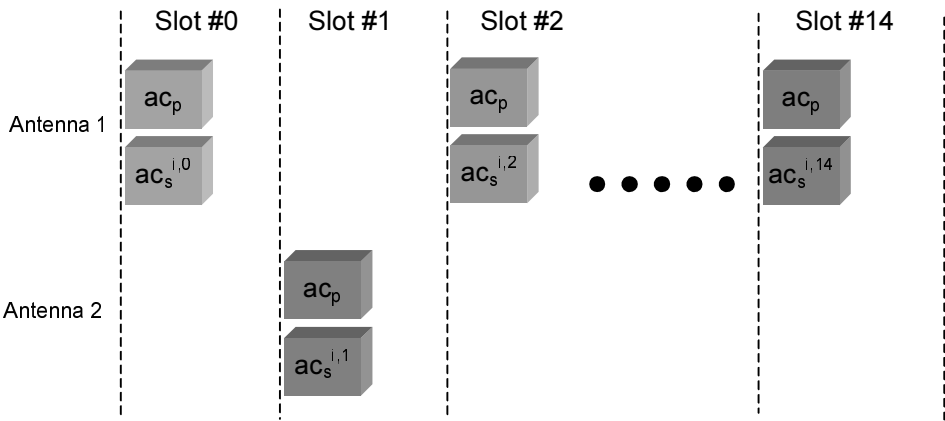


Figure 4.25 Structure of SCH transmitted by TSTD scheme.

4.3.9.5 Physical Downlink Shared Channel (PDSCH)

Users share the PDSCH (carrying the transport Downlink Shared Channel (DSCH)), based on code multiplexing. Since the DSCH associates itself always with one or several DCHs, we also associate the PDSCH with one or several downlink DPCHs. More precisely, we associate each PDSCH radio frame with one downlink DPCH. Figure 4.26 illustrates the PDSCH frame and slot structure.

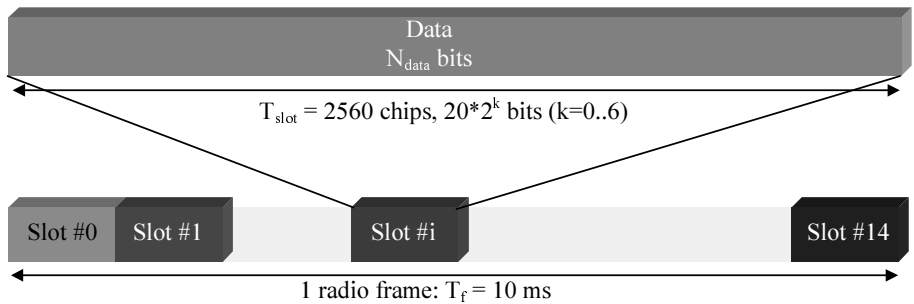


Figure 4.26 Frame structure for the PDSCH.

Two signalling methods indicate if the UE has data to decode on the DSCH, i.e. through the TFCI field or higher layer signalling. For example, when the spreading factor and other physical layer parameters vary on a frame-by-frame basis, the TFCI informs the UE of PDSCH instantaneous parameters including the channelization code from the PDSCH OVSF code tree.

Although the PDSCH and DPCH do not necessarily have the same spreading factors, and the PDSCH spreading factor may vary from frame to frame, a PDSCH transmission with associated DPCH is a special case of multi-code transmission. Thus, when mapping a DSCH to multiple parallel PDSCHs the spreading factor of all PDSCH codes will be the same. The PDSCH does not carry physical layer info, but all relevant L1 control information. Table 4.26 illustrates PDSCH T bit rates and symbol rates, where spreading factors may vary from 256 to 4.

Table 4.26 PDSCH* Fields

Slot format #i	Channel bit rate (kbps)	Channel symbol rate (ksps)	SF	Bits/frame	Bits/slot	N_{data}
0	30	15	256	300	20	20
1	60	30	128	600	40	40
2	120	60	64	1200	80	80
3	240	120	32	2400	160	160
4	480	240	16	4800	320	320
5	960	480	8	9600	640	640
6	1920	960	4	19200	1280	1280

* When open loop transmit diversity is employed for the PDSCH, STTD encoding is used on the data bits.

4.3.9.6 Acquisition Indicator Channel (AICH)

As a physical channel, having the primary CPICH for phase reference, the AICH carries Acquisition Indicators (AI) corresponding to signature s on the PRACH. Figure 4.27 illustrates its structure consisting of a repeated sequence of 15 consecutive *Access Slots* (AS), each of length 40 bit intervals. In turn, every access slot consists of two parts: an *Acquisition-Indicator* (AI) part containing 32 real-valued symbols $a_0, \dots, a_{31}$ and a part of 1024 chips duration without transmission.

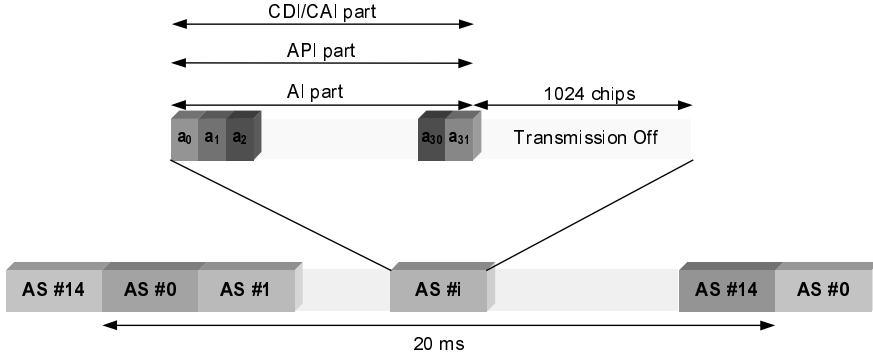


Figure 4.27 Structure of the AICH, AP-AICH and CD/CA-ICH.

Equation (4.40a) defines the real-valued symbols $a_0, a_1, \dots, a_{31}$ seen in Figure 4.27. Table 20 in [1] provides the acquisition indicator AI_s with values +1, -1, and 0 corresponding to signature s and the sequence $b_{s,0}, \dots, b_{s,31}$. When the AICH has STTD-based open-loop transmit diversity, STTD encoding applies to each $b_{s,0}, b_{s,1}, \dots, b_{s,31}$ sequence separately before these sequences combined into AICH symbols $a_0, \dots, a_{31}$.

$$(a) \ a_j = \sum_{s=0}^{15} AI_s b_{s,j}, \quad (b) \ a_j = \sum_{s=0}^{15} API_s \times b_{s,j}. \quad (4.40)$$

4.3.9.7 CPCH Access Preamble Acquisition Indicator Channel (AP-AICH)

The physical AP-AICH carries CPCH Access Preamble acquisition Indicators (API). The AP acquisition indicator API corresponds to the AP signature s transmitted by UE. The AP-AICH, with the primary CPICH as reference, and the AICH may use the same or different channelization codes. Figure 4.27 with the corresponding 'API part' label illustrates the structure of AP-AICH. The AP-AICH has a part of 4096 chips duration to transmit the AP acquisition indicator (API), followed by a part of 1024 chips duration with no transmission.

Equation (4.40b) defines the real-valued symbols $a_0, a_1, \dots, a_{31}$ seen in Figure 4.27, where API_s , taking the values +1, -1, and 0, are the AP acquisition indicators corresponding to the access preamble signature s transmitted by UE and Table 20 in [1] provides the sequence $b_{s,0}, \dots, b_{s,31}$. As in the AICH, when the AP-AICH has STTD-based open-loop transmit diversity, STTD encoding applies to each $b_{s,0}, b_{s,1}, \dots, b_{s,31}$ sequence separately before these sequences combine into AICH symbols $a_0, \dots, a_{31}$.

4.3.9.8 CPCH Collision Detection/Channel Assignment Indicator Channel

As a physical channel, the Collision Detection Channel Assignment Indicator Channel (CD/CA-ICH) carries the CD Indicator (CDI) only if the CA is not active, or CD Indicator/CA Indicator (CDI/CAI) at the same time if the CA is active. Figure 4.27 illustrates the CD/CA-ICH structure with the corresponding label, where the CD/CA-ICH transmits 4096 chips duration, followed by 1024 chips duration without transmission. The same or different channelization codes may apply to the CD/CA-ICH and AP-AICH.

As in the preceding indicator channels, when the CD/CA-ICH has STTD-based open-loop transmit diversity, STTD encoding applies to each $b_{s,0}, b_{s,1}, \dots, b_{s,31}$ sequence separately before these sequences combined into AICH symbols $a_0, \dots, a_{31}$.

Equation (4.41a) defines the real-valued symbols $a_0, a_1, \dots, a_{31}$ shown in Figure 4.27 for non-active CA, where CDI_s , with values +1, and 0, is the CD indicator corresponding to CD preamble signature s transmitted by the UE. Table 20 in [1] lists the sequence $b_{s,0}, \dots, b_{s,31}$.

$$(a) \ a_j = \sum_{s=0}^{15} \text{CDI}_s \times b_{s,j}, \quad (b) \ a_j = \sum_{i=0}^{15} \text{CDI}_i \times b_{s_i,j} + \sum_{k=0}^{15} \text{CAI}_k \times b_{s_k,j}. \quad (4.41)$$

Equation (4.41b) defines the real-valued symbols $a_0, a_1, \dots, a_{31}$ when CA is active, where the subscript s_i, s_k depend on the indexes i, k according to Table 4.27, respectively, and indicate the signature number s in Table 20 [1]. The sequence $b_{s,0}, \dots, b_{s,31}$ is also given in Table 20 [1]. CDI_i , with values +1/0 or -1/0, is the CD indicator corresponding to the CD preamble i transmitted by the UE, and CAI_k , with values +1/0 or -1/0, is the CA indicator corresponding to the assigned channel index k as given in Table 4.27.

Table 4.27 Generation of $\text{CDI}_i/\text{CAI}_k$

UE transmitted CD preamble i	CDI_i	Signature s_i	Channel assignment index k	CAI_k	Signature s_k
0	+1/0	1	0	+1/0	0
1	-1/0		1	-1/0	
2	+1/0	3	2	+1/0	8
3	-1/0		3	-1/0	
4	+1/0	5	4	+1/0	4
5	-1/0		5	-1/0	
6	+1/0	7	6	+1/0	12
7	-1/0		7	-1/0	
8	+1/0	9	8	+1/0	2
9	-1/0		9	-1/0	
10	+1/0	11	10	+1/0	6
11	-1/0		11	-1/0	
12	+1/0	13	12	+1/0	10
13	-1/0		13	-1/0	
14	+1/0	15	14	+1/0	14
15	-1/0		15	-1/0	

4.3.9.9 Paging Indicator Channel (PICH)

As a physical channel the PICH has a fixed rate (SF = 256) and carries the Paging Indicators (PI). It is always associated with a S-CCPCH to which a PCH transport channel is mapped.

Figure 4.28 illustrates PICH frame structure, where one PICH radio frame of 10 ms length consists of 300 bits ($b_0, b_1, \dots, b_{299}$). Of these, 288 bits ($b_0, b_1, \dots, b_{287}$) are used to carry paging indicators. The remaining 12 bits ($b_{288}, b_{289}, \dots, b_{299}$) are undefined. Each PICH frame transmits N paging indicators $\{PI_0, \dots, PI_{N-1}\}$, where $N = 18, 36, 72$, or 144.

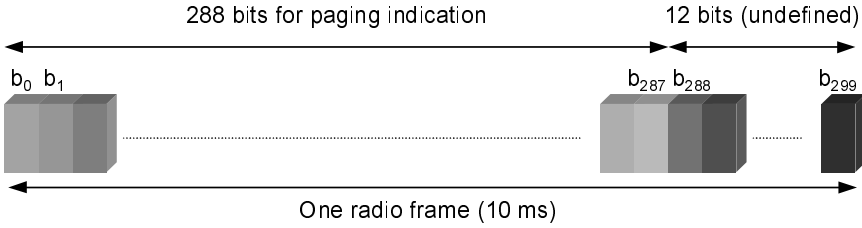


Figure 4.28 Structure of Paging Indicator Channel (PICH).

Higher layers calculate the PI mapped to the paging indicator PI_p , where p is computed as a function of the PI, the SFN of the P-CCPCH radio frame during which the start of the PICH radio frame occurs, and the number of paging indicators per frame (N) see Equation (4.3). Table 4.28 shows the mapping from $\{PI_0, \dots, PI_{N-1}\}$ to the PICH bits $\{b_0, \dots, b_{287}\}$.

$$p = \left(PI + \left\lfloor \left((18 \times (\text{SFN} + \lfloor \text{SFN}/8 \rfloor + \lfloor \text{SFN}/64 \rfloor + \lfloor \text{SFN}/512 \rfloor)) \bmod 144 \right) \times \frac{N}{144} \right\rfloor \right) \bmod N \quad (4.42)$$

Table 4.28 Mapping of Paging Indicators (PI) to PICH Bits

Number of PI per frame (N)	$PI_p = 1$	$PI_p = 0$
$N = 18$	$\{b_{16p}, \dots, b_{16p+15}\} = \{1, 1, \dots, 1\}$	$\{b_{16p}, \dots, b_{16p+15}\} = \{0, 0, \dots, 0\}$
$N = 36$	$\{b_{8p}, \dots, b_{8p+7}\} = \{1, 1, \dots, 1\}$	$\{b_{8p}, \dots, b_{8p+7}\} = \{0, 0, \dots, 0\}$
$N = 72$	$\{b_{4p}, \dots, b_{4p+3}\} = \{1, 1, \dots, 1\}$	$\{b_{4p}, \dots, b_{4p+3}\} = \{0, 0, \dots, 0\}$
$N = 144$	$\{b_{2p}, b_{2p+1}\} = \{1, 1\}$	$\{b_{2p}, b_{2p+1}\} = \{0, 0\}$

If a paging indicator in a certain frame has the value '1', it indicates that UEs associated with this paging indicator should read the corresponding frame of the associated S-CCPCH. In the event of transmit diversity for the PICH, STTD encoding applies on the PICH bits.

4.3.9.10 CPCH Status Indicator Channel (CSICH)

The CPCH Status Indicator Channel (CSICH) is also a fixed rate (SF=256) physical channel carrying CPCH status information. It has always association with a physical channel used for transmitting a CPCH AP-AICH and uses the same channelization and scrambling codes. Figure 4.29 illustrates the CSICH frame structure. It consists of 15 consecutive access slots (AS) each 40 bits long and having two parts, i.e. one of duration 4096 chips without transmission, and a Status Indicator (SI) part consisting of 8 bits $b_{8i}, \dots, b_{8i+7}$, where i is the access slot number. The CSICH uses the same modulation of the PICH, and has the primary CPICH phase as reference.

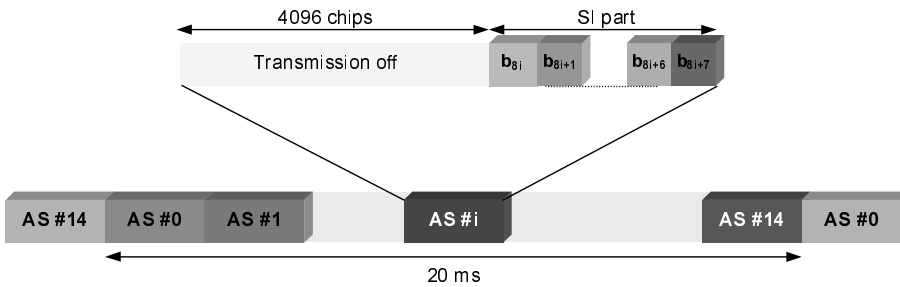


Figure 4.29 Structure of CPCH Status Indicator Channel (CSICH).

Each CSICH frame transmits N status indicators $\{SI_0, \dots, SI_{N-1}\}$, i.e. all the access slots of the CSICH frame transmit status indicator even if some signatures and/or access slots are shared between CPCH and RACH. The mapping from $\{SI_0, \dots, SI_{N-1}\}$ to the CSICH bits $\{b_0, \dots, b_{119}\}$ follows Table 4.29. In the event of transmit diversity for the CSICH, STTD encoding applies on the CSICH bits.

Table 4.29 Mapping of Status Indicators (SI) to CSICH Bits

Number of SI per frame (N)	$SI_n = 1$	$SI_n = 0$
$N = 1$	$\{b_0, \dots, b_{119}\} = \{1, 1, \dots, 1\}$	$\{b_0, \dots, b_{119}\} = \{0, 0, \dots, 0\}$
$N = 3$	$\{b_{40n}, \dots, b_{40n+39}\} = \{1, 1, \dots, 1\}$	$\{b_{40n}, \dots, b_{40n+39}\} = \{0, 0, \dots, 0\}$
$N = 5$	$\{b_{24n}, \dots, b_{24n+23}\} = \{1, 1, \dots, 1\}$	$\{b_{24n}, \dots, b_{24n+23}\} = \{0, 0, \dots, 0\}$
$N = 15$	$\{b_{8n}, \dots, b_{8n+7}\} = \{1, 1, \dots, 1\}$	$\{b_{8n}, \dots, b_{8n+7}\} = \{0, 0, \dots, 0\}$
$N = 30$	$\{b_{4n}, \dots, b_{4n+3}\} = \{1, 1, 1, 1\}$	$\{b_{4n}, \dots, b_{4n+3}\} = \{0, 0, 0, 0\}$
$N = 60$	$\{b_{2n}, b_{2n+1}\} = \{1, 1\}$	$\{b_{2n}, b_{2n+1}\} = \{0, 0\}$

Higher layers set the Status Indicator values for UTRAN. Thus, the higher layers provide Layer 1 with the mapping between the values of the status indicators and the availability of CPCH resources. At the UE the number of status indicators per frame is also a higher layer parameter.

4.3.10 Mapping Transport Channels onto Physical Channels

Table 4.30 summarizes the mapping of transport channels onto physical channels. The DCHs are coded and multiplexed as described in [3], and the resulting data stream is mapped sequentially (first-in-first-mapped) directly to the physical channel(s). The

mapping of BCH and FACH/PCH is equally straightforward, where the data stream after coding and interleaving is mapped sequentially to the primary and secondary CCPCH, respectively. Also for the RACH, the coded and interleaved bits are sequentially mapped to the physical channel, in this case the message part of the PRACH.

Table 4.30 Mapping Transport Channels to Physical Channels

Transport channels	Physical channels
Dedicated Channel (DCH)	Dedicated Physical Data Channel (DPDCH)
	Dedicated Physical Control Channel (DPCCH)
Random Access Channel (RACH)	Physical Random Access Channel (PRACH)
Common Packet Channel (CPCH)	Physical Common Packet Channel (PCPCH)
	Common Pilot Channel (CPICH)
Broadcast Channel (BCH)	Primary Common Control Physical Channel (P-CCPCH)
Forward Access Channel (FACH)	Secondary Common Control Physical Channel (S-CCPCH)
Paging Channel (PCH)	
Downlink Shared Channel (DSCH)	Synchronization Channel (SCH)
	Physical Downlink Shared Channel (PDSCH)
	Acquisition Indicator Channel (AICH)
	Access Preamble Acquisition Indicator Channel (AP-AICH)
	Paging Indicator Channel (PICH)
	CPCH Status Indicator Channel (CSICH)
	Collision-Detection/Channel-Assignment Indicator Channel (CD/CA-ICH)

4.3.11 Timing Relationship Between Physical Channels

The P-CCPCH, which carries a cell's SFN, serves as timing reference for all the physical channels, directly for downlink and indirectly for uplink. Figure 4.30 describes the frame timing of the downlink physical channels, for the AICH it includes the access slot timing. Uplink physical channels get their transmission timing from the received timing of the downlink physical channels. In general the following applies from [1]:

- SCH (primary and secondary), CPICH (primary and secondary), P-CCPCH, and PDSCH have identical frame timings.
- The S-CCPCH timing may vary for different S-CCPCHs, but the offset from the P-CCPCH frame timing is a multiple of 256 chips, i.e. $\tau_{\text{S-CCPCH},k} = T_k \times 256 \text{ chip}$, $T_k \in \{0, 1, \dots, 149\}$.

- The PICH timing is $\tau_{\text{PICH}} = 7680$ chips prior to its corresponding S-CCPCH frame timing, i.e. the timing of the S-CCPCH carrying the PCH transport channel with the corresponding paging information.
- AICH access slots #0 starts the same time as P-CCPCH frames with (SFN modulo 2) = 0.
- Any DPCH frame is associated to one PDSCH frame through the relation $46\,080 \text{ chips} \leq T_{\text{PDSCH}} - T_{\text{DPCH}} < 84\,480 \text{ chips}$.
- The DPCH timing may be different for different DPCHs, but the offset from the P-CCPCH frame timing is a multiple of 256 chips, i.e. $\tau_{\text{DPCH},n} = T_n \times 256 \text{ chips}$, $T_n \in \{0,1,\dots,149\}$.

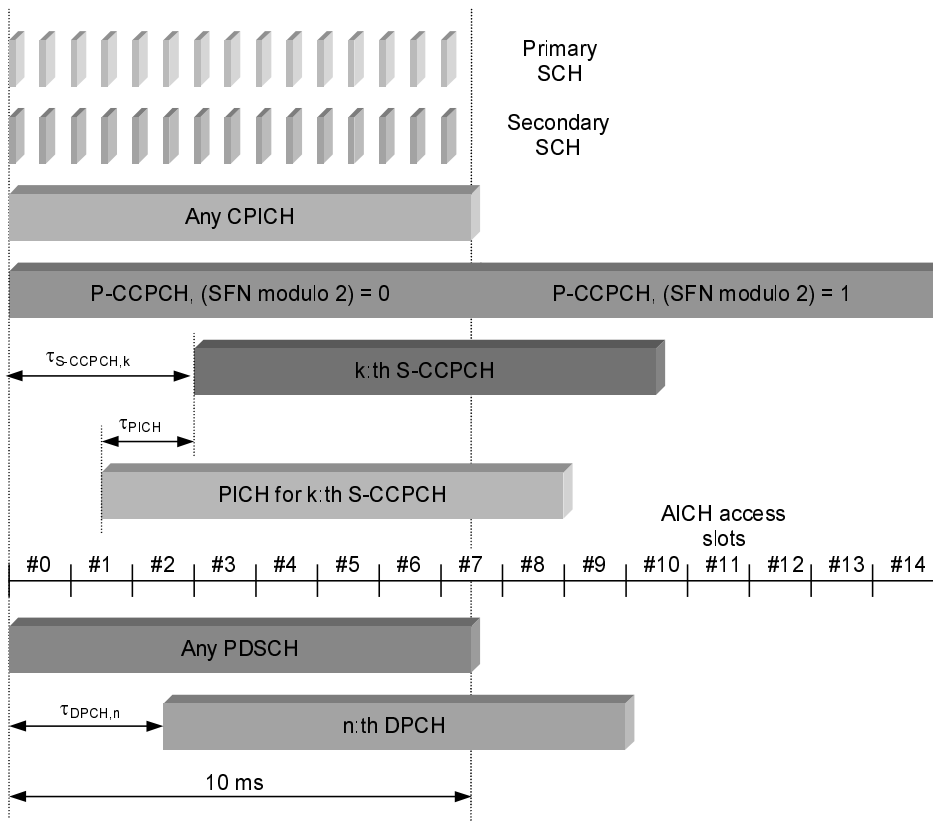


Figure 4.30 Frame timing and access slot timing of downlink physical channels.

4.3.11.1 PICH/S-CCPCH Timing Relation

Figure 4.31 illustrates the timing between a PICH frame and its associated S-CCPCH frame, i.e. the S-CCPCH frame carrying paging information related to the paging indi-

cators in the PICH frame. A paging indicator set in a PICH frame means that the paging message is transmitted on the PCH in the S-CCPCH frame starting τ_{PICH} chips after the transmitted PICH frame.

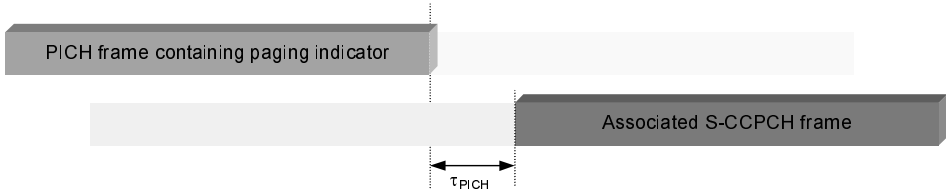


Figure 4.31 Timing relation between PICH frame and associated S-CCPCH frame.

4.3.11.2 PRACH/AICH Timing Relation

The downlink AICH has two downlink access slots, each with 5120 chips length and time aligned with the P-CCPCH. The uplink PRACH has uplink access slots, each with 5120 chips length. The UE transmits Uplink access slot number n , $\tau_{\text{p-a}}$ chips prior to the reception of downlink access slot number n , $n = 0, 1, \dots, 14$.

Downlink acquisition indicators may only start at the beginning of a downlink access slot. Likewise, transmission of uplink RACH preambles and RACH message parts may only start at the beginning of an uplink access slot.

Figure 4.32 illustrates the PRACH/AICH timing relation, where the preamble-to-preamble distance $\tau_{\text{p-p}}$ shall be larger than or equal to the minimum preamble-to-preamble distance $\tau_{\text{p-p,min}}$, i.e. $\tau_{\text{p-p}} \geq \tau_{\text{p-p,min}}$.

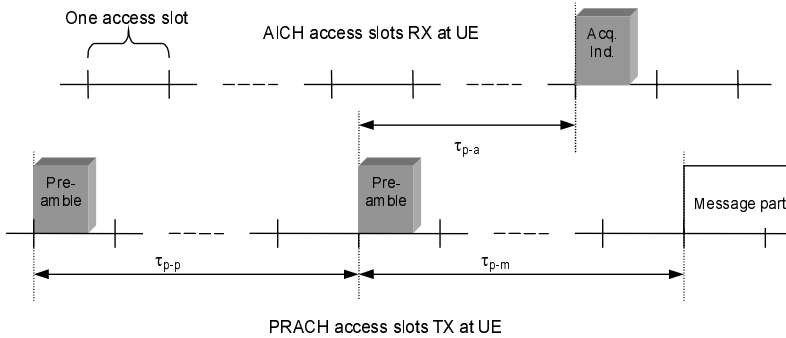


Figure 4.32 Timing relation between PRACH and AICH as seen at the UE.

In addition to $\tau_{\text{p-p,min}}$, [1] defines the *preamble-to-AI distance* $\tau_{\text{p-a}}$ and *preamble-to-message distance* $\tau_{\text{p-m}}$ as follows:

when *AICH_transmission_timing = 0, then	when *AICH_transmission_timing = 1, then
$\tau_{p-p, \min} = 15\,360$ chips (3 access slots)	$\tau_{p-p, \min} = 20\,480$ chips (4 access slots)
$\tau_{p-a} = 7680$ chips	$\tau_{p-a} = 12\,800$ chips
$\tau_{p-m} = 15\,360$ chips (3 access slots)	$\tau_{p-m} = 20\,480$ chips (4 access slots)
* Higher layers signal the parameter AICH_Transmission_Timing.	

4.3.11.3 PCPCH/AICH Timing Relation

AICH, the message and the PRACH/AICH have an identical timing relationship between preambles, where the collision resolution preambles follow the access preambles of the latter. The RACH preamble and AICH have the same timing relationship of the CD preamble and CD-ICH. Likewise, AICH to message in RACH and CD-ICH power control preamble in CPCH, have identical timing relationships. Finally, the PRACH/AICH transmission timing parameter and the T_{cpch} timing parameter are identical. See that a_1 corresponds to AP-AICH and a_2 corresponds to CD-ICH.

When $T_{cpch} = 0$ or $=1$, the following PCPCH/AICH timing values apply:

τ_{p-p}	=	Time to next available access slot, between access preambles Minimum time = $15\,360$ chips + 5120 chips $\times T_{cpch}$ Maximum time = 5120 chips $\times 12 = 61\,440$ chips Actual time = time to next slot (which meets minimum time criterion) in allocated access slot sub-channel group
τ_{p-a1}	=	Time between access preamble and AP-AICH has two alternative values: 7680 chips or $12\,800$ chips depending on T_{cpch}
τ_{a1-cdp}	=	Time between receipt of AP-AICH and CD preamble τ_{a1-cdp} transmission has a minimum value of: $\tau_{a1-cdp, \min} = 7680$ chips
τ_{p-cdp}	=	Time between the last AP and CD preamble. It has a minimum value of: $\tau_{p-cdp, \min} = 3$ or 4 access slots, depending on T_{cpch}
τ_{cdp-a2}	=	Time between the CD preamble and the CD-ICH has two alternative values: 7680 chips or $12\,800$ chips, depending on T_{cpch}
$\tau_{cdp-pcp}$	=	Time between CD preamble and the start of the power control preamble is either 3 or 4 access slots, depending on T_{cpch} .

The message transmission starts at 0 or 8 slots after the start of the power control preamble depending on the length of the power control preamble. Figure 4.33 illustrates the PCPCH/AICH timing relationship when $T_{cpch} = 0$ and all access slot sub-channels are available for the PCPCH.

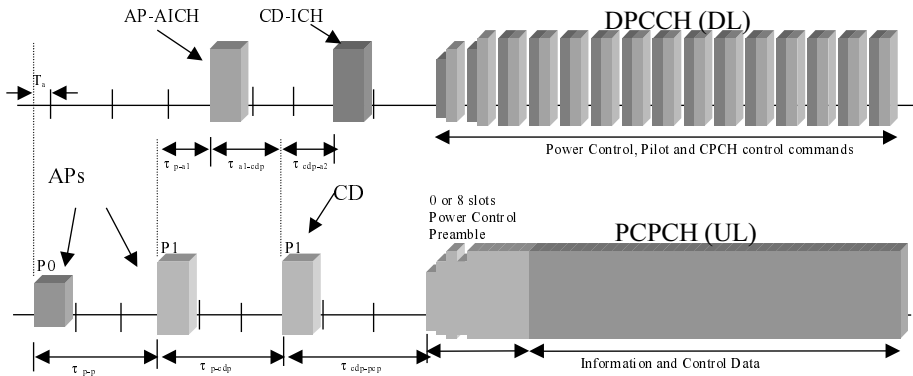


Figure 4.33 Timing of PCPCH and AICH transmission as seen by the UE, with $T_{\text{cpch}} = 0$.

4.3.11.4 DPCH/PDSCH Timing

Figure 4.34 illustrates relative timing between a DPCH frame and the associated PDSCH frame, where the start of a DPCH and of an associated PDSCH frames are denoted T_{DPCH} and T_{PDSCH} , respectively. Any DPCH frame associates itself to one PDSCH frame through the relation $46080 \text{ chips} \leq T_{\text{PDSCH}} - T_{\text{DPCH}} < 84480 \text{ chips}$, i.e. the associated PDSCH frame starts anywhere between three slot after the end of the DPCH frame up to 18 slots behind the end of the DPCH frame [1].

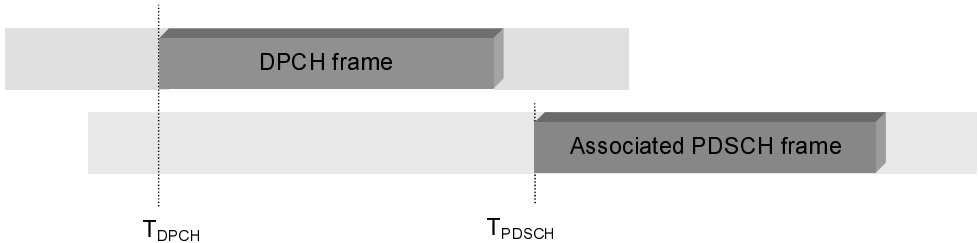


Figure 4.34 Timing relation between DPCH frame and associated PDSCH frame.

4.3.11.5 DPCCH/DPDCH Timing Relations

In the uplink the DPCCH and all the DPDCHs transmitted from one UE have the same frame timing. Likewise, in the downlink the DPCCH and all the DPDCHs carrying CCTrCHs of dedicated type to one UE have the same frame timing.

4.3.11.6 UE Uplink/Downlink Timing

At the UE, the uplink DPCCH/DPDCH frame transmission takes place approximately T_0 chips after the reception of the first significant path of the corresponding downlink

DPCCH/DPDCH frame, where T_0 is a constant defined as 1024 chips. Other timing relations for initialization of channels are in [1] and [5].

4.3.12 Downlink Spreading

All downlink physical channels (i.e. P-CCPCH, S-CCPCH, CPICH, AICH, PICH, PDSCH, and downlink DPCH) but excluding the SCH, follow the spreading operation represented by Figure 4.35. The non-spread physical channel consists of a sequence of three (i.e. +1, -1, and 0) real-valued symbols, where 0 indicates DTX. However, for the AICH, the symbol values depend on the exact combination of acquisition indicators to be transmitted.

In the first step each pair of two consecutive symbols pass from serial-to-parallel and get mapped to an I and Q branch, where even and odd numbered symbols are mapped to the I and Q branch, respectively. In all channels excluding AICH, we define symbol number zero as the first symbol in each frame. In the AICH, we define symbol number zero as the first symbol in each access slot. Then we spread I and Q branches at the chip rate by the same real-valued channelization code $C_{ch,SF,m}$. Afterwards, we treat the sequences of real-valued chips on the I and Q branch as a single complex-valued sequence of chips. This sequence of chips is scrambled (complex chip-wise multiplication) by a complex-valued scrambling code $S_{dl,n}$. For the P-CCPCH, we apply the scrambling code aligned with the P-CCPCH frame boundary, i.e. we multiply the first complex chip of the spread P-CCPCH frame with chip number zero of the scrambling code. For other downlink channels, we apply the scrambling code aligned with the scrambling code directed to the P-CCPCH. In this case, the scrambling code application does not align with the frame boundary of the physical channel under scrambling.

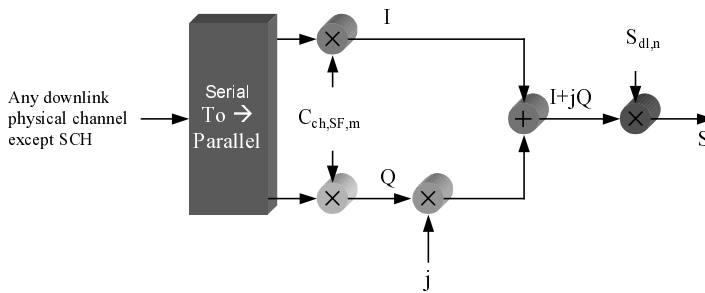


Figure 4.35 Spreading for all downlink physical channels, excluding the SCH.

4.3.12.1 Downlink Channelization Codes

The same uplink channelization codes presented in Section 4.3.4 apply to the downlink, i.e. Orthogonal Variable Spreading Factor (OVSF) codes that preserve the orthogonality between downlink channels of different rates and spreading factors. The specifications fix the channelization code for the primary CPICH to $C_{ch,256,0}$ and the channelization

code for the primary CCPCH to $C_{\text{ch},256,1}$. The UTRAN assigns the channelization codes for all other physical channels.

Spreading Factor (SF) 512 has specific restriction in its application. For example, if we use the code word $C_{\text{ch},512,n}$, with $n = 0, 2, 4, \dots, 510$ in soft handover, then we do not allocate the code word $C_{\text{ch},512,n+1}$ in the Node B, because we need the usage of timing adjustment. Likewise, if we use $C_{\text{ch},512,n}$, with $n = 1, 3, 5, \dots, 511$, then we do not allocate the code word $C_{\text{ch},512,n-1}$ in the Node B, again because we need time adjustment usage. However, this restriction does not apply to softer handover operation or when the UTRAN synchronizes to such a level that timing adjustments in soft handover are not applied in conjunction with SF 512.

If we do implement compressed mode by reducing the spreading factor by 2, the OVFSF code used for compressed frames is: $C_{\text{ch},\text{SF}/2,\lfloor n/2 \rfloor}$ when applying ordinary scrambling code, and $C_{\text{ch},\text{SF}/2,n \bmod \text{SF}/2}$ when applying alternative scrambling, where $C_{\text{ch},\text{SF},n}$ corresponds to the channelization code used for non-compressed frames.

If the OVFSF code on the PDSCH varies from frame to frame, the OVFSF codes allocation will occur in a way that the OVFSF code(s) below the smallest spreading factor will be from the branch of the code tree pointed by the smallest spreading factor used for the connection. This implies that all the UE-PDSCH codes connecting can originate according to the OVFSF code generation principle from smallest spreading factor code used by the UE on PDSCH.

When mapping the DSCH to multiple parallel PDSCHs, the same rule applies. However, all of the branches identified by the multiple codes, corresponding to the smallest spreading factor, may be used for higher spreading factor allocation [8].

4.3.12.2 Downlink Scrambling Codes

In principle, we can generate about $2^{18} - 1 = 262\,143$ scrambling codes, numbered $0, \dots, 262\,142$. However, we do not use all the scrambling codes. The specifications divide these scrambling codes into 512 sets each of a primary scrambling code and 15 secondary scrambling codes. The primary scrambling codes consist of $n = 16 \times i$ where $i = 0, \dots, 511$. The i th set of secondary scrambling codes consists of $16 \times (i + k)$, where $k = 1, \dots, 15$. There exists a one-to-one mapping between each primary scrambling code and 15 secondary scrambling codes in a set such that the i th primary scrambling code corresponds to the i th set of secondary scrambling codes.

Thus, based on the principles above, we can use $k = 0, 1, \dots, 8191$ scrambling codes, where for compressed frames, each of these codes are associated with a left alternative scrambling code and a right alternative scrambling code. The left alternative scrambling code corresponding to scrambling code k has scrambling code number $k + 8192$, while the right alternative scrambling code corresponding to scrambling code k has scrambling code number $k + 16\,384$. When we use alternative scrambling codes for compressed frames, left alternative scrambling code applies if $n < \text{SF}/2$ and the right alternative scrambling code applies if $n \geq \text{SF}/2$. Upper layers signal the usage of alternative

scrambling code in compressed frames for each physical channel. We use channelization code $c_{\text{ch,SF},n}$ for non-compressed frames.

The set of primary scrambling codes gets further divided into 64 scrambling code groups, each consisting of 8 primary scrambling codes. The j th scrambling code group consists of primary scrambling codes $16 \times 8 \times j + 16 \times k$, where $j = 0, \dots, 63$ and $k = 0, \dots, 7$. *Each cell receives one and only one primary scrambling code allocation.* The primary CCPCH and primary CPICH always use the primary scrambling code to transmit. On the other hand, the other downlink physical channels can use either the primary scrambling code or a secondary scrambling code from the set associated with the primary scrambling code of the cell to transmit. The combination of primary and secondary scrambling codes in one CCTrCH may be possible. However, if the CCTrCH has a DSCH type then all the PDSCH channelization codes that one UE may receive must be under a single scrambling code, i.e. either the primary or a secondary scrambling code.

The scrambling code sequences result from combining two real sequences into a complex sequence. We build each of the two real sequences as the position wise modulo 2 sum of 38 400 chip segments of two binary m sequences generated by means of two generator polynomials of degree 18. The resulting sequences thus constitute segments of a set of gold sequences. The scrambling codes are repeated for every 10 ms radio frame.

If we assume that x and y are two sequences, then x sequence originates from the primitive (over GF(2)) polynomial $1 + X^7 + X^{18}$, and y sequence results from using the polynomial $1 + X^5 + X^7 + X^{10} + X^{18}$. The sequence, which depends on the chosen scrambling code number n , we denote z_n , in the sequel. In addition, we let $x(i)$, $y(i)$ and $z_n(i)$ denote the i th symbol of the sequence x , y , and z_n , respectively. Then from the specifications in [8] we summarize the m sequences x and y as:

Initial conditions:

x is constructed with $x(0) = 1, x(1) = x(2) = \dots = x(16) = x(17) = 0$

$y(0) = y(1) = \dots = y(16) = y(17) = 1.$

Recursive definition of subsequent symbols:

$$x(i+18) = x(i+7) + x(i) \mod 2, \quad i = 0, \dots, 2^{18} - 20,$$

$$y(i+18) = y(i+10) + y(i+7) + y(i+5) + y(i) \mod 2, \quad i = 0, \dots, 2^{18} - 20.$$

The n th gold code sequence z_n , $n = 0, 1, 2, \dots, 2^{18} - 2$, is then defined as

$$z_n(i) = x((i+n) \mod (2^{18}-1)) + y(i) \mod 2, \quad i = 0, \dots, 2^{18} - 2. \quad (4.43)$$

These resulting binary sequences get converted to real valued sequences Z_n by the following transformation:

$$Z_n(i) = \begin{cases} +1 & \text{if } z_n(i) = 0 \\ -1 & \text{if } z_n(i) = 1 \end{cases} \text{ for } i = 0, 1, \dots, 2^{18} - 2. \quad (4.44)$$

Ultimately, we define the n th complex scrambling code sequence $S_{dl,n}$ as:

$$S_{dl,n}(i) = Z_n(i) + jZ_n((i + 131072) \bmod (2^{18} - 1)), \quad i = 0, 1, \dots, 38399, \quad (4.45)$$

where the pattern from phase 0 up to the phase of 38 399 gets repeated.

4.3.12.3 Synchronization Codes

The primary synchronization code (PSC), C_{psc} results from a so-called generalized hierarchical Golay sequence. It gets chosen to have good aperiodic autocorrelation properties. We express it as:

$$a = \langle x_1, x_2, x_3, \dots, x_{16} \rangle = \langle 1, 1, 1, 1, 1, 1, -1, -1, 1, -1, 1, -1, -1, 1, -1, 1 \rangle. \quad (4.46)$$

Thus, we generate the PSC by repeating the sequence a modulated by a Golay complementary sequence, and creating a complex-valued sequence with identical real and imaginary components. The PSC C_{psc} is defined as

$$C_{psc} = (1 + j) \times \langle a, a, a, -a, -a, a, -a, -a, a, a, -a, a, -a, a, a \rangle,$$

where the leftmost chip in the sequence corresponds to the chip transmitted first in time.

The 16 secondary synchronization codes (SSCs), $\{C_{ssc,1}, \dots, C_{ssc,16}\}$, are complex-valued with identical real and imaginary components, and are constructed from position wise multiplication of a Hadamard sequence and a sequence z , defined as

$$z = \langle b, b, b, -b, b, b, -b, -b, b, -b, b, -b, -b, -b, -b, -b \rangle,$$

where

$$b = \langle x_1, x_2, x_3, x_4, x_5, x_6, x_7, x_8, -x_9, -x_{10}, -x_{11}, -x_{12}, -x_{13}, -x_{14}, -x_{15}, -x_{16} \rangle$$

and $x_1, x_2, \dots, x_{15}, x_{16}$, are same as in the definition of the sequence a above.

The Hadamard sequences are obtained as the rows in a matrix H_8 constructed recursively by:

$$H_0 = (1) \\ H_k = \begin{pmatrix} H_{k-1} & H_{k-1} \\ H_{k-1} & -H_{k-1} \end{pmatrix}, \quad k \geq 1.$$

The rows are numbered from the top starting with row 0 (the all ones sequence).

Denote the n th Hadamard sequence as a row of H_8 numbered from the top, $n = 0, 1, 2, \dots, 255$, in the sequel.

Furthermore, let $h_n(i)$ and $z(i)$ denote the i th symbol of the sequence h_n and z , respectively where $i = 0, 1, 2, \dots, 255$ and $i = 0$ corresponds to the leftmost symbol.

The k th SSC, $C_{\text{SSC},k}$, $k = 1, 2, 3, \dots, 16$ is then defined as

$$C_{\text{SSC},k} = (1+j) \times \langle h_m(0) \times z(0), h_m(1) \times z(1), h_m(2) \times z(2), \dots, h_m(255) \times z(255) \rangle,$$

where $m = 16 \times (k - 1)$ and the leftmost chip in the sequence corresponds to the chip transmitted first in time.

The 64 secondary SCH sequences are constructed such that their cyclic shifts are unique, i.e. a non-zero cyclic shift less than 15 of any of the 64 sequences is not equivalent to some cyclic shift of any other of the 64 sequences. Also, a non-zero cyclic shift less than 15 of any of the sequences is not equivalent to itself with any other cyclic shift less than 15. Table 4.?? describes the sequences of SSCs used to encode the 64 different scrambling code groups. The entries in Table 4.?? denote what SSC to use in the different slots for the different scrambling code groups, e.g. the entry ‘7’ means that SSC $C_{\text{SSC},7}$ shall be used for the corresponding scrambling code group and slot. See SSCs for secondary SCH in [8].

4.3.13 Downlink Power Control Procedure

The network determines the transmit power of the downlink channels. Generally, the transmit power ratio between different downlink channels does not have specification and may change with time.

4.3.13.1 DPCCH/DPDCH Downlink Power Control

The downlink transmit power control procedure controls simultaneously the DPCCH power and that of its corresponding DPDCHs. The power control loop adjusts the power of the DPCCH and DPDCHs with the same relative power difference. The network determines the relative transmit power offset between DPCCH fields and DPDCHs. The TFCI, TPC and pilot fields of the DPCCH are offset relative to the DPDCHs power by PO1, PO2 and PO3 dB, respectively. The power offsets may vary in time.

The Downlink Power Control Function

The UE generates TPC commands to control the network¹ transmit power and send them in the TPC field of the uplink DPCCH. In the absence of UE is soft handover UE TPC command generated get transmitted in the first available TPC field in the uplink DPCCH. In the presence of soft handover, the UE checks the downlink power control mode (DPC_MODE)² before generating the TPC command as follows [11]:

- if DPC_MODE = 0, then the UE sends a unique TPC command in each slot and the TPC_cmd generated gets transmitted in the first available TPC field in the uplink DPCCH;

¹ As a response to the received TPC commands, UTRAN may adjust the downlink DPCCH/DPDCH power.

² The DPC_MODE parameter is a UE specific parameter controlled by the UTRAN.

- if $DPC\_MODE = 1$, then the UE repeats the same TPC_cmd over 3 slots and we transmit the new TPC_cmd aiming to have a new command at the beginning of the frame.

The average DPDCH power symbols³ transmitted over one time slot do not exceed the $Maximum\_DL\_Power$ (dBm), neither do they fall below the $Minimum\_DL\_Power$ (dBm). These two powers become power limits for one spreading code. When the UE cannot generate TPC commands due to lack of synchronization, the transmitted TPC command gets set to '1' during the period of out-of-synchronization.

Power changes occur in a multiples of the minimum step size $\Delta_{TPC,min}$ (dB), where it is mandatory for UTRAN to support $\Delta_{TPC,min}$ of 1 dB, but optional to support 0.5 dB. The UTRAN may further employ the following method.

When the *limited power raise* parameter value applies, the UTRAN will not increase the DL power of the Radio Link (RL) if it would exceed by more than $Power\_Raise\_Limit$ (dB) the averaged DL power used in the last $DL\_Power\_Averaging\_Window\_Size$ time slots of the same RL. The latter applies only after the first $DL\_Power\_Averaging\_Window\_Size$ time slots preceding the activation of this method. The $Power\_Raise\_Limit$ and $DL\_Power\_Averaging\_Window\_Size$ parameter configuration occur in the UTRAN [11].

4.3.13.2 Power Control in the Downlink Compressed Mode

Compressed mode or slotted power control in uplink or/and downlink aims to recover as fast as possible the SIR close to the target SIR after each transmission gap. Practically, compressed mode intervenes when making measurements into other frequency ranges from single mode WCDMA systems. The same UE behaviour of the preceding section applies to the compressed mode. Since the specifications do not describe the details the UTRAN behaviour during the compressed downlink mode, algorithms of the UL compress mode may apply. Downlink DPCCCH and DPDCH(s) transmission stops during DL compressed mode or in simultaneous DL and UL compressed mode. As [14] puts it, the use of compressed mode has an impact on the link performance as studied in [15] for the uplink compressed mode and for the downlink in [16]. The largest impact occurs at the cell edge, where the difference in the uplink performance between compressed mode and non-compressed mode cases is very small until headroom is less than 4 dB.

4.3.13.3 Power Control in Site Selection Diversity Transmit (SSTD)

The UE in SSTD, the optional macro diversity method in soft handover mode, selects one of the cells from its active set to be 'primary', all other cells are classed as 'non primary'. In this context, there are two goals, first to transmit on the downlink from the primary cell, minimizing thereby the interference resulting from multiple transmissions in a soft handover mode. Second, to achieve fast site selection without network intervention, thereby maintaining the advantage of the soft handover. To select a primary

³ Transmitted DPDCH symbols imply complex QPSK symbols before spreading which does not contain DTX.

cell, each cell gets a temporary identification (ID) and the UE periodically informs a primary cell ID to the connecting cells via the uplink FBI field. The non-primary cells selected by the UE, switch off their transmission power. Upper layer signalling carries out SSDT activation, termination and ID assignment. See [11] for details on cell identification.

4.3.13.4 Power Control in The PDSCH, AICH, PICH, and S-CCPCH

- The network can select *inner(fast) loop power control* based on the power control commands sent by the UE on the uplink DPCCH or *slow power control* to realize the PDSCH power control.
- The UE gets information about the relative transmit power of the AICH⁴, compared to the primary CPICH transmit power by the higher layers.
- The UE gets also information about the relative transmit power of the PICH⁵, compared to the primary CPICH transmit power by the higher layers.
- The TFCI and pilot fields may have time-varying offset relative to the power of the data field.

4.3.14 The Compressed Mode Procedure

In the compressed mode we do not use TGL slots from N_{first} to N_{last} for data transmission. As shown in Figure 4.36 the instantaneous compress-frame transmit power increases to keep quality (e.g. BER, FER, etc.) despite reduced processing gain. The power increase depends on the transmission time reduction method under the network decision. Compressed frames may occur periodically or on demand. Compressed-frame rate and type varies and depends on the environment and the measurement requirements.

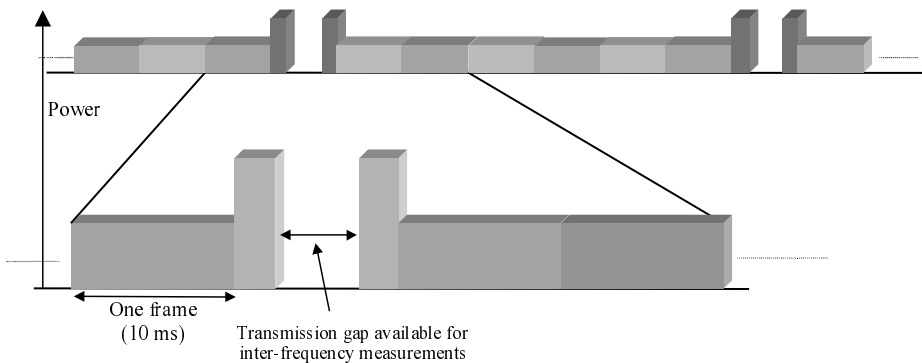


Figure 4.36 Compressed mode transmission (after [10]).

⁴ Measured as the power per transmitted acquisition indicator.

4.3.14.1 Uplink Frame Structure

Figure 4.37 illustrates the UL compressed mode structure showing the control slots and data transmission slots.

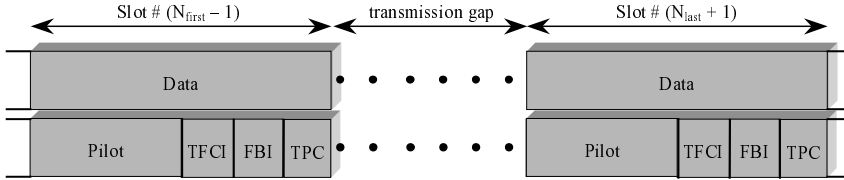
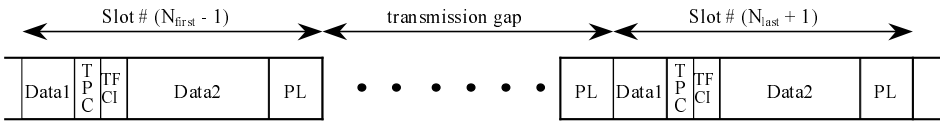


Figure 4.37 Uplink compressed frame structure.

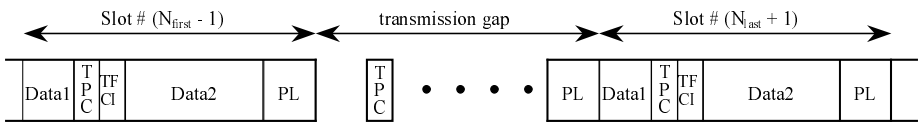
4.3.14.2 Downlink Frame Structure

We have two different types of downlink frame structures. Type A maximizes the TGL and type B optimizes power control.

- In type A we transmit the pilot field of the last slot in the transmission gap, and transmission gets turned off during the remaining of the transmission gap, see Figure 4.38a.
- In type B we transmit the TPC field of the first slot and the pilot field of the last slot in the transmission gap. As in type A, transmission gets turned off during the remaining of the transmission gap, see Figure 4.38a.



(a) Frame structure type A



(b) Frame structure type B

Figure 4.38 Frame structure types in downlink compressed transmission.

4.3.14.3 Transmission Time Reduction Method

In compressed mode we transmit in less time the information typically transferred in 10 ms frames. We achieve the latter through puncturing, halving spreading factor, and higher layer scheduling. The DL compression supports all methods, while the DL ex-

⁵ Measured as the power over the transmitted paging indicators, excluding the undefined part of the PICH frame.

cludes compression by puncturing. We define the maximum idle length as 7 slots per one 10 ms frame.

By puncturing. Puncturing compression takes place through rate matching while using the rate matching or puncturing algorithm for rate matching.

By halving the SF. In this mode we halve the SF during one radio frame to enable the transmission of the information bits in the remaining time slots of a compressed frame. In the DL the UTRAN may also command the UE to use a different scrambling code than the usual one. If the latter occurs then there exists a one-to-one mapping between the scrambling code used in normal mode and the one used in compressed mode.

Compression through higher layer scheduling. Higher layers set conditions so that only a subset of the allowed TFCs can apply in compressed mode.

4.3.14.4 Transmission Gap Position

We can place transmission gaps at different positions (Figure 4.39) for inter-frequency power measurement, acquisition of the control channel of another system/carrier, and actual handover operation. Thus, when using the single frame method, we locate the transmission gap within the compressed frame depending on the transmission gap length (TGL) as illustrated in Figure 4.39a. When applying the double frame method, we locate the transmission gap at the centre of two connected frames as illustrated in Figure 4.39b.

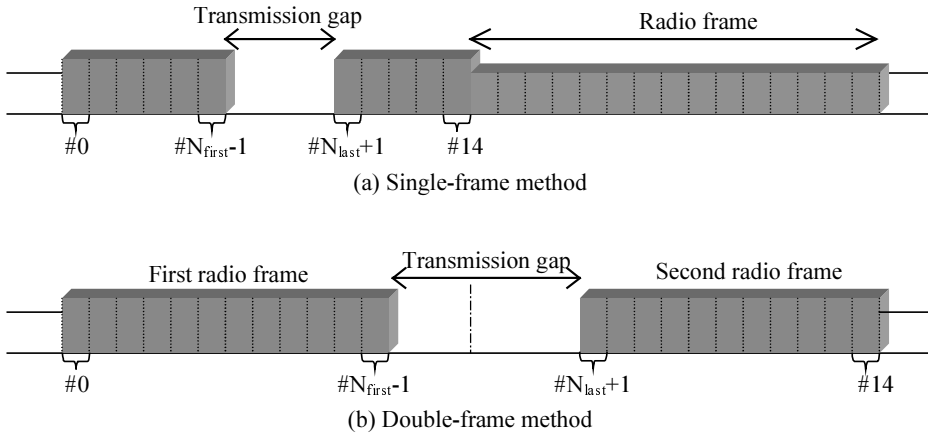


Figure 4.39 Transmission gap position [10].

We calculate the parameters of the transmission gap positions as follows: TGL is the number of consecutive idle slots during the compressed mode transmission gap, i.e.

$$\text{TGL} = 3, 4, 5, 7, 10, 14.$$

N_{first} specifies the starting slot of the consecutive idle slots,

$$N_{\text{first}} = 0, 1, 2, 3, \dots, 14.$$

N_{last} shows the number of the final idle slot and is calculated as follows:

- If $N_{\text{first}} + \text{TGL} \leq 15$, then $N_{\text{last}} = N_{\text{first}} + \text{TGL} - 1$ (in the same frame),
- If $N_{\text{first}} + \text{TGL} > 15$, then $N_{\text{last}} = (N_{\text{first}} + \text{TGL} - 1) \bmod 15$ (in the next frame).

When the transmission gap spans by two consecutive radio frames, we choose N_{first} and TGL so that at least 8 slots in each radio frame get transmitted. Table 4.31 illustrates the detailed parameters for each transmission gap length for the different transmission time reduction methods.

Table 4.31 Parameters for Compressed Mode⁶ [10]

TGL	Frame type	Spreading factor	Idle length (ms)	Transmission time reduction method	Idle frame combining
3	A	512–4	1.73–1.99	Puncturing Spreading factor division by 2 or Higher layer scheduling	(S) = Single frame method (D) = (1,2) or (2,1)
	B	256–4	1.60–1.86		
4	A	512–4	2.40–2.66		(S) (D) = (1,3), (2,2) or (3,1)
	B	256–4	2.27–2.53		
5	A	512–4	3.07–3.33		(S) (D) = (1,4), (2,3), (3, 2) or (4,1)
	B	256–4	2.94–3.20		
7	A	512–4	4.40–4.66		(S) (D) = (1,6), (2,5), (3,4), (4,3), (5,2) or (6,1)
	B	256–4	4.27–4.53		
10	A	512–4	6.40–6.66		(D) = (3,7), (4,6), (5,5), (6,4) or (7,3)
	B	256–4	6.27–6.53		
14	A	512–4	9.07–9.33		(D) = (7,7)
	B	256–4	8.93–9.19		

S, single-frame method as illustrated Figure 4.39a; D, double-frame method as illustrated in Figure 4.39a; (x,y) indicates: x, the number of idle slots in the first frame; y, the number of idle slots in the second frame.

4.3.15 Handover Procedures

The essential FDD mode handover types include: intra-mode handover (i.e. soft, softer, and hard⁷ handover); inter-mode handover (i.e. handover to TDD); intersystem handover (e.g. handover to GSM).

Intra-mode handover. This handover depends on CPICH⁸ power level measurements, which include the Received Signal Code Power (RSCP), Received Signal Strength Indicator (RSSI), and the E_c/N_o resulting from the RSCP/RSSI ratio. The other measurement involved is the relative timing information between the cells. Cells within a 10 ms window generally get relative timing from the primary scrambling code phase since the code period used is 10 ms. If large inaccuracies occur, the terminal decodes the System Frame Number (SFN) from the primary CCPCH.

⁶ Compressed mode by spreading factor reduction is not supported when SF = 4 is used in normal mode.

⁷ Hard handover may also take place as intra- or inter-frequency handover.

⁸ That is the E_c/N_o measurement performed at the common pilot channel.

Inter-mode handover. Multimode or dual FDD-TDD mode terminals will afford inter-mode handover. While in the FDD, a MS will measure the power level in synchronized TDD cells with useful reference midambles to execute the handover process.

Inter-system handover. Intersystem handover for UTRA takes an important proportion of all the interoperability and system integration tasks. It plays an important role in the evolution process of 2nd and 3rd generation mobile systems. In particular, UTRA needs to inter-operate seamlessly with the family of IMT2000 networks and co-exist for some time with all types of deployment scenarios. However, complete interoperability will not necessarily occur at the introduction of UMTS, but will follow a gradual process. Thus, in this section we mainly discuss essential handover issues with GSM. Handover with other systems such as MC-CDMA and IS-136, e.g. will not be cover at this time.

We introduce *inter-system handover* based on the handover recommendations from GSM to UMTS. Specifications in this area continue at this writing. Thus, here we look at it primarily from the requirement side. The handover may consist of the following aspects [19]: cell selection/reselection and handover features.

4.3.15.1 Cell Selection/Reselection Requirements

The MS compares GSM quality with UMTS neighbour cells to determine the most appropriate cell for cell selection/reselection. The MS in a GSM cell obtains system information of its UMTS neighbour cells through BSC broadcasting all the required information in the GSM cell. Broadcasting UMTS cell information depends on the definition of new system information messages or the adaptation of existing system information messages together with the transmission of area-based UMTS system information (e.g. UTRAN frequency/ies used in LA) [19].

4.3.15.2 Handover Requirements

The requirements for GSM to UMTS handover as an extract from [19] can be outlined as follows:

1. Synchronization requirements

- a MS will synchronize with a UTRAN cell using GSM idle frame(s)
- a MS will be capable of blind detection; thus, it will not reject a handover command to an UMTS or GSM cell which it has not reported and to which it is not synchronized

2. GSM MS requirements

- R99 and newer MSs will support a 'blind handover' to GSM or UMTS
- when a handover fails, the MS will remain camped onto the original cell, and continue its measurement reporting as defined prior to the attempted handover.

3. GSM BSC requirements

- the call will go to the most suitable cell for the service that the user requested

- the BSC needs to know the service in progress to determine suitable cells to direct the call to
 - the BSC will provide the MS with mapping parameters, which will enable the MS to obtain the set of most suitable cells for the measurement report
 - the BSC may provide the MS with the Q search parameters to trigger measurement of other radio access technologies; a separate bit may be used to indicate whether the UMTS measurements are triggered when the GSM RXLEV measurement for the current cell is below or above Q search
 - in GSM to UMTS handover, it is the working assumption that the source GSM BSC shall provide the ID of the target RNC; it is also assumed that the source RNC to target RNC transparent container will be created by the BSC
 - in order to optimize non-synchronized ('blind') handover, the GSM BSC shall provide additional information about the target cell (e.g. scrambling code, synchronization)
4. *UMTS RNC*
- the RNC will broadcast the parameter M-offset on its BCCH; this offset is set by the network, and is used to adjust the comparison of UMTS with GSM measurements
5. *UMTS measurements from GSM MS*

The requirements on the measurements made by the MS are:

- measurement on UTRAN cells by the MS will not have a significant impact on the measurement ability and performance of the MS for support of GSM to GSM handover; it is assumed that the MS uses search frames for UTRAN measurements and that the UTRAN cells are only monitored during idle search frames
 - the maximum time for detecting a new suitable UTRAN cell relates to the number of UTRAN frequencies under monitoring
 - the time it takes to detect, confirm BSIC and report a new suitable GSM cell applies also for detecting and reporting a new suitable UTRAN cell when one UTRAN frequency remains under monitoring
 - the UE, when in connected mode, does not need to read BCCH on the UTRAN cells under measurement
 - the operator will be able to provide the UE with information that enables the UE to activate the actual physical measurements only when considered needed; e.g. when the quality of the GSM cell falls below a certain threshold
6. *Measurement reports*

The requirements on the measurement reporting are:

- the UE will include both UTRAN and GSM measurements in the periodic measurement reports which are sent to the BSS

- the operator will have the capability to control the minimum number of GSM cells in the serving band, the minimum number of cells in UMTS, and the minimum number of cells in other GSM bands which are included in the measurement report
- if there is any space remaining in the measurement report after inserting the minimum number of entries for each reporting type, the MS will use this space to insert further measurements in decreasing order of priority
- both the existing format measurement report, and the extended format measurement report (e.g. 76/00 and 2B00-009) will be supported by the MS
- Release 99 and newer MSs will support extended measurement reporting
- the network will inform the MS whether extended measurement reporting is supported; the default measurement reporting type is the normal measurement reporting

7. Signalling

The signalling between the GSM network and UMTS network to perform handover needs the following modifications:

- the GSM system will provide the UMTS system with the target ID of the RNC to which the call is being directed

Table 4.32 Allowed Handover for GSM Release 99 [19]

	GSM CS	GSM GPRS	GSM ECSD	GSM EGPRS nRT	GPRS COMPACT	UTRA FDD	UTRA TDD	Multi-Carrier CDMA	IS-136
GSM CS	✓	✗	✓	✗	✗	✓	✓	✓	✗
GSM GPRS	✗	✗	✗	✗	✗	✗	✗	✗	✗
GSM ECSD	✓	✗	✓	✗	✗	✓	✓	✓	✗
GSM EGPRS nRT	✗	✗	✗	✗	✗	✗	✗	✗	✗
GSM GPRS COMPACT	✗	✗	✗	✗	✗	✗	✗	✗	✗

✓, handover permitted; ✗, handover not permitted.

4.3.16 Other FDD Mode Physical Layer Procedures

Paging Channel (PCH). The MS gets a Paging Indicator (PI) belonging to a paging group once it registers in a network. The PI appears periodically on the Paging Indicator Channel (PICH) whenever paging messages exists, and the MS decodes⁹ the next PCH frame transmitted on the secondary CCPCH, seeking for the messages corresponding to it.

⁹ Battery life duration will increase with the lowest amount of PI detection events.

The RACH. To cope with the power control uncertainty and near-far impacts, e.g. the following events correspond to the RACH procedures in the terminal: (1) identifying and eventual selection of available RACH sub-channels with scrambling codes and signatures through BCH decoding; (2) measurement of DL power level and setting of initial RACH power level; (3) sending of 1 ms RACH preamble with chosen signature; (4) AICH decoding to verify preamble detection¹⁰ by the BS; (5) transmitting¹¹ 10 or 20 ms RACH message part at the AICH detection.

CPCH. The CPCH follows basically the same events as the RACH, differing only on L1 collision detection (see [11]). Applying fast power control on the CPCH we minimize interference due to the data transmission.

Cell search. The cell search procedure employing the synchronization channel uses different scrambling codes with different phase shifts of the code. The events include: (A) searching 256 chips primary synchronization code;¹² since the latter is the same in every slot, the peak detected corresponds to the slot boundary; (B) on the peak detection of the primary synchronization code, the MS will look at the largest peak from the secondary SCH code word, i.e. from among the 64 options; (C) at the detection of the secondary SCH code word the frame timing is known.

4.4 CONFIGURATION OF TDD PHYSICAL CHANNELS

TDD physical channels illustrated in Figure 4.40 have three-layer structure with respect to time slots (TS), radio frames and system frame numbering (SFN). The radio frame configurations or time slots vary according to the resource allocation. All physical channels use guard symbols in every time slot. The latter serve as the TDMA component to separate different user signals in time and code domains.

A TDD physical channel is burst (i.e. a combination of a data part, a midamble and a guard period), transmitted in a particular time slot within allocated radio frames. A burst lasts one time slot and its allocation can be either continuous (i.e. in every frame), or discontinuous (i.e. only one in a subset of radio frames). Several bursts can be transmitted at the same time from one transmitter. In this case, the data part must use different OVSF channelization codes, but the same scrambling code. The midamble part has to use the same basic midamble code, but can use different midambles [6].

The data part of the burst has a combined spread of channelization and scrambling codes. The OVSF channelization code can have a spreading factor of 1, 2, 4, 8, or 16, where the data rate of the physical channel will depend on the spreading factor used.

The midamble part of the burst may contain two different types of midambles, i.e. a short one with a length of 256 chips, or a long one with 512 chips. The midamble size also has an impact on the data rate of the physical channel.

¹⁰ In the absence of AICH the terminal resends preamble with higher power in the next available slot.

¹¹ When the RACH transmits data, the SF and thereby the data rate fluctuate.

¹² Identical to all cells.

Thus, we define a TDD physical channel by frequency, time slot, channelization code, burst type and radio frame allocation. Scrambling and basic midamble codes broadcast may be constant within a cell. After the physical channel establishment a frame start event occurs with infinite or limited duration.

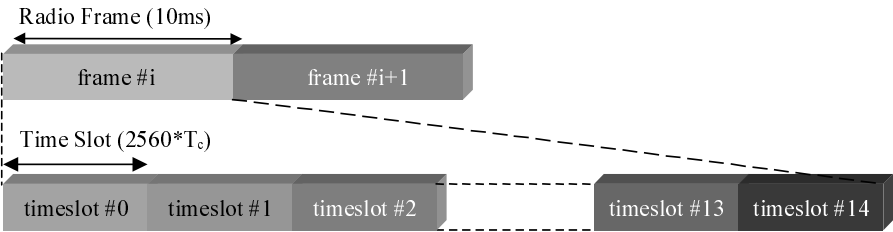


Figure 4.40 Physical channel signal format.

4.4.1 Frame Structure

As in the FDD mode, a TDMA frame in the TDD mode has a duration of 10 ms with a sub-division into 15 time slots (TS) of $2560 \times T_c$ duration each. Hence, a TS corresponds to 2560 chips, each allocated to either the uplink or the downlink as illustrated in Figure 4.41. This flexibility allows the TDD mode to adapt itself to different environments and deployment scenarios. Nonetheless, in any configuration there must be at least one time slot on the downlink and at least one time slot in the uplink.

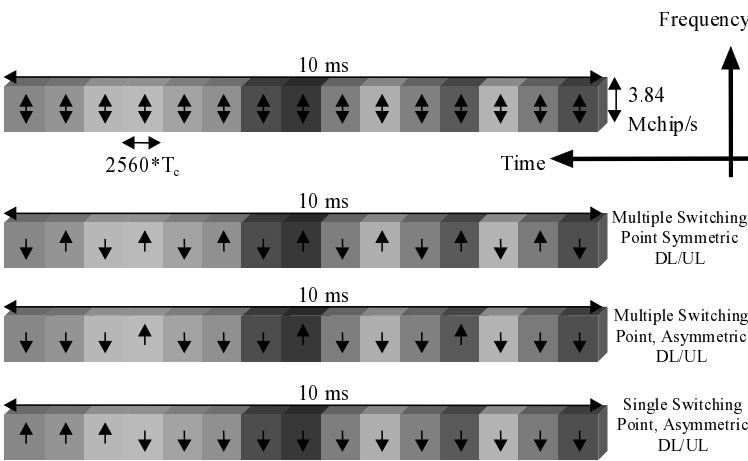


Figure 4.41 The TDD frame structure.

4.4.2 Dedicated Physical Channel (DPCH)

4.4.2.1 Downlink and Uplink Physical Channel Spreading

We map DCH onto the dedicated physical channel. The two step spreading of the data

part in the physical channels include channelization and scrambling operations. The first operation transforms every data symbol into a number of chips, thereby increasing the bandwidth of the signal. We call this number of chips per data symbol the Spreading Factor (SF). The second operation scrambles the spread signal through a scrambling code [7].

Downlink physical channels use SF = 16 as described in [7], and operation with a single code with SF = 1 can apply to downlink physical channels. To support higher data rates we can utilize multiple parallel physical channel transmission using different channelization codes.

Uplink physical channels have SF ranging from 16 down to 1. In multi-code transmission a UE simultaneously uses two physical channels per time slot maximum. These parallel physical channels transmit using different channelization codes [7].

4.4.3 Burst Types

The two types of dedicated physical channel bursts, i.e. Burst1 (B1) and Burst2 (B2), consist of two data symbol fields, a midamble and a guard period. Burst1 because of its longer midamble of 512 chips, suits the uplink better by allowing up to 16 channel estimations. Burst2, which has only 256 chips, applies more to the downlink. We can summarize the burst use as follows:

	Burst1	Burst2
Uplink	Independent of the number of active users in one time slot	When bursts within a time slot are allocated to < 4 users
Downlink	Independent of the number of active users in one time slot	Independent of the number of active users in one time slot

Burst1 has data fields of 976 chips, whereas Burst2 has data fields of 1104 chips. The corresponding number of symbols depends on the spreading factor, as seen in Table 4.33. Both bursts have 96 chip long Guard Periods (GP).

Table 4.33 TDD Burst Characteristics

SF	Symbols per data field		Chip number		Field length		Field content (symbols)	
	B1	B2	B1	B2	B1	B2	B1	B2
1	976	1104	0–975	0–1103	976	1104	Data	Data
2	488	552	976–1487	1104–1359	512	256	Mid	Mid
4	244	276	1488–2463	1360–2463	976	1104	Data	Data
8	122	138	2464–2559	2464–2559	96	96	GP	GP
16	61	69						

The two different bursts illustrated in Figure 4.42 and characteristics noted in Table 4.33, can support a different set of applications and also allow optimization for particular operational environments within the unlicensed frequency range.

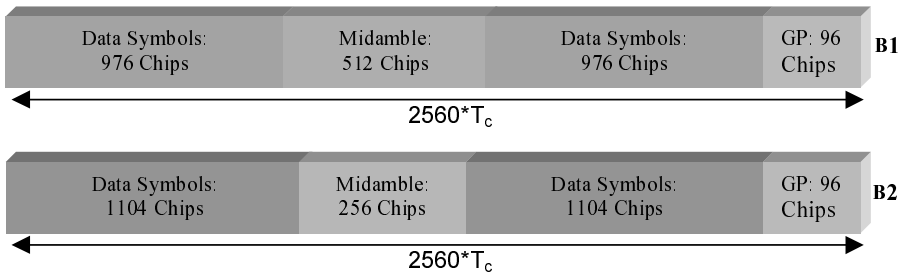


Figure 4.42 TDD burst types.

4.4.3.1 Transmission of TFCI

Both bursts (B1 and B2) afford uplink and downlink TFCI transmission. This transmission is negotiated at call setup and re-negotiation may occur during a call. Upper layer signalling indicates TFCI formats for each CCTrCH with information on the presence or absence of TFCI. When a time slot contains a TFCI, transmission takes place using the first allocated channelization code in the time slot.

The data parts of a corresponding physical channel realize the TFCI transmission following the same spreading procedures outlined in [7], and while keeping the midamble structure illustrated in Figure 4.42. We transmit the TFCI information directly adjacent to the midamble, and after the TPC in the presence of power control commands. Figure 4.43 illustrates the two cases

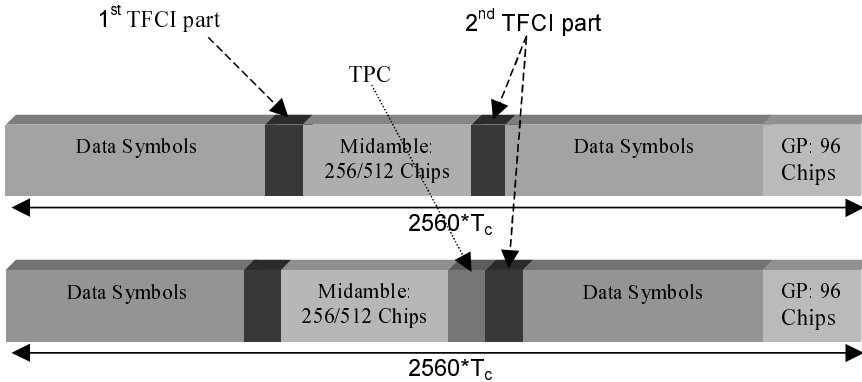


Figure 4.43 TFCI information positions in traffic bursts.

Both burst types 1 and 2 for dedicated channels provide the possibility for transmission of TPC in uplink.

The transmission of TPC is negotiated at call setup and can be re-negotiated during the call. If applied, transmission of TPC is done in the data parts of the traffic burst. Hence the midamble structure and length is not changed. The TPC information is to be transmitted directly after the midamble. Figure 4.43 shows the position of the TPC in a traffic burst.

For every user the TPC information is to be transmitted once per frame. If the TPC is applied, then it is always transmitted using the first allocated channelization code and the first allocated time slot, according to the order in the higher layer allocation message. The TPC is spread with the same Spreading Factor (SF) and spreading code as the data parts of the respective physical channel. Specifications in [6] cover time slot formats and training sequences for spread bursts.

4.4.3.2 *Midamble Transmit Power and Beam-forming/Transmit Diversity*

When all one-time slot downlink users have a common midamble, this common midamble’s transmit power has no power offset between the data part and the midamble part of the transmit signal within the given slot. Likewise, transmit power of users with specific midambles does not have power offset between the data parts and the midamble part. In the event of DL beam-forming or Tx diversity, the user who has beam-forming/Tx diversity and a dedicated channel, will get one individual midamble.

4.4.4 **Common Physical Channels**

4.4.4.1 *Primary and Secondary Common Control Physical Channels*

We map the BCH onto the Primary Common Control Physical Channel (P-CCPCH), and obtain the position (time slot/code) of the P-CCPCH from the Physical Synchronization Channel (PSCH).

We also map PCH and FACH onto one or more Secondary Common Control Physical Channels (S-CCPCH). Through the PCH, the FACH adapts itself to different requirements. Table 4.34 summarizes the key P-CCPCH and S-CCPCH features.

Table 4.34 P-CCPCH Features

	P-CCPCH	S-CCPCH
Spreading	It uses fixed spreading with a spreading factor SF = 16 and always channelization code $c_{Q=16}^{(k=1)}$	It uses fixed spreading with a spreading factor SF = 16
Burst types	Burst 1 type with no TFCI applies	Utilizes bursts type 1 and 2 with TFCI
Training sequences, i.e. midambles	Time slots carrying P-CCPCH transmission use midambles $m^{(1)}$, $m^{(2)}$, $m^{(9)}$ and $m^{(10)}$ in order to support block STTD antenna diversity and the beacon function; see a description in [6].	The training sequences described in [6] apply to the S-CCPCH
Block STTD antenna diversity	Its support is mandatory for the UE as follows: (1) if no antenna diversity, it uses $m^{(1)}$ while $m^{(2)}$ remains unused; (2) if in block STTD antenna diversity exists, the first antenna uses $m^{(1)}$ and the diversity antenna uses $m^{(2)}$	

4.4.5 The Physical Random Access Channel (PRACH)

The RACH maps onto one or more uplink physical random access channels (PRACH) affording thereby flexible and scalable capacity to the RACH.

	PRACH
Spreading	The uplink PRACH uses either spreading factor $SF = 16$ or $SF = 8$, where the BCH broadcasts the set of admissible spreading codes and the associated spreading factors.
Burst types	Mobiles send uplink access bursts randomly through the PRACH, which has a burst consisting of two data symbol fields, a midamble and a guard period as seen in Figure 4.44. The 2nd data symbol field has only 880 chips instead 976 to allow larger guard time [6]. The collision groups depend on the selected RACH configuration. For example, with $SF = 8$, the first data field has 122 symbols and the second 110. Likewise, with $SF = 16$, field1 has 61 and field2 55 symbols, respectively. Table 7 in [6] illustrates the PRACH burst field content.
Training Sequences (TS)	Different active users in the same time slot have time-shifted TS or midamble versions of single periodic basic code described in [6]. We choose time shifts from <i>all</i> $k = 1, 2, 3, \dots, K'$ (for cells with small radius) or <i>uneven</i> $k = 1, 3, 5, \dots, \leq K'$ (for cells with large radius).
TS and channelization code association	We base the generic rule to define this association on the channelization codes $\mathbf{c}_Q^{(k)}$ order given by k and midambles $\mathbf{m}_j^{(k)}$ order given by k , firstly, and j , secondly, with the constraint that the midamble for a spreading SF factor is the same as in the upper branch for the spreading factor $2SF$. The index $j = 1$ or 2 indicates whether we use the original basic midamble sequence ($j = 1$) or the time-inverted basic midamble sequence ($j = 2$) [6].

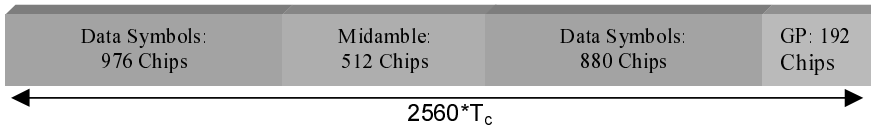


Figure 4.44 PRACH burst configuration.

4.4.6 The Synchronization Channel (SCH)

The synchronization channel provides the code group of a cell. To prevent uplink/downlink asymmetry limitations we map the SCH on one or two downlink slots per frame only. The two cases of SCH and P-CCPCH allocation include: first, SCH and P-CCPCH allocated in $TS\#k$, $k = 0, \dots, 14$; and second SCH allocated in two TS ($TS\#k$ and $TS\#k + 8$, $k = 0, \dots, 6$; P-CCPCH allocated in $TS\#k$). The position of SCH (value of k) in the frame can change in the long term in either of the two cases and allow knowledge of the position of P-CCPCH from the SCH. Specifications in [6] and [7] give more details for the SCH.

4.4.7 Physical Uplink/Downlink Shared Channels

The Physical Uplink Shared Channel (PUSCH), which provides uplink TFCI transmission possibilities, uses the DPC burst structure, where user specific physical layer parameters, e.g. power control, timing advance or directive antenna settings come from the associated channel (i.e. FACH or DCH).

The Physical Downlink Shared Channel (PDSCH), which provides downlink TFCI transmission possibilities, uses the DPCH burst structure. As in the PUSCH, specific L1 parameters, e.g. power control or directive antenna settings come from the associated channel (FACH or DCH).

The DSCH utilizes three signalling methods to inform the UE that it has data to decode: (a) using the TFCI field of the associated channel or PDSCH; (b) using the DSCH user specific midamble derived from the set of midambles used for that cell; and (c) using higher layer signalling. In the last method, the UE decodes the PDSCH if the PDSCH was transmitted with the midamble assigned to the UE by UTRAN.

4.4.8 The Page Indicator Channel (PICH)

The Page Indicator Channel¹³ (PICH) carries the Page Indicators (PI), which indicate a paging message for one or more UEs associated with it, and is always transmitted at the same reference power level as the P-CCPCH. The PICH substitutes one or more paging sub-channels mapped on a S-CCPCH.

Figure 4.45 illustrates normal bursts that carry PIs of length $L_{PI} = 2$, $L_{PI} = 4$ or $L_{PI} = 8$ symbols, and Table 4.35 illustrates the number of page indicators N_{PI} per time slot given by the number L_{PI} symbols for the page indicators and the burst type.

Table 4.35 Number N_{PI} of PI per Time Slot for the Different Burst Types and PI Lengths L_{PI}

	$L_{PI} = 2$	$L_{PI} = 4$	$L_{PI} = 8$
Burst type 1	61	30	15
Burst type 2	69	34	17

The same burst type is used for the PICH in every cell. As illustrated in Figure 4.45 when $L_{PI} = 4$ or $L_{PI} = 8$, we leave behind one symbol in each data part adjacent to the midamble and fill it by dummy bits transmitted with the same power as the PI [6].

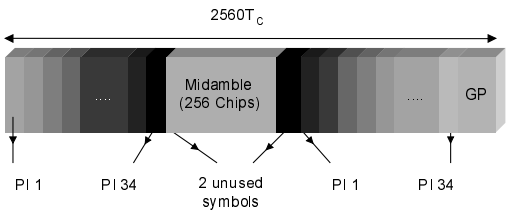


Figure 4.45 Example of PI transmission in PICH bursts ($L_{PI} = 4$).

¹³ Physical channel.

4.4.9 Beacon Function in Physical Channels

Depending on its allocation case, the SCH determines the location of the physical channels with beacon function for the purpose of measurements. In Case 1, all physical channels with channelization code $c_{0-16}^{(k+1)}$ and in TS# k , $k = 0, \dots, 14$ allocation provide the beacon function. In Case 2 all physical channels with channelization code $c_{0-16}^{(k+1)}$ and in TS# k and TS# $k + 8$, $k = 0, \dots, 6$, allocation also provide the beacon function. Thereby, the P-CCPCH always provides the beacon function.

The physical channels providing the beacon function transmit with reference power and without beam-forming, use type 1 burst employing midamble $m^{(1)}$ and $m^{(2)}$ exclusively, while midambles $m^{(9)}$ and $m^{(10)}$ remain unused in this time slot when the cell allows 16 midambles.

The reference power equals the sum of the power allocated to both midambles $m^{(1)}$ and $m^{(2)}$. According to [6] two options are:

- In the absence of block STTD antenna diversity application to the P-CCPCH, all the reference power of any physical channel providing the beacon function goes to $m^{(1)}$.
- When block STTD antenna diversity applies to the P-CCPCH, physical channels providing beacon function midambles $m^{(1)}$ and $m^{(2)}$ share the reference power, i.e. midamble $m^{(1)}$ applies to the first antenna and $m^{(2)}$ applies to the diversity antenna. The data in P-CCPCH uses block STTD encoding [9]. For all other physical channels, both antennas transmit identical data sequences.

4.4.10 Allocating Midamble to Physical Channels

Generally high layers configure DL physical channels with midambles. Otherwise, they allocate default midambles by fixed association between midambles and channelization codes. Different associations apply for different burst types and cell configurations with respect to the maximum number of midambles. Physical channels providing the beacon function shall always use the reserved midambles. For all other DL physical channels the midamble allocation is signalled or given by default.

In the UL, if the physical channel has a midamble as part of its configuration, we assign an individual midamble to all UEs in one time slot. Otherwise, when higher layers do not allocate midambles, the UE will derive the midamble from the assigned channelization code as for DL physical channels. If the UE changes the SF according to the data rate, it shall always vary the channelization code along the lower branch of the OVSF tree. See more midamble details in [6].

4.4.11 Mapping Transport Channels onto Physical Channels

Table 4.36 summarizes the mapping of the transport channels onto the physical channels.

4.4.11.1 Dedicated Transport Channels

We map a dedicated transport channel onto one or more physical channels, where an interleaving period association occurs with each allocation. The frame is subdivided into slots that are available for uplink and downlink information transfer.

For NRT packet data services, shared channels (USCH and DSCH) can be used to allow efficient allocations for a short period of time.

Table 4.36 Mapping of Transport Channels to Physical Channels

Transport channels	Map to	Physical channels
DCH	→	Dedicated Physical Channel (DPCH)
BCH	→	Primary Common Control Physical Channel (P-CCPCH)
FACH	→	Secondary Common Control Physical Channel (S-CCPCH)
PCH	→	
RACH	→	Physical Random Access Channel (PRACH)
USCH	→	Physical Uplink Shared Channel (PUSCH)
DSCH	→	Physical Downlink Shared Channel (PDSCH)
	→	Page Indicator Channel (PICH)
	→	Synchronization Channel (SCH)

4.4.12 Mapping Common Transport Channels

4.4.12.1 The Broadcast Channel (BCH)

We map the BCH onto the P-CCPCH, where the secondary SCH indicates in which time slot a mobile can find the P-CCPCH containing a BCH. For additional resources the BCH in P-CCPCH will comprise a pointer to additional FACH S-CCPCH resources in which this additional broadcast information will occur.

4.4.12.2 The Paging Channel (PCH)

We map the PCH onto one or several S-CCPCHs while matching capacity to requirements, indicate its location on the BCH, and always transmit it at a reference power level. To allow an efficient DRX, the PCH is divided into several paging sub-channels within the allocated multi-frame structure. See examples of multi-frame structures in [6]. Each paging sub-channel comes mapped onto two consecutive frames allocated to the PCH on the same S-CCPCH. Layer 3 information to a particular paging group arrives through the associated paging sub-channel. UE assignment to paging groups occurs independently of the assignment of UEs to paging indicators.

4.4.12.3 The Forward Channel (FACH)

We map the FACH onto one or several S-CCPCHs. Indication of FACH location comes on the BCC, where both capacity and location can be changed when necessary. The FACH may or may not have power control.

4.4.12.4 The Random Access Channel (RACH)

The RACH, which we map onto the PRACH, has intra-slot interleaving only. One or more cells may use the same slots for PRACH. However, more than one slot per frame may be administered for the PRACH. The BCH broadcasts the location of slots allocated to PRACH. The latter uses open loop power control with algorithms, which may differ from the ones used on other channels. Multiple transmissions using different spreading codes may be received in parallel [6].

4.4.12.5 Shared Channels

We map the Uplink Shared Channel (USCH) on one or several PUSCH. Likewise we map the Downlink Shared Channel (DSCH) on one or several PDSCH.

4.5 SPREADING AND MODULATION IN TDD

4.5.1 Modulation and Symbol Rate

Table 4.37 illustrates the TDD basic modulation parameters. Notice that it has a low chip rate option at 1.28 Mchip/s. The complex-valued chip sequence is QPSK modulated as illustrated in Figure 4.46.

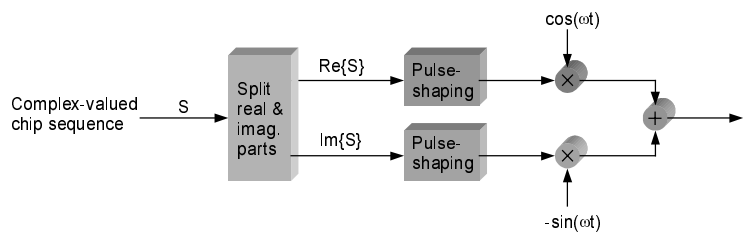


Figure 4.46 Modulation of complex valued chip sequences.

Table 4.37 Basic Modulation Parameters [7]

Chip rate	Same as FDD basic chip rate: 3.84 Mchip/s	Low chip rate: 1.28 Mchip/s
Data modulation	QPSK	QPSK
Spreading characteristics	Orthogonal Q chips/symbol, where $Q = 2^p$, $0 \leq p \leq 4$	Orthogonal Q chips/symbol, where $Q = 2^p$, $0 \leq p \leq 4$

In this section we use Q for the spreading, while SF denotes spreading in the FDD mode.

The symbol duration T_s depends on the spreading factor Q and the chip duration T_c : $T_s = Q \times T_c$, where $T_c = 1/\text{chip rate}$.

4.5.2 Mapping of Bits onto Signal Point Constellation

4.5.2.1 Mapping for Burst Type 1 and 2

We perform data modulation on the bits from the output of the physical channel mapping procedure in [20] and combine always two consecutive binary bits to a complex valued data symbol. Each user burst has two data carrying parts, termed data blocks:

$$\underline{d}^{(k,i)} = (\underline{d}_1^{(k,i)}, \underline{d}_2^{(k,i)}, \dots, \underline{d}_{N_k}^{(k,i)})^T, \quad i = 1, 2; \quad k = 1, \dots, K. \quad (4.47)$$

N_k corresponds to the number of symbols per data field for the user k . We link this number to the spreading factor Q_k as described in Table 1 of [6].

Data block $\underline{d}^{(k,1)}$ gets transmitted before the midamble and data block $\underline{d}^{(k,2)}$ after the midamble. Each of the N_k data symbols $\underline{d}_n^{(k,i)}$; $i = 1, 2$; $k = 1, \dots, K$; $n = 1, \dots, N_k$; of equation (4.47) has the symbol duration $T_s^{(k)} = Q_k T_c$ as already given.

The data modulation is QPSK, thus the data symbols $\underline{d}_n^{(k,i)}$ are generated from two consecutive data bits from the output of the physical channel mapping procedure in [20]:

$$b_{l,n}^{(k,i)} \in \{0, 1\}, \quad l = 1, 2; \quad k = 1, \dots, K; \quad n = 1, \dots, N_k; \quad i = 1, 2 \quad (4.48)$$

using the following the mapping to complex symbols illustrated in Table 4.38.

Table 4.38 Mapping Complex Symbols

Consecutive binary bit pattern	Complex symbol
$b_{1,n}^{(k,i)} \ b_{2,n}^{(k,i)}$	$\underline{d}_n^{(k,i)}$
00	$+j$
01	$+1$
10	-1
11	$-j$

The mapping corresponds to a QPSK modulation of the interleaved and encoded data bits $b_{l,n}^{(k,i)}$ of equation (4.48).

4.5.2.2 Mapping for PRACH Burst Type

When mapping the PRACH burst type the preceding logic applies with a modified number of symbols in the second data block. Thus, for the PRACH burst type, the number of symbols in the second data block $\underline{d}^{(k,2)}$ is decreased by $96/Q_k$ symbols.

4.5.3 Spreading Parameters and Channelization Codes

Data spreading includes two steps, i.e. channelization and scrambling. First, each complex valued data symbol $\underline{d}_n^{(k,i)}$ of equation (4.47) gets spread with a real valued channelization code $\mathbf{c}^{(k)}$ of length $Q_k \in \{1, 2, 4, 8, 16\}$. We then scramble the resulting sequence by a complex sequence $\underline{v}$ of length 16.

The elements $c_q^{(k)}$; $k = 1, \dots, K$; $q = 1, \dots, Q_k$; of the real valued channelization codes, i.e.

$$\mathbf{c}^{(k)} = (c_1^{(k)}, c_2^{(k)}, \dots, c_{Q_k}^{(k)}), \quad k = 1, \dots, K \quad (4.49)$$

will be taken from the set

$$V_c = \{1, -1\}. \quad (4.50)$$

The $\mathbf{c}_{Q_k}^{(k)}$ belong to Orthogonal Variable Spreading Factor (OVSF) codes, which allow mixing in the same time slot channels with different spreading factors while preserving the orthogonality. We define the OVSF codes using the code tree illustrated in Figure 4.47.

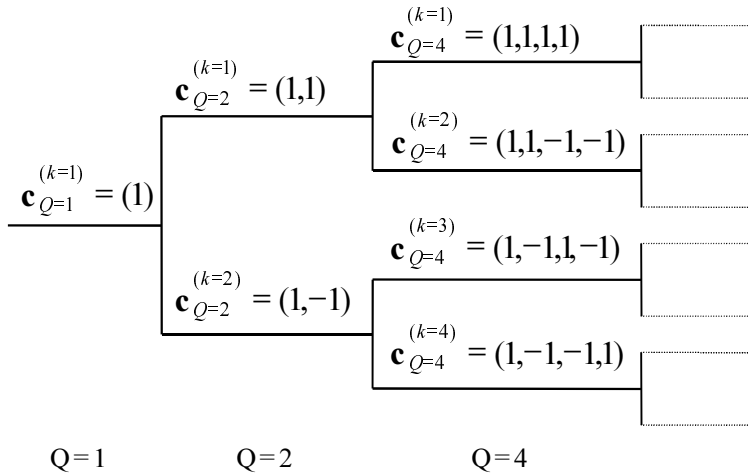


Figure 4.47 Code-tree generating OVSF codes for channelization.

Each level in the code tree defines a SF indicated by the value of Q in Figure 4.47. We may not use all codes within the code tree simultaneously in a given time slot. We can use a code in a time slot if and only if no other code on the path from the specific code to the root of the tree, or in the sub-tree below the specific code is used in this time slot. This implies that the number of available codes in a slot depends on the rate and spreading factor of each physical channel. The SF goes up to $Q_{\text{MAX}}=16$ [7].

4.5.4 Scrambling Codes

Data spreading by a real valued channelization code $\mathbf{c}^{(k)}$ of length Q_k gets followed by a cell specific complex scrambling sequence $\mathbf{v} = (v_1, v_2, \dots, v_{16})$. The elements v_i , $i = 1, \dots, 16$ of the complex valued scrambling codes originates from the complex set

$$V_v = \{1, j, -1, -j\}. \quad (4.51)$$

where j denotes the imaginary unit.

We generate a complex scrambling code $\underline{v}$ from the binary scrambling codes $\mathbf{v} = (v_1, v_2, \dots, v_{16})$ of length 16 described in the Annex of [7]. The relation between the elements $\underline{v}$ and $\mathbf{v}$ is given by:

$$\underline{v}_i = (j)^i \cdot v_i, \quad v_i \in \{1, -1\}, \quad i = 1, \dots, 16. \quad (4.52)$$

Thus, the elements $\underline{v}_i$ of the complex scrambling code $\underline{v}$ have alternating real and imaginary values.

We obtain length matching by concatenating Q_{MAX}/Q_k spread words before the scrambling event as illustrated in Figure 4.48.

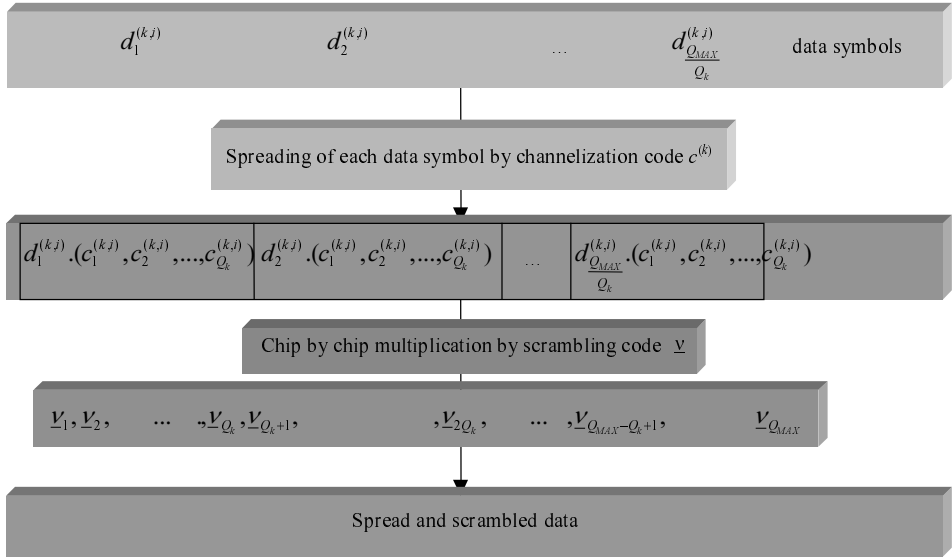


Figure 4.48 Spreading of data symbols.

4.5.5 Spreading Data Symbols and Data Blocks

We can see the combination of the user specific channelization and cell specific scrambling codes as a user and cell specific spreading code $\mathbf{s}^{(k)} = (s_p^{(k)})$ with

$$s_p^{(k)} = c_{1+[(p-1) \bmod Q_k]} \cdot \underline{v}_{1+[(p-1) \bmod Q_{\text{MAX}}]}, \quad k = 1, \dots, K, \quad p = 1, \dots, N_k Q_k. \quad (4.53)$$

With the root raised cosine chip impulse filter $Cr_0(t)$ the transferred signal belonging to the data block $\underline{d}^{(k,1)}$ of equation (4.47) transmitted before the midamble gets expressed as:

$$\underline{d}^{(k,1)}(t) = \sum_{n=1}^{N_k} \underline{d}_n^{(k,1)} \sum_{q=1}^{Q_k} s_{(n-1)Q_k+q}^{(k)} \cdot Cr_0(t - (q-1)T_c - (n-1)Q_k T_c) \quad (4.54)$$

and for the data block $\underline{d}^{(k,2)}$ of equation (1) transmitted after the midamble

$$\underline{d}^{(k,2)}(t) = \sum_{n=1}^{N_k} \underline{d}_n^{(k,2)} \sum_{q=1}^{Q_k} s_{(n-1)Q_k+q}^{(k)} \cdot Cr_0(t - (q-1)T_c - (n-1)Q_kT_c - N_kQ_kT_c - L_mT_c), \quad (4.55)$$

where L_m is the number of midamble chips.

4.5.6 Synchronization Codes

The primary code sequence, C_p results from a generalized hierarchical Golay sequence. The Primary Synchronization Channel (SCH), in addition has a good aperiodic auto correlation properties.

We define

$$a = \langle x_1, x_2, x_3, \dots, x_{16} \rangle = \langle 1, 1, 1, 1, 1, 1, -1, -1, 1, 1, -1, 1, -1, 1, -1, 1 \rangle.$$

The PSC code word gets generated by repeating the sequence 'a' modulated by a Golay complementary sequence and creating a complex-valued sequence with identical real and imaginary components.

Then we define the PSC code word C_p as:

$$C_p = \langle y(0), y(1), y(2), \dots, y(255) \rangle,$$

where

$$y = (1 + j) \times \langle a, a, a, -a, -a, a, -a, -a, a, a, -a, a, -a, a, a \rangle$$

and the left most index corresponds to the chip transmitted first in each time slot.

The 16 secondary synchronization code words, $\{C_0, \dots, C_{15}\}$ constitute complex valued with identical real and imaginary components, and they originate from the position wise multiplication of a Hadamard sequence and a sequence z , defined as

$$z = \langle b, b, b, -b, b, b, -b, -b, b, -b, b, -b, -b, b, -b, -b \rangle, \quad (4.56)$$

where

$$b = \langle x_1, \dots, x_8, -x_9, \dots, -x_{16} \rangle = \langle 1, 1, 1, 1, 1, 1, -1, -1, -1, 1, -1, 1, -1, 1, 1, -1 \rangle.$$

We build the Hadamard sequences as the rows in a matrix H_8 constructed recursively by:

$$H_0 = (1) \\ H_k = \begin{pmatrix} H_{k-1} & H_{k-1} \\ H_{k-1} & H_{k-1} \end{pmatrix}, \quad k \geq 1. \quad (4.57)$$

The rows are numbered from the top starting with row 0 (the all zeros sequence).

We denote the n th Hadamard sequence as a row of H_8 numbered from the top, $n = 0, 1, 2, \dots, 255$, in the sequel.

In addition, we let $h_m(i)$ and $z(i)$ and denote the i th symbol of the sequence h_m and z , respectively where $i = 0, 1, 2, \dots, 255$ and $i = 0$ corresponds to the leftmost symbol.

The i th SCH code word, $C_{\text{SCH},i}$, $i = 0, \dots, 15$ is then defined as

$$C_{\text{SCH},i} = (1+j) \times \langle h_m(0) \times z(0), h_m(1) \times z(1), h_m(2) \times z(2), \dots, h_m(255) \times z(255) \rangle, \quad (4.58)$$

where $m = (16 \times i)$ and the leftmost chip in the sequence corresponds to the chip transmitted first in time.

This code word gets selected from every 16th row of the matrix H_8 , which yields 16 possible code words. We define the secondary SCH code words in terms of $C_{\text{SCH},i}$ and the definition of $\{C_0, \dots, C_{15}\}$ now follows as:

$$C_i = C_{\text{SCH},i}, \quad i = 0, \dots, 15. \quad (4.59)$$

Finally, more details and code allocations and evaluation of synchronization codes can be found in [7].

4.6 MULTIPLEXING AND CHANNEL CODING

We encode/decode information from/to upper layers to afford transport services over the air interface. Thus, through channel coding we protect data flow by combining error detection and correction and adapting to transmission needs by means of rate matching, interleaving, and mapping of transport channels to physical channels.

Multiplexing and channel coding techniques in the UTRA physical layer specifications apply to both FDD and TDD modes in most aspects. Thus, for completeness here we will introduce primarily the description for the FDD because it has more cases (e.g. description for uplink and downlink), and indicate where they differ with the TDD. Specifications in [10] and [18] provide all the details for each mode, respectively.

In UTRA data arrives at the coding/multiplexing unit in transport block sets once every transmission time interval. The transmission time interval depends on the transport channel from the set $\{10 \text{ ms}, 20 \text{ ms}, 40 \text{ ms}, 80 \text{ ms}\}$. The main steps valid for both FDD and TDD from [10] are:

- add CRC to each transport block
- transport block concatenation and code block segmentation
- channel coding
- rate matching
- insertion of discontinuous transmission (DTX) indication bits
- interleaving
- radio frame segmentation

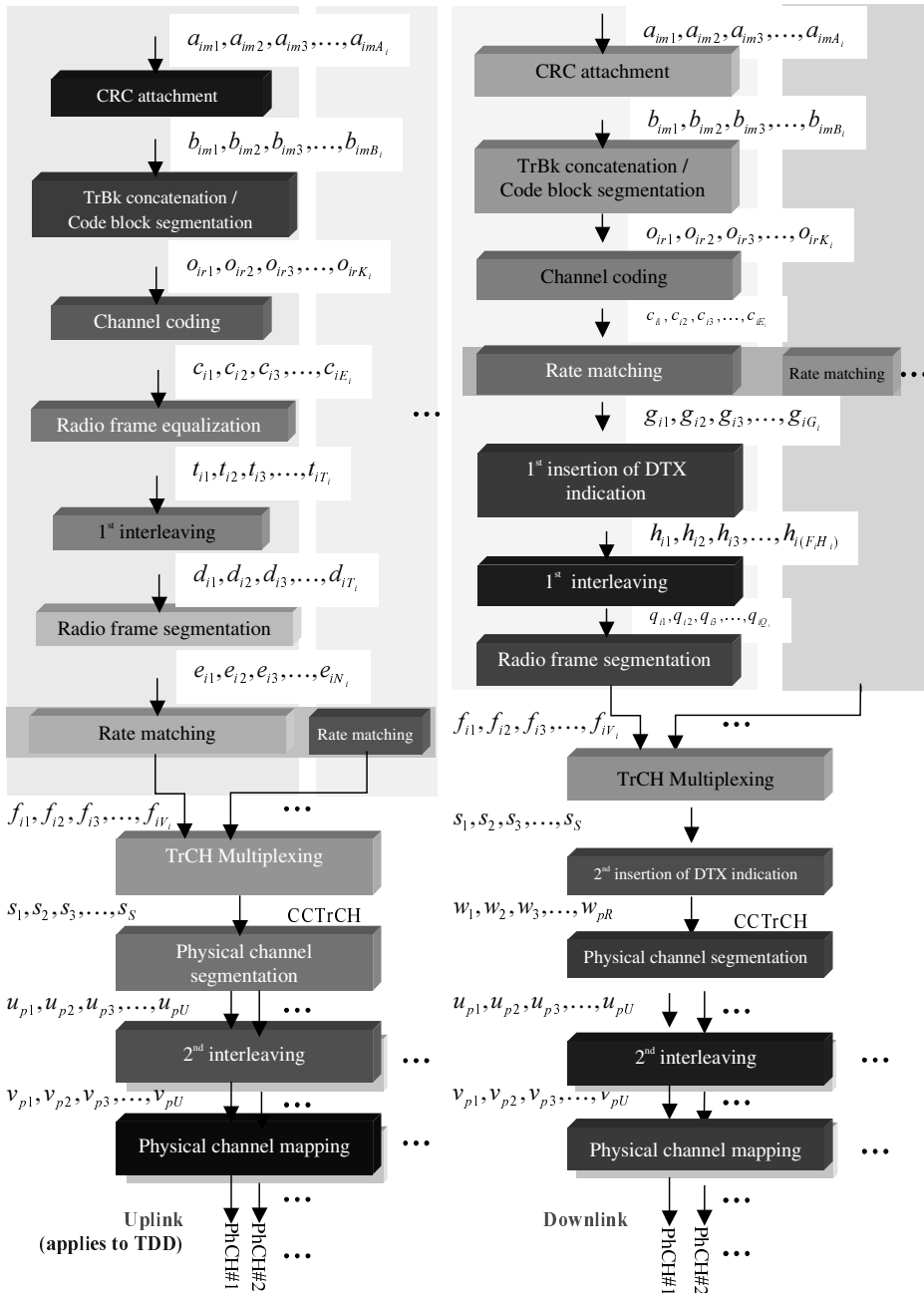


Figure 4.49 FDD uplink and downlink transport channel multiplexing structure.

- multiplexing of transport channels
- physical channel segmentation
- mapping to physical channels

Figure 4.49 illustrates the coding/multiplexing steps for FDD uplink and downlink. Clearly, the uplink also applies to the TDD mode. Hence the uplink description will basically cover the needs for the TDD. However, as mentioned above we will highlight where differences exist. We should also note here that in this section for consistency we keep the structure and nomenclature of the technical specifications by incorporating direct extracts and using the same type of equations.

Note: In the downlink we denoted *Coded Composite Transport Channel* (CCTrCH) the single output data stream from the TrCH multiplexing, including DTX indication. This CCTrCH can get mapped to one or several physical channels.

4.6.1 Error Detection and CRC Calculations

In the sequel we cover the DL and UL in an integrated manner by indicating differences where appropriate.

Cyclic Redundancy Check (CRC) affords error detection on transport blocks. Higher layers signal what CRC (24, 16, 12, 8 or 0) bit length shall be used for each TrCH. We use the entire transport block to calculate the CRC parity bits for each transport block. The following cyclic generator polynomials generate these parity bits:

$$g_{\text{CRC24}}(D) = D^{24} + D^{23} + D^6 + D^5 + D + 1, \quad (4.60)$$

$$g_{\text{CRC16}}(D) = D^{16} + D^{12} + D^5 + 1, \quad (4.61)$$

$$g_{\text{CRC12}}(D) = D^{12} + D^{11} + D^3 + D^2 + D + 1, \quad (4.62)$$

$$g_{\text{CRC8}}(D) = D^8 + D^7 + D^4 + D^3 + D + 1. \quad (4.63)$$

In relation with Figure 4.49 and [10], we denote the bits in a transport block delivered to layer 1 by

$$a_{im1}, a_{im2}, a_{im3}, \dots, a_{imA_i}$$

and the parity bits by

$$p_{im1}, p_{im2}, p_{im3}, \dots, p_{imL_i},$$

where A_i is the length of a transport block of TrCH i , m is the transport block number, and L_i is either 24, 16, 12, 8, or 0 depending on what the upper layers signal. Then encoding follows systematically; which means that in GF(2) we express the above polynomials as:

$$a_{im1}D^{A_i+23} + a_{im2}D^{A_i+22} + \dots + a_{imA_i}D^{24} + p_{im1}D^{23} + p_{im2}D^{22} + \dots + p_{im23}D^1 + p_{im24}, \quad (4.64)$$

$$a_{im1}D^{A_i+15} + a_{im2}D^{A_i+14} + \dots + a_{imA_i}D^{16} + p_{im1}D^{15} + p_{im2}D^{14} + \dots + p_{im15}D^1 + p_{im16}, \quad (4.65)$$

$$a_{im1}D^{A_i+11} + a_{im2}D^{A_i+10} + \dots + a_{imA_i}D^{12} + p_{im1}D^{11} + p_{im2}D^{10} + \dots + p_{im11}D^1 + p_{im12}, \quad (4.66)$$

$$a_{im1}D^{A_i+7} + a_{im2}D^{A_i+6} + \dots + a_{imA_i}D^8 + p_{im1}D^7 + p_{im2}D^6 + \dots + p_{im7}D^1 + p_{im8}. \quad (4.67)$$

The preceding polynomials yield a remainder equal to 0 when divided by $g_{CRC24}(D)$, $g_{CRC16}(D)$, $g_{CRC12}(D)$, and $g_{CRC8}(D)$, respectively. In the absence of transport block inputs to the CRC calculation ($M_i = 0$), CRC attachment does not occur. However, if transport blocks inputs exist in the CRC calculation (i.e. $M_i \neq 0$), and the size of a transport block equals zero ($A_i = 0$), CRC attachment occurs, i.e. all parity bits equal to zero. Denoting the bits after CRC attachment by

$$b_{im1}, b_{im2}, b_{im3}, \dots, b_{imB_i},$$

where $B_i = A_i + L_i$; the relation between a_{imk} and b_{imk} can be defined as

$$b_{imk} = a_{imk} \quad \text{where } k = 1, 2, 3, \dots, A_i; \quad (4.68)$$

$$b_{imk} = p_{im(L_i+1-(k-A_i))} \quad \text{where } k = A_i + 1, A_i + 2, A_i + 3, \dots, A_i + L_i. \quad (4.69)$$

4.6.2 Transport Block Concatenation and Code Block Segmentation

All transport blocks in a Transmission Time Interval (TTI) have serial concatenation. When the number of bits in a TTI is larger than Z , i.e. the maximum size of a code block in question, then code block segmentation takes place after the concatenation of the transport blocks. The maximum size of the code blocks depends on whether convolutional coding, turbo coding or no coding occurs. We denote

$$b_{im1}, b_{im2}, b_{im3}, \dots, b_{imB_i}$$

the bits input to the transport block concatenation, where i is the TrCH number, m is the transport block number, and B_i is the number of bits in each block (including CRC). M_i represents the number of transport blocks on TrCH i .

$$x_{i1}, x_{i2}, x_{i3}, \dots, x_{iX_i}$$

denotes the bits after concatenation, where i is the TrCH number and $X_i = M_i B_i$. Then, the following relations apply:

$$x_{ik} = b_{ik} \quad k = 1, 2, \dots, B_i \quad (4.70)$$

$$x_{ik} = b_{i,2,(k-B_i)} \quad k = B_i + 1, B_i + 2, \dots, 2B_i \quad (4.71)$$

$$x_{ik} = b_{i,3,(k-2B_i)} \quad k = 2B_i + 1, 2B_i + 2, \dots, 3B_i \quad (4.72)$$

...

$$x_{ik} = b_{i,M_i,(k-(M_i-1)B_i)} \quad k = (M_i-1)B_i + 1, (M_i-1)B_i + 2, \dots, M_i B_i. \quad (4.73)$$

Segmentation of the bit sequence from transport block concatenation transpires when $X_i > Z$, where the segmented blocks have the same size. If the number of bits input to the segmentation (i.e. X_i) is not a multiple of C_i (the number of code blocks on TrCH i), we add filler bits (0s) to the beginning of the first block. The specifications in [10] define maximum code block sizes as:

- convolutional coding: $Z = 504$
- turbo coding: $Z = 5114$
- no channel coding: $Z = \text{unlimited}$

From Figure 4.49 and [10] we denote

$$O_{ir1}, O_{ir2}, O_{ir3}, \dots, O_{irK_i}$$

the bit output from code block segmentation, where i is the TrCH number, r is the code block number, and K_i is the number of bits. Then number of code blocks: $C_i = \lceil X_i/Z \rceil$, and for the number of bits in each code block and filler bits, the following logic applies:

number of bits in each code block	number of filler bits: $Y_i = C_i K_i - X_i$
if $X_i < 40$ and turbo coding is used, then	if $X_i \leq Z$, then
$K_i = 40$	$O_{rlk} = 0, k = 1, 2, \dots, Y_i$
else	$O_{rlk} = x_{i, (k - Y_i)}, k = Y_i + 1, Y_i + 2, \dots, K_i$
$K_i = \lceil X_i / C_i \rceil$	end if
end if	

If $X_i > Z$, then

$$\begin{aligned}
 O_{rlk} &= 0 & k &= 1, 2, \dots, Y_i \\
 O_{rlk} &= x_{i, (k - Y_i)} & k &= Y_i + 1, Y_i + 2, \dots, K_i \\
 O_{rlk} &= x_{i, (k + K_i - Y_i)} & k &= 1, 2, \dots, K_i \\
 O_{rlk} &= x_{i, (k + 2K_i - Y_i)} & k &= 1, 2, \dots, K_i \\
 &\dots & & \\
 O_{rlk} &= x_{i, (k + (C_i - 1)K_i - Y_i)} & k &= 1, 2, \dots, K_i
 \end{aligned}$$

end if

4.6.3 Channel Coding

The concatenation or segmentation process delivers code blocks

$$O_{ir1}, O_{ir2}, O_{ir3}, \dots, O_{irK_i}$$

to the channel coding block, where i is the TrCH number, r is the code block number, and K_i is the number of bits in each code block. We denote C_i the number of code blocks on TrCH i , and the encoded bits

$$Y_{ir1}, Y_{ir2}, Y_{ir3}, \dots, Y_{irY_i}$$

where Y_i is the number of encoded bits. The relation between o_{irk} and y_{irk} and between K_i and Y_i depends on the following channel coding scheme: convolutional coding; turbo coding; and no coding. Table 4.39 illustrate the usage of these schemes and the values of Y_i in connection with each coding scheme are:

- convolutional coding with rate 1/2: $Y_i = 2K_i + 16$; rate 1/3: $Y_i = 3K_i + 24$;
- turbo coding with rate 1/3: $Y_i = 3K_i + 12$;
- no coding: $Y_i = K_i$.

Table 4.39 Usage of channel coding scheme and coding rate

Type of TrCH	Coding scheme	Coding rate
BCH	Convolutional coding	1/2
PCH		
RACH		
CPCH, DCH, DSCH, FACH + USCH (tdd)		1/3, 1/2
	Turbo coding	1/3
	No coding	

4.6.3.1 Convolutional Coding

In UTRA we define convolutional codes with constraint length 9 and coding rates 1/3 and 1/2. Figure 4.50 illustrates the convolutional coder configuration. Output from the rate 1/3 convolutional coder follows the order output0, output1, output2, output0, output1, output 2, output 0,...,output2; while output from the rate 1/2 convolutional coder follows the order output 0, output 1, output 0, output 1, output 0, ..., output 1. We add 8 tail bits with binary value 0 at the end of the code block before encoding, and when starting to encode the initial value of the shift register of the coder = ‘all 0’.

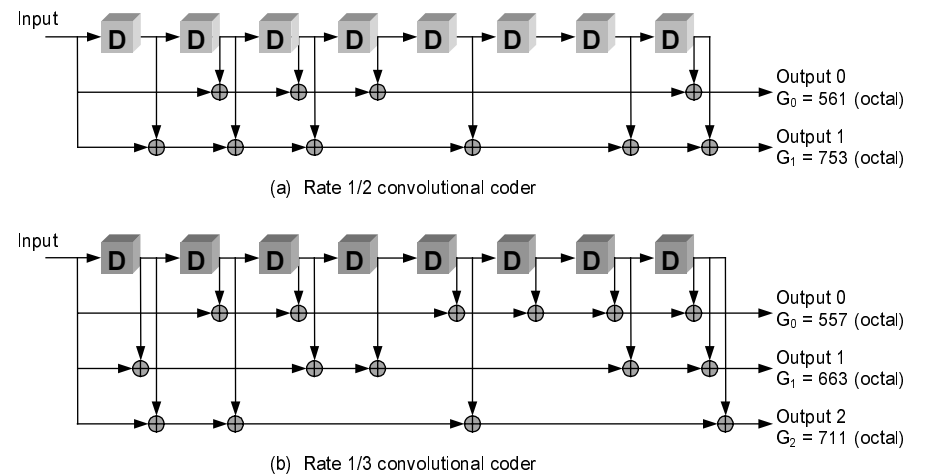


Figure 4.50 Rate 1/2 and rate 1/3 convolutional coders.

4.6.3.2 Turbo Coding

From the two types of code concatenation, i.e. serial and parallel, the latter suits well for high quality of services in 3rd generation systems. This can provide very low-maximum bit error ratio, e.g. 10^{-6} has the lowest S/N. In parallel code concatenation we feed the information stream into a second encoder and encode data stream generating by multiplexing (and puncture) the encoded sequences resulting from both encoding processes [17].

Coder. In UTRA we apply the Parallel Concatenated Convolutional Code (PCCC) scheme of the turbo coder with two 8-state constituent encoders and one turbo code internal interleaver [10]. The turbo coder structure illustrated in Figure 4.51 has a coding rate of 1/3. We express the transfer function of the 8-state constituent code for PCCC as:

$$G(D) = \begin{bmatrix} 1, \frac{g_1(D)}{g_0(D)} \end{bmatrix}, \quad (4.74)$$

where $g_0(D) = 1 + D^2 + D^3$, and $g_1(D) = 1 + D + D^3$.

The initial value of the shift registers in the 8-state constituent encoders = all zeros when starting to encode the input bits, and the output from the turbo coder is:

$$x_1, z_1, z'_1, x_2, z_2, z'_2, \dots, x_K, z_K, z'_K, \quad (4.75)$$

where $x_1, x_2, \dots, x_K$ are the bits input to the turbo coder, i.e. both first 8-state constituent encoder and turbo code internal interleaver, and K is the number of bits, and $z_1, z_2, \dots, z_K$ and $z'_1, z'_2, \dots, z'_K$ are the bits output from first and second 8-state constituent encoders, respectively. The bits output from turbo code internal interleaver are denoted by $x'_1, x'_2, \dots, x'_K$, and these bits are to be input to the second 8-state constituent encoder [10].

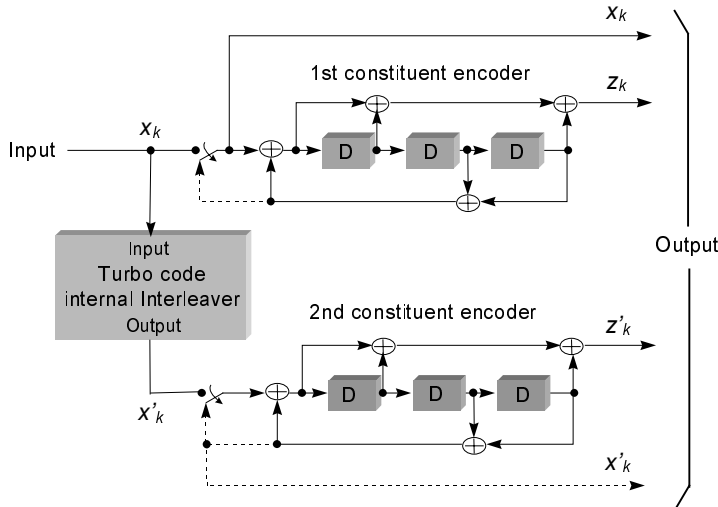


Figure 4.51 Turbo coder structure of rate 1/3 (dotted lines indicate $\rightarrow$ termination only).

Other details such as trellis termination, internal interleaver, of the turbo coder can be found in [10].

4.6.4 Radio Frame Size Equalization

Radio frame size equalization implies padding the input bit sequence in order to ensure that the output can be segmented in F_i data segments of same size, e.g. rate matching. Radio frame size equalization occurs only in the UL (DL rate matching output block length is always an integer multiple of F_i). We denote by

$$c_{i1}, c_{i2}, c_{i3}, \dots, c_{iE_i}$$

the input bit sequence to the radio frame size equalization, where i is TrCH number and E_i the number of bits. We denote the output bit sequence by

$$t_{i1}, t_{i2}, t_{i3}, \dots, t_{iT_i},$$

where T_i is the number of bits. Then the output bit sequence follows as:

$$t_{ik} = c_{ik}, \text{ for } k = 1, \dots, E_i \quad \text{and} \quad t_{ik} = \{0, 1\} \text{ for } k = E_i + 1, \dots, T_i, \text{ if } E_i < T_i, \quad (4.76)$$

where $T_i = F_i \times N_i$; and $N_i = \lceil E_i / F_i \rceil$ is the number of bits per segment after size equalization.

4.6.5 First Interleaving

In a compressed mode through puncturing, bits marked with a 4th value on top of $\{0, 1, \delta\}$ and noted p , get introduced in radio frames to be compressed at positions corresponding to the first bits of the radio frames. They will be removed in a later stage of the multiplexing chain to create the actual gap. We perform additional puncturing in the rate matching step, over the Transmission Time Interval (TTI) containing the compressed radio frame, to create room for these p bits. Specifications in [10] provide this and other first interleaving details.

4.6.6 Radio Frame Segmentation

When the TTI is longer than 10 ms, the input bit sequence get segmented and mapped onto consecutive F_i radio frames. Following rate matching in the DL and radio frame size equalization in the UL, we warrant that the input bit sequence length is an integer multiple of F_i . We denote this input bit sequence by

$$x_{i1}, x_{i2}, x_{i3}, \dots, x_{iX_i},$$

where i is the TrCH number and X_i is the number bits. Likewise, we denote the F_i output bit sequences per TTI by

$$y_{i,n1}, y_{i,n2}, y_{i,n3}, \dots, y_{i,nY_i},$$

where n_i is the radio frame number in the current TTI and Y_i is the number of bits per radio frame for TrCH i . Then we define the output sequence as:

$$y_{i,n,k} = x_{i,((n-1)Y_i)+k}, \quad n_i = 1, \dots, F_i, \quad k = 1, \dots, Y_i, \quad (4.77)$$

where $Y_i = (X_i/F_i)$ is the number of bits per segment. The n_i th segment is mapped to the n_i th radio frame of the transmission time interval.

4.6.6.1 Input–Output Relationship of the Radio Frame Segmentation Block in Uplink

We denote the input bit sequence to the radio frame segmentation by

$$d_{i1}, d_{i2}, d_{i3}, \dots, d_{iT_i},$$

where i is the TrCH number and T_i the number of bits. Thus, $x_{ik} = d_{ik}$ and $X_i = T_i$. Likewise, we denote the output bit sequence corresponding to radio frame n_i by

$$e_{i1}, e_{i2}, e_{i3}, \dots, e_{iN_i},$$

where i is the TrCH number and N_i is the number of bits. Thus, $e_{i,k} = y_{i,n,k}$ and $N_i = Y_i$.

4.6.6.2 Input–Output Relationship of Radio Frame Segmentation Block in Downlink

As in the preceding section, we denote the bits input to the radio frame segmentation by

$$q_{i1}, q_{i2}, q_{i3}, \dots, q_{iQ_i},$$

where i is the TrCH number and Q_i the number of bits. Hence, $x_{ik} = q_{ik}$ and $X_i = Q_i$. Again, we denote the output bit sequence corresponding to radio frame n_i is by

$$f_{i1}, f_{i2}, f_{i3}, \dots, f_{iV_i},$$

where i is the TrCH number and V_i is the number of bits. Then, $f_{i,k} = y_{i,n,k}$ and $V_i = Y_i$.

4.6.7 Rate Matching

By rate matching we mean the repetition or puncturing of bits on a transport channel based on attributes assigned by higher layers. An attribute is semi-static and can only get changed through higher layer signalling. The rate matching attribute assignment occurs after the calculation of the number of bits to be repeated or punctured.

The number of bits on a transport channel can vary between different TTIs. In the DL the transmission gets interrupted if the number of bits is lower than maximum. When the number of bits between different uplink TTIs changes, bits get repeated or punctured to ensure that the total bit rate after TrCH multiplexing is identical to the total channel bit rate of the allocated dedicated physical channels. If the rate matching event does not get input bits for all TrCHs within a CCTrCH, the rate matching does not out-

put bits for all TrCHs within the CCTrCH and no uplink DPDCH will mean no selection of uplink rate matching. See the detailed description of rate matching characteristics, such as determination of rate matching in uplink/downlink, as well as separation and collection in uplink/downlink in [10].

4.6.8 TrCH Multiplexing

The TrCH delivers one radio frame every 10 ms to the TrCH multiplexing, which are serially multiplexed into a Coded Composite Transport Channel (CCTrCH).

We denote by

$$f_{i1}, f_{i2}, f_{i3}, \dots, f_{iV_i}$$

the input bits going to the TrCH multiplexing, where i is the TrCH number and V_i is the number of bits in the radio frame of TrCH i . Likewise, we denote by I the number of TrCHs, and by $s_1, s_2, s_3, \dots, s_S$ the output bits from TrCH multiplexing, where S is the number of bits, i.e. [10].

$$S = \sum_i V_i a. \quad (4.78)$$

The TrCH multiplexing is defined by the following relations:

$$s_k = f_{1k}, \quad k = 1, 2, \dots, V_1 \quad (4.79)$$

$$s_k = f_{2, (k-V_1)}, \quad k = V_1 + 1, V_1 + 2, \dots, V_1 + V_2 \quad (4.80)$$

$$s_k = f_{3, (k-(V_1+V_2))}, \quad k = (V_1 + V_2) + 1, (V_1 + V_2) + 2, \dots, (V_1 + V_2) + V_3 \quad (4.81)$$

...

$$s_k = f_{I, (k-(V_1+V_2+\dots+V_{I-1}))} \quad (4.82)$$

$$k = (V_1 + V_2 + \dots + V_{I-1}) + 1, (V_1 + V_2 + \dots + V_{I-1}) + 2, \dots, (V_1 + V_2 + \dots + V_{I-1}) + V_I \quad (4.83)$$

4.6.9 Discontinuous Transmission (DTX) Bits Insertion

We use DL DTX to fill up the radio frame with bits, where the insertion point of these bits can have either fixed or flexible positions of the TrCHs in the radio frame. It depends on the UTRAN to decide for each CCTrCH whether it will have fixed or flexible positions during the connection. DTX indication bits communicate only when the transmission will get turned off, i.e. they are not transmitted themselves.

4.6.9.1 First Insertion of DTX Indication Bits

First DTX indication bits insertion occurs only if the positions of the TrCHs in the radio frame are fixed. In the fixed position scheme, we reserve a fixed number of bits for each TrCH in the radio frame. We denote the bits from rate matching by

$$g_{i1}, g_{i2}, g_{i3}, \dots, g_{iG_i}$$

where G_i is the number of bits in one TTI of TrCH i . Likewise, we denote the number of bits in one radio frame of TrCH i by H_i . Finally, we also denote by D_i the number of bits output of the first DTX insertion block.

In normal or compressed mode using spreading factor reduction, H_i is constant and corresponds to the maximum number of bits from TrCH i in one radio frame for any transport format of TrCH i , and $D_i = F_i \times H_i$.

Within compressed mode using puncturing techniques, additional puncturing occurs in the rate matching block. The empty positions resulting from the additional puncturing get p bits inserted in the first interleaving block, the DTX insertion is thus limited to allow later insertion of p bits. Consequently, DTX bits get inserted until the total number of bits is D_i where:

$$D_i = F_i \times H_{i,*} + \Delta N_{\text{cm},j,\text{max}}^{\text{TTI}} \quad \text{and} \quad H_i = N_{i,*} + \Delta N_i \quad (4.84)$$

We denote the output bits from the DTX insertion by $h_{i1}, h_{i2}, h_{i3}, \dots, h_{iD_i}$, where these three valued bits can be expressed by following relations:

$$h_k = g_{ik}, \quad k = 1, 2, 3, \dots, G_i \quad (4.85)$$

$$h_k = \delta, \quad k = G_i + 1, G_i + 2, G_i + 3, \dots, D_i \quad (4.86)$$

where we denote DTX indication bits by δ . Here $g_{ik} \in \{0, 1\}$ and $\delta \notin \{0, 1\}$.

4.6.9.2 Second Insertion of DTX Indication Bits

The DTX indication bits inserted in the 2nd insertion get placed at the end of the radio frame, and the DTX will be distributed over all slots after 2nd interleaving. The input bits to the DTX insertion block get denoted by $s_1, s_2, s_3, \dots, s_S$, where S is the number of bits from TrCH multiplexing. We denote by P the number of PhCHs and the number of bits in one radio frame, including DTX indication bits, for each PhCH by R . In a normal mode

$$R = \frac{N_{\text{data},*}}{P} = 15N_{\text{data}1} + 15N_{\text{data}2}, \quad (4.87)$$

where $N_{\text{data}1}$ and $N_{\text{data}2}$ are defined in the first part of this chapter and [1]. For compressed mode, $N'_{\text{data},*}$ is defined as

$$N'_{\text{data},*} = P(15N'_{\text{data}1} + 15N'_{\text{data}2}), \quad (4.88)$$

where $N'_{\text{data}1}$ and $N'_{\text{data}2}$ are the number of bits in the data fields of the slot format used for the current compressed mode, i.e. slot format A or B as defined in [1] corresponding to the spreading factor and the number of transmitted slots in use [10].

When compressed mode by puncturing and fixed positions occurs, DTX get inserted until $N'_{data,*}$ bits, because the exact room for the gap is already reserved thanks to the earlier insertion of the p bits. Thus, R is defined as

$$R = N'_{data,*} / P. \quad (4.89)$$

If compressed mode by SF reduction and by higher layer scheduling occurs, additional DTX get inserted when the transmission time reduction method does not exactly create a transmission gap of the desired Transmission Gap Length (TGL). The number of bits available to the CCTrCH in one radio frame of this compressed mode gets denoted by

$$N_{data,*}^{cm} \quad \text{and} \quad R = \frac{N_{data,*}^{cm}}{P}.$$

The exact value of $N_{data,*}^{cm}$ is dependent on the TGL and the transmission time reduction method signalled from higher layers.

For transmission time reduction by SF/2 method in compressed mode

$$N_{data,*}^{cm} = \frac{N'_{data,*}}{2}$$

and for other methods it can be calculated as

$$N_{data,*}^{cm} = N'_{data,*} - N_{TGL}.$$

For every transmission time reduction method

$$N'_{data,*} = P(15N'_{data1} + 15N'_{data2}),$$

where N'_{data1} and N'_{data2} are the number of bits in the data fields of a slot for slot format A or B as defined in [1].

N_{TGL} is the number of bits that are located within the transmission gap and defined as:

$$N_{TGL} = \begin{cases} \frac{TGL}{15} N'_{data,*} & \text{if } N_{first} + TGL \leq 15, \\ \frac{15 - N_{first}}{15} N'_{data,*} & \text{in first frame if } N_{first} + TGL > 15, \\ \frac{TGL - (15 - N_{first})}{15} N'_{data,*} & \text{in second frame if } N_{first} + TGL > 15. \end{cases}$$

N_{first} and TGL are part of the description of the compressed mode section.

Furthermore, notice that in compressed mode by SF/2 method, we also add DTX in the physical channel mapping stage. During the 2nd DTX insertion the number of CCTrCH bits remains the same as in the normal mode. We denote the bits output from the DTX insertion block by $w_1, w_2, w_3, \dots, w_{(PR)}$. Notice also that these bits have four values in case

of compressed mode by puncturing, and three otherwise. We can define them by the following relations:

$$w_k = s_k, \quad k = 1, 2, 3, \dots, S \quad \text{and} \quad w_k = \delta, \quad k = S+1, S+2, S+3, \dots, PR, \quad (4.90)$$

where DTX indication bits are denoted by δ . Here $s_k \in \{0, 1, p\}$ and $\delta \notin \{0, 1\}$ [10].

4.6.10 Physical Channel Segmentation

When using more than one PhCH, the physical channel segmentation event divides the bits among the different PhCHs. The bits input to the physical channel segmentation are $x_1, x_2, x_3, \dots, x_Y$, where Y is the number of bits input to the physical channel segmentation block. P denotes the number of PhCHs.

The bits after the physical channel segmentation get denoted $u_{p1}, u_{p2}, u_{p3}, \dots, u_{pU}$, where p is PhCH number and U is the number of bits in one radio frame for each PhCH, i.e. $U = (Y - N_{\text{TGL}})/P$ for compressed mode by puncturing, and $U = Y/P$ otherwise.

For all modes, we map some bits of the input flow to each code until the number of bits on the code reach V . For modes other than *compressed mode by puncturing*, we take all bits of the input flow for mapping to the codes. For compressed mode by puncturing, only the bits of the input flow not corresponding to bits p are taken for mapping to the codes, each bit p is removed to ensure creation of the gap required by the compressed mode, as described next.

Bits on the 1st PhCH after physical channel segmentation: $u_{1,k} = x_{i f(k)}, k = 1, 2, \dots, U$

Bits on 2nd PhCH after physical channel segmentation: $u_{2,k} = x_{i f(k+U)}, k = 1, 2, \dots, U$

Bits on P th PhCH after physical channel segmentation: $u_{P,k} = x_{i f(k+(P-1)U)}, k = 1, 2, \dots, U$

where f is such that:

- in modes other than compressed mode by puncturing, $x_{i f(k)} = x_{i k}$, i.e. $f(k) = k$, for all k ;
- in the compressed mode by puncturing, bit $u_{1,1}$ corresponds to the bit $x_{i k}$ with smallest index k when the bits p are not counted, bit $u_{1,2}$, it corresponds to the bit $x_{i k}$ with second smallest index k when the bits p are not counted, and so on for bits $u_{1,3}, \dots, u_{1,V}, u_{2,1}, u_{2,2}, \dots, u_{2,V}, \dots, u_{P,1}, u_{P,2}, \dots, u_{P,V}$.

We denote the bits input to the physical segmentation by $s_1, s_2, s_3, \dots, s_S$. Hence, $x_k = s_k$ and $Y = S$. We denote the bits input to the physical segmentation by $w_1, w_2, w_3, \dots, w_{(PU)}$. Hence, $x_k = w_k$ and $Y = PU$ [10].

4.6.11 Second Interleaving

The 2nd interleaving consists on a block interleaver with inter-column permutations. We denote the input bits to the 2nd interleaver by $u_{p1}, u_{p2}, u_{p3}, \dots, u_{pU}$, where p is PhCH number and U is the number of bits in one radio frame for one PhCH. The matrix configuration can be as follows:

- Set the number of columns $C_2 = 30$. Number columns $0, 1, 2, \dots, C_2 - 1$ from left to right.
- Determine the number of rows R_2 by finding minimum integer R_2 such that: $U \leq R_2 C_2$.
- The bits input to the 2nd interleaving are written into the $R_2 \times C_2$ rectangular matrix row by row.

$$\begin{bmatrix} u_{p1} & u_{p2} & u_{p3} & \dots & u_{p30} \\ u_{p31} & u_{p32} & u_{p33} & \dots & u_{p60} \\ \vdots & \vdots & \vdots & \dots & \vdots \\ u_{p,((R_2-1)30+1)} & u_{p,((R_2-1)30+2)} & u_{p,((R_2-1)30+3)} & \dots & u_{p,(R_2 30)} \end{bmatrix}. \quad (4.91)$$

- Perform the inter-column permutation based on the pattern $\{P_2(j)\}$ ($j = 0, 1, \dots, C_2 - 1$) which Table 4.39 illustrates, and where $P_2(j)$ is the original column position of the j th permuted column. After permutation of the columns, the bits are denoted by y_{pk} [10].

$$\begin{bmatrix} y_{p1} & y_{p,(R_2+1)} & y_{p,(2R_2+1)} & \dots & y_{p,(29R_2+1)} \\ y_{p2} & y_{p,(R_2+2)} & y_{p,(2R_2+2)} & \dots & y_{p,(29R_2+2)} \\ \vdots & \vdots & \vdots & \dots & \vdots \\ y_{pR_2} & y_{p,(2R_2)} & y_{p,(3R_2)} & \dots & y_{p,(30R_2)} \end{bmatrix}. \quad (4.92)$$

The output of the 2nd interleaving corresponds to the bit sequence read out column by column from the inter-column permuted $R_2 \times C_2$ matrix. We prune the output by deleting bits that were not present in the input bit sequence, i.e. bits y_{pk} that correspond to bits u_{pk} with $k > U$. We denote the bits after 2nd interleaving by $v_{p1}, v_{p2}, \dots, v_{pU}$, where v_{p1} corresponds to the bit y_{pk} with smallest index k after pruning, v_{p2} to the bit y_{pk} with second smallest index k after pruning, and so on.

Table 4.40 Inter-column Permutations [10]

Number of column C_2	Inter-column permutation pattern
30	{0, 20, 10, 5, 15, 25, 3, 13, 23, 8, 18, 28, 1, 11, 21, 6, 16, 26, 4, 14, 24, 19, 9, 29, 12, 2, 7, 22, 27, 17}

4.6.12 Physical Channel Mapping

Specifications in [1] and the earlier sections in this chapter define the PhCH for both uplink and downlink. We denote the input bits to the physical channel mapping by $v_{p1}, v_{p2}, \dots, v_{pU}$, where p is the PhCH number and U is the number of bits in one radio frame for one PhCH. We map the bits v_{pk} to the PhCHs so that the bits for each PhCH are transmitted over the air in ascending order with respect to k .

In the compressed mode, no bits get mapped to certain slots of the PhCH(s). Likewise, if $N_{\text{first}} + \text{TGL} \leq 15$, no bits get mapped to slots N_{first} to N_{last} . If $N_{\text{first}} + \text{TGL} > 15$, i.e. the transmission gap spans two consecutive radio frames, the mapping is as follows:

- in the first radio frame, no bits are mapped to slots $N_{\text{first}}, N_{\text{first}}+1, N_{\text{first}}+2, \dots, 14$.
- in the second radio frame, no bits are mapped to the slots $0, 1, 2, \dots, N_{\text{last}}$.

We describe TGL, N_{first} , and N_{last} while presenting the compressed mode section.

4.6.12.1 Uplink and Downlink

Uplink. PhCHs used during a radio frame can go either full of bits transmitted over the air or not used at all. However, with UE in compressed mode the transmission gets turned off during consecutive slots of the radio frame.

Downlink. PhCHs do not need to be transmitted full of bits over the air, e.g. bits $v_{pk} \notin \{0, 1\}$ do not get transmitted. During compressed mode when reducing the SF by 2, no bits get mapped to the DPDCH field. See the logic of this event and more details in the DL physical channel mapping in [10].

The preceding sections complete the functional description of the multiplexing structure illustrated in Figure 4.49. Additional details on the presentation of each block can be found in [10] and [18].

4.6.13 Detection of the Transport Format

When the transport format set of a TrCH i contains more than one transport format, we can detect them according to one of the following schemes [10]:

- *Transport Format Combination Indicator (TFCI)* based detection: this scheme applies when the transport format combination signals using the TFCI field;
- *explicit blind detection*: consists of detecting the TF of TrCH i by means of channel decoding and CRC check;
- *guided detection*: applies when there exists at least one other TrCH i' , hereafter called guiding TrCH, such that:
 - the guiding TrCH has the same TTI duration as the TrCH under consideration, i.e. $F_{i'} = F_i$;
 - different TFs of the TrCH under consideration correspond to different TFs of the guiding TrCH;
 - we can use explicit blind detection on the guiding TrCH.

If the *transport format set* for a TrCH i contains one transport format only, we do not need a transport format detection event for this TrCH. In the uplink, the blind transport format detection corresponds to network controlled option. In the downlink, the UE will perform blind transport format detection, when given conditions on the configured transport channels comply. For a DPCH associated with a PDSCH, the DPCCCH includes TFCI.

4.6.13.1 Blind Transport Format Detection

In the absence of TFCI explicit blind detection or guided detection takes place on all TrCHs within the CCTrCH that have more than one transport format. However according to [10], the UE will support blind transport format detection only if all of the following conditions apply:

1. the number of CCTrCH bits received per radio frame ≤ 600 ;
2. the number of transport format combinations of the CCTrCH ≤ 64 ;
3. the CCTrCH under detection use fixed positions of the transport channels;
4. all explicitly detected TrCHs use convolutional coding;
5. we append CRC to all transport blocks on all explicitly detected TrCHs;
6. the number of explicitly detected TrCHs ≤ 3 ;
7. for all explicitly detected TrCHs i , the number of code blocks in one TTI (C_i) does not exceed 1;
8. the sum of the transport format set¹⁴ sizes of all explicitly detected TrCHs, ≤ 16 ;
9. there is at least one usable TrCH in guiding a transport channel for all transport channels using guided detection. See examples in [10].

4.6.13.2 Transport Format Detection Based on TFCI

When a TFCI exists, TFCI based detection applies to all TrCHs within the CCTrCH, where the TFCI informs the receiver about the transport format combination of the CCTrCHs. Right after the TFCI detection we know the transport format combination as well as the transport formats of the individual transport channels.

TFCI coding. As illustrated in Figure 4.52, we encode TFCI bits using a (32,10) sub-code of the second order Reed–Muller code.

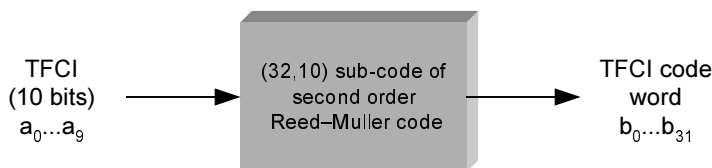


Figure 4.52 Channel coding of TFCI bits.

- If the TFCI < 10 bits, we pad it with zeros to 10 bits, by setting the most significant bits to zero. The length of the TFCI code word = 32 bits. The code words of the (32,10) sub-code of second order Reed–Muller code has a linear combination of 10 basis sequences.
- If we define the TFCI information bits as $a_0, a_1, a_2, a_3, a_4, a_5, a_6, a_7, a_8, a_9$, where a_0 = LSB and a_9 = MSB, the TFCI information corresponds to the TFC defined by the

¹⁴ The transport format set size is defined as the number of transport formats within the transport format set.

RRC layer to reference the TFC of the CCTrCH in the associated DPCH radio frame. The output code word bits b_i are then given by:

$$b_i = \sum_{n=0}^9 (a_n \times M_{i,n}) \bmod 2, \quad (4.93)$$

where $i = 0, \dots, 31$, and the output bits are denoted by b_k , $k = 0, 1, 2, \dots, 31$.

In the downlink if the SF < 128, the encoded TFCI code words get repeated yielding 8 encoded TFCI bits per slot in normal mode and 16 encoded TFCI bits per slot in compressed mode.

4.6.13.3 TFCI Operation in the Split Mode

If one of the DCH has association with a DSCH, the TFCI code word gets split in such a way that the code word relevant for TFCI activity indication is not transmitted from every cell. Higher layer signalling indicates usage of this latter functionality.

In this case we encode the TFCI bits using a (16,5) bi-orthogonal (or first order Reed–Muller) code as illustrated in Figure 4.53. Table 4.42 illustrates the code words of the (16,5) bi-orthogonal code, which are linear combinations of 5 basic sequences.

If we define a 1st set of TFCI information bits as $a_{1,0}, a_{1,1}, a_{1,2}, a_{1,3}, a_{1,4}$, where $a_{1,0}$ = LSB and $a_{1,4}$ = MSB, we can assume this set of TFCI information bits will correspond to the TFC index¹⁵ defined by the RRC layer to reference the TFC of the DCH CCTrCH in the associated DPCH radio frame. Likewise, if we define a 2nd set of TFCI information bits as $a_{2,0}, a_{2,1}, a_{2,2}, a_{2,3}, a_{2,4}$, where $a_{2,0}$ = LSB and $a_{2,4}$ = MSB, we can assume that this set of TFCI information bits will correspond to the TFC index defined by the RRC layer to reference the TFC of the associated DSCH CCTrCH in the corresponding PDSCH radio frame. Then, the output code word bits b_k are given by [10]:

$$b_{2i} = \sum_{n=0}^4 (a_{1,n} \times M_{i,n}) \bmod 2, \quad b_{2i+1} = \sum_{n=0}^4 (a_{2,n} \times M_{i,n}) \bmod 2, \quad (4.93)$$

where $i = 0, \dots, 15$, $j = 0, 1$ and the output bits are: b_k , $k = 0, 1, 2, \dots, 31$.

Table 4.41 Basis Sequences for (32,10) TFCI Code [10]

i	$M_{i,0}$	$M_{i,1}$	$M_{i,2}$	$M_{i,3}$	$M_{i,4}$	$M_{i,5}$	$M_{i,6}$	$M_{i,7}$	$M_{i,8}$	$M_{i,9}$
0	1	0	0	0	0	1	0	0	0	0
1	0	1	0	0	0	1	1	0	0	0
2	1	1	0	0	0	1	0	0	0	1
3	0	0	1	0	0	1	1	0	1	1
4	1	0	1	0	0	1	0	0	0	1
5	0	1	1	0	0	1	0	0	1	0
6	1	1	1	0	0	1	0	1	0	0
7	0	0	0	1	0	1	0	1	1	0

¹⁵ Expressed in unsigned binary.

8	1	0	0	1	0	1	1	1	1	0
9	0	1	0	1	0	1	1	0	1	1
10	1	1	0	1	0	1	0	0	1	1
11	0	0	1	1	0	1	0	1	1	0
12	1	0	1	1	0	1	0	1	0	1
13	0	1	1	1	0	1	1	0	0	1
14	1	1	1	1	0	1	1	1	1	1
15	1	0	0	0	1	1	1	1	0	0
16	0	1	0	0	1	1	1	1	0	1
17	1	1	0	0	1	1	1	0	1	0
18	0	0	1	0	1	1	0	1	1	1
19	1	0	1	0	1	1	0	1	0	1
20	0	1	1	0	1	1	0	0	1	1
21	1	1	1	0	1	1	0	1	1	1
22	0	0	0	1	1	1	0	1	0	0
23	1	0	0	1	1	1	1	1	0	1
24	0	1	0	1	1	1	1	0	1	0
25	1	1	0	1	1	1	1	0	0	1
26	0	0	1	1	1	1	0	0	1	0
27	1	0	1	1	1	1	1	1	0	0
28	0	1	1	1	1	1	1	1	1	0
29	1	1	1	1	1	1	1	1	1	1
30	0	0	0	0	0	1	0	0	0	0
31	0	0	0	0	1	1	1	0	0	0

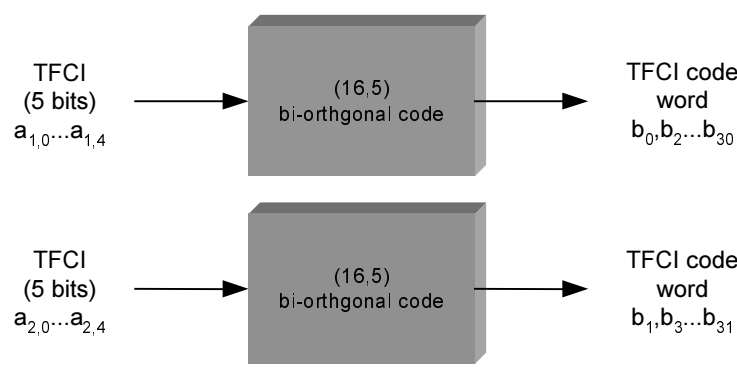


Figure 4.53 Channel coding of split mode TFCI bits.

Table 4.42 Basis Sequences for (16,5) TFCI Code [10]

i	$M_{i,0}$	$M_{i,1}$	$M_{i,2}$	$M_{i,3}$	$M_{i,4}$
0	1	0	0	0	1
1	0	1	0	0	1
2	1	1	0	0	1
3	0	0	1	0	1
4	1	0	1	0	1

5	0	1	1	0	1
6	1	1	1	0	1
7	0	0	0	1	1
8	1	0	0	1	1
9	0	1	0	1	1
10	1	1	0	1	1
11	0	0	1	1	1
12	1	0	1	1	1
13	0	1	1	1	1
14	1	1	1	1	1
15	0	0	0	0	1

4.6.14 Mapping of TFCI Words

In non-compressed mode, we map cord word bits directly to the slots of the radio frame, the bit with lower index gets transmitted before the bit with higher index. The coded bits b_k , get mapped to the transmitted TFCI bits d_k , as follows: $d_k = b_{k \bmod 32}$.

For the UL physical channels, despite the SF and DL physical channels, if the $SF \geq 128$, $k = 0, 1, 2, \dots, 29^{16}$. In like manner, for the DL physical channels whose $SF < 128$, $k = 0, 1, 2, \dots, 119$. The latter implies that bits b_0 to b_{23} get transmitted four times and bits b_{24} to b_{31} get transmitted three times [10].

In the uplink compressed-mode, we map TFCI bits differently for downlink with $SF \geq 128$ and downlink with $SF < 128$. The slot format gets changed so that we do not lose TFCI bits. We repeat TFCI bits because the different slot formats in compressed mode do not match the exact number of TFCI bits for all possible TGLs.

Denoting the number of bits available in the TFCI fields of one compressed radio frame by D and the number of bits in the TFCI field in a slot by N_{TFCI} , we obtain the first bit to get repeated, $E = N_{\text{first}} N_{\text{TFCI}}$. When $N_{\text{last}} \neq 14$, then E corresponds to the number of the first TFCI bit in the slot directly after the TG. The following expressions define the uplink mapping:

$$d_k = b_{k \bmod 32} \quad \text{where } k = 0, 1, 2, \dots, \min(31, D-1). \quad (4.95)$$

If $D > 32$, the remaining positions get filled in reverse order by repetition:

$$d_{D-k-1} = b_{(E+k) \bmod 32} \quad \text{where } k = 0, \dots, D-33. \quad (4.96)$$

In the downlink compressed mode we change the slot format to prevent TFCI bit losses. When the slot formats do not match the exact number of TFCI bits for all possible TGLs and the number of TFCI fields exceeds the number of TFCI bits we use DTX. The block of fields, where we use DTX starts on the first field after the gap. If fewer TFCI fields exist after the gap than DTX bits, the last fields before the gap can also get filled with DTX. Denoting the number of bits available in the TFCI fields of one com-

¹⁶ This implies that bits b_{30} and b_{31} do not get transmitted.

pressed radio frame by D and the number of bits in the TFCI field in a slot by N_{TFCI} , then we can express E , the first bit to be repeated as:

$$E = N_{\text{first}} N_{\text{TFCI}}, \quad \text{if } N_{\text{first}} + \text{TGL} \leq 15, \text{ else } E = 0. \quad (4.97)$$

When the transmission gap does not extend to the end of the frame, then E corresponds to the number of the first TFCI bit in the slot directly after the TG. We denote the total number of TFCI bits to be transmitted by N_{tot} . Thus, if $\text{SF} \geq 128$ then $N_{\text{tot}} = 32$, else $N_{\text{tot}} = 128$. Afterwards, the following relations define the mapping:

$$d_k = b_{(k \bmod 32)} \quad \text{where } k = 0, 1, 2, \dots, \min(E, N_{\text{tot}}) - 1 \text{ and if } E < N_{\text{tot}}, \quad (4.98)$$

$$d_{k+D-N_{\text{tot}}} = b_{(k \bmod 32)} \quad \text{where } k = E, \dots, N_{\text{tot}} - 1. \quad (4.99)$$

DTX bits are sent on d_k where $k = \min(E, N_{\text{tot}}), \dots, \min(E, N_{\text{tot}}) + D - N_{\text{tot}} - 1$ [10].

4.6.15 Examples on Channel Coding and Multiplexing

In the following we illustrate channel coding and multiplexing examples from [21] following the principles outlined in [10,20]. The examples aim to practically show the patterns and fields to code different frames in the UTRA FDD and TDD modes. Thus, the number and variables in the forthcoming figures show the number of bits in the corresponding fields.

4.6.15.1 Downlink FDD BCH

The parameters for the BCH shown in Table 4.43 indicate CRC bits, coding type, TTI, the number of codes used, and the spreading factor (SF).

Figure 4.54 illustrates the patterns of the bits in the corresponding fields of the DL FDD BCH example. Notice that we do not necessarily follow all the steps outlined in the preceding section. Thus, each particular channel will use only the corresponding steps.

Table 4.43 Downlink FDD BCH Parameters

Transport block size	246
CRC	16 bits
Coding	CC, coding rate = 1/2
TTI	20 ms
The number of codes	1
SF	256

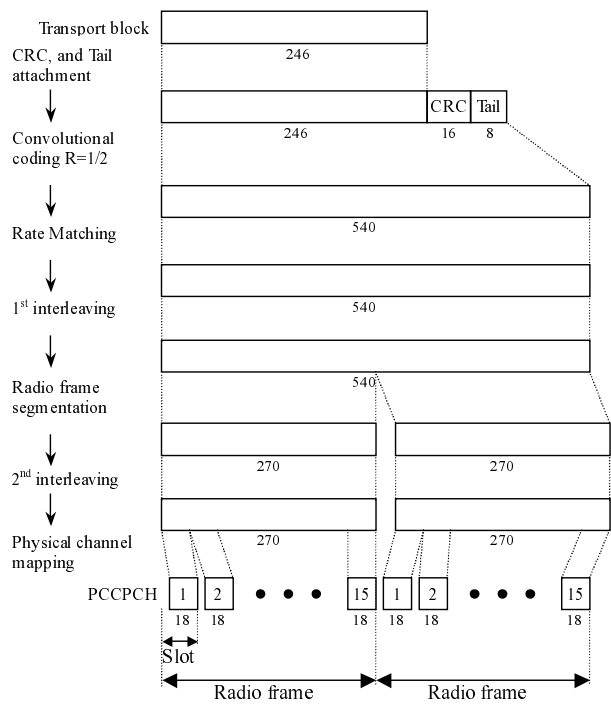


Figure 4.54 Coding the FDD downlink BCH.

4.6.15.2 Speech Channel Coding in the FDD Mode

We next illustrate the coding of transport channel for CS data or speech services. Notice how this example applies to 12.2 kbps AMR speech.

Table 4.44 and Figure 4.55 illustrate the key parameters for the 12.35 kbps AMR speech data. See the number of TrChs coding example when compared to the control channel in the previous example.

Table 4.44 Parameter Examples for 12.35 kbps Speech Information

The number of TrChs	3
Transport block size	81, 103, and 60 bits
CRC	12 bits (attached only to TrCh#1)
Coding	CC, coding rate = 1/3 for TrCh#1, 2 coding rate = 1/2 for TrCh#3
TTI	20 ms

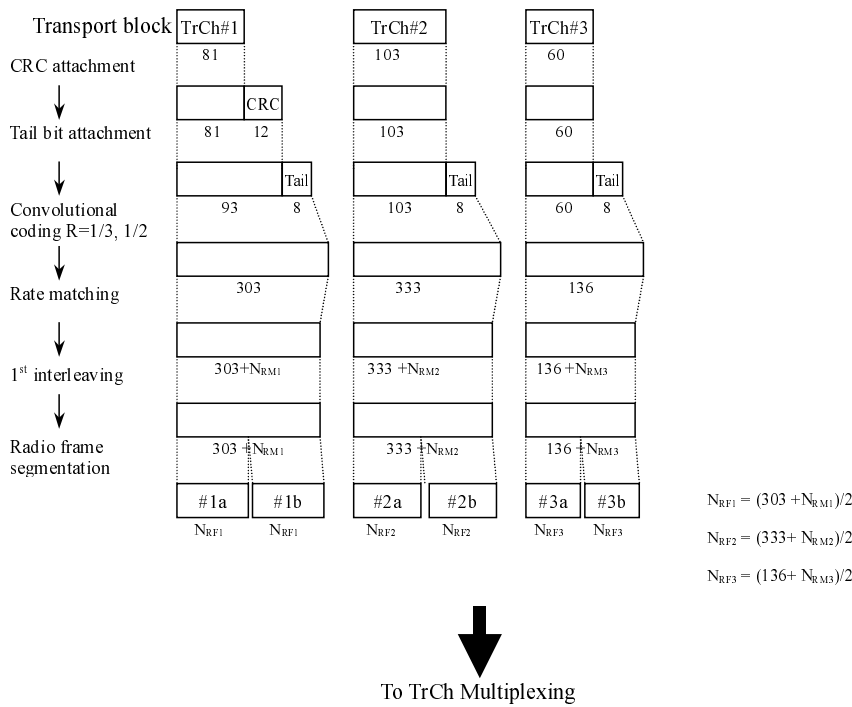


Figure 4.55 Channel coding and multiplexing 12.2 kbps speech.

4.6.15.3 Coding FDD 64/128/384 kbps Packet Data Channels

Table 4.45 and Figure 4.56 show the key coding and multiplexing parameters for the aforementioned packet data channels. Here the number of blocks used logically depends on the transmission data rate required.

Notice also that when coding these transport channels we use turbo coding instead of the convolution codes in the preceding examples.

Table 4.45 Packet Data Parameters for 64/128/384 kbps Services

The number of TrChs		1
Transport block size		640 bits
Transport block	64 kbps	$640 \times B$ bits ($B = 0, 1$)
Size size	128 kbps	$640 \times B$ bits ($B = 0, 1, 2$)
	384 kbps	$640 \times B$ bits ($B = 0, 1, 2, \dots, 6$)
CRC		16 bits
Coding		Turbo coding, coding rate = 1/3
TTI		10 ms

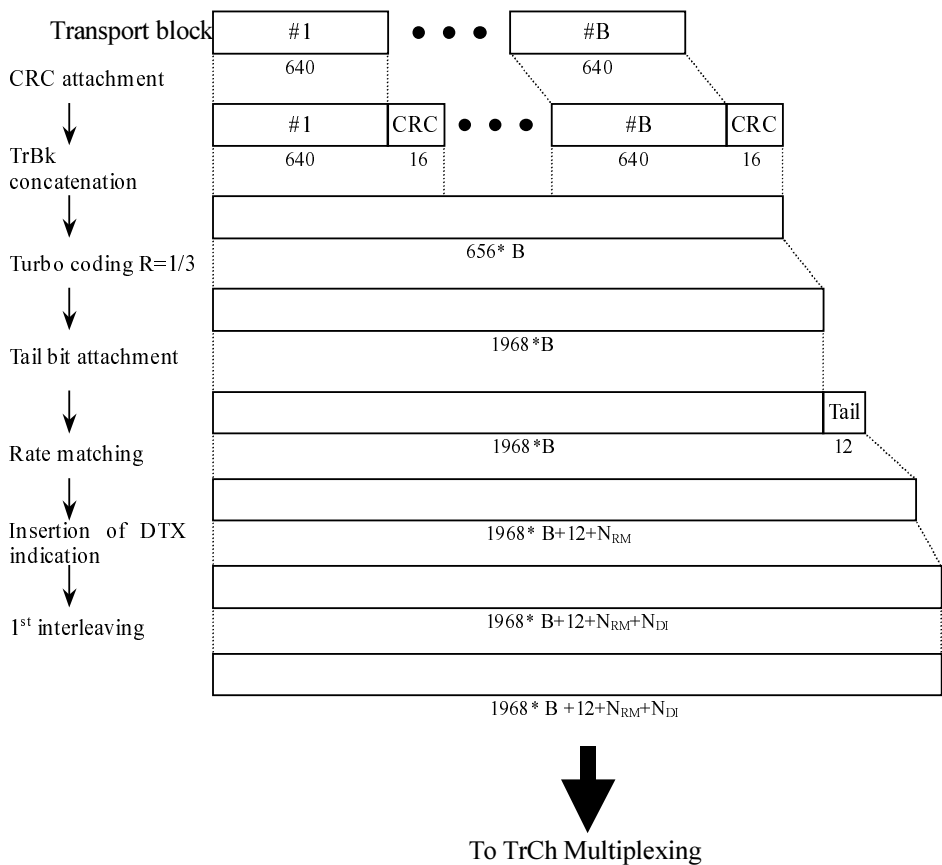


Figure 4.56 Channel coding and multiplexing 64/128/384 kbps packet data channels.

4.6.15.4 Multiplexing of 64/128/384 kbps Packet Data and 4.1 kbps Data

This example applies to multiplexing 64/128/384 kbps packet data and DCCH. Table 4.46 and Figure 4.57 show a second view of the key physical channel parameters for multiplexing of 64/128/384 kbps packet data and 4.1 kbps data.

Table 4.46 Physical Channel Parameters to Multiplex 64/128/384 kbps Packet Data and 4.1 kbps Data

Data rate (kbps)	Symb. Rate (ksps)	No. of physical channel: P	N_{pilot} (bits)	N_{TFCI} (bits)	N_{TPC} (bits)	N_{data1} (bits)	N_{data2} (bits)
64	120	1	8	8	4	4	56
128	240	1	16	8	8	48	240
384	240	3	16	8	8	48	240

Other examples for FDD and TDD can be found in [21].

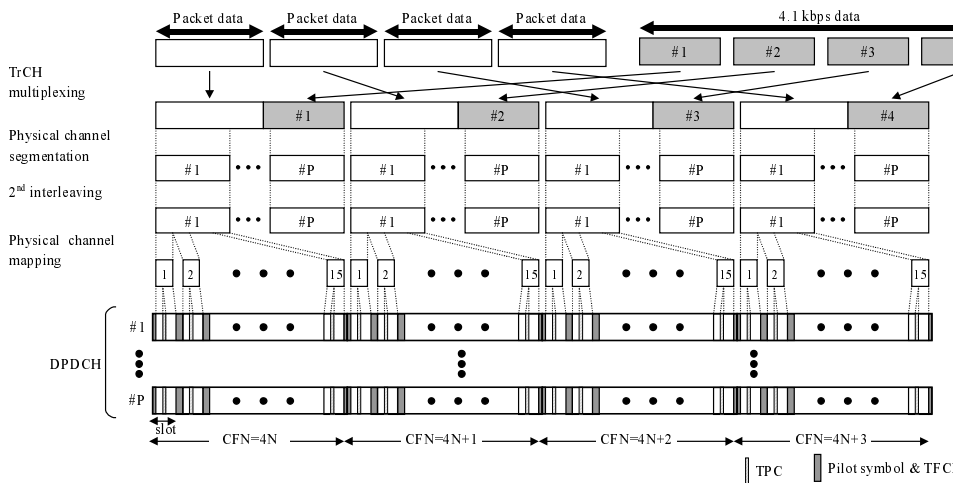


Figure 4.57 Channel coding and multiplexing 64/128/384 kbps packet data and 4.1 kbps data.

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APPENDIX A: DPDCH AND DPCCH FIELDS

Table 4.47 DPDCH and DPCCH Fields

Slot format # <i>i</i>	Channel bit rate (kbps)	Channel symbol rate (ksps)	SF	Bits/ slot	DPDCH bits/slot		DPCCH bits/slot			Transmitted slots per radio frame, N_{Tr}
					N_{Data1}	N_{Data2}	N_{TPC}	N_{TFCI}	N_{Pilot}	
0	15	7.5	512	10	0	4	2	0	4	15
0A	15	7.5	512	10	0	4	2	0	4	8–14
0B	30	15	256	20	0	8	4	0	8	8–14
1	15	7.5	512	10	0	2	2	2	4	15
1B	30	15	256	20	0	4	4	4	8	8–14
2	30	15	256	20	2	14	2	0	2	15
2A	30	15	256	20	2	14	2	0	2	8–14
2B	60	30	128	40	4	28	4	0	4	8–14
3	30	15	256	20	2	12	2	2	2	15
3A	30	15	256	20	2	10	2	4	2	8–14
3B	60	30	128	40	4	24	4	4	4	8–14
4	30	15	256	20	2	12	2	0	4	15
4A	30	15	256	20	2	12	2	0	4	8–14
4B	60	30	128	40	4	24	4	0	8	8–14
5	30	15	256	20	2	10	2	2	4	15
5A	30	15	256	20	2	8	2	4	4	8–14
5B	60	30	128	40	4	20	4	4	8	8–14
6	30	15	256	20	2	8	2	0	8	15
6A	30	15	256	20	2	8	2	0	8	8–14
6B	60	30	128	40	4	16	4	0	16	8–14
7	30	15	256	20	2	6	2	2	8	15
7A	30	15	256	20	2	4	2	4	8	8–14
7B	60	30	128	40	4	12	4	4	16	8–14
8	60	30	128	40	6	28	2	0	4	15
8A	60	30	128	40	6	28	2	0	4	8–14
8B	120	60	64	80	12	56	4	0	8	8–14
9	60	30	128	40	6	26	2	2	4	15
9A	60	30	128	40	6	24	2	4	4	8–14
9B	120	60	64	80	12	52	4	4	8	8–14
10	60	30	128	40	6	24	2	0	8	15
10A	60	30	128	40	6	24	2	0	8	8–14
10B	120	60	64	80	12	48	4	0	16	8–14
11	60	30	128	40	6	22	2	2	8	15
11A	60	30	128	40	6	20	2	4	8	8–14
11B	120	60	64	80	12	44	4	4	16	8–14
12	120	60	64	80	12	48	4	8*	8	15
12A	120	60	64	80	12	40	4	16*	8	8–14

12B	240	120	32	160	24	96	8	16*	16	8–14
13	240	120	32	160	28	112	4	8*	8	15
13A	240	120	32	160	28	104	4	16*	8	8–14
13B	480	240	16	320	56	224	8	16*	16	8–14
14	480	240	16	320	56	232	8	8*	16	15
14A	480	240	16	320	56	224	8	16*	16	8–14
14B	960	480	8	640	112	464	16	16*	32	8–14
15	960	480	8	640	120	488	8	8*	16	15
15A	960	480	8	640	120	480	8	16*	16	8–14
15B	1920	960	4	1280	240	976	16	16*	32	8–14
16	1920	960	4	1280	248	1000	8	8*	16	15
16A	1920	960	4	1280	248	992	8	16*	16	8–14

* If TFCI bits are not used, then DTX shall be used in TFCI field.

NOTE1: Compressed mode is only supported through spreading factor reduction for SF = 512 with TFCI.

NOTE2: Compressed mode by spreading factor reduction is not supported for SF = 4.

APPENDIX B: BIT PATTERNS COMPRESSED MODE AND $N_{\text{PILOT}} = 4$

For slot formats 2B and 3B, i.e. compressed mode through spreading factor reduction and $N_{\text{pilot}} = 4$, the pilot bits on antenna 1 are STTD encoded. Thus, the pilot bit pattern is as shown in the most right set of Table 4.14.

Slot	$N_{\text{pilot}} = 4$	
	0	1
0	01	10
1	10	01
2	11	00
3	10	01
4	00	11
5	01	10
6	01	10
7	00	11
8	11	00
9	01	10
10	11	00
11	00	11
12	00	11
13	10	01
14	10	01

5 THE UTRA¹ TRANSMISSION SYSTEM

5.1 UMTS SPECTRUM ALLOCATION

The UMTS frequency ranges are part of the world wide spectrum allocation for 3rd or evolving 2nd generation systems. Figure 5.1 illustrates the representation of the spectrum from major regions (e.g. Europe, Japan, Korea, and USA).

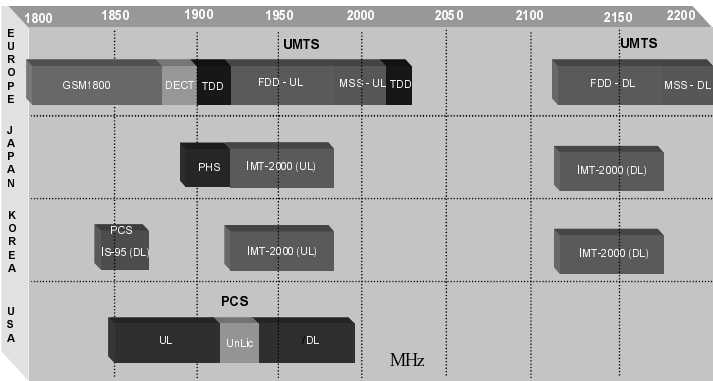


Figure 5.1 Spectrum allocation representation for 3G systems.

The distribution of the frequency bands from the allocated spectrum for the UTRA system is covered next. We present the ranges for the FDD and the TDD in parallel in order to unveil a complete view of the UMTS frequency assignment.

5.1.1 UTRA Frequency Bands

Table 5.1 summarizes the frequency bands for the TDD and FDD modes, as well as the frequency distribution for the User Equipment (UE) and the Base Station (BS). Although, in some cases the frequency ranges may be the same for both UE and BS, they are noted separately for completeness.

Additional spectrum allocations in ITU region 2 are FFS, and deployment of UMTS in existing and other frequency bands is not precluded. Furthermore, co-existence of TDD and FDD in the same bands (now under study) may be possible.

¹ The UMTS Terrestrial Radio Access.

Table 5.1 UTRA Frequency Bands in the MS and BS Side

Case	FDD (MHz)			TDD (MHz) up- and downlink	
		User equip- ment	Base station	User equip- ment	Base station
(a)	Uplink				
	(MS to BS)	1920–1980	1920–1980	1900–1920	1900–1920
	Downlink				
	(BS to MS)	2110–2170	2110–2170	2010–2025	2010–2025
Region 2 – e.g. Europe					
(b)	Uplink				
	(MS to BS)	1850–1910	1850–1910	1850–1910	1850–1910
	Downlink				
	(BS to MS)	1930–1990	1930–1990	1930–1990	1930–1990
(c)				1910–1930	1910–1930

5.2 RADIO TRANSMISSION AND RECEPTION ASPECTS

After the allocation of the frequency ranges for the UTRA modes in the preceding section, in the following we present the transceiver parameters from the technical specifications, [1–4]. These parameters will set the necessary background to consider equipment and network design, including traffic engineering issues.

5.2.1 Transmit to Receive (TX-RX) Frequency Separation

While the TDD mode does not need Frequency Separation (FS), the FDD mode does in both the EU and the BS.

Table 5.2 UTRA TX-RX Frequency Separation

	FDD	TDD
	User Equipment (UE) and Base Station (BS)	UE and BS
1.	Minimum value = 134.8 MHz Maximum value = 245.2 MHz All UE(s) shall support 190 MHz FS in case (a) ¹	No TX-RX frequency separation is required
2.	All UE(s) shall support 80 MHz FS in case (b) ¹	Each TDMA frame has 15 time slots
3.	FDD Can support both fixed and variable TX-RX FSs	Each time slot can be allocated to either transit (TX) or receive (RX)
4.	Use of other TX-RX FSs in existing or other frequency bands shall not be precluded	

¹ When operating within spectrum allocations of cases (a) and (b) Table 5.1, respectively.

5.2.2 Channel Configuration

The channel spacing, raster and numbering arrangements aim to synchronize in both FDD and TDD modes as well as keep certain compatibility with GSM, in order to facilitate multi-mode system designs. This applies, e.g. to the raster distribution where 200 kHz corresponds to all (UE and BS in FDD and TDD modes). Table 5.3 summarizes the specified channel configurations:

Table 5.3 UTRA Channel Configurations

	FDD (MHz)		TDD (MHz)
Channel:	UE and BS		UE and BS
Spacing	5 MHz		5 MHz
Raster	200 kHz		200 kHz
Number	UL	$N_u = 5 \times ({}^1F_{\text{uplink}} \text{ MHz})$ $0.0 \text{ MHz} \leq F_{\text{uplink}} \leq 3276.6 \text{ MHz}$	$N_t = 5 \times (F - \text{MHz})$ $0.0 \text{ MHz} \leq F \leq 3276.6 \text{ MHz}$ F is the carrier frequency in MHz
	DL	$N_d = 5 \times ({}^1F_{\text{downlink}} \text{ MHz})$ $0.0 \text{ MHz} \leq F_{\text{downlink}} \leq 3276.6 \text{ MHz}$	

¹ F_{uplink} and F_{downlink} are the uplink and downlink frequencies in MHz, respectively.

The nominal channel spacing (i.e. 5 MHz) can be adjusted to optimize performance depending on the deployment scenarios; and the channel raster (i.e. 200 kHz) implies the centre frequency which must be an integer multiple of 200 kHz.

In the case of the channel number, the carrier frequency is designated by the UTRA Absolute Radio Frequency Channel Number (UARFCN), Table 5.3 shows those defined in the IMT2000 band.

5.3 TRANSMITTER CHARACTERISTICS

As in the UE or otherwise stated, we specify transmitter characteristics at the BS antenna connector (test port A) with a full complement of transceivers for the configuration in normal operating conditions. When using external apparatus (e.g. TX amplifiers, diplexers, filters or a combination of such devices, requirements apply at the far end antenna connector (port B).

5.3.1 Maximum Output Power

5.3.1.1 User Equipment (UE)

At this time detailed transmitter characteristics of the antenna connectors in the UE are not available; thus, a reference UE with integral antenna and antenna gain of 0 dBi is assumed. For the definition of the parameters to follow we use the UL reference measurement channel (12.2 kbps) illustrated in Table 5.4, other references can be found in [1,2].

Table 5.4 UL Reference Measurement Channel Physical Parameters (12.2 kbps)

FDD		TDD	
Parameter	Level	Parameter	Level
Information bit rate (kbps)	12.2	Information data rate	12.2 kbps
DPDCH (kbps)	60	RUs allocated	2 RU
DPCCH (kbps)	15	Mid-amble	512 chips
DPCCH/DPDCH (dB)	−6	Interleaving	20 ms
TFCI	On	Power control	2 bit/user
Repetition (%)	23	TFCI	16 bit/user
		Inband signalling DCCH	2 kbps
		Puncturing level at code rate 1/3 : DCH / DCCH	5%/0%

About four UE power classes have been defined (Table 5.5). The tolerance of the maximum output power is below the suggested level even when we would use multi-code transmission mode in the FDD and TDD modes.

Other cases applying to the TDD mode from [2] are:

- Maximum output power refers to the measure of power while averaged over the useful part of transmit time slots with maximum power control settings.
- In multi-code operation the maximum output power decreases by the difference of the peak to average ratio between single and multi-code transmission.
- UE using directive antennas for transmission, will have a class dependent limit placed on the maximum Equivalent Isotropic Radiated Power (EIRP).

Table 5.5 UE Power Classes

	FDD		TDD	
Power Class	Maximum output power (dBm)	Tolerance (dB)	Maximum output power (dBm)	Tolerance (dB)
1	+33	+1/-3		
2	+27	+1/-3	+24	+1/-3
3	+24	+1/-3	+21	+2/-2
4	+21	± 2		

5.3.1.2 Base Station Output Power

In the TDD mode, *BS output power*, P_{out} , represents the one carrier mean power delivered to a load with resistance equal to the nominal load impedance of the transmitter during one slot. Likewise, *BS rated output power*, P_{RAT} , indicates the manufacturer declared mean power level per carrier over an active timeslot available at the antenna connector [4].

In FDD or TDD *BS maximum output power*, P_{max} , implies the mean power level per carrier measured at the antenna connector in specified reference conditions. In normal conditions, BS maximum output power remains within +2 dB and -2dB of the manufacturer's rated output power. In extreme conditions, BS maximum output power remains within +2.5 dB and -2.5 dB of the manufacturer's rated output power.

5.3.2 Frequency Stability

Here frequency stability applies to both FDD and TDD modes. The required accuracy of the UE modulated carrier frequency lies within ± 0.1 ppm when compared to the carrier frequency received from the BS. The signals have apparent errors as a result of BS frequency error and Doppler shift; hence signals from the BS need averaging over sufficient time.

The BS modulated carrier frequency is accurate to within ± 0.05 ppm for RF frequency generation.

5.3.3 Output Power Dynamics

5.3.3.1 User Equipment

In the FDD as well as TDD we use power control to limit interference. The *Minimum Transmit Output Power* is better than -44 dBm measured with a Root-Raised Cosine (RRC) filter having a roll-off factor $\alpha = 0.22$ and a bandwidth equal to the chip rate.

5.3.3.1.1 Open Loop Power Control

Open loop power control enables the UE transmitter to sets its output power to a specific value, where in normal conditions it has tolerance of ± 9 dB and ± 12 dB in extreme conditions. We defined it as the average power in a time slot or ON power duration depending on the availability. The two options are measured with a filter having a RRC response with a roll off $\alpha = 0.22$ and a bandwidth equal to the chip rate.

5.3.3.1.2 Uplink Inner Loop Power Control

Through the uplink inner loop power control the UE transmitter adjusts its output power according to one or more TPC command steps received in the downlink. The UE transmitter will change the output power in step sizes of 1, 2 and 3 dB, depending on derived Δ_{TPC} or $\Delta_{\text{RP-TPC}}$ values in the slot immediately after the TPC_cmd. Tables 5.6 and 5.7 illustrate the transmitter power control range and average output power, respectively.

Table 5.6 Transmitter Power Control Range

TPC_cmd	1 dB step size		2 dB step size		3 dB step size	
	Lower	Upper	Lower	Upper	Lower	Upper
+1	+0.5	+1.5	+1	+3	+1.5	+4.5
0	-0.5	+0.5	-0.5	+0.5	-0.5	+0.5
-1	-0.5	-1.5	-1	-3	-1.5	-4.5

We define the inner loop power as the relative power differences between averaged power of original (reference) time slot and averaged power of the target time slot without transient duration. The UE has minimum controlled output power with the power control set to its minimum value. This applies to both inner loop and open loop power control, where the minimum transmit power is better than -50 dBm [1]. They are measured with a filter that has a RRC filter response with a roll off $\alpha = 0.22$ and a bandwidth equal to the chip rate.

Table 5.7 Transmitter Average Power Control Range

TPC_cmd	Transmitter power control range after 10 equal TPC_cmd groups				Transmitter power control range after 7 equal TPC_cmd groups	
	1 dB step size		2 dB step size		3 dB step size	
	Lower	Upper	Lower	Upper	Lower	Upper
+1	+8	+12	+16	+24	+16	+26
0	-1	+1	-1	+1	-1	+1
-1	-8	-12	-16	-24	-16	-26
0,0,0,0,+1	+6	+14	N/A	N/A	N/A	N/A
0,0,0,0,-1	-6	-14	N/A	N/A	N/A	N/A

5.3.3.1.3 Uplink Power Control TDD

Through the uplink power control, the UE transmitter sets its output power taking into account the measured downlink path loss, values determined by higher layer signalling and filter response α . This power control has an *initial error accuracy* of less than ± 9 dB under normal conditions and ± 12 dB under extreme conditions.

From [2] we define the *power control differential accuracy* as the error in the UE transmitter power step, originating from a step in SIR_{TARGET} when the parameter $\alpha = 0$. The step in SIR_{TARGET} is rounded to the closest integer dB value. The error does not exceed the values illustrated in Table 5.8.

Table 5.8 Transmitter Power Step Tolerance in Normal Conditions¹

ΔSIR_{TARGET} (dB)	Transmitter power step tolerance (dB)
$\Delta SIR_{TARGET} \leq 1$	± 0.5
$1 < \Delta SIR_{TARGET} \leq 2$	± 1
$2 < \Delta SIR_{TARGET} \leq 3$	± 1.5
$3 < \Delta SIR_{TARGET} \leq 10$	± 2
$10 < \Delta SIR_{TARGET} \leq 20$	± 4
$20 < \Delta SIR_{TARGET} \leq 30$	± 6
$30 < \Delta SIR_{TARGET}$	$\pm 9^1$

¹For extreme conditions the value is ± 12 .

5.3.3.2 Base Station

In FDD the transmitter uses a quality-based power control on both the uplink and downlink to limit the interference level. In TDD the transmitter uses a quality-based power control primarily to limit the interference level on the downlink.

Through *inner loop power control* in the downlink the FDD BS transmitter has the ability to adjust the transmitter output power of a code channel in accordance with the corresponding TPC symbols received in the uplink. In the TDD inner loop control is based on SIR measurements at the UE receiver and the corresponding TPC commands are generated by the UE, although the latter may or does also apply to the FDD.

5.3.3.2.1 Power control steps

The *power control step change* executes stepwise variation in the DL transmitter output power of a code channel in response to a corresponding power control command. The *aggregated output power change* represents the required total change in the DL transmitter output power of a code channel while reacting to multiple consecutive power control commands corresponding to that code channel. The BS transmitter will have the capability of setting the inner loop output power with a step size of 1 dB mandatory and 0.5 dB optional [3]. The power control step and the aggregated output power change due to inner loop power control shall be within the range illustrated in Table 5.9.

In TDD, power control steps change the DL transmitter output power in response to a TPC message from the UE in steps of 1, 2, and 3 dB. The tolerance of the transmitter output power and the greatest average rate of change in mean power due to the power control step will remain within the range illustrated in Table 5.10.

Table 5.9 FDD Transmitter Power Control Steps and Aggregated Output Power Change Range

Power control commands in the down link	Transmitter power control step range			
	1 dB step size		0.5 dB step size	
	Lower	Upper	Lower	Upper
Up (TPC command “1”)	+0.5	+1.5	+0.25	+0.75
Down (TPC command “0”)	−0.5	−1.5	−0.25	−0.75
	Transmitter aggregated output power change range after 10 consecutive equal commands (up or down)			
	1 dB step size		0.5dB step size	
	Lower	Upper	Lower	Upper
Up (TPC command “1”)	+8	+12	+4	+6
Down (TPC command “0”)	−8	−12	−4	−6

Table 5.10 TDD Power Control Step Size Tolerance

Step size	Tolerance	Range of average rate of change in mean power per 10 steps	
		Minimum	Maximum
1dB	±0.5dB	±8dB	±12dB
2dB	±0.75dB	±16dB	±24dB
3dB	±1dB	±24dB	±36dB

5.3.3.2.2 Power Control Dynamic Range and Primary CPICH–CCPCH Power

We refer to the difference between the maximum and the minimum transmit output power of a code channel for a specified reference condition as the *power control dynamic range*. This range in the downlink (DL) has a maximum power → BS maximum output power of −3 dB or greater, and minimum power → BS maximum output power of −28 dB or less.

By *total power dynamic range* we mean the difference between the maximum and the minimum total transmit output power for a specified reference condition. In this case, the upper limit of the dynamic range is the BS maximum output power and the lower limit the lowest minimum power from the BS when no traffic channels are activated. The DL total power dynamic range is 18 dB or greater [3].

We call *Primary CPICH* power to the transmission power of the common pilot channel averaged over one frame and indicated in a BCH. This power is within ± 2.1 dB of the value indicated by a signalling message [3].

In TDD, the power control dynamic range, i.e. the difference between the maximum and the minimum transmit output power for a specified reference condition has a DL minimum requirement of 30 dB. The minimum transmit power, i.e. the minimum controlled BS output power with the power control setting set to a minimum value, has DL maximum output power of -30 dB. The primary CCPCH power is averaged over the transmit time slot and signalled over the BCH. The error between the BCH-broadcast value of the primary CCPCH power and the primary CCPCH power averaged over the time slot does not exceed the values illustrated in Table 5.11. The error is a function of the total power averaged over the timeslot, P_{out} , and the manufacturer’s rated output power, P_{RAT} [4].

Table 5.11 Errors Between Primary CCPCH Power and the Broadcast Value (TDD)

Total power in slot (dB)	PCCPCH power tolerance (dB)
$P_{\text{RAT}} - 3 < P_{\text{out}} \leq P_{\text{RAT}} + 2$	± 2.5
$P_{\text{RAT}} - 6 < P_{\text{out}} \leq P_{\text{RAT}} - 3$	± 3.5
$P_{\text{RAT}} - 13 < P_{\text{out}} \leq P_{\text{RAT}} - 6$	± 5

5.3.4 Out-of-Synchronization Output Power Handling

- The UE monitors the DPCCCH quality to detect L1 signal loss. The thresholds Q_{out} and Q_{in} specify at what DPCCCH quality levels the UE shall shut its power off and when it may turn its transmitter on, respectively. The thresholds are not defined explicitly, but are defined by the conditions under which the UE shuts its transmitter off and turns it on.

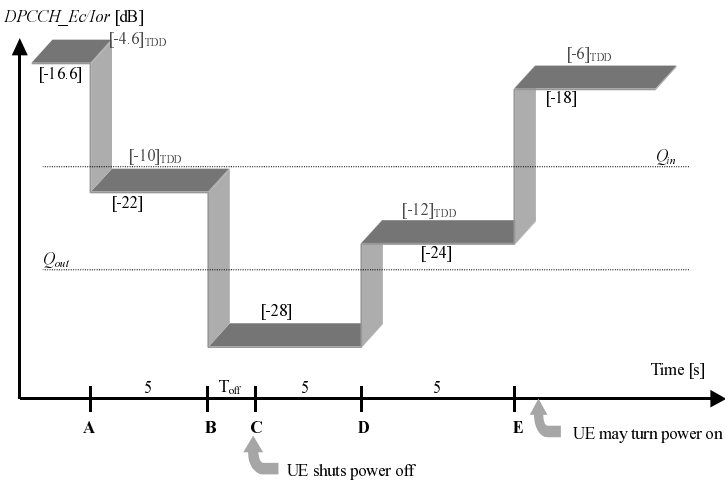


Figure 5.2 UE out-of-synch handling. Q_{out} and Q_{in} thresholds are for reference only [1].

Figure 5.2 illustrates the DPCH power level and the shutting off and on, where the requirements for the UE from Refs. [1,2] are that:

- The UE shall not shut its transmitter off before point B.
- The UE shall shut its transmitter off before point C, which is $T_{\text{off}} = [200]$ ms after point B.
- The UE shall not turn its transmitter on between points C and E.

The UE may turn its transmitter on after point E.

5.3.5 Transmit ON/OFF Power

Transmit OFF power state occurs when the UE does not transmit, except during UL DTX mode (see Figure 5.3). We define this parameter as the maximum output transmit power within the channel bandwidth when the transmitter is OFF. The requirement for transmit OFF power shall be better than -56 dBm for FDD and -65 dBm for TDD, defined as an averaged power within at least one time slot duration measured with a RRC filter response having a roll off factor $\alpha = 0.22$ and a bandwidth equal to the chip rate.

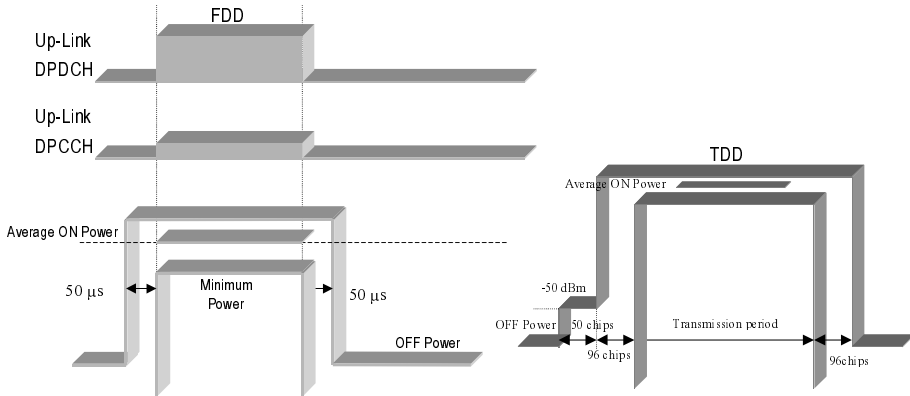


Figure 5.3 Transmit ON/OFF template.

The time mask for transmit ON/OFF defines the UE ramping time allowed between transmit OFF power and transmit ON power. This scenario may include the RACH, CPCH or UL slotted mode. We define ON power as one of the following cases [1]:

- first preamble of RACH: open loop accuracy;
- during preamble ramping of the RACH and compressed mode: accuracy depending on size of the power step;
- power step to maximum power: maximum power accuracy.

Specifications in Ref. [1] describes power control events in Transport Format Combination (TFC) and compressed modes.

5.3.5.1 BS Transmit OFF Power (TDD)

When the BS does not transmit, it remains in *transmit off power* state, which we defined as the maximum output transmit power within the channel bandwidth when the transmitter states OFF. Its required level shall be better than -79 dBm measured with a RRC filter response having a roll off $\alpha = 0.22$ and a bandwidth equal to the chip rate.

The time mask transmit ON/OFF defines the ramping time allowed for the BS between transmit OFF power and transmit ON power. The transmit power level vs. time meets the mask illustrated in Figure 5.4.

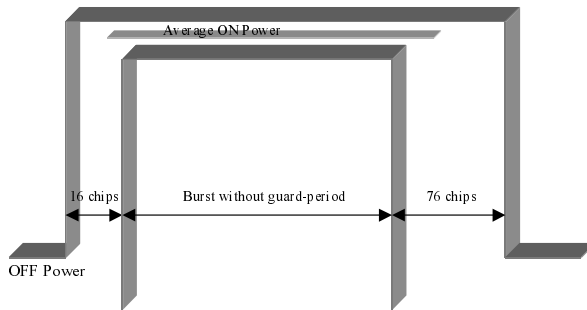


Figure 5.4 BS Transmit ON/OFF template (TDD).

5.3.6 Output RF Spectrum Emissions

5.3.6.1 Occupied Bandwidth and Out of Band Emission

Occupied bandwidth implies a measure of the bandwidth containing 99% of the total integrated power of the transmitted spectrum, centred on the assigned channel frequency. In the TDD as well as FDD, the occupied channel bandwidth shall be less than 5 MHz based on a chip rate of 3.84 Mcps.

Out of band emissions are unwanted emissions immediately outside the nominal channel originating from the imperfect modulation process and non-linearity in the transmitter but excluding spurious emissions. A Spectrum emission mask and adjacent channel leakage power ratio specify out of band emission limits.

5.3.6.2 Spectrum Emission Mask

The UE spectrum emission mask applies to frequencies that are between 2.5 MHz and 12.5 MHz away from the UE carrier frequency centre. The out of channel emission is specified relative to the UE output power measured in a 3.84 MHz bandwidth. Table 5.12 illustrates UE power emission values, which shall not exceed specified levels.

Table 5.12 Spectrum Emission Mask Requirement

Frequency offset from carrier Δf (MHz)	Minimum requirement (dBc)	Measurement bandwidth (MHz)
2.5–3.5	$-35-15 (\Delta f - 2.5)$	30 kHz
3.5–7.5	$-35-1 (\Delta f - 3.5)$	1
7.5–8.5	$-39-10 (\Delta f - 7.5)$	1
8.5–12.5	-49	1

The first and last measurement position with a 30 kHz filter is 2.515 MHz and 3.485 MHz.
The first and last measurement position with a 1 MHz filter is 4 MHz and 12 MHz.
The lower limit shall be -50 dBm/3.84 MHz or which ever is higher.

The BS spectrum emission mask illustrated in Figure 5.5 and outlined in Table 5.13 may be mandatory in some regions and may not apply to others. Where it applies, BS transmitting on a single RF carrier and configured according to the manufacturer’s specification shall meet specified requirements. The mask basically applies to the FDD and TDD.

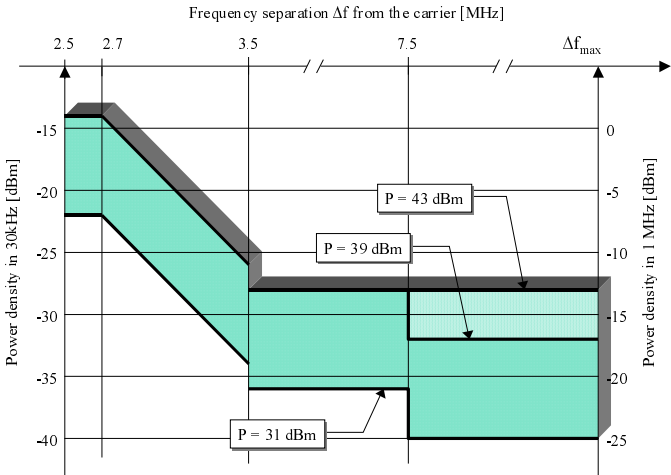


Figure 5.5 BS spectrum emission mask [3].

For example, emissions for the appropriate BS maximum output power, in the frequency range from $\Delta f = 2.5$ MHz to $f_{\text{offset\_max}}$ from the carrier frequency, shall not exceed the maximum level specified in Table 5.13 [3–4], where:

- Δf = separation between the carrier frequency and the nominal -3 dB point of the measuring filter closest to the carrier frequency.
- F_{offset} = separation between the carrier frequency and the centre of the measuring filter.
- $f_{\text{offset\_max}}$ = 12.5 MHz or is the offset to the UMTS Tx band edge, whichever is the greater.

Table 5.13 BS Spectrum Emission Mask Values

Δf of measurement filter –3 dB point (MHz)	Δf of filter measurement at centre frequency (MHz)	Maximum level (dBm)	Measurement bandwidth
<i>BS maximum output power $P \geq 43$ dBm</i>			
$2.5 \leq \Delta f < 2.7$	$2.515 \leq \Delta f < 2.715$	–14	30 kHz
$2.7 \leq \Delta f < 3.5$	$2.715 \leq \Delta f < 3.515$	$-14 - 15 \cdot (\Delta f - 2.715)$	30 kHz
	$3.515 \leq \Delta f < 4.0$	–26	30 kHz
$3.5 \leq \Delta f$	$4.0 \leq \Delta f < \Delta f_{\max}$	–13	1 MHz
<i>BS maximum output power $39 \leq P < 43$ dBm</i>			
$2.5 \leq \Delta f < 2.7$	$2.515 \leq \Delta f < 2.715$	–14	30 kHz
$2.7 \leq \Delta f < 3.5$	$2.715 \leq \Delta f < 3.515$	$-14 - 15 \cdot (\Delta f - 2.715)$	30 kHz
*	$3.515 \leq \Delta f < 4.0$	–26	30 kHz
$3.5 \leq \Delta f < 7.5$	$4.0 \leq \Delta f < 7.5$	–13	1 MHz
$7.5 \leq \Delta f$	$7.5 \leq \Delta f < \Delta f_{\max}$	$P - 56$	1 MHz
<i>BS maximum output power $31 \leq P < 39$ dBm</i>			
$2.5 \leq \Delta f < 2.7$	$2.515 \leq \Delta f < 2.715$	$P - 53$	30 kHz
$2.7 \leq \Delta f < 3.5$	$2.715 \leq \Delta f < 3.515$	$P - 53 - 15 \cdot (\Delta f - 2.715)$	30 kHz
*	$3.515 \leq \Delta f < 4.0$	–26	30 kHz
$3.5 \leq \Delta f < 7.5$	$4.0 \leq \Delta f < 7.5$	$P - 52$	1 MHz
$7.5 \leq \Delta f$	$7.5 \leq \Delta f < \Delta f_{\max}$	$P - 56$	1 MHz
<i>BS maximum output power $P < 31$ dBm</i>			
$2.5 \leq \Delta f < 2.7$	$2.515 \leq \Delta f < 2.715$	–22	30 kHz
$2.7 \leq \Delta f < 3.5$	$2.715 \leq \Delta f < 3.515$	$-22 - 15 \cdot (\Delta f - 2.715)$	30 kHz
*	$3.515 \leq \Delta f < 4.0$	–26	30 kHz
$3.5 \leq \Delta f < 7.5$	$4.0 \leq \Delta f < 7.5$	–21	1 MHz
$7.5 \leq \Delta f$	$7.5 \leq \Delta f < \Delta f_{\max}$	–25	1 MHz

*This frequency range ensures that the range of values of Δf is continuous.

5.3.6.3 Adjacent Channel Leakage Power Ratio (ACLR)

The ratio of the transmitted power to the power measured in an adjacent channel corresponds to the Adjacent Channel Leakage Power Ratio (ACLR). Both the transmitted and the adjacent channel power measurements use a RRC filter response with roll-off $\alpha = 0.22$ and a bandwidth equal to the chip rate. If the adjacent channel power greater than –50 dBm then the ACLR shall be higher than the value specified in Table 5.14 [1].

Table 5.14 UE ACLR

Power class	Adjacent channel relative to UE channel (MHz)	ACLR limit (dB)
3	± 5	33
3	± 10	43
4	± 5	33
4	± 10	43

5.3.6.4 Spurious Emissions

Spurious emissions or unwanted transmitter effects result from harmonics emission, parasitic emission, inter-modulation products and frequency conversion products, but not from band emissions. The frequency boundary and the detailed transitions of the limits between the requirement for out band emissions and spectrum emissions are based on ITU-R Recommendations SM.329. These requirements illustrated in Table 5.15 apply only to frequencies which are greater than 12.5 MHz away from the UE carrier frequency centre [1].

Table 5.15 General spurious emissions requirements

Frequency bandwidth	Resolution bandwidth (kHz)	Minimum requirement (dBm)
$9 \text{ kHz} \leq f < 150 \text{ kHz}$	1	-36
$150 \text{ kHz} \leq f < 30 \text{ MHz}$	10	-36
$30 \text{ MHz} \leq f < 1000 \text{ MHz}$	100	-36
$1 \text{ GHz} \leq f < 12.75 \text{ GHz}$	1 MHz	-30

Measurements integer multiples of 200 kHz.

5.3.6.5 Transmit Modulation and Inter-modulation

The transmit modulation pulse has a RRC shaping filter with roll-off $\alpha = 0.22$ in the frequency domain. The impulse response of the chip impulse filter $RC_0(t)$ is:

$$RC_0(t) = \frac{\sin\left(\pi \frac{t}{T_c}(1-\alpha)\right) + 4\alpha \frac{t}{T_c} \cos\left(\pi \frac{t}{T_c}(1+\alpha)\right)}{\pi \frac{t}{T_c} \left(1 - \left(4\alpha \frac{t}{T_c}\right)^2\right)}, \quad (5.1)$$

where the roll-off factor $\alpha = 0.22$ and the chip duration is $T = 1/\text{chip rate} \approx 0.26042\mu$.

5.3.6.5.1 Vector Magnitude and Peak Code Domain Error

The Error Vector Magnitude (EVM) indicates a measure of the difference between the measured waveform and the theoretical modulated waveform (the error vector). A square root of the mean error vector power to the mean reference signal power ratio expressed as a % defines the EVM. One time slot corresponds to the measurement interval of one power control group. The EVM is less or equal to 17.5% for the UE output power parameter (≥ -20 dBm) operating at normal conditions in steps of 1 dB.

The *code domain error* results from projecting the *error vector power* onto the code domain at the maximum spreading factor. We define the *error vector* for each power code as the ratio to the mean power of the reference waveform expressed in dB, and the *peak code domain error* as the maximum value for the *code domain error*. The measurement interval is one power control group (time slot). The requirement for the peak code domain error applies only to multi-code transmission, and it shall not exceed

-15 dB at a spreading factor of 4 for the UE output power parameter having a value (≥ -20 dBm) and operating at normal conditions [1].

5.3.6.5.2 *Inter-modulation*

By transmit Inter-modulation (IM) performance we meant the measure of transmitter capability to inhibit signal generation in its non-linear elements in the presence of wanted signal and an interfering signal arriving to the transmitter via the antenna. For example, user equipment(s) transmitting in close vicinity of each other can produce inter-modulation products, which can fall into the UE, or BS receive band as an unwanted interfering signal.

We define UE inter-modulation attenuation as the output power ratio of wanted signal to the output power of inter-modulation product when an interfering CW signal adds itself at a level below a wanted signal. Both the wanted signal power and the IM product power measurements use a RRC filter response with roll-off $\alpha = 0.22$ and a bandwidth equal to the chip rate. Table 5.16 illustrates IM requirement when transmitting with 5 MHz carrier spacing.

Table 5.16 Transmit Inter-modulation

Interference signal frequency offset (MHz)	5	10
Interference CW signal level (dBc)	-40	
Inter-modulation product (dBc)	-31	-41

5.4 RECEIVER CHARACTERISTICS

We specify receiver characteristics at the UE antenna connector, and for UE(s) with an integral antenna only, we assume a reference antenna with a gain of 0 dBi. Receiver characteristics for UE(s) with multiple antennas/antenna connectors are FFS.

5.4.1 Diversity

We assume appropriate receiver structure using coherent reception in both channel impulse response estimation and code tracking procedures. The UTRA/FDD includes three types of diversity:

- time diversity → channel coding and interleaving in both up- and downlink;
- multi-path diversity → rake receiver or other appropriate receiver structure with maximum combining;
- antenna diversity → occurs with maximum ratio combining in the BS and optionally in the MS.

5.4.2 Reference and Maximum Sensitivity Levels

Reference sensitivity implies the minimum receiver input power measured at the antenna port at which the Bit Error Ratio (BER) does not exceed a specific value, e.g.

BER = 0.001, the DPCH_Ec has a level of -117 dBm/3.48 MHz, and the $\hat{I}_{\text{or}}$ a level of -106.7 dBm/3.84 MHz.

For the maximum input level, also with BER not exceeding 0.001, $\hat{I}_{\text{or}} = -25$ dBm/3.84 MHz, and $\text{DPCH\_Ec}/\hat{I}_{\text{or}} = -19$ dB.

In the TDD mode reference sensitivity levels for $\Sigma\text{DPCH\_Ec}/\hat{I}_{\text{or}}$ and $\hat{I}_{\text{or}}$ are 0 dB and -105 dBm/3.84 MHz, respectively, while the maximum sensitive level requirements are -7 dB and -25 dBm/3.84 MHz.

5.4.3 Adjacent Channel Selectivity (ACS)

Adjacent Channel Selectivity (ACS) refers to the measure of a receiver's ability to receive a W-CDMA signal at its assigned channel frequency in the presence of an adjacent channel signal at a given frequency offset from the centre frequency of the assigned channel. We define the ACS as the ratio of receive filter attenuation on the assigned channel frequency to the receive filter attenuation on the adjacent channel(s) [1].

The ACS shall be better than 33 dB in Power Class 2(TDD), 3 and 4 for the test parameters specified in Table 5.17, where the BER shall not exceed 0.001.

Table 5.17 Test parameters for Adjacent Channel Selectivity

Parameter	Unit	Level
DPCH_Ec	dBm/3.84 MHz	-103
$\hat{I}_{\text{or}}$	dBm/3.84 MHz	-92.7
I_{oac} (modulated)	dBm/3.84 MHz	-52
F_{uw} (offset)	MHz	± 5

The $(\Sigma\text{DPCH\_Ec}/\hat{I}_{\text{or}})_{\text{TDD}}$ has 0 dB as test parameter for the adjacent channel selectivity.

5.4.4 Blocking

The blocking characteristic indicates the measure of the receiver's ability to receive a wanted signal at its assigned channel frequency in the presence of an unwanted interference on frequencies other than those of the spurious response or the adjacent channels. The unwanted input signal shall not cause a degradation of the performance of the receiver beyond a specified limit, and the blocking performance shall apply at all frequencies except those at which a spurious response occur.

The BER shall not exceed 0.001 for the parameters specified in Tables 7.6 and 7.7. For Table 7.7 up to (24) exceptions are allowed for spurious response frequencies in each assigned frequency channel when measured using a 1 MHz step size.

Table 5.18 In-band Blocking FDD and TDD

Parameter	Unit	Offset	Offset
Wanted signal _{TDD}	dBm/3.84 MHz	<RefSens> + 3 dB	<RefSens> + 3 dB
DPCH_Ec	dBm/3.84 MHz	-114	-114
$\hat{I}_{or}$	dBm/3.84 MHz	-103.7	-103.7
$I_{blocking}$ (modulated) applies to FDD and TDD	dBm/3.84 MHz	-56	-44
F_{uw} (offset) FDD and TDD	MHz	±10	±15

Table 5.19 Out of Band Blocking FDD

Parameter	Unit	Band 1	Band 2	Band 3
DPCH_Ec	dBm/3.84 MHz	-114	-114	-114
$\hat{I}_{or}$	dBm/3.84 MHz	-103.7	-103.7	-103.7
$I_{blocking}$ (CW)	dBm	-44	-30	-15
F_{uw}	MHz	2050 < f < 2095 2185 < f < 2230	2025 < f < 2050 2230 < f < 2255	1 < f < 2025 2255 < f < 12750
F_{uw}	MHz	1870 < f < 1915 2005 < f < 2050	1845 < f < 1870 2050 < f < 2075	1 < f < 1845 2075 < f < 12750

Table 5.20 Out of Band Blocking TDD

Parameter	Unit	Band 1	Band 2	Band 3
Wanted signal	dBm/ 3.84 MHz	<RefSen> + 3 dB	<RefSen> + 3 dB	<RefSen> + 3 dB
Unwanted signal level (CW)	dBm	-44	-30	-15
F_{uw}	MHz	1840 < f < 1885 1935 < f < 1995 2040 < f < 2085	1815 < f < 1840 2085 < f < 2110	1 < f < 1815 2110 < f < 12750
F_{uw}	MHz	1790 < f < 1835 2005 < f < 2050	1765 < f < 1790 2050 < f < 2075	1 < f < 1765 2075 < f < 12750
F_{uw}	MHz	1850 < f < 1895 1945 < f < 1990	1825 < f < 1850 1990 < f < 2015	1 < f < 1825 2015 < f < 12750

The TDD out of band blocking differs from the FDD because they do not have the same frequency range allocation.

5.4.5 Spurious Response

Through the spurious response, a receiver has the ability to receive a desired signal on its assigned channel frequency, without exceeding a given degradation originating from an undesired CW interfering signal. The latter occurs at any other frequency at which the blocking limit is not met. Table 5.21 illustrates the spurious responses, where the BER does not exceed 0.001.

Table 5.21 Spurious Response FDD and TDD

Parameter	Unit	Level
Wanted signal _{TDD}	dBm/3.84 MHz	<RefSens> + 3 dB
DPCH_Ec _{FDD}	dBm/3.84 MHz	−114
$\hat{I}_{or}$ (FDD)	dBm/3.84 MHz	−103.7
$I_{blocking}$ (CW) (FDD and TDD)	dBm	−44
F_{uw} (FDD and TDD)	MHz	Spurious response frequencies

5.4.6 Inter-Modulation

Inter-modulation response rejection enables the receiver to receive a wanted signal on its assigned channel frequency in the presence of two or more interfering² signals, which have a specific frequency relationship to the wanted signal. Table 5.22 illustrates the inter-modulation characteristics, where BER does not exceed 0.001.

In the notation of tables, the TDD subscript implies that it applies to the TDD mode. If there is not a TDD subscript or a FDD subscript exist it applies to the FDD mode.

Table 5.22 Receive Inter-Modulation Characteristics FDD and TDD

Parameter	Unit	Level
DPCH_Ec	dBm/3.84 MHz	−114
$\hat{I}_{or}$	dBm/3.84 MHz	−103.7
$\hat{I}_{or}$ (TDD)	dBm/3.84 MHz	<RefSens> + 3 dB
$(\sum DPCH\_Ec/\hat{I}_{or})$ (TDD)	DB	0
I_{ouw1} (CW)	dBm	−46
I_{ouw2} (modulated)	dBm/3.84 MHz	−46
F_{uw1} (offset)	MHz	10
F_{uw2} (offset)	MHz	20

5.4.7 Spurious Emissions Power

We refer to the power of emissions generated or amplified in a receiver and appearing at the UE antenna connector as *spurious emissions power*. The spurious emission shall be [1]:

- Less than −60 dBm/3.84 MHz at the UE antenna connector, for frequencies within the UE receive band. In URA_PCH-, Cell_PCH- and IDLE- stage the requirement applies also for the UE transmit band.
- Less than −57 dBm/100 kHz at the UE antenna connector, for the frequency band from 9 kHz to 1 GHz.

² Two interfering RF signals of 3rd and higher order mixing can produce interfering signal in the desired channel band.

- Less than -47 dBm/100 kHz at the UE antenna connector, for the frequency band from 1 GHz to 12.75 GHz.

Table 5.23 TDD Receiver Spurious Emission Requirements [2]

Band	Maximum level (dBm)	Measurement Bandwidth
9 kHz–1 GHz	-57	100 kHz
1 GHz–1.9 GHz and 1.92 GHz–2.01 GHz and 2.025 GHz–2.11 GHz	-47	1 MHz
1.9 GHz–1.92 GHz and 2.01 GHz–2.025 GHz and 2.11 GHz–2.170 GHz	-60	3.84 MHz
2.170 GHz–12.75 GHz	-47	1 MHz

The UE uses the last carrier frequency, except for frequencies between 12.5 MHz below the first carrier frequency and 12.5 MHz above the last carrier frequency.

Specifications in [1,2] describe the performance for the transmitter and receiver characteristics.

5.5 UTRA RF PERFORMANCE EXAMPLES

In the sequel we provide RF system scenarios based on the studies reported in [5]. Here we aim primarily to illustrate the principles outlined in the preceding sections in order to present practical applications of the recommended parameters. The examples may not strictly apply to actual designs; however, they could serve as reference for initial analysis.

5.5.1 Coexistence FDD/FDD: ACIR

Before we describe a methodology, we first define some of the essential terminology as in [5] for the context of the examples to follow:

Outage – in this context an outage occurs when, due to a limitation on the maximum TX power, the measured Eb/No of a connection is lower than the Eb/No target.

Satisfied user - a user is satisfied when the measured Eb/No of a connection at the end of a snapshot, is higher than a value equal to Eb/No target -0.5 dB.

ACIR - the Adjacent Channel Interference Power Ratio (ACIR) is defined as the ratio of the total power transmitted from a source (base station or UE) to the total interference power affecting a victim receiver, resulting from both transmitter and receiver imperfections.

5.5.1.1 Overview of Simulation Assumptions

Simulations use snapshots where we place subscribers randomly in a predefined deployment scenario; each snapshot simulates a power control loop until it reaches a tar-

get E_b/N_0 ; a simulation is made of several snapshots. We obtain the measured E_b/N_0 by the measured C/I multiplied by the processing gain.

UEs do not reach the target E_b/N_0 *at the end* of a PC loop in the outage state. We consider satisfied users those able to reach at least ($E_b/N_0 - 0.5$ dB) at the end of a Power Control (PC) loop. Statistical data related to outage (satisfied users) are collected at the end of each snapshot.

We model soft handover allowing a maximum of 2 BTS in the active set, where we set the window size of the candidate to 3 dB, and the cells in the active set are chosen randomly from the candidate set. We use selection combining in the uplink and maximum ratio combining in DL, and simulate uplink and downlink independently.

5.5.1.2 Simulated Scenarios

We have already outlined the background of the simulated scenarios in Chapter 2. Nonetheless, here we briefly describe them again to introduce the proper context of the different environments considered, e.g. macro-cellular and micro-cellular environments with their respective cases, i.e. macro to macro multi-operator case and macro to micro case.

5.5.1.3 Macro to Macro Multi-Operator Case

In a *single operator* layout we place BS on a hexagonal grid with distance of 1000 m; the cell radius is then equal to 577 m (see e.g. Figure 5.6). We assume BSs with omnidirectional antennas in the middle of the cell. In practice we use either 3 or 6 sector antennas. We also assume 19 cells (or higher) for each operator in the macro-cellular environment. This number appears suitable when using the wrap around technique.

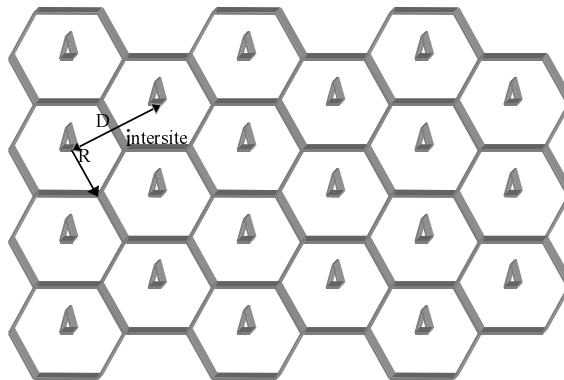


Figure 5.6 Macro-cellular deployment.

In the *multi-operator* case, we consider two shifting BSs shifting two operators, e.g. (worst case scenario) 577 m BS shift, and (intermediate case) 577/2 m BS shift. We do not consider the best case scenario (i.e. 0 m shifting = co-located sites).

5.5.1.4 Macro to Micro Multi-Operator Case

5.5.1.4.1 Single Operator Layout, Micro Cell Layer

For the micro-cell deployment in a Manhattan deployment scenario we place the BSs so that they stand at street crossings in every second junction as illustrated in Figure 5.7 [6]. Although the model does not reflect efficient planning, it does provide sufficient amount of inter cell interference generation with reasonably low number of micro cell BSs. The parameters of the micro cells are thus: block size = 75 m, road width = 15 m, inter-site distance between line of sight = 180 m, and the number of micro cells in the micro-cellular scenario is 72.

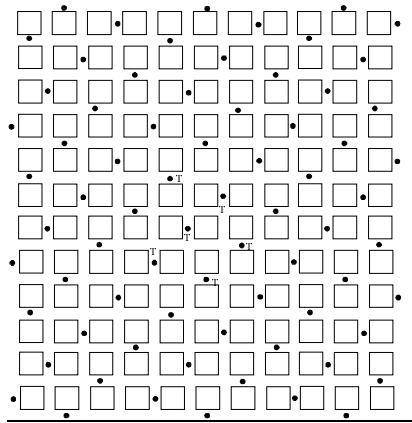


Figure 5.7 Micro-cell deployment.

5.5.1.4.2 Multi-Operator Layout

In this micro cell layout we use the parameters proposed earlier, i.e. (72 BSs in every second street junction, block size 75 m, road width 15 m). We also apply a macro cell radius of 577 m with a distance of 1000 between BSs.

Figure 5.8 illustrates the cellular layout to simulate Hierarchical Cell Structures (HCS). This layout allows large enough macro cells and a low number of micro cells so that computing simulation times remain reasonable. Furthermore, we select macro cell BS positions to observe handovers and many other conditions (e.g. border conditions, etc.).

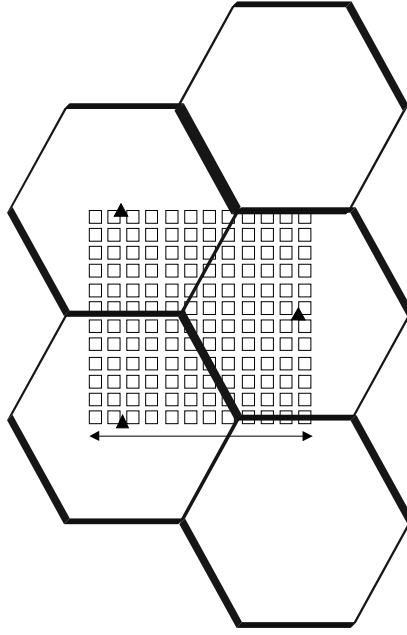


Figure 5.8 Macro-to micro deployment.

When measuring interference at macro cell BSs in uplink, we measure same channel interference *only* from those users connected to the observed BS. Then we multiply the measured same channel interference by $1/F$, where F is the ratio of intra-cell interference to total interference, i.e.

$$F = I_{\text{intra}}(i) / (I_{\text{intra}}(i) + I_{\text{inter}}(i)). \quad (5.2)$$

F depends on the assumed propagation model; earlier studies suggest a typical value of around 0.6. However, an appropriate value for F can also be derived from specific macro cell only simulations. We measure interference from micro cells to macro cell by using the wraparound technique. We can then define the interference that a macro cell BS receives as

$$I = \text{ACIR} \times I_{\text{micro}} + (1/F) \times I_{\text{macro}}, \quad (5.3)$$

where ACIR is the adjacent channel interference rejection ratio, and I_{macro} is same channel interference measured from users connected to the base station.

When we measure DL interference, same channel and adjacent channel interference gets measured from all base stations. To measure interference from micro cells we use the wraparound technique. When measuring interference at micro cells in UL and DL, same channel and adjacent channel interference gets measured from all BS, and when measuring same channel interference we apply the wraparound technique.

When measuring simulation results we consider all micro cell users and those macro cell users that are in the area covered by micro cells. We also need to plot figures depicting the position of bad quality calls, in order to see how they are distributed in the network. In addition, noise rise should be measured at every BS and from that data, a probability density function should be generated [5].

5.5.1.5 *Simulated Services*

The following services were considered:

- speech 8 kbps
- data 144 kbps

Speech and data services were simulated in separate simulations, i.e. no traffic mix was simulated.

5.5.2 Description of the propagation models

Two propagation environments were considered in the ACIR analysis, i.e. macro-cellular and micro-cellular. For each environment a propagation model was used to evaluate the propagation path loss due to the distance. As noted earlier, these propagation models are described in Chapter 2 and are also briefly presented in the forthcoming sections.

5.5.2.1 *Received Signal*

Before describing the propagation environments, a key parameter to be defined is the Minimum Coupling Loss (MCL), i.e. the minimum distance loss including antenna gain measured between antenna connectors. This represents the minimum loss in signal due to fact that the BSs are always placed much higher than the UE(s). The following values are assumed in our example for MCL:

- 70 dB for the macro-cellular environment
- 53 dB for the micro-cellular environment

With the above definition, the received power in DL or UL can be expressed for the macro environment as:

$$RX\_PWR = TX\_PWR - \text{Max}(\text{pathloss\_macro} - G\_Tx - G\_RX, \text{MCL}) \quad (5.4)$$

and for the micro environment as:

$$RX\_PWR = TX\_PWR - \text{Max}(\text{pathloss\_micro} - G\_Tx - G\_RX, \text{MCL}) \quad (5.5)$$

where

- RX_PWR is the received signal power
- TX_PWR is the transmitted signal power
- G_Tx is the Tx antenna gain

- G_{RX} is the Rx antenna gain

For the set of simulations in this section we have assumed 11 dB antenna gain (including cable losses) in base station and 0 dB in UE [5]. In Chapter 7 we use other assumptions.

5.5.2.2 Macro Cell Propagation Model

The macro cell propagation model serves here to test scenarios in urban and suburban areas outside the high rise core where the buildings are of nearly uniform height as discussed in Chapter 2 and [7].

$$L = 40(1 - 4 \times 10 - 3D_{hb}) \log_{10}(R) - 18 \log_{10}(D_{hb}) + 21 \log_{10}(f) + 80 \text{ dB}, \quad (5.6)$$

where R is the base station – UE separation in kilometres, f is the carrier frequency of 2000 MHz and D_{hb} is the base station antenna height, in metres, measured from the average rooftop level.

When the BS antenna height gets fixed at 15 m above the average rooftop (i.e. $D_{hb} = 15$ m) and when considering a carrier frequency of 2000 MHz, the macro cell propagation model formula becomes:

$$L = 128.1 + 37.6 \log_{10}(R). \quad (5.7)$$

Once we calculate L we add a log-normally distributed shadowing ($\log F$) with standard deviation of 10 dB to obtain the path loss as follows:

$$\text{Pathloss}_{\text{macro}} = L + \log F. \quad (5.8)$$

To complete the definition of the path loss for our analysis we should note from [5] that:

- L shall in no circumstances be less than free space loss, and this model applies only to the Non-Line-of-Sight (NLOS) case and describes worse case propagation for the examples in this section.
- The path loss model is valid for a range of D_{hb} from 0 to 50 m.
- This model concerns designs mainly for distance from a few hundred metres to kilometres. Thus, it may not accurately apply to short distances.

5.5.2.3 Micro Cell Propagation Model

We use the micro cell propagation model (also covered in Chapter 2) for spectrum efficiency evaluations in urban environments modelled through a Manhattan-like structure. It allows us to appropriately evaluate the performance in micro cell situations that will be typical for example in European like cities at the time of UMTS deployment.

In this case, this recursive model calculates the path loss as a sum of LOS and NLOS segments. We find the shortest path along streets between the BS and the UE within the Manhattan environment. The path loss in dB is thus given by the well-known formula

$$L = 20 \log_{10} \frac{4\pi d_n}{\lambda}, \quad (5.9)$$

where d_n is the 'illusory' distance, λ is the wavelength, and n is the number of straight street segments between BS and UE (along the shortest path).

The illusory distance is the sum of these street segments and can be obtained by recursively using the expressions $k_n = k_{n-1} + d_{n-1}c$ and $d_n = k_n s_{n-1} + d_{n-1}$ where c is a function of the angle of the street crossing. For a 90° street crossing the value c should be set to 0.5. Further, s_{n-1} is the length in metres of the last segment. A segment is a straight path. The initial values are set according to: k_0 is set to 1 and d_0 is set to 0. The illusory distance is obtained as the final d_n when the last segment has been added.

The model is extended to cover the micro cell dual slope behavior, by modifying the expression to:

$$L = 20 \log_{10} \left(\frac{4\pi d_n}{\lambda} D \left(\sum_{j=1}^n s_{j-1} \right) \right),$$

where

$$D(x) = \begin{cases} x/x_{br}, & x > x_{br}, \\ 1, & x \leq x_{br}. \end{cases} \quad (5.10)$$

Before the break point x_{br} the slope is 2, after the break point it increases to 4. The break point x_{br} is set to 300 m. x is the distance from the transmitter to the receiver.

To take into account effects of propagation going above rooftops the path loss according to the shortest geographical distance must also be calculated. This is done by using the commonly known COST Walfish-Ikegami Model and with antennas below rooftops:

$$L = 24 + 45 \log(d + 20), \quad (5.11)$$

where d is the shortest physical geographical distance from the transmitter to the receiver (m).

The final path loss corresponds to the minimum value between the path loss value from the propagation through the streets and the path loss based on the shortest geographical distance, plus the log-normally distributed shadowing ($\log F$) with standard deviation of 10 dB.

$$\text{Pathloss\_micro} = \min(\text{Manhattan path loss, macro path loss}) + \log F \quad (5.12)$$

The above path loss model applies only to the micro cell coverage with antenna located below rooftop. For the urban structure covered by the macro cells, the path loss defined in the preceding section applies.

5.5.3 The Simulation Process

Only one link gets considered in a single simulation, i.e. we simulate UL and DL independently. A simulation, aiming to cover most of the possible UEs of single placement in the network, consists of several simulation steps (snapshot).

5.5.3.1 Single Step (Snapshot) Simulation

In general a simulation step (snapshot) constitutes placement of MSs, path loss calculations, handover, power control and collection of statistics.

In particular:

- Each simulation step begins with the uniform random distribution of UE(s) across the network.
- For each UE (e.g. in the case of macro to macro simulation) we randomly select an operator, so that the number of users per base station is the same for both operators.
- After the placement of UEs, we calculate the path loss between each UE and BS, then add the log-normal fading and stored to a so-called G-matrix (gain matrix).

We keep constant the distance attenuation and log-normal fading during the execution of a snapshot.

- Using the gain matrix and based on the HO algorithm we select the transmitting BSs for each UE.
- Then a power control loop³ stabilization period gets started. During this stabilization span we execute power control as long as the used powers reach the level required to meet the expected quality.
- The acceptable number of power control commands in each power control loop, can exceed 150.
- We collect statistical data at the end of a power control loop. UEs with quality below the target remain in outage state; UEs with quality higher than the target -0.5 dB are in satisfied state.

5.5.3.2 Multiple Steps (Snapshots) Execution

Multiple steps occur when a single step (snapshot) finishes, UE(s) are re-located to the system and the above processes get executed again. During a simulation, we execute as many simulation steps (snapshots) as required in order to achieve the ideal amount of local-mean-SIR values. The ideal number of snapshots for 8 kbps speech service amounts to 10 000 values or more. For data service we require a higher number of snapshot, e.g. 10 times the value used for 8 kbps speech.

³ During the power control loop, the gain matrix remains constant.

During one simulation step (i.e. snapshot) we obtain as many local-mean-SIR values as UE(s) in the simulation. The outputs from a simulation include SIR-distribution, outage probability, and capacity figures, etc.

5.5.4 Modeling of Handover and Power Control

5.5.4.1 Handover Modelling

Here we model non-ideal soft handover, where an active set for the UE gets selected from a pool of candidate BSs for handover. The candidate set consists of BSs whose path loss is within handover margin, i.e. BSs whose received pilot is stronger than the received pilot of the strongest BS subtracted by the handover margin. We select the active set of BS randomly from the candidate BSs, where a single UE may be connected to maximum of two base stations simultaneously. We assume 3 dB as a soft handover margin.

5.5.4.1.1 Uplink and Downlink Combining

In the uplink, selection combining among active BSs takes place to use the frame with the highest average SIR for statistics collecting purposes, while the other frames get discarded.

In the downlink, we model macro diversity to sum together the signal received from active BSs. Thus, we realize maximal ratio combining by summing measured SIR values i.e.:

$$\text{SIR} = \frac{C_1}{I_1 + N} + \frac{C_2}{I_2 + N}. \quad (5.13)$$

5.5.4.2 Power Control Modelling of Uplink Traffic Channels

In these simulations Power Control (PC) corresponds to the SIR based fast inner loop power control. Here we assume perfect PC, i.e. during the power control loop, each UE achieves perfectly the E_b/N_0 target, assuming that the maximum TX power is not exceeded. Assuming perfect power control, we imply that PC error equals 0%, and PC delay equals 0 s.

As noted earlier, UEs, which cannot achieve the target E_b/N_0 at the end of a power control loop, are in outage. We base the initial TX power for the PC loop of the UL traffic channel on path loss, thermal noise and 6 dB noise rise. However, the initial TX power should not affect the convergence process (PC loop) to the target E_b/N_0 [5].

5.5.4.2.1 Simulation Parameters

UE Max TX power. The maximum UE TX power is 21 dBm (both for speech and data), and the UE power control range is 65 dBm; the minimum TX power is thus -44 dBm.

For Uplink E_b/N_0 target, based on [7] we assume

- macro-cellular environment: speech 6.1 dB, data 3.1 dB;
- micro-cellular environment: speech 3.3 dB, data 2.4 dB.

5.5.4.2.2 SIR Calculation in Uplink

We calculate the local mean SIR by dividing the received signal by the interference, and multiplying by the processing gain. Signals from the other users are summed together and seen as interference. Signal-to-interference ratio for our analysis is thus:

$$\text{SIR}_{\text{UL}} = \frac{G_p S}{(1 - \beta) I_{\text{own}} + I_{\text{other}} + N_o}, \quad (5.14)$$

where S is the received signal, G_p is processing gain, I_{own} is interference generated by those users that are connected to the same base station as the observed user, I_{other} is interference from other cells, N_o is thermal noise and β is an interference reduction factor due to the use of, e.g. Multi User Detection (MUD) in UL. However, MUD is NOT included in these simulations, therefore $\beta = 0$.

We calculate thermal noise for the 4.096 MHz band by assuming 5-dB system noise. Thermal noise power is thus equal to -103 dBm. In the multi-operator case, I_{other} also includes the interference coming from the adjacent operator; which is decreased by ACIR dB.

5.5.4.3 Power Control Modelling of Traffic Channels in Downlink

As in the UL case, downlink power control corresponds to the SIR based fast inner loop power control. Here too we assume perfect PC, i.e. during the power control loop, each DL traffic channel achieves perfectly the Eb/No target, assuming that the maximum TX power is not exceeded. Assuming perfect power control, we imply that PC error equals 0%, and PC delay equals 0 s. The UEs whose DL traffic channel is not able to achieve the Eb/No target at the end of a power control loop are considered also in outage as in the UL.

We choose randomly the initial TX power for the PC loop of the DL traffic channel in the TX power range; however, the initial TX power should not affect the convergence process (PC loop) to the target Eb/No.

5.5.4.3.1 Reference Simulation Parameters

Traffic channel TX power. We assume 25 dBm for DL traffic channel power control-range, and for the maximum power for each DL traffic channel (i.e. both speech and data) we assume 30 and 20 dBm in the macro-cellular and micro-cellular environment, respectively.

The downlink Eb/No target following directives in [7] assumes 7.9 and 2.5 dB for speech and data, respectively, with DL TX or RX diversity⁴ in the macro-cellular environment. The micro-cellular environment the DL Eb/No target assumes 6.1 dB and 1.9 dB for speech and data, respectively, with DL TX or RX diversity.

⁴ 4.5 dB without diversity.

5.5.4.3.2 The Downlink SIR Calculation

The DL signal-to-interference-ratio (SIR) can be expressed as:

$$\text{SIR}_{\text{DL}} = \frac{G_p S}{\alpha I_{\text{own}} + I_{\text{other}} + N_o}, \quad (5.15)$$

where S is the received signal, G_p is processing gain, I_{own} is the interference generated by those users linked to the same BS that the observed user (it includes also interference caused by perch channel and common channels), I_{other} ⁵ is interference from other cells, α is the orthogonality factor and N_o is thermal noise, which is calculated for the 4.096 MHz band by assuming 9 dB system noise figure. Thermal noise power is then equal to -99 dBm. Transmission powers for them are in total 30 and 20 dBm for macro cells and micro cells, respectively.

As noted in earlier chapters, the orthogonality factor takes into account the fact that the DL is not perfectly orthogonal due to multi-path propagation. An orthogonality factor of 0 implies perfectly orthogonal intra-cell users, while the value of 1 implies that the intra-cell interference has the same effect as inter-cell interference. Here the orthogonality factor α assumes 0.4 and 0.06 for macro cells and micro cells, respectively.

5.5.4.3.3 The Downlink Maximum TX Power

For the maximum BS TX power, i.e. when the sum of all DL traffic channels in a cell exceeds the maximum base station TX power, here we assume 43 and 33 dBm in the macro cell and micro cell environments, respectively. Thus, during simulations if in the PC loop of each snapshot the overall TX power of each BS gets higher than the maximum power allowed, we record the event and validate it to guide a future DL approach.

The scheme used to maintain the output level of the BS equal or below the maximum BS TX power, is similar to an analog mechanism to protect the power amplifier. At each iteration, the MSs request more or less power, depending on their C/I values. A given BS will be requested to transmit the common channels and the sum of the TCHs for all the MSs it is in communication with. If this total output power exceeds the maximum allowed for the PA, an attenuation gets applied in order to set the output power of the BS equal to its maximum level. As an RF variable attenuator would operate, this attenuation gets applied on the output signal with the exception of common channels, i.e. all the TCHs are reduced by this amount of attenuation. The power of the TCH for a given mobile will be [5]:

$$\text{TCH}(n+1) = \text{TCH}(n) \pm \text{Step} - \text{RF\_Attenuation}. \quad (5.16)$$

⁵ In the multi-operator case, I_{other} also includes adjacent operator interference, which is decreased by ACIR (dB).

5.5.5 System Loading

5.5.5.1 Uplink

The steps for *single operator loading* in these simulation examples can be outlined from [5] as follows: We define the number of users in the uplink of the single operator case as N_{UL_single} . The latter gets evaluated according to a 6 dB noise rise over the thermal noise in the UL (6 dB noise rise is equivalent to 75% of the pole capacity of a CDMA system).

We measure a simulation run with a predefined number of users at the end of the average noise rise (over the thermal noise). If lower than 6 dB, we increase the number of users until we reach the 6 dB noise. Thus, we define here the number of users corresponding to a 6 dB noise rise as N_{UL_single} .

5.5.5.1.1 Multi-Operator Scenario with Macro to Macro Cellular Environment

We define the number of users in the uplink of the multi-operator case as N_{UL_multi} . It gets evaluated, as in the single case, according to a 6 dB noise rise over the thermal noise in the UL. We run a simulation with a predefined number of users, and measure it at the end of the average noise rise (over the thermal noise). If lower than 6 dB, we increase the number of users until we reach the 6 dB noise rise. Thus, here we also define the number of users corresponding to a 6 dB noise rise as N_{UL_multi} .

Then, for a given value of ACIR, the obtained N_{UL_multi} gets compared to N_{UL_single} to evaluate the capacity loss due to the presence of a second operator [5].

5.5.5.1.2 Multi-Operator Scenario with Macro to Micro Cellular Environment

In general, the noise rise does not change by the same amount for micro and macro cell layers if the number of users do change in the system. Thus, [5] proposes that loading in this case takes the following steps with two different numbers of input of users included in the simulator, i.e. $N_{users\_UL\_macro}$ and $N_{users\_UL\_micro}$. Then the steps are:

0. selection of an ACIR value;
1. begin a simulation (made of several snapshots) with an arbitrary number of $N_{users\_UL\_micro}$ and $N_{users\_UL\_macro}$;
2. system loading measurement;
3. run a 2nd simulation (made of several snapshots) by increasing the number of users (i.e. $N_{users\_UL\_macro}$ or micro) in the cell layer having lower noise rise than the layer-specific threshold, and decreasing number of users ((i.e. $N_{users\_UL\_micro}$ or macro) in the cell layer in which noise rise is higher than the layer-specific threshold, etc.;
4. redo phases 1 and 2 until noise rise is equal to the specific threshold for both layers;
5. when each layer reaches on average the noise rise threshold, the input values of $N_{UL\_users\_UL\_macro}$ and micro are taken as an output and compared to the values obtained in the single operator case for the ACIR value chosen at step 0.

We investigate two options (e.g. Option A and Option B) in relation to the noise rise threshold. In the first option, the noise rise threshold for the macro layer is equal to 6 dB whilst the threshold for the micro layer is set to 20 dB. The noise rise results from the combination of interference coming from the micro and the macro cell layers. Micro and macro cell layers interact, i.e. micro cell interference affects the macro cell layer and vice versa. In the second option, we set the noise rise threshold to 6 dB for both the macro and the micro layer, but the micro cells are de-sensitized at 14 dB.

5.5.5.2 Downlink

5.5.5.2.1 Single Operator Loading

As in the UL, the number of users in the downlink for the single operator case gets defined as N_{DL_single} . Then DL simulations occur in a way that a single operator network gets loaded so that 95% of the users achieve an E_b/N_o of at least (target E_b/N_o -0.5 dB) (i.e. 95% of users are satisfied) and the supported number of users N_{DL_single} is then measured [5].

5.5.5.2.2 Multi-Operator for Macro to Macro and Macro to Micro

In the *macro to macro* multi-operator case, the networks get loaded so that 95% of users are satisfied and the obtained number of users is defined as N_{DL_multi} . For a given value of ACIR, the measured N_{DL_multi} is obtained and compared to the N_{DL_single} obtained in the single operator case. The multi-operator case (macro to micro) follows similar reasoning to that of the UL case.

5.5.5.3 Simulation Output

Finally, the expected outputs include: capacity figures (N_{UL} and N_{DL}), DL and UL capacity versus ACIR in the multi-operator case, as well as outage (non-satisfied users) distributions.

5.5.6 BTS Receiver Blocking and Simulation Assumptions

The simulations are static Monte Carlo simulations using a methodology consistent with the ACIR approach described in the preceding sections. For our examples, we construct the simulations using two uncoordinated networks at different frequencies. The frequencies assume a separation by 10–15 MHz or more so that the BS receiver selectivity will not limit the simulation, and so that the UE spurious and noise performance will dominate over its adjacent channel performance. These are factors that distinguish a blocking situation from an adjacent channel situation in which significant BS receiver degradation can be caused at very low levels due to the poor ACP from the UE [5].

During each trial of the simulations, we make uniform drops of the UE, adapt power levels, and record data⁶. From these results, we plot CDF of the total signal appearing at the receivers' inputs to be covered in the results sections.

⁶ A thousand such trials are made.

5.5.6.1 *Simulation Scenario Assumptions for 1 and 5 km Cell Radius*

The assumptions here are extracts from [5] to be consistent with the example results. The primary simulation assumptions for the 1 km radius are then:

1. both networks are operated with the average number of users (50) that provide a 6 dB noise rise;
2. the two networks have maximal geographic offset (a worst case condition);
3. cell radius is 1 km;
4. maximum UE power is 21 dBm;
5. UE spurious and noise in a 4.1 MHz bandwidth is 46 dB;
6. BS selectivity is 100 dB (to remove its effect);
7. C/I requirement is -21 dB;
8. BS antenna gain is 11 dB;
9. UE antenna gain is 0 dB; and
10. minimum path loss is 70 dB excluding antenna gains.

The primary assumptions that are common to all simulations in the 5 km radius are:

1. the two networks have maximal geographic offset (a worst case condition);
2. cell radius is 5 km;
3. UE spurious and noise in a channel bandwidth is 46 dB;
4. BS selectivity is 100 dB (to remove its effect);
5. BS antenna gain is 11 dB;
6. UE antenna gain is 0 dB;
7. minimum path loss is 70 dB including antenna gains; in addition,
8. for the speech simulations, maximum UE power is 21 dBm and the C/I requirement is -21 dB;
9. for the data simulations, maximum UE power is 33 dBm and the C/I requirement is -11.4 dB.

Note that this is different from the basic assumption in the ACIR section, since its data power level is 21 dBm, just like the speech level.

5.5.6.2 *Simulation Parameters for 24 dBm Terminals*

The only difference with respect to the parameters listed in the previous sections are:

- 3.84 Mcps chip rate considered;
- 24 dBm Max TX power for the UE (results provided for 21 dBm terminals as well);
- 68 dB dynamic range for the power control;

- number of snapshots per simulation (3000).

Therefore, the considered parameters are shown in Table 5.24.

Table 5.24 Simulation Parameters 24 dBm Terminals

MCL	70 dB
BS antenna gain	11 dBi
MS antenna gain	0 dBi
Log normal shadowing	Standard Deviation of 10 dB
No. of snapshots	3000
Handover threshold	3 dB
Noise figure of BS receiver	5 dB
Thermal noise (NF included)	-103.16 dBm@3.84 MHz
Max TX power of MS	21 dBm/24 dBm
Power control dynamic range	65 dB/68 dB
Cell radius	577 m (for both systems)
Inter-site distance	1000 m (for both systems)
BS offset between two systems (x, y)	Intermediate: (0.25 km, 0.14425 km) → 0.289 km shift Worst: (0.5 km, 0.2885 km) → 0.577 km shift
User bit rate	8 kbps and 144 kbps
Activity	100%
Target Eb/10	6.1 dB (8 kbps), 3.1dB?(144 kbps)
ACIR	25–40 dB

5.5.6.3 Summary of Simulation Parameters

For completeness in Table 5.25 we list the same assumptions simulation parameters as in [5] to be consistent with the example results in the forthcoming sections.

Table 5.25

Parameter	UL value	DL value
Simulation type	Snapshot	Snapshot
Propagation parameters		
MCL macro (including antenna gain)	70 dB	70 dB
MCL micro (including antenna gain)	53 dB	53 dB
Antenna gain (including losses)	11 dBi	0 dBi
	0 dBi	11 dBi
Log Normal fade margin	10 dB	10 dB
<i>PC modelling</i>		
No. of snapshots	>10000 for speech >10 × no. of snapshots for speech for 144 kbps service	>10000 for speech >10 × no. of snapshots for speech in the 144 kbps case >20 000 for data
No. of PC steps per snapshot	>150	>150
Step size PC	Perfect PC	Perfect PC

PC error	0%	0%
Margin with respect to target C/I	0 dB	0 dB
Initial TX power	Path loss and noise, 6 dB noise rise	Random initial
Outage condition	Eb/N0 target not reached due to lack of TX power	Eb/N0 target not reached due to lack of TX power
Satisfied user		Measured Eb/N0 higher than Eb/N0 target -0.5 dB
<i>Handover modelling</i>		
Handover threshold for candidate set	3 dB	
Active set	2	
Choice of cells in the active step	Random	
Combining	Selection	Maximum ratio combining
<i>Noise parameters</i>		
Noise figure	5 dB	9 dB
Receiving bandwidth	4.096 MHz proposed	4.096 MHz proposed
Noise power	-103 dBm proposed	-99 dBm proposed
TX power		
Maximum BTS power		43 dBm macro 33 dBm micro
Common channel power		30 dBm macro 20 dBm micro
Maximum TX power speech	21 dBm	30 dBm macro 20 dBm micro
Maximum TX power data	21 dBm	30dBm macro 20dBm micro
Power control range	65 dB	25 dB
<i>Handling of downlink maximum TX power</i>		
Admission control	Not included	Not included
User distribution		Random and uniform across the network
<i>Interference reduction</i>		
MUD	Off	N/A
Non-orthogonality factor macro cell	N/A	0.4
Non-orthogonality micro cell	N/A	0.06
Common channel orthogonality		Orthogonal
<i>Deployment scenario</i>		
Macro cell		Hexagonal with BTS in the middle of the cell
Micro cell		Manhattan (from 30.03)
BTS type		Omnidirectional

Cell radius macro		577 macro
Inter-site single operator		1000 macro
Cell radius micro		Block size = 75 m, road 15 m
Inter-site single micro		Intersite between LoS = 180 m
Intersite shifting macro		577 and 577/2 m
No. of macro cells		>19 with wrap around technique)
Intersite shifting macro-micro		See scenario
Number of cells per each operator		See scenario
Wrap around technique		Should be used
<i>Simulated services</i>		
Bit-rate speech	8 kbps	8 kbps
Activity factor speech	100%	100%
Multi-path environment macro	Vehicular macro	Vehicular macro
Eb/N0 target	6.1 dB	7.9 dB
Multi-path environment macro	Outdoor micro	Outdoor micro
Eb/N0 target	3.3 dB	6.1 dB
Data rate	144 kbps	144 kbps
Activity factor speech	100 %	100 %
Multi-path environment macro	Vehicular macro	Vehicular macro
Eb/N0 target	3.1 dB	2.5 dB with DL TX or RX diversity, 4.5 dB no diver- sity
Multi-path environment macro	Outdoor micro	Outdoor micro
Eb/N0 target	2.4 dB	1.9 dB with DL TX or RX

5.5.7 Example Results FDD/FDD

Here we illustrate example results primarily for the FDD to FDD mode. In [5] we can see additional cases. The goal of the preceding section and this section is simply to practically visualize some of the procedures covered in the first part of the chapter. Thus, this section aims to collect results on carrier spacing evaluations to illustrate deployment coordination, and multi-layer deployment considerations.

5.5.7.1 ACIR for 21 dBm Terminals

Figure 5.9 illustrates the UL speech ACIR for Intermediate and Worst case scenarios for the Macro to Macro cellular environment, while Figure 5.10 shows the DL case.

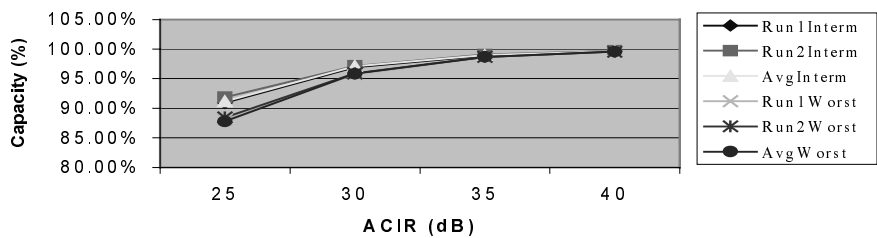


Figure 5.9 ACIR intermediate and worst case for macro to macro UL speech 8 kbps.

The examples include UL and DL 8 kbps speech service with the following characteristics:

- Intermediate case scenario where the 2nd system is located at a half-cell radius shift.
- Worst case scenario where the second system base stations are located at the cell border of the first system.
- Average results for intermediate and worst case.

We can see clearly in Figure 5.9 that as the ACIR increases the capacity also increases. The impact applies to the intermediate as well as to the worst case situations. Likewise for the DL case, as the ACIR value increase the capacity keeps high. The worst case run still has impact at 32.5 dB but remains at minimum at 35 dB.

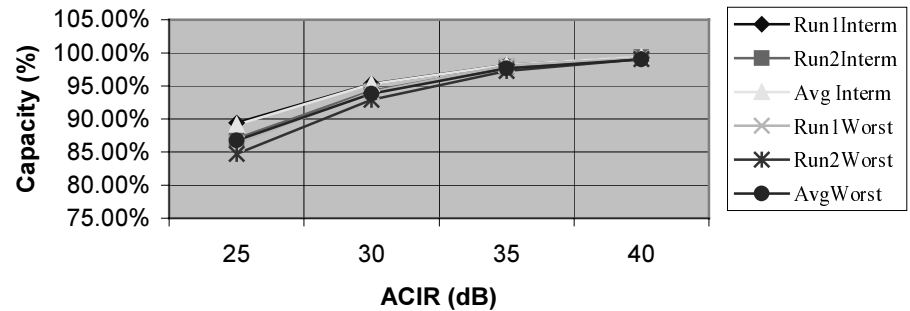


Figure 5.10 ACIR intermediate and worst cases for macro to macro DL speech 8 kbps.

5.5.7.2 ACIR for 24 dBm terminals

Other simulation results also following [5] include outputs for UL ACIR with 24 dBm terminals, for both speech (8 kbps) and data (144 kbps). We compare the results with those obtained with 21 dBm terminals. Figure 5.11 illustrates the UL ACIR in 24 dBm terminals.

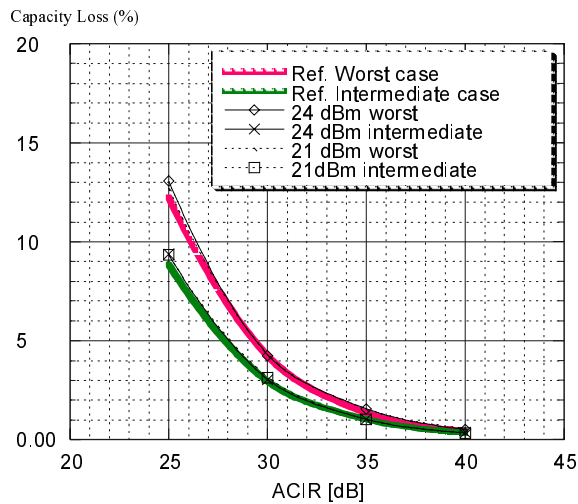


Figure 5.11 UL ACIR speech 24 dB: macro to macro.

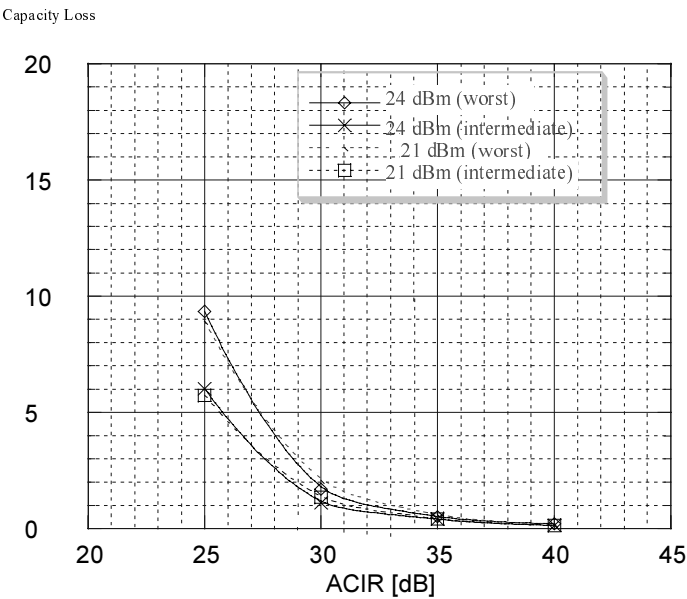


Figure 5.12 UL data (144 kbps) macro to macro.

When comparing the results illustrated Figures 5.11 and Figure 5.12, we can see that with lower ACIR values in speech, the capacity degrades much more than with the same ACIR values in data.

5.5.8 BTS Receiver Blocking

5.5.8.1 Simulation Results for 1 km Cell Radius

Figure 5.13 shows the overall Cumulative Distribution Function (CDF) of the input signals to the receivers using 21 dBm terminals. Based on the preceding simulation assumptions and parameters, we can perceive that the largest signal appears at -54 dBm amplitude while occurring in less than 0.01% of the cases. Although simulations have not been done for a higher power terminal, according to [5] it is reasonable to assume approximated scaling of the power levels by 12 dB (i.e. from 21 to 33 dBm). Thus, it is proposed that $-54 + 12 = -42$ dBm should be considered a reasonable (if not slightly pessimistic) maximum value for the largest W-CDMA blocking signals.

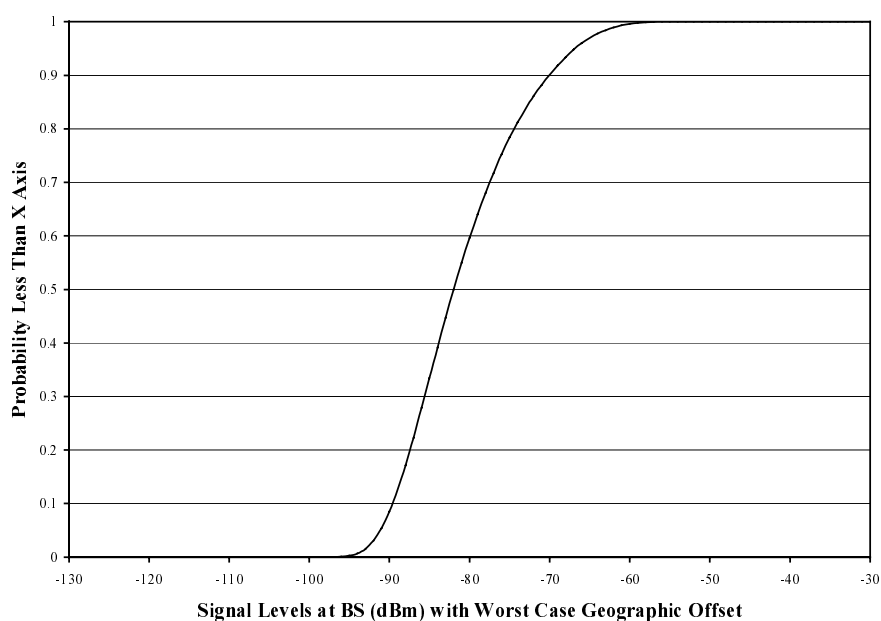


Figure 5.13 BS signals levels [5].

5.5.8.2 Simulation Results for 5 km Cell Radius

Figure 5.14 illustrates the overall CDF of the input signals to the receivers using speech only. Discontinuity occurs, e.g. at -49 dBm input level because in large cells there are a few occurrences of users operating at their maximum transmitted power level of 21 dBm while remaining close enough to another network's cell to produce a minimum coupling loss condition.

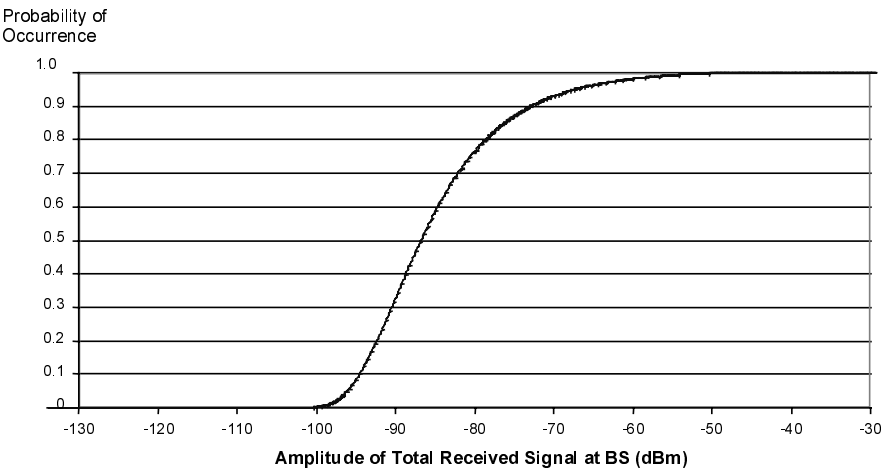


Figure 5.14 CDF of total signal for speech only system with 5 km cells and worst case geographic offset (after [5]).

Therefore, for this large cell, the received signal power level corresponding to 99.99% of the occurrences is very close to the level dictated by MCL and is about -49 dBm ($= 21\text{ dBm} - 70\text{ dB}$) [5]. The preceding event may show the same phenomenon with mixed speech and data systems, i.e. it would produce approximately the same result if the maximum power level for a data terminal were also 21 dBm .

Figure 5.15 illustrates the CDF of the input signals to the receivers in mixed speech and data systems. This indicates that 99.99% of occurrences of the input signals to the receivers are about -40 dBm or less. Because of the large cell, the MCL dictates the absolute maximum signal; and it is only a few dB higher, i.e. $(33\text{ dBm} - 70\text{ dB} = -37\text{ dBm})$.

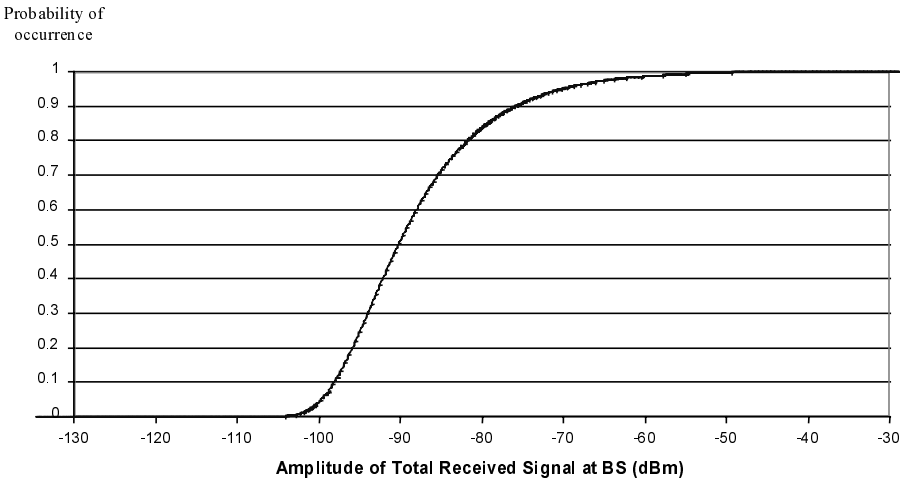


Figure 5.15 CDF of total signal for mixed speech and data system with 5 km cells and worst case geographic offset (after [5]).

Discussion in [5] indicates that it may be desirable to allow more than the 3 dB degradation in sensitivity to that which is typically used in the measurement of a blocking specs. This can be justified because:

- the interfering UE's spurious and noise are going to dominate the noise in the victim cell in a real system, and
- the measurement equipment is approaching the limit of its capability in the performance of this test.

The first reason seems evident from observing that the interfering UE's noise two channels from its assigned frequency is probably typically in the range of -90 dBm ($= -40$ dBm $- 50$ dB), which is much larger than the typical noise floor of the receiver at -103 dBm. The 2nd reason appears evident from observing that the typical noise floor of most high quality signal generators equals $65-70$ dBc with a W-CDMA signal. The latter results in test equipment generated noise of -105 to -110 dBm, which can produce a significant error in the blocking measurement.

In view of these concerns, it is probably reasonable to allow more than a 3 dB increase in the specified sensitivity level under the blocking conditions. In conclusion, it seems reasonable to assume that the in-band blocking specification for UTRA should be -40 dBm (considering that 33 dBm terminals will exist), and the interfering (blocking) test signal should be an HPSK carrier. A 6 dB degradation in sensitivity under the blocking condition should be allowed [5].

5.5.9 Transmit Intermodulation for the UE

User equipment(s) transmitting in close vicinity of each other can produce Intermodulation (IMD) products, which can fall into the UE, or BS receive band as an unwanted interfering signal. The transmit intermodulation performance indicates the ability of a transmitter to inhibit the generation of signals in its non-linear elements caused by presence of the wanted signal and an interfering signal reaching the transmitter via the antenna.

We define the UE intermodulation attenuation by the ratio of the output power of the wanted signal to the output power of the intermodulation product when an interfering CW signal gets added at a level below the wanted signal. We measure, both the wanted signal power and the IMD product power with a filter that has a Root-Raised Cosine (RRC) filter response with roll-off of 0.22 and a bandwidth equal to the chip rate. Such test procedure is identical to the ACLR requirement with the exception of the interfering signal. Thus, when performing the aforementioned test, we cannot separate the ACLR impact due to the wanted signal, which would fall into the 1st and 2nd adjacent channel from the IMD product as a result of the interfering signal. Consequently, the IMD cannot be specified to be the same value as the ACLR. It has to be a lower value (e.g. 2 dB) to account for the worst case ACLR contribution [5].

5.6 CONCLUSIONS

In this chapter we outlined the UTRA transmission system characteristics and also provided reference examples quantifying some of the recommended parameters or threshold values. The examples use the principles described in Chapter 2 through simulation techniques carried out during the specifications. Thus they serve primarily as illustrations to visualize some of the impacts while designing actual UMTS networks.

References

- [1] 3GPP, Technical Specification Group (TSG) RUN WG4, UE Radio Transmission and Reception (FDD) (3G TS 25.101. version 3.1.0, 1999-12).
- [2] 3GPP, Technical Specification Group (TSG) RUN WG4, UTRA (UE) TDD; Radio Transmission and Reception (3G TS 25.102. version 3.1.0, 1999-12).
- [3] 3GPP, Technical Specification Group (TSG) RUN WG4, UTRA (BS) FDD; Radio Transmission and Reception (3G TS 25.104. version 3.1.0, 1999-12).
- [4] 3GPP, Technical Specification Group (TSG) RUN WG4, UTRA (BS) TDD; Radio Transmission and Reception (3G TS 25.105. version 3.1.0, 1999-12).
- [5] 3GPP, Technical Specification Group (TSG) Radio Access Networks; RF System Scenarios, Release 1999 (3G TS 25.942. Version 2.1.3, 2000-03).
- [6] Pizarrosa, M., Jimenez, J. (eds.), "Common Basis for Evaluation of ATDMA and CODIT System Concepts", MPLA/TDE/SIG5/DS/P/001/b1, September 1995.
- [7] "Universal Mobile Telecommunications System (UMTS); Selection Procedures for the Choice of Radio Transmission Technologies of the UMTS", TR 101 112 V3.1.0 (1997-11), UMTS30.03 version 3.1.0.

6 SERVICE COMPONENTS IN UMTS

6.1 BACKGROUND

UMTS services will not only offer mobile services supported by 2nd generation systems such as GSM, but will also expand these services to higher rates and greater flexibility. The services evolving in the GSM platform through its Circuit Switched (CS) and Packet Switched (PS) services will continue in UMTS while new services are introduced.

Thus, future UMTS services will have user transmission rates from low bit up to 2 Mbps. Although, high rates will occur primarily within indoor environments, there will be substantial increases in rates throughout all environments when compared to the typical 9.4 kbps. Table 6.1 (an extract from Table 2.1) illustrate this increase.

Table 6.1 Range of Transmission Rates

High level description	Maximal bit rate (kbits/s)	Maximal speed (km/h)	Cell coverage
Rural outdoor	144s	500	Macrocell
Suburban outdoor	384	120	Microcell
			Macrocell
Indoor/ Low range outdoor	2048	10	Picocell
			Microcell

Then the question of the transmission range for UTMS services, is no longer just what transmission rates, but what type of services, when and where. It is no longer “communications any where any time”, but “what I want when I want where ever I want”.

Practically, the exploitation of wider transmission rates will facilitate the expansion of data traffic. As illustrated Table 6.2 there exists a clear trend for the convergence of IP protocol to wireless, or to what we now call *wireless IP*. The latter will lead to the *Wireless Internet*, where about 200 M Internet and 300 M mobile subscribers will merge into 1 billion Wireless Internet users.

Table 6.2 Convergence of Internet Protocol (IP) to Wireless

Computer: mobility high speed services	Telecommunications: mobility wide services	Media: mobility personal services
Internet access	ISDN services	Streaming audio
Electronic mail	Video telephony	Video on demand
Real time images	Wideband data services	Interactive video services
Multimedia	Location services coupled with application servers	TV/radio/data contribution and distribution

Non-voice services will make demands not only on manufacturers and operators but also from supporting industries, creating a need for new service enablers. However,

such demand will also introduce new challenges and the need for pragmatic integration of services and devices, as well as new data processing and managing techniques. These demands can be summarized as needs as illustrated in Table 6.3.

Table 6.3 Needs for Service Providers and Technology Enablers

Needs for service providers	Needs for technology enablers
Strategy for innovative services	Well integrated CS and PS system
Economic and spectrum efficiency data pipe	Advanced value added platforms (e.g. WAP, IS, location services, unified messaging, etc.)
Standard interface to phone display	Power efficient handsets
Dynamic management control points	Effective yet very light device OSs
New and flexible billing systems	Text → speech
Perception of market needs	Speech → text
Personalization	Intelligent voice recognition
Addressing all user segments	Multi-band terminals exploiting software radio
New data processing and management techniques	Synchronization
Cost efficient terminals and devices	Pragmatic user interfaces (e.g. efficient portals)

Clearly, the challenges cover all main areas of SW/HW and management technology. In the forthcoming sections we will see how UMTS addresses these needs and outline the main approaches and requirements to meet the challenges.

6.2 THE UMTS BEARER ARCHITECTURE

As illustrated in Figure 6.1 after [1], UMTS proposes a layered bearer service architecture, where each bearer service on a specific layer offers its individual services based on lower layers. Thus, the UMTS bearer service architecture serves as an ideal platform for end-to-end services providing key features in preceding layers.

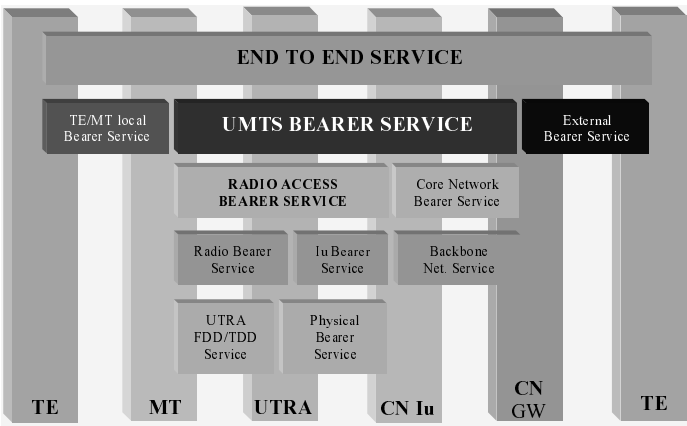


Figure 6.1 UMTS bearer service architecture.

Because of its layered-bearer service architecture UMTS permits users or applications to negotiate, re-negotiate or change appropriate bearer characteristics to carry their information. Negotiations take place based on application needs, network resource availability, and demands of quality of service (QoS).

6.3 QUALITY OF SERVICE IN 3RD GENERATION NETWORKS

The main four classes of UMTS traffic differentiated by their delay sensitivity are conversational, streaming, interactive, and background. Conversational classes have higher delay sensitivity than background classes. The first two classes correspond to real time classes, while the 2nd two to non-real time. Table 6.4 illustrates these classes.

Table 6.4 QoS Classes in UMTS

Traffic class	Conversational	Streaming	Interactive	Background
Characteristics and applications	Preserve time relation between information entities – low delay (e.g. voice, video-telephony)	Preserve also time relation between information entities (e.g. multi-media)	Request response pattern preserving data integrity. (e.g. Internet or web browsing)	Connectionless, generally packet communications preserving data integrity (e.g. ftp, email, etc.)

6.4 MULTIMEDIA TRANSMISSION – UMTS TRAFFIC CLASSES

6.4.1 Conversational

Typical speech over CS bearers, voice over IP (VoIP) and video-telephony represent the conversational class, which in turn represent real-time services. The latter corresponds to symmetric traffic with end-to-end delay thresholds below 399 ms.

6.4.1.1 Enabling Speech

The adaptive multi-rate (AMR) techniques will enable the UMTS speech codec. This codec consists of single integrated speech codec with eight source rates controlled by the RAN, i.e: 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 use of the average required bit rate has impacts on interference levels, thereby on capacity, and battery life. Logically, lower rates will favour capacity and battery life duration, but not necessary quality.

The AMR coder [4] works with speech frames of 20 ms, i.e. 160 samples at a sampling rate of 8000 sample/s. It may switch its bit rate at every frame through in-band signaling or through a dedicated channel. It uses Multi-rate Algebraic Code Excited Linear Prediction Coder (MR-ACELP) as a coding scheme. We extract CELP parameters at each 160 speech samples for error sensitive tests. The latter consist of three error classes (A–C), where class A has the highest sensitivity and requires strong channel coding. The AMR speech codec can tolerate about 1% frame error rate (FER) of class A bits without any deterioration of the speech quality. For classes B and C bits a higher FER can be allowed. The corresponding bit error rate (BER) of class A bits will be about 10^{-4} .

AMR allows an activity factor of 50% (while parties have a telephone conversation), through a set of basic functions:

- background acoustic noise evaluation on the Tx to transmit key parameters to the Rx;
- Voice Activity Detector (VAD) on the Tx;
- a Silence Descriptor (SID) frame that passes transmission comfort noise information to the Rx at regular intervals. This noise gets generated on the Rx in the absence of normal speech frames.

6.4.1.2 *Enabling Circuit Switched Video Telephony*

Video telephony has higher BER requirements than speech due to its video compression features; however, it has the same delay sensitivity of speech. Technical specifications in [2] UMTS recommend ITU-T Rec. H.324M for video telephony in CS links, while at present there exists two video telephony options for PS links, i.e. ITU-T Rec. H.323 [5] and IETF SIP [7]. The H.323 has similar characteristics to H.324M.

The adapted¹ H.324 includes essential elements such as H.223 for multiplexing, H.245 for control. It also includes H.263 video codec, G.723.1 speech codec, and V.8bis. It may have MPEG-4 video and AMR to better suit UMTS services as illustrated in Figure 6.2. Technical specifications include seven phases for a call, i.e. set-up, speech only, modem learning, initialization, message, end, and clearing. Backward compatibility occurs through level 0 of the H.223 multiplexing, which is the same as H.324.

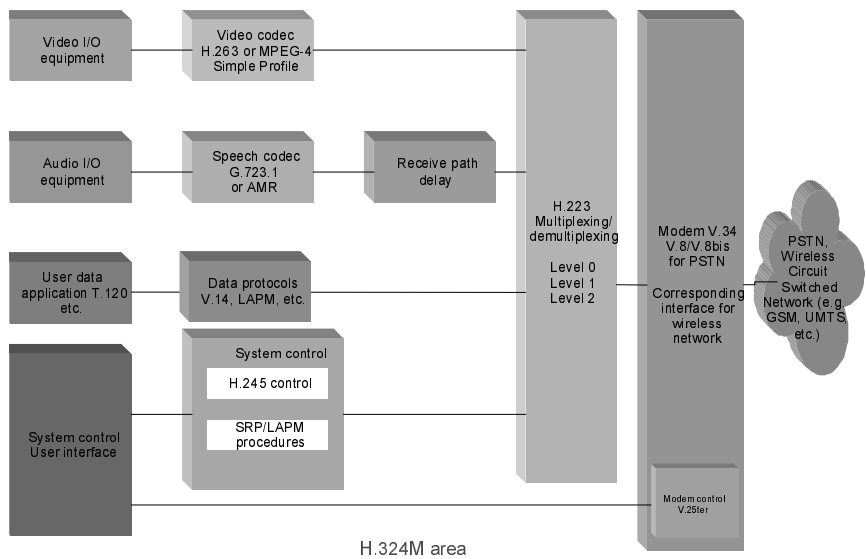


Figure 6.2 The ITU Rec. H.324 model.

¹ Adapted to wireless from what was originally meant for fixed networks.

The H.324 terminal has an operation mode for use over ISDN links. Annex D in the H.324 recommendations defines this mode of operation as H.324/I [3]. H.324/I offers direct inter-operability with the H.320 terminals, H.324 terminals on the GSTN, H.324 terminals operating on ISDN, and voice telephones.

For seamless data communications between UMTS and PSTNs, the UMTS call control mechanism takes into account V.8bis messages. These messages get interpreted and converted into UMTS messages and V.8bis, respectively. The latter contains identification procedures and selection of common modes of operation between data circuit-terminating equipment (DCE) and between data terminal equipment (DTE). Essential V.8bis features include:

- flexible communication mode selection by either the calling or answering party;
- enabling automatic identification of common operating modes;
- enabling automatic selection between multiple terminals sharing common telephone channels;
- friendly user interface to switch from voice telephony to a modem based communications.

6.4.1.3 Enabling Packet Switched Video Telephony

The H.323 ITU-T protocol standard for multimedia (and IP telephony) call control enables PS multimedia communications in UMTS. The standard:

- employs a peer-to-peer model in which the source terminal and/or GW is the peer of the destination terminal and/or GW,
- treats gateways (GW) and terminals alike;
- requires GWs and terminals to provide their own call control/processing functions;
- provides multiple options for voice, data and video communications;
- it may employ a gatekeeper function to provide telephone number-to-IP address translation, zone admission control and other resource management functions.

Figure 6.3 illustrates the H.323 architecture, which incorporates a family of standards including H225, H245 and H450. As an international standard for conferencing over packet networks H.323:

- acts as a single standard to permit Internet telephony products to inter-operate;
- also serves as base for standard interoperability between ISDN- and telephony-based conferencing systems; and
- has the flexibility to support different HW/SW and network capabilities.

The logical channels in H.323 get multiplexed at the destination port transport address level. The transport address results from the combination of a network address and a port identifying a transport level endpoint, e.g., an IP address and a UDP port. Packets having different payload types go to different transport address, thereby eliminating

usage of separate multiplexing/demultiplexing layer in H.225.0. The H.225 standard uses RTP/RTCP² for media stream packetization and synchronization for supporting LANs. This usage depends on the usage of UDP/TCP/IP. BER control takes place at lower layers; thus, incorrect packets do not reach the H.225 level.

When both audio and video media act in a conference, they transmit using separate RTP sessions, and RTCP packets get transmitted for each medium using two different UDP port pairs and/or multicast addresses. Thus, it does not exist direct coupling at the RTP level between audio and video sessions, and synchronised playback of a source's audio and video takes place using timing information carried in the RTCP packets for both sessions.

Point-to-point H.323 conference occurs with two TCP connections between the two terminals, i.e. one for call set-up connection and one for conference control and feature exchange. The first connection carries the call set-up messages defined in H.225.0, i.e. the Q.931 channel. After a 1st TCP connection on a dynamic port, the calling parties establish the second TCP connection to the given port, where the 2nd connection carries the conference control messages defined in H.245. Thus, the H.245 serves to exchange audio and video features in the master/slave context.

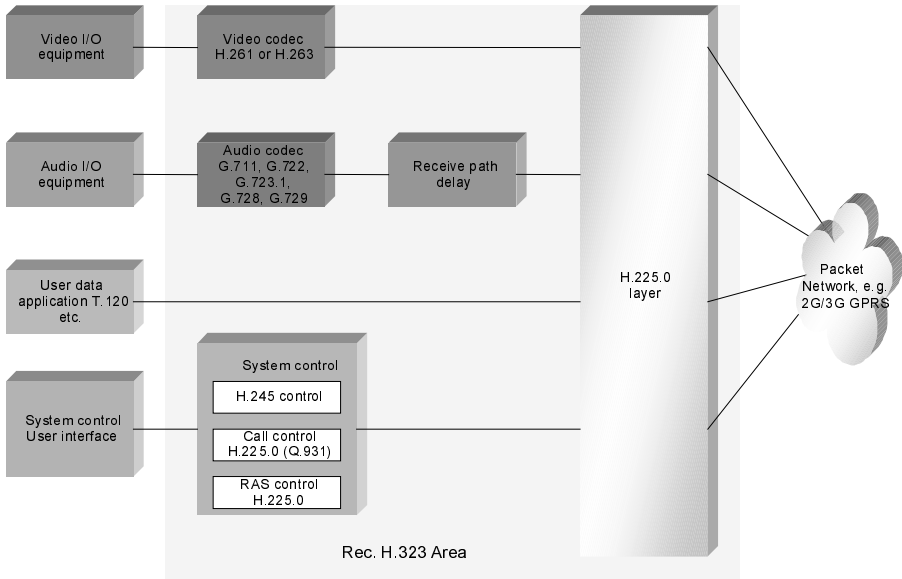


Figure 6.3 The ITU Rec. H.323 model.

6.4.1.4 Session Initiation Protocol (SIP)

The Session Initiation Protocol (SIP) is another alternative to enable PS video-telephony. Developed in IETF by the MMSIC Multiparty Multimedia Session Control group, SIP is an application layer control signalling protocol for creating/modifying and

² Real-time transport protocol/real-time transport control protocol.

terminating sessions with one or more participants, e.g. Internet multimedia conferences, Internet telephone calls and multimedia distribution. Participants in a session can communicate via multicast or via a mesh of unicast relations, or a combination of these. See Figure 6.4. SIP corresponds to:

- the overall IETF multimedia data and control architecture currently incorporating protocols such as Resource Reservation Protocol – RFC 2205 (RSVP) for reserving network resources;
- the real-time transport protocol (RTP – RFC 1889) for transporting real-time data and providing QoS feedback;
- the real-advertising multimedia sessions via multicast and the session description protocol (SDP – RFC 2327) for describing multimedia sessions.

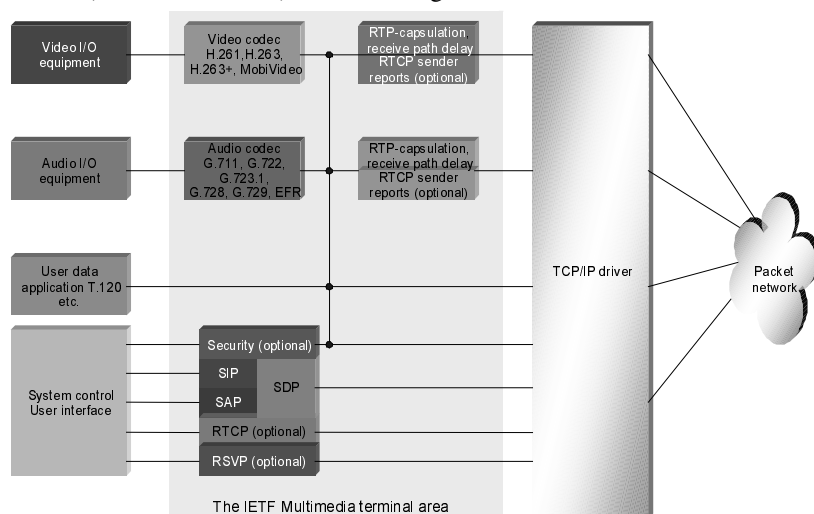


Figure 6.4 The IETF multimedia model.

Nevertheless, it does not depend in any of the above for its functionality and operation.

SIP transparently supports name mapping and redirection services, thereby allowing the implementation of ISDN and IN telephony subscriber services, and enabling personal mobility. Technically SIP has the following characteristics

- called and calling peers can specify their preference of where they would like calls to be connected;
- use of user@domain as call addresses and http look alike messages;
- only deals with tracking down users and delivering a call to an endpoint, i.e. it is orthogonal to other signalling protocols;
- uses servers for redirection (redirect server), user location tracking (registrar), fork request (proxy server);
- it does not have address initiation and termination like H.323, but it is widely accepted;

- simple and easy to implement by IP developers.

SIP supports five phases of establishing and terminating multimedia calls:

- user location → determination of the end system for connection;
- user capabilities → determination of the media and media parameters for usage;
- user availability → determination of the willingness of the called party to engage in communications;
- call setup → ringing establishment of call parameters at both called and calling party;
- call handling → including transfer and termination of calls.

SIP can also initiate multi-party calls using a multi-point control unit (MCU) or fully meshed interconnection instead of multi-cast.

SIP vs. H.323

Standards body	H.323 ITU-TSG – 16	SIP IETF MMUSIC
Properties	Based on H.320 conferencing and ISDN Q.931 legacy Difficult to extend and update No potential beyond telephony Complex, monolithic design	Based on Web principals (Internet friendly) Easily to extend and update Readily extensible beyond telephony Modular simplistic design
Standards status	H.450.x series provides minimal feature set (pure peer approach) Adding mixed peer/stimulus approach (inefficient architecture) Slow moving	No real end-device feature standard yet Many options for advanced telephony features Good velocity
Industry acceptance	Established now, primarily system level Few if any H:323 base telephones End-user primarily driven by Micro-soft (NetMeeting), Intel, etc.	Rapidly growing industry momentum (system level) Growing interest in SIP phones and soft clients

Undoubtedly, SIP is poised as the most appropriate protocol to enable PS video telephony in UTMS. At this writing, technical bodies are debating the final outcome. From the author’s point of view, it seems evident that SIP would lead to better results and widespread usage of video telephony.

6.4.1.5 Layer Structure Enabling for Multimedia – MEGACO/H.248

MEDIA Gateway Control (MEGACO) or H.248 is part of the protocols that will facilitate the control of video telephony on the PS side. Megaco/248 jointly developed by ITU TG-16 and IETF, covers all gateway applications moving information streams from IP networks to PSTN, ATM and others. These include: PSTN trunking, gateways,

ATM interfaces, analog line and telephone interfaces, announcement servers, IP phones, and many others.

The Megaco IP phone master/slave approach is entirely compatible with peer-level call control approaches such as SIP and H.323. It acts orthogonal to the last two protocols. Megaco/H.248 allows:

- profiles to be defined, i.e. permits application level agreements on gateway organization and behaviour to be made for specific application types, thereby reducing complexity;
- allows support of multiple underlying transport types (e.g. ALF reliability layer over UDP, TCP), and both text and binary encoding; the latter enables more appropriate support for a broader range of application scales (e.g. big vs. small gateways) and more direct support for existing systems.

6.4.1.6 IETF Signalling Transport (SIGTRAN)

SIGTRAN develops an essential Simple Control Transmission Protocol (SCTP), which we view as a layer between the SCTP user application and an unreliable end-to-end datagram service such as UDP. Thus, the main function of SCTP amounts to reliable transfer of user datagrams between peer SCTP users. It performs this service within the context of an association between SCTP nodes, where APIs exist at the boundaries.

SCTP has connection-oriented characteristics but with broad concept. It provides means for each SCTP endpoint to provide the other during association startup with a list of transport addresses (e.g. address/UDP port combinations) by which that endpoint can be reached and from which it will originate messages. The association carries transfers over all possible source/destination combinations, which may be generated from two end lists. As result SCTP offers the following services:

- application-level segmentation;
- acknowledged error-free non-duplicated transfer of user data;
- sequenced delivery of user datagrams within multiple streams;
- enhanced reliability through support of multi-homing at either or both ends of the association;
- optional multiplexing of user datagram into SCTP datagrams.

6.4.2 Streaming

Streaming implies transmitting information continuously in streams. This technique facilitates Internet browsing by allowing displays even before the completion of information transfer. It has higher tolerance for jitter to support the large asymmetry of Internet applications. Through buffering, the streaming technique smoothes out packet traffic and offers it as it becomes available. Thus, it can support video on demand as well as web broadcast. While both types of video applications can benefit from the same video compression technologies, they differ in the usage of coding, protocols, etc. Thus, we can offer two types of video applications and address or offer services to more than one type of user depending on the transmission rate or delay sensitivity.

6.4.3 Interactive

Logically, we denote interactive to be the dynamic exchange of information through a man-machine interface or machine-to-machine interconnection. The tempo of the dynamics will depend on the application or the purpose of the device under interaction. In the context of Internet applications like web browsing, the response time will depend on the type of information requested and the quality of the link as well as protocols in use. Delay sensitive applications will demand faster interaction, e.g. emergency devices, system controls, etc. Other applications such location services, games, passive information centres, etc., will operate within flexible round trip delays. In the forthcoming section we cover other applications.

6.4.4 Background

While the background class still grows with innovative solutions, it remains as one of the traditional data communications techniques. It serves for e-mail, SMS, database inquiry, and information service platforms. Delay does not have critical consequence in this class, although delays of more than a minute will be highly noticeable. But despite the non-demanding round-trip delays, accuracy becomes critical. Thus, the background users expect error-free communications. For example, control mechanisms measuring performance or monitoring actions will need a reliable accuracy when sending or transmitting information.

6.4.5 Sensitivity to IP Transmission Impairments

To conclude the UMTS traffic classes, in the following we briefly outline some criteria for different applications in the context of the aforementioned classes.

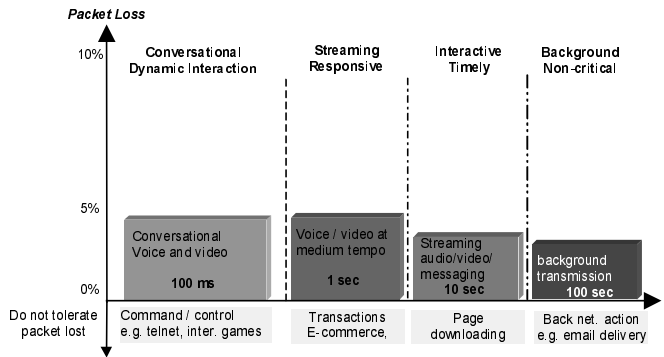


Figure 6.5 Sensitivity of applications to delay in IP environments.

From the algorithmic representation of delays (x-axis) and linear scale of packet loss estimation (y-axis) in Figure 6.5, we can see the sensitivity of applications to IP impairments. Clearly, entries below the vertical axis do not tolerate any type of packet lost; e.g. command/control actions in Telnet or interactive games, on-line-banking, e-commerce, etc. Which means that reliable service transmission will imperatively include both delay control and packet transfer integrity.

Controlling delay implies keeping end-to-end one way delay below 250 ms, otherwise this impairment will annoy users and service quality perception will diminish. When packets get lost due to late arrival or discarded as result of congestion, the missing information degrades multimedia transmission, demanding Packet Loss Concealment (PLC) techniques in voice type transmission and error correction or re-send on data transmission.

Although PS or IP networks have flexibility when using codecs, we still need to add encoding time to the end-to-end delay. While the delay for different types of codecs illustrated in Figure 6.6 vary depending on their physical parameters, e.g., type, bit rate, and frame size as noted in Table 6.5; all must allow normal CS voice quality.

Table 6.5 Selected Codec Parameters

Codec	Type	Bit rate (kbps)	Frame size (ms)	Total delay (ms)*
G.711	PCM	64	Based on packet size	
G.726	ADPCM	32	Based on packet size	
G.729/A	CS-ACELP	8	10	25
G.732.1	MP-MLQ	6.3	30	67.7
GSM-EFR	ACELP	12.2	20	40

*Total delay assumes one frame per packet

Voice quality obtained through test methods, e.g. Mean Opinion Score (MSO) described in ITU recommendations P.800, rate the GSM-EFR codec quite acceptable (Figure 6.6). This codec corresponds to the AMR family selected for UMTS. Hence, when it comes to delay limits for future VoIP services, e.g. 3G networks will not add unnecessary delays.

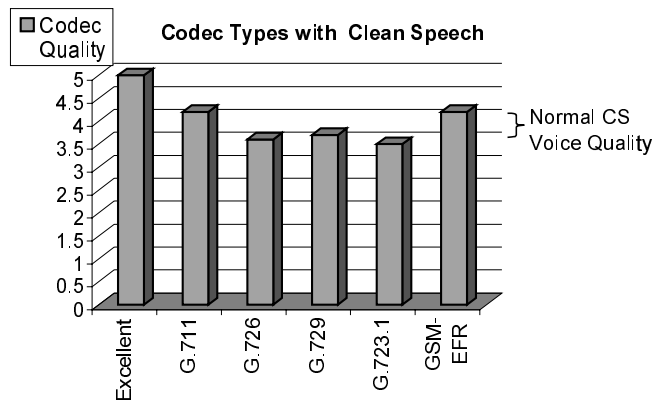


Figure 6.6 Codec quality perception.

Finally for completeness, we list the ITU G.114 recommendations on delay limits. Depending on the applications delay ranges can be noted as follows:

- 50 ms limit for processing delay, → will vary based on processing power;
- 0–150 ms one way delay → acceptable;

- 150–400 ms one way delay → acceptable depending on the applications;
- >400 ms one way → unacceptable.

In summary sources of delay in PS network include:

- propagation (while the signal moves through the channel);
- processing (encoding/transcoding → 50–140 ms, packetization → 0–60 ms, DSP functions e.g. filtering → 0–25 ms);
- packet loss mitigation (queuing and jitter buffers → 20–50 ms, interleaving → 5–90 ms).

6.5 APPLICATIONS AND SERVICE OFFERINGS

The questions arising from the exploitation of wireless networks, more in particular IP based network or non-voice services, can be summarized as follows:

- What are these services?
- Who are they targeted at?
- How much do we offer them for?
- How do we apply technology? Or what technology do we require?

6.5.1 UMTS Generic Services

The strength of UMTS services will not reside in one or two applications, but in the conjunction and complementation of a series of application and technologies, which will generate different sets of services. Figure 6.7 illustrates a generic set of application targets primarily for PS networks including multimedia features. In this illustration, we can see the characteristics of connectionless and connection oriented services, i.e. variable and constant bit rate.

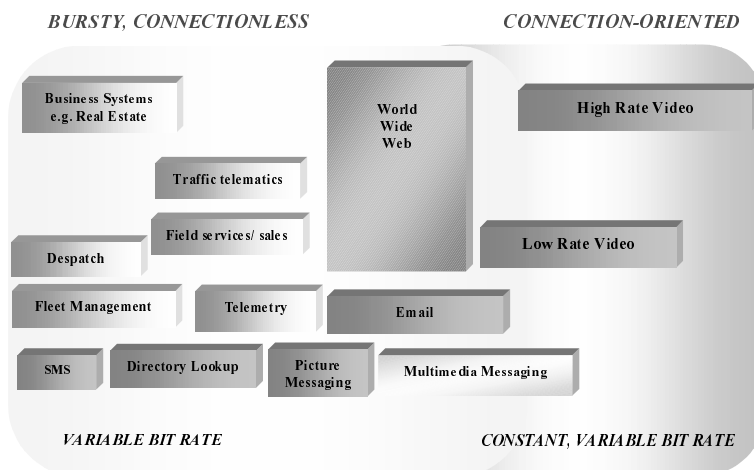


Figure 6.7 Generic range non-voice applications.

We not only need to know to what groups we can address these services (e.g. enterprises, communication firms, telematic centres, content and location based providers,

commerce organizations, and typical wireless operators aiming to minimize operational costs and churn). We also need to know where is the end user and how does he/she apply technology.

6.5.2 Family of UMTS Users

In the process of identifying the potential 3G users we can segment the subscriber body based in the population distribution, e.g. business, residential and mass market. We can further break these groups down into heavy and light users. However, our interest lies in finding who does actually correspond to each group and how much traffic they generate.

6.5.2.1 Business Subscribers

Business users will follow their enterprises and set the pace according to the wealth of resources and activity intensity. While a simple distinction would fall into large, medium and small corporations, it will not identify the true nature of business users. Thus, for all practical purposes we will group (non-exhaustively) into:

- *Information technologist* → involved in generating or transferring all types of modern information in communications and computers, software, etc.
- *Designers and producers* → working in manufacturing, heavy industry, product lines, etc.
- *Distributors and retailers* → active in marketing, sales, product distribution
- *Financial and legal people* → banking and financing work, legal world activities, etc.

The classification above aims to group activities while identifying the type of business subscribers will foster. Then, based on the profile we can see the volumes of traffic and demands they will generate.

6.5.2.2 Residential Subscribers

We can characterize these subscribers by their life style. The latter in turn will provide a window to the amount of traffic they will generate. To make it simple and logical we can classify them into:

- *communicators* → those continuously involved in social activities, communicating at all times;
- *always prepared* → keeping up with the trends and having all means of modern communications;
- *world travellers* → relocating often, an international citizen;
- *well to do* → the wealthy and established pillars of the community owning the national capital.

6.5.2.3 *Mass Market Subscribers*

All the remaining population groups not listed in the preceding segments correspond to this category, e.g. children and young people 5–22 years, the labour force, educational groups (i.e. university), institutions, government bodies, etc. All of us while not classified in the above categories may also correspond to this segment.

6.5.3 **Cost and Services**

Regardless of who the subscriber is or to what subscriber segment he/she belongs, a user will always be looking to cost–value investments. Costly or too sophisticated communication services will not appeal to any of the aforementioned segments. Expensive services like the ones proposed by the Iridium³ group will not gain sufficient penetration to justify investments. Thus, 3G services will not only need to be affordable, but also efficient to generate interest in all segments. No doubt, acceptance level will vary from group to group and region to region, but affordability and utility will go before wide acceptance. Hence, the key issue is to meet the needs of whatever segment of the population group.

6.5.4 **UMTS Services Technology**

To meet the needs implies making available the correct tools and environment. Now, if we assume that the infrastructure arrangements will take care of the environment, it remains a big task to find a tool or user equipment device to satisfy users.

A terminal not only needs to be a smart device capable of accessing a PS network, support bandwidth on demand, audio streaming, multimedia, it will also need versatility and have multiple capabilities.

A multi-functional device will make the difference in future usage and acceptance of higher transmission rates offered through UMTS. Market penetration and widespread usage of these of multimedia services will depend on the available and affordable terminals, as well as the pragmatic applications.

Wireless device interconnections, intelligent voice recognition, wireless e-mail, simultaneous voice and data, user defined closed user group, location services [6,8], personal profile portal, location based delivery and marketing will only occur with efficient integration and inter-working of multiple technologies.

During 2002–2003, more than 50% of terminals will be replaced ranging from low end to high end, with about 80% penetration of mobile users in some regions; today's smartphones will be tomorrow's low end terminals.

Thus, the minimum features for a UMTS handset at the start of 3G services will consist of:

- dual mode UMTS/GSM 900, 1800, 1900 MHz, including GPRS and HSCSD for seamless compatibility and roaming with 2G networks;

³ Satellite mobile services offering mainly voice and low data with world coverage.

- integrated, WAP, Bluetooth;
- voice control and intelligent voice recognition (e.g. VoxML);
- large colour display and limited multimedia features;
- simultaneous UMTS sessions from 64 kbps up to 384 kbps;
- approx. $100 \times 50 \times 18$ mm and <100 g;
- accessories – headset, camera, GPS and all existing accessories.

Furthermore, information centric devices like PDAs will have additional options, e.g.

- advanced multimedia capabilities;
- video clip and play support with easy man–machine interface;
- video and music stream support;
- standard and open OS;
- WAP and Java application capabilities;
- HTML and XML browser, e-mail client, personal portal configuration capabilities;
- intelligent phone management features, e.g. cmd completion;
- advanced colour touch-screens;
- Bluetooth and all necessary features integrated, e.g. pull-able headset, camera, etc.

6.5.4.1 Applications

Applications may not necessarily come from the technology design. However, the final blend will depend on the available and accessible technology. Therefore, the creation and implementation of applications will require large complicity between those providing technology solutions, those generating application platforms (including SW), and those planning to offer services. For example, the quality and utilization feature of location services will not depend only on the information services server, but also on the capabilities of the terminal to display the information.

6.6 CONCLUSIONS

The set of service components for UMTS will continue to evolve, e.g. Chapter 9 outlines further requirements for future UMTS specification releases. Thus, this chapter points out only the key elements that we cannot neglect as we follow the trends for non-voice services over integrated 3G CS and PS networks.

The essential issues discussed in the preceding sections indicate that despite the large amount of standardization work and intensive industry activity, we are still in the process of consolidating solutions for terminals and applications.

The massive investment in license fees for the spectrum⁴, may in some way be limiting future UMTS service providers from practical dedication to the creation of services in

⁴ Operators in UK and Germany have each invested more than US\$5 billion in UMTS licenses; operators in other countries may invest less. However, there is a trend for governments to get the maximum from the 3G spectrum.

closer collaboration with content providers, developers, and manufactures. The truth is that we may also need massive investment for the implementation of innovative applications and services to take full advantage of the potential of UMTS and its forthcoming technology.

On the service provider side, again, it does not matter within what segment subscribers are, at the end, with the penetration of mobile services, free-Internet and the choice⁵ of service provider, users will only care about quality, price and value. So far, the services illustrated in earlier sections (e.g. Figure 6.1), are still early evolutions of 2G services. Ideal platforms for service differentiation do not yet exist. The implementation process for new services exploiting full 3G capabilities appears slow.

On the other hand, great challenges also remain for manufactures on the terminal side, to produce intelligent multi-functional terminals with efficient power consumption. This without even mentioning the amount of innovation required in the infrastructure side to maximize capacity and spectrum usage. Thus, why do we not invest more generously in technology, development/research, and the creation of applications than in spectrum license fees? It seems that this capital circulation would also create revenues for needy governments.

In conclusion, there exist expanding possibilities for service innovation, technology applications and research and development to solve challenging telecommunication demands.

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⁵ The increase in operators during the 3G licensing process in many countries will create higher competition yet.

7 DEPLOYING 3G NETWORKS

7.1 BACKGROUND

Logically deploying 3G networks implies dimensioning and implementing corresponding elements within a geographical area, where an operator would desire to offer advanced mobile communications services, e.g. voice, mobile Internet, video-telephony, etc.

In the preceding chapters we have outlined the service requirements and technical specifications of the UMTS solution. In this chapter we aim to describe the application of the proposed solutions and go through the process of designing a network to provide UMTS services.

Before describing the results of a field study with reference-parameters based on real scenarios, we provide the necessary principles for dimensioning and implementing a 3G network using UMTS technology. We then present results of dimensioning and introduce the functional capabilities of the selected elements.

7.2 NETWORK DIMENSIONING PRINCIPLES

Figure 7.1 identifies non-exhaustively the major areas to dimension a 3G network. It summarizes the essential tasks to obtain the necessary count of elements for implementation and network deployment.

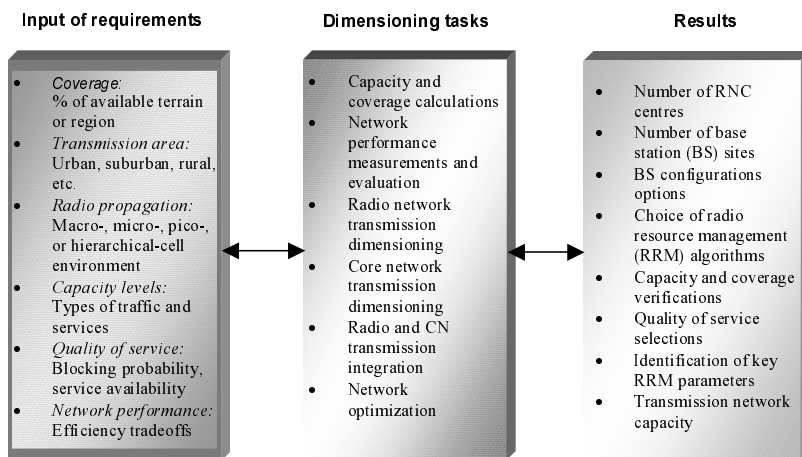


Figure 7.1 Essential network dimensioning tasks.

To simplify the whole process we group the dimensioning tasks into four key iterative actions, i.e.

- radio coverage and traffic flow identification;
- system dimensioning;
- network configuration and verification (i.e. radio, core, transmission);
- implementation and deployment.

In the first action, *radio coverage* depends on both propagation environment, (i.e. service population areas) and the traffic flow expected. Through a computerized process and classical optimization, the main output consists of the identification of sites for BS (or node B) location. The latter will depend on the projected service strategy and the BS range and capacity. The service strategy will take into account the traffic flow generated based on the subscriber profiles of service utilization levels and population densities. The radio coverage task will include or use the multi-path channel models, and reference service rates illustrated in Chapter 2.

System dimensioning involves the optimization of coverage and capacity based on macrocells and microcells in densely populated areas. It aims to take into account the asymmetry of traffic in the UL and DL and includes in the optimization the TDD mode to maximize capacity and flexibility in micro- and picocells.

Network configuration and verification consolidates the coverage and site location exercise by starting a process for the integrated solution of radio and core elements. Based on the capacity and service target requirements, the 3G system architecture is set for the node Bs and CS and PS elements in the core network side. It also looks at the impact on the transmission subsystem.

Implementation and deployment completes the 3G-network design process by realizing the projected site locations, service target requirements and time to service. It takes into account the solution adopted for the network deployment, e.g. sharing sites with existing 2G BSs and evolution of CN elements, or a complete new overlay network on the top of the existing 2G system. It may also apply to a totally green field network, i.e. a new deployment. It will also take into account the hierarchy of the network, i.e. the macro- and microlayers where applicable.

When deploying in the macrocell environment primarily with the FDD mode or WCDMA technology, the implementation will take into account the coverage dependency on the transmission rates and technology availability in terms of antenna configuration and interference minimizing features. Thus, the four actions or steps outlined above do have an iterative process.

7.2.1 Coverage and Capacity Trade-off in the FDD Mode

From the practical side as mentioned in earlier chapters and Section 7.4 of this chapter, in the FDD mode, which uses WCDMA techniques, the interference increases with the number of active users, thereby limiting capacity. Within this soft limitation, the system quality decreases continuously until service performance degrades to an intolerable state. This state leads to the *breathing cells* phenomenon, i.e. when user numbers gets too high, the quality of users at the *cell-edge* degrades rapidly to the point to drop the link or the call. Such event implies that cell radio coverage shrinks. On the other hand, when call drops occur, interference decreases for the remaining users and cell area cov-

erage grows again. This is what we call the trade off between capacity and coverage in the FDD mode.

Cell coverage and capacity thus depend on the received bit energy to total noise plus interference ratio $E_b/(N_0 + I_0)$ on each cell part for the DL and in the BS for the UL. This means that any parameter, which affects the signal level and/or the interference¹, or reduces the $E_b/(N_0 + I_0)$ requirements², has impact on cell coverage and capacity, as well as on the overall system.

7.2.1.1 Soft Handover and Orthogonality

We described soft handover in Chapter 4 from the design side; here we look at it from the performance and dimensioning side. In this context, a MS performs handover when the signal strength of a neighbouring cell exceeds the signal strength of the current cell with a given threshold. In *soft handover* position, a MS connects to more than one BS simultaneously. Thus, the FDD mode uses *soft handover*³ to *minimize interference* into neighbouring cells and thereby *improve performance* through macro diversity, i.e. we combine all the paths together to get a better signal quality. We also reduce power originating from two or more BSs to reach the same mobile's E_b/N_0 requirement while we combine the paths.

We separate the information signal of different users by assigning to each one a different broadband and time limited, user specific carrier signal derived from orthogonal code sequences (e.g. OVSF codes). When completely orthogonal⁴, we can perfectly separate synchronously transmitted and received signals. However, this does not occur in the UL for example, due to different propagation paths, i.e. different distances with different time delays. In the DL even if all signals originate from a single point and the parallel code channels can be synchronized there is still not perfect signal separation. As a result, we cannot maintain complete orthogonality due to multipath propagation, and we have to use orthogonality compensation factors as noted in Chapter 2.

7.3 PARAMETERS FOR MULTISERVICE TRAFFIC

While some earlier⁵ 2G mobile systems measure network quality mainly for one service, e.g. speech, UMTS has many different bearer services with varying quality requirements. We characterize these differing services by parameters such as the bit rate, the maximal delay, connection symmetry, and tolerable maximum BER. As result to accurately dimension or design a network for multiple services, we need to use different traffic models and settings. We have to plan the BS numbers to handle the expected service mix. The multiple set of services will have *different impact on capacity and coverage*. For example, user bit rate will have large impact on coverage as illustrated in

¹ Interference = intracell interference and intercell interference.

² Interference here implies intracell interference and intercell interference.

³ Softer handover is a soft handover between two sectors of a site.

⁴ Two function orthogonality, e.g. $g(t)$ and $s(t)$, occurs when their cross-correlation functions equal zero.

⁵ Today GSM evolved to a more than just speech network, it does also GPRS and HSCSD.

Figure 7.2. On the other hand, we can often adjust all services to the same cell range by individually adjusting the emitted power of each service.

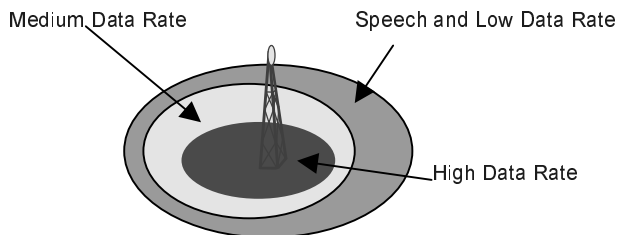


Figure 7.2 Transmission rates and coverage.

7.3.1 Circuit and Packet Switched Services

When dimensioning a 3G network in the FDD mode, e.g. the number of concurrent channels derived to cope with the different service requirements becomes the main input of the link budget analysis. Thus, if we have to manage traffic beyond a cell loading of 30%, any small load variation will have direct impact on the cell radius. We then have to achieve a dimensioning to *meet the peak traffic during the busy hour* in order to obtain a stable network. This stability will depend on how we treat the different types of service, i.e. Real Time (RT) or Circuit Switched and Non Real Time (NRT) or packet switched types.

7.3.1.1 Circuit Switched (CS)

To dimension capacity for CS services we can follow the classical approach, i.e. given the offered load (Erlangs) and the blocking rate, we derive from the traffic assumptions the offered traffic at the busy hour per cell (Erlang). Here we would assume the cell radius gets optimized iteratively with the link budget. Then, from Erlang B table we would determine the number of concurrent channels required during the busy hour for a given blocking rate.

Although the traditional solution may allow us to estimate CS capacity easily, it may also over dimension the required number of channels. Thus, it seems imperative that we use the multi-service Erlang B formulation and pool the resources for better availability on demand. This implies that we offer the CS channels depending on the required number, e.g. if one service requires 2 channels and the other 10, both can benefit from the pool, which may contain 20 channels. The latter would also imply that we could use different blocking rates for each service. For example, voice calls can tolerate degradation better than video calls.

7.3.1.2 Packet Switched Services

As in the CS, although with more sophistication, we also need to estimate the number of concurrent channels required for PS traffic. This number of channels will correspond to

the peak traffic during a Busy Hour (BH), which as in the CS, we determine also from the traffic assumptions of the offered load during the busy hour per cell expressed in kbits. In general, we treat each service independently to meet the different grade of service or asymmetry required.

We calculate the number of PS service channels by accounting a duration window corresponding to an acceptable delay (e.g. $d \approx 0.5\text{--}0.7$ s) for a given service. From the principles outlined in Chapter 2, we can illustrate the calculation for WWW application⁶ as follows.

We take 384 kps service with packet length $\zeta = 480$ bytes. From the total BH traffic for a given reference area we calculate the mean offered data rate m in kbps. Translating this into a mean packet arrival p rate, i.e. $p = (m \times d)/\zeta$. Then assuming a Poisson packet arrival distribution for all users, with a mean p , we obtain the probability density function (PDF), as well as the cumulative density function (CDF). Figure 7.3 illustrates the peak packet arrival rate h at 95% time probability [7].

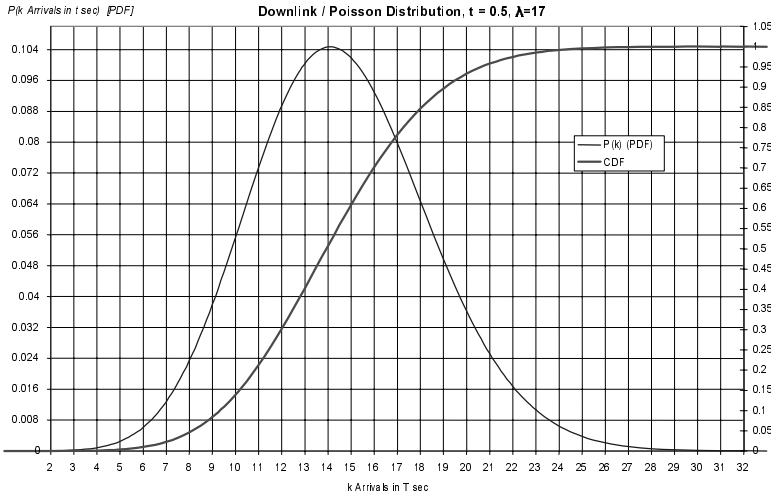


Figure 7.3 Peak arrival rate.

Utilizing the upper 95% time probability of the packet arrival rate (Figure 7.3) and applying the typical packet length we translated back into kbps. We then calculate the number of channels (ch) dividing by the service bearer rate r , i.e. $ch = h \text{ (kbps)}/r$. We can summarize the process as: $Chs = (1/\text{Serv Rate}) \times (1/\text{Serv Delay}) \times \text{CDFp}\{(m/\text{Serv delay} \times \zeta), 95\}$, where $\text{CDFp}(x, y)$ corresponds to the point of probability on the CDF associated with Poisson's law of mean x , and where m represents the mean offered data rate in kbps. We should note here that this process can be inefficient with low traffic in the cell, resulting in over-dimensioning for PS services. Thus, other types of distribution should also be considered.

⁶ For example e-commerce, on line banking, file transfer, information DB access, etc.

7.4 ESTABLISHING SERVICE MODELS

Before deploying new elements in a mobile telecommunications network, whether it is an existing system based on 2nd generation (2G) technology like GSM, or a new one like UMTS, we will need a projection for the potential number of subscribers. In this chapter, we consider a field study to extrapolate some subscriber numbers from two growth forecast⁷ assumptions. Although these projections will not necessarily apply to a particular deployment scenario, it will serve to illustrate network dimensioning based on the split of voice only and combined (voice + data) services.

In Table 7.1 we illustrate estimations for a 10-year period where 2G values correspond primarily to GSM voice services and 3G values to data starting with GPRS in the 1st 2 years. Thereafter, full multimedia services expand rapidly at the introduction of UMTS in existing GSM networks. A major breaking point occurs around 2005 with high predominance of 3G type services.

Table 7.1 Subscriber Growth Within a 10-year Period (in 1000s)

	Subscribers									
Year:	2000	2001	2002	2003	2004	2005	2006	2007	2008	2009
2G	750	1000	900	600	400	300	200	150	100	50
3G	0	0	300	700	1000	1200	1400	1500	1600	1700
Total	750	1000	1200	1300	1400	1500	1600	1650	1700	1750

In Table 7.2 we illustrate the subscriber growth beginning in 2002 when penetration of data has already reached about 30% of the total traffic. Here we assume that GPRS carrying wireless IP type services has grown to non-negligible levels right before the introduction of UMTS. Despite the stretch to a 15-year period, 2005 stands again as the breaking point towards full predominance of multimedia services. Nonetheless, as in the projections of the 10-year period, voice only services will remain a good 25% of all traffic.

Table 7.2 Subscriber Growth Within a 15-year Period (in 1000s)

	Subscribers									
Year:	2002	2003	2004	2005	2007	2009	2011	2013	2015	2017
Voice	900	600	400	300	150	100	50	0	0	0
Voice + data	300	700	1000	1200	1450	1550	1650	1750	1800	1850
Total	1200	1300	1400	1500	1600	1650	1700	1750	1800	1850

After 2005 in both cases the subscriber growth appears low. This can reflect the fact that the overall penetration of mobile services in the region begins to reach its limits or that the market share between operators starts to stabilize. Thus, for all practical purposes, in particular for the network dimensioning exercise in this field study, we consider primarily the data from 2002 to 2005 from Table 7.2.

⁷ The forecast has harmonized numbers, which do not apply to any operator or service provider in particular.

7.5 PROJECTING CAPACITY NEEDS

Based on the preceding section dimensioning in this field study would then begin for about 1.5 million subscribers all using either voice only or multimedia services. The proportion will depend on the business strategy and the type of service products offered. Business strategy will have a strong relationship with the market segment addressed and the penetration of the type of services proposed. If we take Switzerland, for example, penetration of mobile services will reach 60% in all segments by the time we complete this writing. Clearly, voice appears as the predominant service, although data through SMS and HSCSD and early GPRS may grow. This means that the market for multimedia services remains quite open even up to 100%. Thus, following a pragmatic approach, network dimensioning and capacity projections will imperatively be done for multimedia services addressing all segments.

Now for all practical purposes we identify three main segments, i.e. business, residential, and mass market (see Chapter 6). The traffic distribution among these segments will depend on the subscriber demand, operator's service⁸ offer, and qualitative thinking. Nevertheless, looking at the data in Table 7.1 and Table 7.2, about 70% of the market stands open for multimedia type services. If we distribute the latter as 40% mass market and 15% business and residential, respectively; then dimensioning should follow conventional wisdom.

Conventional wisdom may tell us that residential and business segments will tend to use larger transmission rates (e.g. 384 kbps) in suburban and urban areas, while mass-market subscribers will use medium rates services (144 kbps) from everywhere.

7.6 CELLULAR COVERAGE PLANNING ISSUES

Before discussing the fundamental parameters, assumptions and planning methodology, we select a region with a typical subscriber population and complex geographical area for cellular planning, e.g. mountainous landscape with large canyons and valleys, as well as hilly cities.

7.6.1 The Coverage Concept

As illustrated in Figure 7.4 the ideal UMTS coverage concerns all types of environments, i.e. in buildings (picocells), urban (microcells), suburban (macrocells), and global (global cells). However, at this time we cover mainly picocells to macrocells. While FDD coverage here may apply primarily⁹ to macrocells, the TDD solution applies more to pico- and microcells. Figure 7.5 shows an option for combining the UTRA technologies for maximum coverage.

⁸ The operator's initiative and creativity on new services offering product packages and a business approach will make a large difference. It will not depend only on Internet traffic.

⁹ The FDD also applies to microcells, and it is not only for use in picocells.

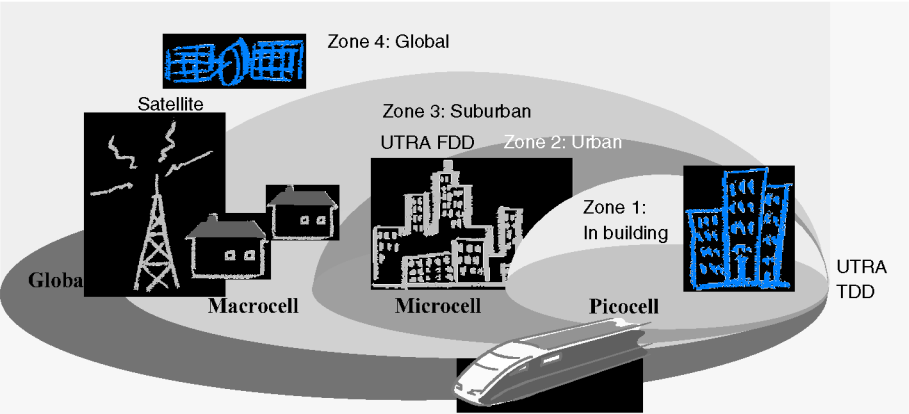


Figure 7.4 Service coverage summary.

Hot spots stand as the main target to complement FDD with TDD. This seems justified if we consider that intensive wideband demanding asymmetric services will take place to greater a extent in dense urban areas. Placing TDD over FDD will handle additional traffic, thereby adding capacity. The absence of TDD in dense areas would imply denser FDD site deployment or simply a microcell FDD implementation and/or additional carriers on the node B hardware.

As indicated in Chapter 6, the motivation behind the introduction of 3G networks lies in innovative applications; a 3G subscriber will not sign up just because of the existing UMTS network technology. At the end, service technology becomes transparent; if a user’s request for service is satisfied, few will notice the underlying technology. Now, most innovative applications tend towards transmission asymmetry (e.g. multimedia Internet). Thus, we can meet areas with small cell sizes and a high data asymmetry more appropriately¹⁰ with TDD than FDD. The lower efficiency of FDD results from larger number of soft handovers in small cells, while the dynamic DL/UL resource allocation makes TDD more efficient.

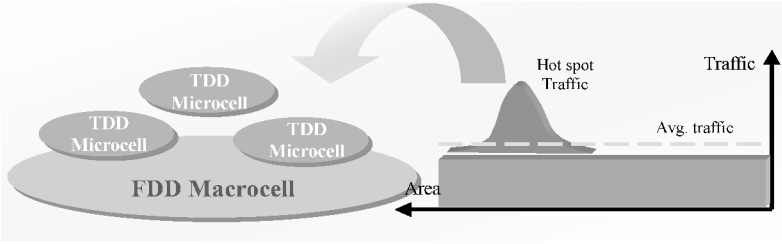


Figure 7.5 Combining FDD and TDD coverage.

Thus, it seems reasonable to think that we do not need to wait for TDD until we need overall network capacity; we can exploit it at the introduction of UMTS in hot spot areas. This means that 3G coverage planning can benefit from TDD from the start. In

¹⁰ FDD needs 2×5 Mhz instead of 1×5 Mhz as for TDD in small cells with asymmetric traffic.

this field study, we assume that TDD can apply to dense urban areas and concentrate on macrocell dimensioning for FDD or WCDMA.

7.6.2 Radio Network Parameter Assumptions

Figure 7.6 illustrates the coverage within a geographical area. Logically, an operator or service provider will aim to have 99% coverage for the populated area while maximizing the geographical coverage. On the other hand, the penetration of UMTS at the introduction will not necessarily include all populated¹¹ environments. Thus, starting in the main cities and suburban areas, 3G network coverage can progress in three phases, i.e. 50%, 75 (80)%, and 99%. For business strategic reasons within a region, e.g. it would be expedient to cover also major vacation centres even if these areas do not have permanent population, but transitory during a quarter of the year. Which means a sound business case for the introduction of UMTS would start with more than just 50% coverage of the populated area.

With the assumptions above, in the following we outline key issues when designing a macrocellular network based on the FDD mode or WCDMA.

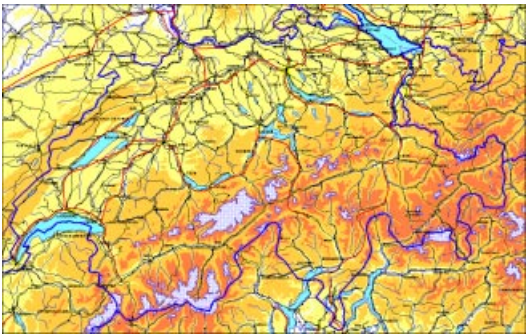


Figure 7.6 Population coverage example.

Figure 7.7 illustrates the conversion of population density to area coverage, where 50% of the population corresponds to about 10% of the coverage area. Thus, we can tailor coverage depending on strategy or demand once basic coverage has been achieved.

Table 7.3 illustrates the morphology distribution of the 50 and 75% population coverage. It indicates area coverage proportion in km² of the different service environments, i.e. dense urban (DU), urban (U), commercial/industrial (CI), suburban (SU), forest (FO), open (OP). It also indicates the service area proportions in % of the total area corresponding to the 50 or 75% population density. These proportions serve as the points of reference to establish the number of subscribers per service area and plan accordingly for the number of sites or cells required for each service environment. It will also allow estimation of RF unit number according to the number of sectors per site.

¹¹ Regulators in some countries are demanding only 50% initial coverage.

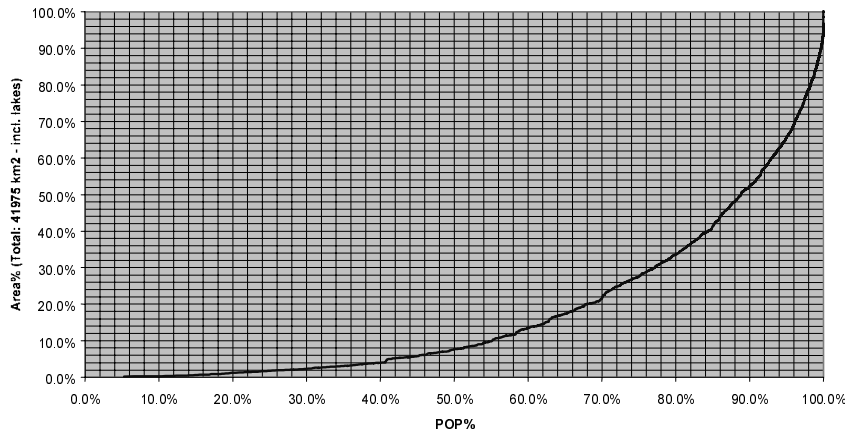


Figure 7.7 Population density conversion to area coverage.

Table 7.3 Morphology Distribution of the Population Density

Coverage area	50% POP	75% POP
Total size (km ²)	4067.00	6741.00
Morphology distribution (km ²)		
Dense urban	2.33	2.37
Urban	9.90	10.60
Commercial/industrial	101.00	138.00
Suburban	387.00	617.00
Forest	1270.00	1961.00
Open	2297.00	4012.00
Morphology distribution		
Dense urban (%)	0.06	0.04
Urban (%)	0.24	0.16
Commercial/industrial (%)	2.48	2.05
Suburban (%)	9.52	9.15
Forest (%)	31.23	29.09
Open (%)	56.48	59.52

Table 7.4 illustrates the service quality assumptions for projected radio bearer services in UMTS. The transmission rates or bearers corresponding to the service environments represent the most common services. On the other hand, we do not necessarily exclude speech, LCD 384, LCD 2048, and UDD 2048. For example, voice service may have the following assumptions: Adaptive Multi Rate (AMR) codec with a bit-rate of 12.2 kbits/s and with 50% voice activity factor. We can also assume 20 mE/subs with the following average holding times per subscriber:

- holding time of a mobile originated call 75 s
- holding time of a mobile terminated call 90 s

The traffic distribution is estimated:

- proportion of call attempts that is mobile originated and mobile terminated 0.60
0.40

Table 7.4 Service Quality Requirements

Area/bearer service	LCD 64	LCD 144	UDD 64	UDD 144	UDD 384
Dense urban	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%
Urban	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%
Commercial/industrial	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 95%	Indoor LCP 90%
Suburban	Indoor LCP 90%	Indoor LCP 90%	Indoor LCP 90%	Indoor LCP 90%	
Forest	In-car LCP 90%	In-car LCP 90%	In-car LCP 90%	In-car LCP 90%	
Open	In-car LCP 90%	In-car LCP 90%	In-car LCP 90%	In-car LCP 90%	

LCD 384 and LCD 2048 can be considered for indoor transmission with LCP 95%. The number of subscriber with these rates in each cell will not exceed a couple of users. The traffic data example illustrated in Table 7.5 shows a possible distribution of the different type of bearer services. Notice it does not include voice services.

Table 7.5 Traffic Data Example for 50 and 75% Population Coverage

	Area					
	DU	U	IND	SU	FO	OP
Active subscribers at 50% population coverage	6000	21000	80000	265000	70000	30800
Active subscribers at 75% population coverage	7000	22000	110000	350000	110000	401000
<i>Busy hour traffic/subscriber UL</i>						
Bearer UDD64 (kbit/s)	0.079	0.079	0.079	0.08	0.08	0.08
Bearer UDD144 (kbit/s)	0.060	0.060	0.060	0.07	0.07	0.07
Bearer UDD384 (kbit/s)	0.015	0.015	0.015			
Bearer LCD64 (mErI)	0.50	0.50	0.50	0.50	0.50	0.50
Bearer LCD144 (mErI)	0.25	0.25	0.25	0.25	0.25	0.25
<i>Busy hour traffic/subscriber DL</i>						
Bearer UDD64 (kbit/s)	0.120	0.120	0.120	0.15	0.15	0.15
Bearer UDD144 (kbit/s)	0.18	0.18	0.18	0.24	0.24	0.24
Bearer UDD384 (kbit/s)	0.08	0.08	0.08			
Bearer LCD64 (mErI)	0.50	0.50	0.50	0.50	0.50	0.50
Bearer LCD144 (mErI)	0.25	0.25	0.25	0.25	0.25	0.25

The traffic data, i.e. Unrestricted Delay Data (UDD) and Low delay Circuit Switch Data (LCD) for the different environments (Dense Urban (DU), Urban (U), Industrial (IND), Suburban (SU), Forest (FO), and Open (OP)), represent the possible traffic flow in the 3G network. We provide them here only as reference to make realistic projections. Notice that the traffic in the DL is higher than in the UL due to the fact the users download

more information than they upload. We can also see that a good part of the subscriber base remains in the open areas in this particular density distribution.

Consolidating 3G BS areas will vary from region to region. Some regions have already strict regulations for the implementation of sites as well as high costs in dense areas. This means that site acquisition will exceed the minimum requirements. Thus, Table 7.5 shows the necessary margins projected for subscriber growth assuming that sites can be available within a short term. The turnaround to prepare sites to increase coverage and capacity may not necessarily match a rapid subscriber growth. If we apply 50% of the population coverage to the 1st case and 75% to the 2nd case, we then have about 750K UMTS subscribers for the initial phase and about 1000K for the latter. This means we dimension the 3G network initially with enough margin for growth towards the latter phase where the subscriber base approaches the predicted numbers for 2005 in Table 7.1 when adding the 2G subscribers, i.e. $\approx 1500\text{K}$ subscribers.

7.6.3 Circuit Switched Data Calls Assumptions

From [1] for 64 kbps UDI we, assumed that 25% of the UMTS subscribers will also be CS data subscribers. We also assume that 50% of the calls will be UL + DL, 25% of the calls will be UL only and 25% of the calls will be DL only. This means, that one call will occupy two channels (one for DL and one for UL) but with a 75% usage each.

CS data users may use multimedia with the following traffic mix:

- 1 data call per 24 h, with a duration of 30 min. We assume that 50% of these calls occur during busy hour (BHCA=0.5); 3% of the CS data users use this service;
- 1 data call per 3 h, with a duration of 5 min. It is assumed that 67% of these calls are done during busy hour (BHCA=0.67); 6% of the CS data users use this service.

CS Data users may use other UDI services with the following traffic:

- 1 data call per 3 h, with a duration of 5 min. It is assumed that 67% of these calls are done during busy hour (BHCA=0.67); 3% of the CS data users use this service.

7.6.4 Packet Switched Applications

Packet data traffic will have different requirements on delays, packet loss, etc. The recommended classes include streaming, conversational, interactive and background. On this basis Table 7.6 illustrates the traffic mix of users and total traffic that may be applied.

Table 7.6 Packet Traffic Mix

Traffic classes	Scenario				
	% Users	% Traffic	Traffic BH (kbytes)		
			DL	UL	Total
Background	59	21	49	16	65
Interactive	156*	39	110	20	130
Streaming	4	18	50	10	60
Conversational	5	22	38	38	76
Total		100	247	84	331

* Note that each subscriber may use several applications.

7.6.5 Characteristic of CDMA Cells

The factors affecting CDMA cell size, capacity, and co-channel parameters in the forward and reverse links include same cell interference and other cell interference. These events also have impact on the link power budgets.

7.6.5.1 Theoretical Capacity

Here we look at capacity from the user interference side. To illustrate a basic case, we use the link reference parameter, i.e. E_b/N_0 , or energy per bit per noise power density, which later will apply to the link budget frame work.

Picking it up from equation (2.6), we consider the generic reverse-link capacity in CDMA¹² as the limiting factor. Thus, assuming perfect power control for this instance, the received powers from all mobiles users are the same. Then

$$\frac{S}{N} = \frac{1}{M-1} \quad (7.1)$$

where M is the total number of active users in a given band, and where the total interference power in the band equals the sum of powers of single users. Now equating the energy per bit to the average modulating signal power we defined

$$E_b = ST = \frac{S}{R}, \quad (7.2)$$

where S is the average modulating signal power, T is bit time duration, and R is the bit rate, i.e. $1/T$. Then, incorporating the noise power density N_0 , which is the total noise power N divided by the bandwidth B (i.e. $N_0 = N/B$), we get

$$\frac{E_b}{N_0} = \frac{S}{N} \frac{B}{R} = \frac{1}{(M-1)} \frac{B}{R}. \quad (7.3)$$

Solving for M yields

$$M-1 = \frac{(B/R)}{(E_b/N_0)} \text{ and for large } M \text{ we get } M = \frac{(B/R)}{(E/N_0)} = \frac{G_p}{(E/N_0)}, \quad (7.4)$$

where G_p corresponds to the system processing gain defined in equation (2.3), and M defines the number of projected users in a single CDMA cell with omnidirectional antenna without interference from neighbouring cells users transmitting continuously.

7.6.5.2 The Cell Loading Effect

Since in real 3G mobile networks there always exists more than one cell and more than one sector, we need to introduce a loading effect due to interference from neighbouring cells as follows:

¹² Mainly in rural areas; in urban area the downlink may/will become the limiting factor.

$$\frac{E_b}{N_o} = \frac{1}{(M-1)} \frac{B}{R} \left(\frac{1}{1+\beta} \right), \quad (7.5)$$

where β is the loading factor (ranging from 0 to 100%) as introduced in equation (2.29). Typical β values will range from 45 to 50%. The inverse of $(1 + \beta)$ has often been defined as the frequency re-use factor, i.e. $F = 1/(1 + \beta)$. The ideal single cell CDMA value of $F = 1$ (i.e. $\beta = 0$) decreases as the loading of multi-cell environments increase.

Sectorization can decrease interference from other users in other cells. Thus, instead of deploying only omnidirectional antennas with 360° a majority (if not) all sites can bear at least three sectors (e.g. 120°), and allow thereby the sectorized antenna to reject interference from users outside its antenna pattern. Such an event will decrease the loading effect and introduce a sectorization gain λ , which can be expressed as:

$$\lambda = \frac{\int_0^{2\pi} I(\theta) d\theta}{\int_0^{2\pi} \left(\frac{A_G(\theta)}{A_G(0)} \right) I(\theta) d\theta}, \quad (7.6)$$

where $A_G(0)$ is the peak antenna gain occurring generally at the bore sight (i.e. $\theta=0$), $A_G(\theta)$ is the horizontal antenna pattern of the sector antenna; $I(\theta)$ represents the received interference power from users of other cells as a function of θ . In practice $\lambda = 2.5$ for a three sector configuration and about 5 for a six sector one. Then, incorporating the sectorization gain in the loading effect, we get:

$$\frac{E_b}{N_o} = \frac{1}{(M-1)} \frac{B}{R} \left(\frac{1}{1+\beta} \right) \lambda. \quad (7.7)$$

Initially for the single cell case, we have assumed continuous transmission. However, this does not occur for voice and some multimedia services; although it does for data. Thus, we will now introduce an activity factor $1/v$ to reflect this event in the UL loading effect. Then, we get

$$\frac{E_b}{N_o} = \frac{1}{(M-1)} \frac{B}{R} \left(\frac{1}{1+\beta} \right) \lambda \left(\frac{1}{v} \right), \quad (7.8)$$

where v may range from 40 to 50% for voice and 1% for data. Therefore, the value of v reduces the overall interference of the UL loading effect equation.

For the downlink (DL) we need an additional parameter ε to reflect the orthogonality of the transmission. Thus empirically, we can express it as:

$$\frac{E_b}{N_o} = \frac{1}{(M-1)} \frac{B}{R} \left(\frac{1}{(1-\varepsilon)+\beta} \right) \lambda \left(\frac{1}{v} \right).$$

7.6.6 Link Budgets

A link budget aims to provide the steps to calculate the ratio of the received bit energy to thermal noise (i.e. E_b/N_o) and the interference density I_o . It considers transmit power, transmit and receive antenna gains, channel capacity factors, propagation environment, and receiver noise figure.

Based on the channel models introduced in Chapter 2 we present the background for link budgets. Following the guidelines from Ref. [3] the formulation assumes that path loss formulas help to determine the maximum range and the coverage area. We also assume that in the case of hexagonal deployment of sectorized cells, the area covered by one sector is:

$$S = 3\sqrt{3} \left(\frac{R}{2} \right)^2 / 2, \quad (7.9)$$

where R is the range obtained in the link budget. This implies that we use hexagonal sectors with base stations placed in the corners of the hexagons. Coverage analysis can thus apply to tri-sectorized antennas for macrocells and with omnidirectional antennas for microcell and picocell coverage.

Before describing the actual reference parameters for the link budgets in Table 7.7, in the following we provide a generic background of the analysis steps for the forward and reverse links.

7.6.6.1 The Forward Link

Applying the logic for the traffic channels analysis in Ref. [2], to the Dedicated Physical Control Channel (DPCCH) and Dedicated Physical Data Channel (DPDCH) we can formulate a generic E_b/N_o for the forward link in a multi-cellular environment.

Starting from a single cell with a single mobile station (MS),

$$\frac{E_b}{N_o} = \frac{P_o L_o A_G}{I_b + I_n + N} G, \quad (7.10)$$

where P_o is the BS sector traffic channel ERP in the direction of the MS within a given antenna pattern with its angle θ_o , L_o equal to the path loss from the home BS in the direction of θ_o within a given distance, A_G is the receive antenna gain the MS, I_n is equals to the interference power received at the MS from non-CDMA origins, N is the thermal noise power, G is the processing gain, and I_b can be defined as:

$$I_b = (1 - \varepsilon) P' L_o A_G, \quad (7.11)$$

where ε is the orthogonality factor, P' is the home BS excess ERP (e.g. paging, sync powers, etc.) in the direction of the MS under consideration.

In the presence of many cells and single MS, interference originates from the powers of the surrounding BSs, in addition to the excess powers of its own cell. Thus, we introduce the interference from the surrounding as I_o :

$$\frac{E_b}{N_o} = \frac{P_o L_o A_G}{I_b + I_n + I_o + N} G.$$

When looking at a single cell with many MSs, the BS serves all MSs plus the MS under consideration. Therefore, the latter gets the interference from the DL powers aimed at the other MSs. We denote this additional interference as I_m :

$$I_m = (1 - \varepsilon) A_G L_o \sum_{i=1}^I P_i, \quad (7.12)$$

where again ε is the orthogonality factor and P_i is the forward traffic channel ERP aimed for MS i , but radiated to the desired MS measuring E_b/N_o . P_i may also denote the traffic channel ERP aimed for MS i but captured by the desired MS. Then

$$\frac{E_b}{N_o} = \frac{P_o L_o A_G}{I_b + I_n + I_m + N} G. \quad (7.13)$$

When a MS measuring E_b/N_o finds itself among many other MSs and many other cells, there is an additional interference term I_t , i.e. the total traffic channel power received from all other BSs. It can be defined as:

$$I_t = A_G \sum_{k=1}^K P_k L_k, \quad (7.14)$$

where P_k is the total traffic channel ERP from BS k . Thus, I_t represents the sum of all traffic channel powers received by the desired MS from all other BS, but excluding its own. K is the total number of cells or sectors in the system under consideration. We can define P_k as

$$P_k = \sum_{j=1}^{J_k} P'_j. \quad (7.15)$$

The P_k expression indicates that, for each BS k , we sum the forward traffic channel ERPs for all MSs corresponding to that BS k .

The expression also implies that P'_j is the traffic channel power aimed to MS j but captured by the MS calculating E_b/N_o . J_k is the total number of MS served by BS k .

Then E_b/N_o for the MS among many MS within many cells can be defined as:

$$\frac{E_b}{N_o} = \frac{P_o L_o A_G}{I_b + I_n + I_o + I_m + I_t + N} G. \quad (7.16)$$

This latter expression will be the most likely environment when calculating the forward link budget.

7.6.6.2 The Reverse Link

In the reverse link or uplink, i.e. MS to BS connection, a single cell serving a single MS has the following E_b/N_o expression:

$$\frac{E_b}{N_o} = \frac{P_R L_R A_{GR}}{I_{nR} + N} G, \quad (7.17)$$

where P_R is the reverse traffic channel ERP of the desired MS assuming an omnidirectional transmit pattern, L_R is the reverse path loss from the desired MS in the direction of θ_o to the home BS at given distance, A_{GR} is the receive antenna gain of the home BS in the direction of θ_o to the desired MS, I_{nR} is the power received at the home BS from other interference from non-CDMA sources.

When considering a single cell with many mobiles one BS serves many MSs, and the MS measuring E_b/N_o gets extra interference (I_{mR}), which can be expressed as:

$$I_m' = \sum_{j=1}^J P_{Rj} L_{Rj} A_{GR} \quad (7.18)$$

where P_{Rj} corresponds to the reverse traffic channel ERP of MS j , L_{Rj} is the reverse path loss from MS j in the direction of θ_j back to the home BS at given distance, A_{GR} is the receiver antenna gain of the home BS in the direction of θ_j to MS j . Thus, I_{mR} represents the total reverse link interference generated by MS served by home BS. P_{Rj} dynamically changes based on the power control algorithm. Then, the reverse link E_b/N_o for a single cell with many MS is:

$$\frac{E_b}{N_o} = \frac{P_R L_R A_{GR}}{I_{nR} + I_{mR} + N} G \quad (7.19)$$

In scenarios involving many MSs and multiple cells, the MS measuring E_b/N_o gets additional interference from MSs served by BSs from neighbouring cells. We can express this interference as:

$$I_{tR} = \sum_{k=1}^K P_{Rk} \quad \text{with} \quad P_{Rk} = \sum_{j=1}^{J_k} P_{Rk,j} L_{Rk,j} A_{GR}, \quad (7.20)$$

where I_{tR} is the total interference from the reverse link generated by MSs served by other BSs other than the home BS of the MS measuring E_b/N_o , P_{Rk} is the total reverse link traffic power received from MSs served by BS k , K is the total number of BSs excluding the home BS of the concerned MS.

We get P_{Rk} by adding the powers of the traffic channels from MSs served by BS k , where for this BS $P_{Rk,j}$ is the reverse traffic channel ERP of MS j ; Likewise for BS k , $L_{Rk,j}$ is the reverse path loss from MS j in the direction of $\theta_{Rk,j}$ at a given distance. A_{GR} is the receiver antenna gain of the home BS in the direction of $\theta_{Rk,j}$ to MS j served by BS k . Then

$$\frac{E_b}{N_o} = \frac{P_R I_R A_{GR}}{I_{nR} + I_{mR} + I_{tR} N} G. \quad (7.21)$$

The sum of the interfering elements divided by the thermal noise power N gives origin to the reverse link factor. This factor η_r represents the rise of the interference level above of the thermal noise level, we can define it as:

$$\eta_r = \frac{I_{nR} + I_{mR} + I_{tR}}{N}. \quad (7.22)$$

Through the value of η_r we can determine the BS loading level. Thus, higher η_r values indicate that the BS can no longer support additional users or MSs.

With generic analytical background of the preceding sections, i.e. the forward and reverse link estimation for E_b/N_o , in the following we outline the main elements for link budgets.

7.6.6.3 Link Budget Elements

Table 7.7 illustrates reference elements typically utilized in the calculations of link budgets. The template after Ref. [4] applies to both forward and reverse links unless specifically stated otherwise. In the forward link the BS acts as the transmitter and the MS as the receiver. In the reverse link the MS acts as the transmitter and the BS as the receiver. For completeness the elements are redefined as follows:

- (a₀) *Average Transmitter Power Per Traffic Channel (dBm)* → the mean of the total transmitted power over an entire transmission cycle with maximum transmitted power when transmitting.
- (a₁) *Maximum Transmitter Power Per Traffic Channel (dBm)* → the total power at the transmitter output for a single traffic¹³ channel.
- (a₂) *Maximum Total Transmitter Power (dBm)* → the aggregate maximum transmit power of all channels.
- (b) *Cable, Connector, and Combiner Losses (Transmitter) (dB)* → the combined losses of all transmission system components between the transmitter output and the antenna input (all losses in + dB values).
- (c) *Transmitter Antenna Gain (dBi)* → the maximum gain of the transmitter antenna in the horizontal plane (specified as dB relative to an isotropic radiator).
- (d₁) *Transmitter e.i.r.p. Per Traffic Channel (dBm)* → the sum of the transmitter power output per traffic channel (dBm), transmission system losses (-dB), and the transmitter antenna gain (dBi) in the direction of maximum radiation.
- (d₂) *Transmitter e.i.r.p. (dBm)* → the sum of the total transmitter power (dBm), transmission system losses (-dB), and the transmitter antenna gain (dBi).
- (e) *Receiver Antenna Gain (dBi)* → the maximum gain of the receiver antenna in the horizontal plane; it is specified in dB relative to an isotropic radiator.

¹³ We define a traffic channel as a communication path between a MS and a BS used for information transfer and signalling traffic. Thus, traffic channel implies a forward traffic channel and reverse traffic channel pair.

- (f) *Cable, Connector, and Splitter Losses (Receiver) (dB)* → includes the combined losses of all transmission system components between the receiving antenna output and the receiver input (all losses in + dB values).
- (g) *Receiver Noise Figure (dB)* → the noise figure of the receiving system referenced to the receiver input.
- (h), (H) *Thermal Noise Density, N_0 (dBm/Hz)* → the noise power per Hertz at the receiver input. Note that (h) is logarithmic units and (H) is linear units.
- (i), (I) *Receiver Interference Density (I_0 (dBm/Hz))* → the interference power per Hertz at the receiver front end. This corresponds to the in-band interference power divided by the system bandwidth. Note that (i) is logarithmic units and (I) is linear units. Receiver interference density I_0 for a forward link is the interference power per Hertz at the MS receiver located at the edge of coverage, in an interior cell.
- (j) *Total Effective Noise Plus Interference Density (dBm/Hz)* → the logarithmic sum of the receiver noise density and the receiver noise figure and the arithmetic sum with the receiver interference density.
- (k) *Information Rate ($10\log(R_b)$) (dBHz)* → the channel bit rate in (dBHz); the choice of R_b must be consistent with the E_b assumptions.
- (l) *Required $E_b/(N_0 + I_0)$ (dB)* → the ratio between the received energy per information bit to the total effective noise and interference power density needed to satisfy quality objectives.
- (m) *Receiver Sensitivity ($j+k+l$) (dBm)* → the signal level needed at the receiver input that just satisfies the required $E_b/(N_0 + I_0)$.
- (n) *Hand-off Gain/Loss (dB)* → the gain/loss factor ($\pm$) brought by hand-off to maintain specified reliability at the boundary.
- (o) *Explicit Diversity Gain (dB)* → the effective gain achieved using diversity techniques. If the diversity gain has been included in the $E_b/(N_0 + I_0)$ specification, it should not be included here.
- (o') *Other Gain (dB)* → additional gains, e.g. Space Diversity Multiple Access (SDMA) may provide an excess antenna gain.
- (p) *Log-Normal Fade Margin (dB)* → defined at the cell boundary for isolated cells corresponds to the margin required to provide a specified coverage availability over the individual cells.
- (q) *Maximum Path Loss (dB)* → the maximum loss that permits minimum SRTT performance at the cell boundary. Maximum path loss = $d1 - m + (e-f) + o + o' + n - p$.
- (r) *Maximum Range, $R_{\max}$ (km)* → computed for each deployment scenario it is given by the range associated with the maximum path loss (see Chapter 2 for details).

Table 7.7 Link Budget Reference Template

Elements	Forward link	Reverse link
Reference: environment, services, multi-path channels	See Section 2.2	See Section 2.2
(a ₀) Average transmitter power per traffic channel	dBm	dBm
(a ₁) Maximum transmitter power per traffic channel	dBm	dBm
(a ₂) Maximum total transmitter power	dBm	dBm
(b) Cable, connector, and combiner losses, etc.	2 dB	0 dB

(c) Transmitter Antenna gain (e.g. 18 dBi vehicular., 10 dBi pedestrian., 2 dBi indoor)	Will vary	0 dBi
(d ₁) Transmitter e.i.r.p. per traffic channel = (a ₁ -b+c)	dBm	dBm
(d ₂) Total transmitter e.i.r.p. = (a ₂ -b+c)	dBm	dBm
(e) Receiver antenna gain (e.g. 18 dBi vehicular., 10 dBi pedestrian., 2 dBi indoor)	0 dBi	Will vary
(f) Cable and connector losses	0 dB	2 dB
(g) Receiver noise figure	5 dB	5 dB
(h) Thermal noise density (H) (linear units)	-174 dBm/Hz 3.98×10^{-18} mW/Hz	-174 dBm/Hz 3.98×10^{-18} mW/Hz
(i) Receiver interference density (I) (linear units)	dBm/Hz mW/Hz	dBm/Hz mW/Hz
(j) Total effective noise plus interference density $= 10 \log (10^{((g+h)/10)} + I)$	dBm/Hz	dBm/Hz
(k) Information rate ($10 \log (R_b)$)	dBHz	dBHz
(l) Required $E_b/(N_o + I_o)$	dB	dB
(m) Receiver sensitivity = (j + k + l)		
(n) Hand-off gain	dB	dB
(o) Explicit diversity gain	dB	dB
(o') Other gain	dB	dB
(p) Log-normal fade margin	dB	dB
(q) Maximum path loss= { d ₁ -m+(e-f)+o+n+o'-p }	dB	dB
(r) Maximum range	m	m

7.6.6.4 Link Budget for Multi-Services

Here we consider how the environment of WCDMA in the FDD mode will influence multi-service provision. In multi-service link budget, the analysis process to calculate the *interference degradation* or the *loading factor* takes into account the interference contribution of all the users with their different services. This results in a common link budget, which aims to provide the same cell radius for all the service by trying to match all the acting UE TX powers. It also aims to balance the two links (i.e. UL and DL) without any a priori knowledge of the limiting link in terms of coverage. This process permits us to estimate the actual system interference degradation without dependency on margins, which may lead to over-dimensioning.

7.6.7 Coverage Analysis

After providing the background to calculate the E_b/N_o values and the link budget in the last two sections, we now look at the practical design factors having impact on coverage.

Coverage may not be an issue at the introduction of UMTS in some regions, because the requirements will be gradual. However, from the service side, to back a pragmatic business case, a network will most likely start with about 50% coverage of populated areas as mentioned at the beginning of this chapter. Thus, such coverage will depend to a good degree on service strategy. From the network design side, this implies that good indoor coverage for high rate services will require dense sites in the urban areas with

downlink limitation and less dense in rural areas with uplink limitation. The latter implies that coverage and capacity trade-off will go hand in hand even at the beginning of UMTS service. Here we are mainly concerned with coverage.

7.6.7.1 Uplink (UL) and Downlink (DL) Coverage

DL coverage depends primarily on the load because the transmission power may remain the same despite the number of MSs active in a given BS, where all share the same power. This means that DL coverage will decrease as a function of the number of MSs and their transmission rates. The latter implies that additional power will afford better coverage for higher rates in the DL.

In WCDMA higher transmission rates imply more spreading, which results in lower processing gain, thereby smaller coverage. On the other hand, higher bit rates (demanding more transmission power), require lower E_b/N_o because the extra power allows better channel estimation, thereby compensating for larger ¹⁴coverage. In relation to the physical channels, i.e. DPCCH/DPDCH, the dependency of the bit rate for E_b/N_o has to do with the mode of channel operation. Figure 7.8 shows that there is a difference in the power utilization for each channel; it is also an overhead difference depending on the transmission rates. When assuming the same E_b/N_o for all rates, e.g. the overhead for 384 kbps does not exceed 6% of the total power in the DPCCH if we define DPCCH overhead as $10\log_{10}(1+10^{(DPDCH - DPCCH)/10})$.

Thus, when looking at the power differences for the reference service rates, logically we can conclude that to support 384 kbps we will need a denser site deployment than we would for 144 kbps.

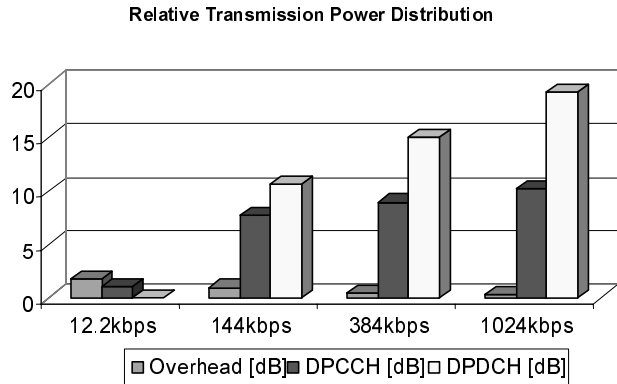


Figure 7.8 DPCCH/DPDCH and overhead power distribution.

¹⁴ In particular this applies to higher rates in data packet transmission.

Other factors having impact on the uplink E_b/N_o values are: multi-path diversity, macro-diversity gain, advanced BS signal processing techniques, and receiver antenna diversity.

In the first case, when looking at characteristics of the reference multi-path channels in Chapter 2, we see that the vehicular channels have more taps than those for the pedestrian ones. More taps implies higher multi-path diversity gain and thereby larger coverage.

In the second case, in the absence of high multi-path diversity gain during soft hand-over, i.e. when the MS receives a signal from at least two BSs, the probability of accurate signal detection increases resulting in higher micro-diversity gain.

Better baseband processing, e.g. adaptive filters for fading environments will improve error rates and thereby lower E_b/N_o values, which in turn will increase coverage.

Finally, through antenna diversity techniques we can also get a coverage gain of 2–3 dB. For example, transmit diversity can use two independent transmit paths from the base station to the mobile, in order to mitigate the effect of fading. The two paths may come from using two spatially separated antennas, or by using the two orthogonal polarizations of one cross-polarised antenna [5,6]. On the uplink, two-branch diversity combining or Maximal Ratio Combining (MRC) is optimal when the traffic consists of voice users only. However, when individual high data rate users are also present a fully adaptive two branch Minimum Mean Squared Estimate (MMSE) algorithm will provide improved performance by cancelling the interference due to these users. This cancellation results in a gain in the order of 1.5 dB.

As mentioned earlier, in the DL we can add power gradually when necessary, thereby increasing coverage for higher rates. However, this may not be the case in the UL because the MS has limited power. For example, a handset with an average power capacity of 21 dBm will have a maximum of 26 or 27 dBm power; the latter if we assume the MS gains 5–6 dBm at the BS due to the high reception sensitivity, antenna diversity and lower noise figures. High rate data terminals¹⁵ or data terminals in general will have 3 dB lower E_b/N_o . Thus, DL coverage for high rates will depend on the DL power amplifier rating, the UL cell dimensioning, and most likely the adjacent cell loading as noted in the preceding section.

7.6.8 Capacity Analysis

In WCDMA, capacity impacts apply to the DL and UL. In the 1st case it has to do with dense areas for high rates as well as subscriber number. In the 2nd case it has to do with rural areas in the context of coverage for high rates. On the other hand, due to the asymmetry of traffic flow, we expect more download information than upload. Hence, DL capacity appears more critical at least at the beginning of UMTS.

Orthogonal codes make the DL more robust against intra-cell interference. However, inter-cell interference does still affect DL capacity, which depends on the load of the

¹⁵ Speech terminals have about 3 dB body loss.

neighbouring cells and the propagation environment. For example, short orthogonal codes are more vulnerable to multi-path channels than single path channels; hence, in the microcell environment orthogonality gets preserved better than it does in the macro-cell environment. Consequently, loading despite the E_b/N_o values on adjacent macro-cells should not exceed 75% in the DL and about 55% in the UL. On the other hand, microcells can probably take 65% UL and 85% loading, respectively. This means we need to apply the appropriate orthogonality factors when utilizing the load equations described generically in Section 7.6.5.

The number of orthogonal codes also has impact on DL capacity despite a good propagation environment and good load sharing. The maximum number of orthogonal codes depends on the Spreading Factor (SF). For example, in general only one scrambling code and thus only one code tree gets used per sector in the BS, where common and dedicated channels share the same tree. On the other hand, the number of orthogonal codes does not imply complete¹⁶ limitation when enabling DL capacity, because we can apply a 2nd scrambling code. However, the 1st and 2nd codes will not remain orthogonal to one another, and channels with the 2nd code interfere more with the channels with the 1st code.

7.7 DIMENSIONING RNC INTERFACES

When dimensioning the RNC Iub interface, i.e. the connection between the Node B and RNC, we also consider the traffic mix in order to determine the number of RNCs required. Thus, RNC interface dimensioning will take into account the number of Node Bs and the projected type of services with the forecasted subscribers and their traffic profiles [7].

Figure 7.9 illustrates the UTRAN interface configuration.

7.7.1 Dimensioning the Iub

The average traffic per Node B provides the *total traffic* based on the service mix statistics, the soft handover traffic and overheads, signaling and O&M traffic.

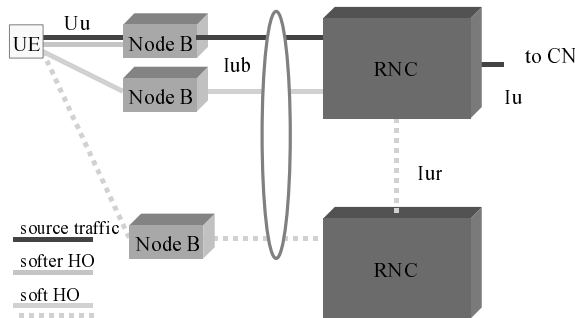


Figure 7.9 The UTRAN interface configuration.

Thus, to determine the total traffic passing through the Iub we consider first, the *peak* aggregate traffic mix calculated analytically taking into account the service parameters, e.g. the number of subscribers (S_i), subscriber bit rate (R_i), session time length (t_i), session inter-arrival time length ($1/\lambda_i$), activity factor α_i , plus signalling overheads and O&M margins. Here we assume that the ratio *peak traffic over average traffic* corresponds to the burstiness factor β .

We then calculate the overall $PDF(R_a)$ and $CDF(R_a)$, where R_a corresponds to the aggregate bit rate to determine the outage probability for each value of the user bit rate. Afterwards we obtain a set of outage probabilities, which corresponds one to each user bit rate R_b . At the end we dimension the channel capacity by fixing a common outage probability value P_0 for each service i .

7.7.1.1 Iub Total Traffic

As indicated in the preceding section, after we calculate the peak traffic per Node B, we take into account additional overheads and signaling loads. Thus, we obtain the *total traffic* at the Iub interface from the user information traffic, soft handover traffic, burstiness factor and overheads as well as signaling margins. Typical assumptions for the margins include: O&M = 10%, signaling = 20%, and ATM overhead = 40% of the Iub peak user traffic, respectively. In summary, we can define the Total Iub traffic as:

Total Iub traffic = peak traffic + O&M + signaling + overhead

or

Total Iub traffic = average traffic $\times \beta \times (1 + 0.4 + 0.2 + 0.1)$

7.7.2 RNC Capacity

To practically determine the number of RNCs required for a network, we generally consider the: maximum number of Node-Bs to be managed; the maximum Iub, Iu and Iur connections supported; and the maximum throughput of both CS and PS services.

We determine the total number of Node Bs based on the type of services offered, the number of subscribers projected and area of coverage desired among other parameters. We express the average traffic forecasted in Erlangs for CS and Mbps for PS.

Plotting nominal values of PS vs. CS traffic, e.g. 64 kbps for both services we can see in Figure 7.10 that the proportion of PS and CS traffic depends on the desired load for either service. In any case, it seems that we cannot have half and half of each service type.

¹⁶ Often referred as hard-blocking.

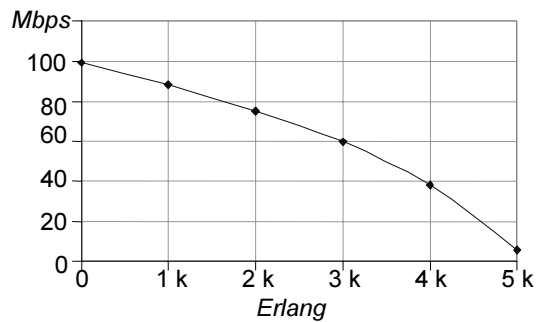


Figure 7.10 Nominal RNC traffic loads in Mbps vs. Erlang (000s).

The Iub, Iu, and Iur interfaces will in general support sufficient capacity margins, and the overheads will not exceed peak rates. Thus, the key RNC dimensioning parameters include the number of Node Bs in the coverage area, and the average traffic in this given area. The first parameter gives the:

$$RNC_{No\_node\_B} = \text{Total no. node Bs/no. Node Bs supported per RNC},$$

and the second one allows us to calculate throughput capacity, i.e. $RNC_{throughput} = \max\{\lceil CS_{avg}/X_1 \rceil, \lceil PS_{avg}/Y_1 \rceil\}$. We can obtain the initial value of $RNC_{throughput}$ from the CS and PS average traffic uniformly distributed in the target area.

We can modify the PS (Mbps) vs. CS (Erlang) output of Figure 7.10 to a PS vs. CS (Mbps) output by translating the CS traffic from Erlang to Mbps (i.e. Erlang $\times$ 12.2 kbps AMR voice codec). Then we can illustrate the $RNC_{throughput}$ in terms of average value of the CS and PS traffic in Mbps as shown in Figure 7.11. However, the average traffic will not take into account the traffic burstiness. Thus, peak values should be projected iteratively using the Gaussian Law. In general, the peak values will mean that the proportion of PS traffic will increase.

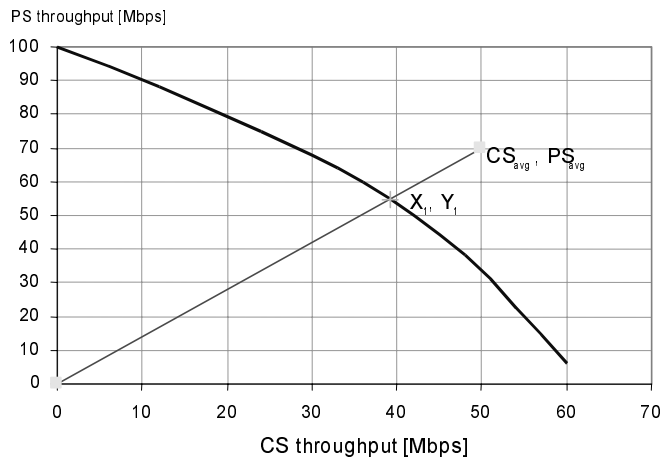


Figure 7.11 Estimation of the RNC throughput.

7.8 RADIO NETWORK DIMENSIONING FIELD STUDY

The analysis assumptions for the projected subscriber growth and traffic flow illustrated in Table 7.5 take the values shown in Table 7.8. These assumptions provide the generic set of information to calculate the number of radio network elements required for the coverage mentioned in Section 7.5.

Notice for example that we set the blocking characteristics to 1%; however, it is often set to 2%. The design assumes UL limitation by setting the load to 50%. With optimized interference techniques, UL load can reach up to 65%. Antenna heights for the MS can vary from 1.5 to 1.7 m and for the BS from 27 to 30 m.

Table 7.8 Analysis Assumptions for Lower-Bound Traffic Flow

	DU	U	IND	SU	FO	OP
Multi-path channels	Reference clutter distributions					
Capacity growth	Based on Table 7.5					
Loading	50% UL load					
Features	Simplified, excludes e.g. MHA					
Blocking on air interface	1% (network quality)					
Average antenna heights	MS 1.55 m, BS 27.5 m					
MS noise figure	8 dB					
MS output power	+24 dBm for data speeds ≥ 64 kbps					
MS antenna gain	2 dB data speeds ≥ 64 kbps					
Body loss	0 dB					
BS Noise figure	5 dB					
BS Antenna gain	18 dBi					
BS output power	43 dBm ≈ 20 Watts					
Soft-handover	40%					

7.8.1 Lower Bound Results

As noted in Section 7.6.2, the reference values illustrated in Table 7.5 represent the parameters to obtain projection for lower bound dimensioning, i.e. the minimum number of elements to meet coverage and traffic requirements. Thus, Table 7.9 shows the results taking into account the morphology distribution of the population density in Table 7.3, the service quality requirements in Table 7.4, and the minimum traffic flow in Table 7.5.

Table 7.9 Lower Bound Results for 50% Coverage

Planned population coverage	50%					
Area type	DU	U	SU	IND	FO	OP
Subscribers	6000	21000	265000	80000	70000	308000
Area size (km ²)	2.33	9.9	387	101	1270	2297
Node B 1+1+1	66	70	191	60	103	111
Sites total	601					
Cells total	1803					
RF units total	1803					
Dominating service class	PS384	PS384	PS144	PS144	PS144	PS144
Cell range (km)	0.155	0.285	1.065	0.870	2.355	3.250

Table 7.10 Lower Bound Results for 75% Coverage

Planned population coverage	75%					
Area type	DU	U	SU	IND	FO	OP
Subscribers	7000	22000	350000	110000	110000	401000
Area size (km ²)	2.37	10.6	617	138	1961	4012
Node B 1+1+1	78	79	306	102	201	173
Sites total	939					
Cells total	2817					
RF units total	2817					
Pre-dominant service class	PS384	PS384	PS144	PS144	PS144	PS144
Cell range (km)	0.155	0.285	1.065	0.870	2.355	3.250

We can observe that the node B configuration in Table 7.9 and Table 7.10 have only one carrier per sector. In practice, in DU areas we would most likely need two carriers per sector in some cities. However, because the results project UL limited design, it does not take into account the DL limitation in dense areas. In Table 7.9 and Table 7.10 we can also see that the predominant services correspond to those with high rates, e.g. 384 kbps in the PS mode within the DU and U areas. As mentioned in Section 7.6.8, the cell range decreases in areas where high transmission rates become predominant, e.g. DU and U.

7.8.2 Upper Bound Results

Realizing a radio network for the minimum traffic and for the minimum subscriber growth in the immediate term will minimize investment costs. However, it may limit the flexibility in service strategy if rapid increase in subscribers and traffic occurs. Hence, in the following we assume upper limit bounds and project the network based on higher traffic profiles. *This does not imply that deployed networks will be operating with the illustrated upper limits, specially at the introduction of UMTS.*

To simplify the propagation environments we group the six areas into three as follows: urban (DU + U); suburban (SU + IND), and rural (OP + FO). We increase traffic profiles at the busy hour (BH) for every service and harmonize the subscriber base for the three areas as well. Clearly, we keep the assumptions on traffic patterns for the DL and UL, i.e. we assume higher volumes in the DL than in the UL. Such asymmetric traffic will allow us to exploit the benefits of the TDD technology in the dense areas by incorporating micro- and picocells already at the introduction of UMTS, the latter assuming the technology has sufficient commercial maturity. Table 7.11 illustrates the consolidated traffic for the three aforementioned propagation environments. In addition, it also shows the total traffic generated per area and the overall traffic proposed for the total number of subscribers proposed and referenced. The activity factor for LCD 12.2 (e.g. voice with AMR codec) is 0.5; for the other LCD and UDD services it is 1.0.

Table 7.11 Upper Bound Traffic Flow with Consolidated Areas

75% population coverage reference projection 2005									
Areas	Urban (DU + U)			Suburban (SU + IND)			Rural (OP + FO)		
Subs no.	127000			663000			210000		
Bearer (kbps)	kbps		mErl	kbps		mErl	kbps		mErl
	UL	DL	UL/DL	UL	DL	UL/DL	UL	DL	UL/DL
UDD64	0.388	0.578		0.388	0.578		0.388	0.578	
UDD144	0.306	0.901		0.306	0.901		0.306	0.901	
UDD384	0.066	0.415		0.066	0.415		0.066	0.415	
LCD12.2	0.122	0.122	20.0	0.122	0.122	20.0	0.122	0.122	20.0
LCD64	0.154	0.154	2.41	0.154	0.154	2.41	0.154	0.154	2.41
LCD144	0.174	0.174	1.21	0.174	0.174	1.21	0.174	0.174	1.21
Total traffic	153731	297749	47.24	802548	1554390	4724	254201	492341	4724
Total traffic per area	451480			2356938			746542		
Total traffic from all areas (kbps) per busy hour (BH) 3,554,960									

The upper bound results illustrated in Table 7.12 take into account the reference inputs from Table 7.11 and dimension the radio network using this time the two UTRA technologies, i.e. FDD and TDD. We consider micro- and picocells of the urban and suburban environments. Thus, enabling the network with 3G services with transmission rates beyond 384 kbps in urban areas.

Table 7.12 Upper Bound Results for 75% Population Coverage in 2005

	Representative propagation environments		
Node B type	Urban	Suburban	Rural
Macro FDD (1+1+1)	480	160	280
Macro cells	1440	480	840
RF units in macro cells	1140	480	280
Micro FDD	310	100	0
Micro TDD	40	90	0
Pico TDD	160	170	0
Total node Bs	990	520	280
Area coverage (km ²)	800	1200	19500
Total existing area (km ²)	800	1600	38800

The TDD nodes aim to cover hot spots from the introduction of UMTS, e.g. shopping and educational centres, railway stations, office conglomerates, airports, etc. This approach will reduce traffic in FDD macrocells and allow coverage extension with overall increased capacity throughout all the network layers, i.e. pico, micro, and macro layers. The capacity and coverage optimization results outlined in Table 7.12 includes the utilization of MUD and intelligent antenna techniques. The first implemented already in the TDD technology as part of its interference solution, and the latter as part of the UL/DL

capacity optimization increase end-to-end performance in terms of coverage and capacity by three.

The TDD microcell will range from approximately 40 m to about 250 m assuming medium mobility. The TDD picocell will cover only indoors within a range of about 10–15 m with low mobility (e.g. below 8 km/h); but will have the capability of offering transmission rates up to 2 Mbps. Thus, the picocell coverage will reach mainly the office and home environment.

7.8.2.1 Upper Bound Examples FDD

To illustrate the radio network dimensioning based on the assumptions and principles of the preceding sections, here we take the upper bound values shown in Table 7.11. Thus, using the figures from the aforementioned table we can deduce average data volume, per subscriber, at busy hour, on the DL as shown in Table 7.13.

Table 7.13 Data Volumes for PS and CS Traffic

	kbps
Average PS* traffic per subscriber at BH (DL)	7445
Average CS traffic per subscriber at BH (DL)	555
Average total voice CS traffic per subscriber at BH (DL)	527

*We assume LCD 144 as packet for the CN.

To have a concrete point of reference in terms of coverage area and number of sites we illustrate values for an existing¹⁷ 2G 1800 MHz GSM network. In such a network, if we can assume that about 70% of sites can be re-used, then we will have co-siting of GSM and UMTS sites. The remaining sites would then be UMTS sites only. Table 7.14 illustrates hypothetical values as examples. Note that the three sector sites with 1 + 1 + 1 carrier configuration; however, some dense areas may need 2 + 2 + 2 or 3 + 3 + 3. On the other hand, it is believed that capacity will not be an issue at the introduction of UMTS.

Table 7.14 Reference Scenario for 75% Coverage

Sites			
GSM sites usable for UMTS	1400		
New UMTS sites (D-urban and urban for capacity):	200		
New sites replacing GSM sites not capable of supporting UMTS	200		
Total	1800		
		Cells	RF carriers
Sites in urban	250	750	750
Sites in suburban	850	2550	2550
Sites in rural	700	2100	2100
Total	1800	5400	5400

¹⁷ A reference network for illustration only

To calculate the multi-service link budget we use the WCDMA parameters assumed in Table 7.15.

Table 7.15 Selected Link Budget Parameters for WCDMA

FDD dedicated parameters	Downlink	Uplink
Chip rate	3.8 Mch/s	3.8 Mch/s
Path loss imbalance	2 dB	
Averaged PSCH Ec/It synchronization channels	-15 dB	
Activity factor in synchronization channel	10%	
PCCPCH Eb/No	6 dB	
PCCPCH bit rate	30 kb/s	
Orthogonality factor	0.4	
DL ratio of MSs at cell edge	30%	
DL inner radius/cell edge ratio	50%	
Io other-cell/Io same-cell DL at inner cell radius	0.06	
Io other-cell/Io same-cell at cell edge	2.3	0.7
Shadowing margin	2 dB	6.35 dB
Loading factor	75%	75%

The assumptions for the transmission parameters include: TMA (MHA), 18 dBi antenna gain, 27 m antenna height, and tri-sector antenna in all sites. Other parameters concern the UE with: 0 dB cable and connector losses, 0 dBi antenna gain, and 8 dB receiver noise. The characteristics for Node B are: 18 dBi antenna gain, Max Tx power 43 dB, cable and connector loss 3 dB, and noise figure of 3.5 dB. The TRUE has MHA noise figure of 2.5 dB, MHA gain of 15 dB, and a global receiver noise figure of 2.8 dB.

While the subscriber response or traffic mix does not depend on the environment, the number of subscribers per cell does depend on the service area or environment. Based on the inputs shown in Table 7.11, the number of subscribers per cell in the urban environment is projected to be about 180 in 2005.

Tables 7.16 and 7.17 illustrate reference parameters for the urban environment.

Table 7.16 Reference Parameters (LCD) in the Urban Environment (I)

Service type parameters	Speech service		LCD 64		LCD 144	
	UL	DL	UL	DL	UL	DL
Eb/No	3.7 dB	5.8 dB	2.2 dB	5.1 dB	0.6 dB	4.1 dB
Bit rate	12.2 kbps	12.2 kbps	64 kbps	64 kbps	144 kbps	144 kbps
Penetration margin	15 dB	15 dB	15 dB	15 dB	15 dB	15 dB
Body loss	3 dB	3 dB	0 dB	0 dB	0 dB	0 dB
No active channels/carrier	3.8	3.2	0.8	0.7	0.4	0.3
Activity factor	60%	60%	100%	100%	100%	100%
Maximum MS TX power	21 dBm		24 dBm		24 dBm	
Soft HO gain	0 dB	2.5 dB	0 dB	2.5 dB	0 dB	2.5 dB

Table 7.17 Reference Parameters (UDD) in the Urban Environment (II)

Service type parameters	UDD 64		UDD 144		UDD 384	
	UL	DL	UL	DL	UL	DL
Eb/No	1.7 dB	4.5 dB	1.1 dB	3.8 dB	1.0 dB	3.0
Bit rate	64 kbps	64 kbps	144 kbps	144 kbps	384 kbps	384 kbps
Penetration margin	15 dB	15 dB	15 dB	15 dB	15 dB	15 dB
Body loss	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB
# active channels / carrier	1.9	2.4	0.7	1.7	.8	2.0
Activity factor	100%	100%	100%	100%	100%	100%
Maximum MS TX power	24 dBm		24 dBm		24 dBm	
Soft HO gain	0 dB	2.5 dB	0 dB	2.5 dB	0 dB	2.5 dB

Based on the preceding tables we can now calculate the maximum allowable path loss for the traffic mix in the urban environment and determine the cell range. Tables 7.20 and 7.21 illustrate uplink and downlink values to serve as comparative reference. Some of these values are round-off from earlier dimensioning exercises to provide generic illustrations of multi-service budget link results [7].

7.8.2.2 RNC Dimensioning for 75% Coverage

Following the principles to dimension the RNC, here we illustrate some results based on the traffic figures illustrated in Table 7.11. We calculated the average throughput per city, at busy hour, both PS and CS traffic. Table 7.18 illustrates the traffic split in the reference cities.

Table 7.18 Subscriber Traffic and Site Split Reference 75% (2005)

Location	User traffic split (%)	Site split	Average total PS throughput (Mbps)	Average total CS throughput (Mbps)
City A	44	800	910	132
City B	29	520	600	87
City C	27	480	560	81
Total	100	1800	2070	300

Table 7.19 illustrates results of the reference examples based on the site distribution illustrated in Table 7.18 and the aforementioned steps for RNC dimensioning.

Table 7.19 RNC Dimensioning Results of Reference Examples

	RNC-A count	Traffic per RNC-A (Iu, + Iur) (Mbps)	STM 1s per RNC-A	RNC-B count	Traffic per RNC-B (Iu + Iur) (Mbps)	STM 1s per RNC-B
City A	3	230	2	4	330	3
City B	3	226	2	2	290	2
City C	1	210	2	3	270	2

Table 7.20 Link Budget Example Uplink

	Units	Uplink					
		Speech	LCD64	LCD144	UDD64	UDD144	UDD384
Radio channels/total channels		49%	10%	5%	25%	9%	2%
Activity factor		60%	100%	100%	100%	100%	100%
DL orthogonality factor							
Chip rate	Mchip	3.84					
PCCPCH bit rate	Kbps						
Service bit rate	Kbps	12.2	64	144	64	144	384
Processing gain		314.6	60	26.5	60	26.5	10
Target Eb/No	dB	3.7	2.2	0.6	1.7	1.1	0
Required PCCPCH Eb/No							
Required SCH Ec/I _t	dB						
TX info							
Cable and combiner losses	DB	0	0	0	0	0	0
Tx antenna gain	dB	0	0	0	0	0	0
DL Tx power	dBm						
SCH Tx power	dBm						
PCCPCH Tx power	dBm						
Total Tx power	dBm	16	18.	20.	18	21	24
Total Tx EIRP	dBm	16	18.	20.	18	21	24
RX info							
Rx antenna gain	dB	18	18	18	18	18	18
Cable and connecto losses	dB	0	0	0	0	0	0
Receiver noise figure	dB	2.8	2.8	2.8	2.8	2.8	2.8
Thermal noise	dBm/Hz	-174	-174	-174	-174	-174	-174
Service Rx sensitivity	dBm	-126	-120	-118	-121	-118	-115
Synchronization Rx sensit.	dBm						
Pilot Rx sensitivity	dBm						
Gains							
Log normal fade margin	dB	6.5	6.5	6.5	6.5	6.5	6.5
Penetration margin	dB	15	15	15	15	15	15
Body loss	dB	3	0	0	0	0	0
Soft HO gain	dB	0	0	0	0	0	0
Max path loss for services	dB	134	134	134	134	134	134
Max pilot and sync channel DL							
Cell range	Km	0.75	0.75	0.75	0.75	0.75	0.75
Loading factor		25%	25%	25%	25%	25%	25%
No. active channels per serv.		3.8	0.7	0.38	1.9	0.7	0.1
Total no. active channels/cell		8	8	8	8	8	8

Table 7.21 Link Budget Example Downlink

	Units	Uplink					
		Speech	LCD64	LCD144	UDD64	UDD144	UDD384
Radio channels/total channels		37%	7%	4%	28%	19%	4%
Activity factor		60%	100%	100%	100%	100%	100%
DL orthogonality factor		0.4	0.4	0.4	0.4	0.4	0.4
Chip rate	Mchip	3.84					
PCCPCH bit rate	Kbps	30	30	30	30	30	30
Service bit rate	Kbps	12.2	64	144	64	144	384
Processing gain		315	60	27	60	27	10
Target Eb/No	dB	3.4	2.7	1.7	2.0	1.4	0.9
Required PCCPCH Eb/No	dB	6	6	6	6	6	6
Required SCH Ec/I _t	dB	-15	-15	-15	-15	-15	-15
TX info							
Cable and combiner losses	dB	3	3	3	3	3	3
Tx antenna gain	dB	18	18	18	18	18	18
DL Tx power per service	dBm	24	20	19	25	26	23
SCH Tx power	dBm	28.7	28.7	28.7	28.7	28.7	28.7
PCCPCH Tx power	dBm	28.6	28.6	28.6	28.6	28.6	28.6
Total Tx power	dBm	34.5	34.5	34.5	34.5	34.5	34.5
Total Tx EIRP	dBm	49.5	49.5	49.5	49.5	49.5	49.5
RX info							
Rx antenna gain	dB	0	0	0	0	0	0
Cable and connecto losses	dB	0	0	0	0	0	0
Receiver noise figure	dB	8	8	8	8	8	8
Thermal noise	dBm/Hz	-174	-174	-174	-174	-174	-174
Service Rx sensitivity	dBm	-122	-115.5	-113	-116	-113	-109.5
Synchronization Rx sensit.	dBm	-115.1	-115.1	-115.1	-115.1	-115.1	-115.1
Pilot Rx sensitivity	dBm	-115.2	-115.2	-115.2	-115.2	-115.2	-115.2
Gains							
Log normal fade margin	dB	2	2	2	2	2	2
Penetration margin	dB	15	15	15	15	15	15
Body loss	dB	3	0	0	0	0	0
Soft HO gain	dB	2.5	2.5	2.5	2.5	2.5	2.5
Max path loss for services	dB	134	134	134	134	134	134
Max pilot and sync channel DL	dB	136	136	136	136	136	136
Cell range	Km	0.75	0.75	0.75	0.75	0.75	0.75
Loading factor		45%	45%	45%	45%	45%	45%
No. active channels per serv.		3.2	0.6	0.3	2.5	1.7	0.4
Total no. active channels/cell		9	9	9	9	9	9

7.9 CORE NETWORK (CN) DESIGN

The basic input for the CN dimensioning is also given in Table 7.5 and Table 7.11. In addition, it includes the results illustrated in Table 7.10 and Table 7.12. For all practical purposes, we illustrate the dimensioning for both the evolving and layered architecture, which is described in the forthcoming sections.

7.9.1 CN Analysis Assumptions

Figure 7.12 [8] illustrates key CN elements which we refer to throughout the dimensioning process. Notice that here we imply a co-existence with 2nd generation elements, e.g. GSM. The next chapter describes the essential functions of this network.

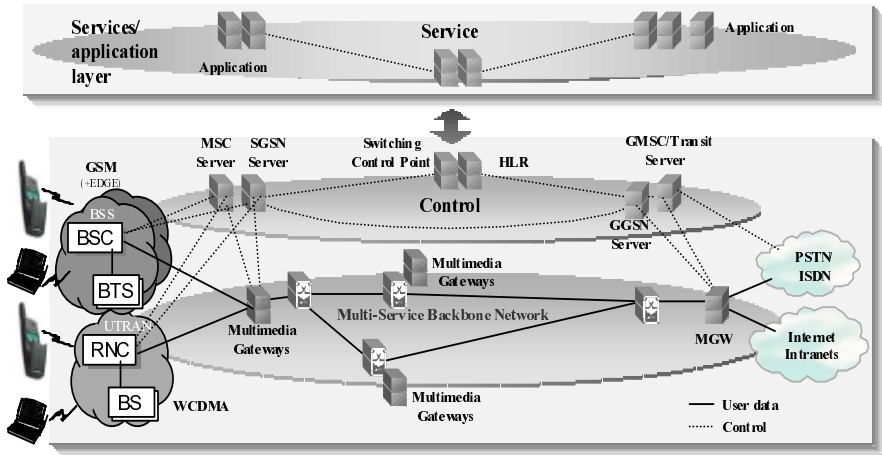


Figure 7.12 The core network layer architecture reference.

7.9.1.1 Locating CN Nodes

Logically, the starting point of the CN design and dimensioning will start by locating the main switching centres. In this field study we consider three sites: let us say they are within cities A–C of our country example in this chapter. Thus, we locate CN nodes in centres A, B, and C.

7.9.1.2 Locating RNC nodes

The number of RNC nodes will depend on the traffic load and number of BS sites connected. Their location in general will vary according to chosen topology; thus, we can have a centralized or distributed network layout. The centralized layout implies placing the RNCs at the switching centres together with the CN nodes. The distributed option will in general expand beyond the CN sites. Each option has its trade-offs and will depend on the availability of the transmission infrastructure as well. For example, some of the advantages of a centralized RNC network, i.e. co-located with the CN nodes include:

- Rapid implementation and integration processes, as well as the lower investment on the RNCs themselves. Today technology allows high capacity systems as better choices than small ones, because a higher grade of statistical multiplexing is gained in larger nodes than in smaller ones.
- Lower overhead costs on O&M and spare parts as well as less restriction of site space.
- With the trend towards more packet data traffic in the UTRAN, a transport network with centralized RNCs will require less bandwidth due to lower overhead over the Iub interface.
- About half of the traffic can be dropped locally at the RNC/MGW site thereby reducing the traffic in the core backbone network to a minimum.
- We do not need Iu transmission resources between RNC and MGW, neither do we need Iur transmission resources for co-located RNCs.

7.9.1.3 Network Overheads

While looking at the traffic flow, we also need to see the corresponding overheads in the various interfaces.

In the RNC different overheads apply to the Iu and Iub interfaces. Table 7.22 illustrates a non-exhaustive summary of the RNC overheads.

Table 7.22 Summary of RNC Overhead Assumptions

	User data	Iu	Iub
Voice activity factor (%)	50	50	50
Voice bit rate	6.22	8.91	10.35
Voice codec (AMR)	12.2	12.2	12.2
Retransmission (%)	10	10	10
Signalling (%)		10	10
Frame OH, CS (%)		21	23
Frame OH, PS (%)		70	24
O&M (%)			2

We can summarize more details on the overhead assumptions considered for the CN dimensioning as follows.

Packet data over Iu (GTP-U/UDP/IP/AAL5/ATM) has 72% OH, assuming an average packet size of 125 bytes, and about 10% signalling OH on top of user data on the Iu and CN. In addition, there is 21% OH on circuit data over Iu and core network, AAL2/ATM.

Overhead in PS interfaces: about 35% OH on packet data for Gb over ATM network, including BSSGP/FR/AAL5/ATM, about 35% OH on packet data on Gn interface, including GTP-U/UDP/IP. Very low OH on Gi interface, on top of the IP address application.

We assume 0.2% grade-of-service for UTRAN links with 90% load factor on the physical payload on ATM links, and 0.1% grade-of-service for voice and circuit data over the CN with also 90% load factor on physical payload on ATM links.

We add 10% retransmission OH on packet data over the Iub on PS user data, assuming 70–75% load factor on IP links in the CN.

We estimate 66% OH, i.e. 10.35 kbps per voice channel over Iub, for AMR 12.2 voice coding and 50% voice activity factor, including frame overhead UP/AAL2/ATM.

There exists also 43% OH, i.e. 8.91 kbps per voice channel over Iu and CN for AMR 12.2 voice coding with 50% activity factor, including frame overhead UP/AAL2/ATM.

The CS and PS may have 20–25% OH on CS over the Iub. This includes TCP/IP header compression and UP/AAL2/ATM. For PS, the average packet length is 125 bytes.

The signalling OH amounts to 10% on top of framed user data, and 2% O&M OH on top of the framed user data on Iub, both with a minimum of 64 kbps per link. In addition, there is 25% frame overhead on signalling and O&M traffic.

7.9.1.4 Analysis Assumptions

Grade-of-service (GoS) applies to the calculation for delay sensitive traffic, and delay varying services, i.e. voice and LCD services, where calculations estimate the required bandwidth to obtain a specific GoS. For ATM links carrying multiple services, the narrow band Erlang formula applies while considering different amounts of multiple rate traffic. For regular TDM links carrying one single service, i.e. voice between BSCs and MSCs, e.g. the narrow band Erlang formula applies.

We assume a load factor for ATM and IP links, i.e. 90% for ATM and 75% for IP links, which describe the maximum allowed theoretical loading on the links. However, we do not assume a load factor for TDM links. Thus, for links carrying both real-time and background type of traffic, the real-time bandwidth results from GoS calculations, and the average packet data originates from the load factor. On Iub and Iur interfaces both voice, LCD and LDD data require the same quality. Thus, we include GoS calculations for LDD data.

For all practical purposes, we assume that packet data terminates in a ISP, or 3rd party network at the local switch site in nodes A, B, or C; this means that no packet data transfer occurs between switch sites. We also assume that 30% of voice and CS data generated in the BSS and UTRAN connected to one switch site, terminates in another switch site, thereby, generating inter-MSC and UMTS inter-MGW traffic. We assume that the remaining traffic terminates locally to PSTN, PLMN or stays within the UMTS network.

7.9.2 Reference Outputs in CN Dimensioning

The final number of elements will depend very much on the type of configuration and capacities a supplier will offer for each CN node. Hence, in the following we provide simple reference results for completeness. [8]

7.9.2.1 RNC Configuration

Tables 7.23 and 7.24 contain reference values for the UTRAN RNC design in terms of total number of RNCs and total traffic volumes per area. The reference traffic flows reside on values provided in Table 7.5.

Table 7.23 RNC Configurations and Locations for 2002 and 2005 (750K and 1000K Subscribers)

Location	Capacity configuration		Node Bs supporting	
	Year 2002	Year 2005	2002	2005
City A	$2 \times L$	$3 \times L$	450	380
City B	$1 \times L, 1 \times M$	$2 \times L$	290	405
City C	$1 \times L, 1 \times S$	$1 \times L, 1 \times M$	280	620
Total (accum.)	$4 \times L, 1 \times M, 1 \times S$	$6 \times L, 1 \times M$	1020	1405

Large (L) = 375K subscribers, medium (M) = 240K subscribers, small (S) = 66K subscribers.

The reference values will vary based on the type and configuration of the RNC proposed. Here we provide generic numbers in order to allow a starting point to illustrate a RNC and CN dimensioning process. The configuration includes grow surge margins and takes into account the overheads described in the preceding sections.

Table 7.24 Total Traffic Volumes Through RNC for 2002 and 2005 (750K and 1000K Subscribers)

Location	User data (Mbps)		Iu-traffic (Mbps)		Iub-traffic (Mbps)	
	Year 2002	Year 2005	2002	2005	2002	2005
City A	206	275	365	487	397	530
City B	134	178	237	316	258	344
City C	126	168	224	298	243	324
Total	466	621	826	1101	898	1198

7.9.2.2 Core Network Elements

As illustrated in Figure 7.12, our reference CN architecture has a layered structure with total separation of the pay load transport and traffic control into the user plane and control plane, respectively. The user plane builds on the MGWs to the CN through the transmission backbone. The control plane contains the common data bases (e.g. HLR/AuC) besides the different servers (e.g. MSC, SGSN, GGSN; GMSC server) which handle traffic.

If we assume decentralized Media Gateway (MGW) configurations, one MGW in each switch centre (i.e. A, B, and C) will handle capacities ≥ 8000 Erlang CS traffic and ≥ 500 kpps packet switched traffic in each city. configurations as needed. Table 7.25 illustrates the value for the assumptions.

Initially, to consolidate the functionality and operation of MGW (new on CN) we can centralize it in one city where there is high-density traffic, e.g. city A. Then we can expand it to other cities as need arises. Or we can start with one MGW in each city to facilitate network lay-out. In this study we choose the latter for the first years starting 2002.

We also estimate the total simultaneous data sessions (SDS) and distribute them throughout the three cities. The SDS allows us to estimate the traffic interaction, e.g. in the PS network, which occurs based on the number of subscribers attached. The latter depends on the GGSN and SGSN capacity for handling PDP contexts and throughput. For example, a GGSN capable of handling 500K PDP context and a throughput of 500 kpps with a SGSN also handling 500K PDP contexts and supporting 400K simultaneous attached users can serve as reference here. We can also assume that the proportion of traffic passing through the cities are: city A 45%, city B 30%, and city C 25%.

Table 7.25 MGW PS and CS, GGSN and SGSN Configurations 2002 and 2005

Location	GGSN configura- tion (min. 300K subscribers)		SGSN configura- tion (min. 300K subscribers)		Simultaneous data sessions (e.g. 5K SDS package)		MGW configura- tion 8400(E) 540 (kpps)	
	2002	2005	2002	2005	2002	2005	2002	2005
City A	1	1	1	2	67.5	90	1	1
City B	1	1	1	1	45	60	1	1
City C	1	1	1	1	37.5	50	1	1
Total	3	3	3	4	150	200	3	3

The SGSN and GGSN servers control the PS traffic. Thus, the number of GSN servers depends on the number of simultaneous packet data sessions. Because the data packets are switched in the media gateways, the data volumes in the network do not influence the dimensioning of the GSN servers.

Likewise, the MSC server controls the circuit switched calls, and only the total circuit switched traffic determines the MSC server dimensioning. Packet switched traffic is controlled by the GSN servers and does not impact the MSC server. The MSC server capacity is influenced by the number of call attempts for circuit switched services and by the total circuit switched Erlangs in the network.

Assuming co-located MSC/VLR/HLR/AuC and signalling functionality in one physical node, Table 7.26 illustrates 3G MSC/HLR locations. The software on the MSC servers is dimensioned according to the simultaneous call capacity (SCC). The SCC equals the CS total traffic (in Erlang).

Table 7.26 Number of 3G MSC/HLR 2002 and 2005

Location	MSC/VLR HLR/AuC		SW for SCC for CS (Erlang)		Signalling HW/SW, e.g. 2Mb boards		Projected HLR capacity ($\times 10^3$ subscribers)	
	2002	2005	2002	2005	2002	2005	2002	2005
City A	1	1	7911	9375.7	2	4	330	450
City B	1	1	4674	6250.5	2	2	220	300
City C	1	1	3895	5208.75	2	2	200	300
Total	3	3	15580	20835.0	6	8	750	1050

7.10 TRANSMISSION NETWORK ASSESSMENT

To adapt or implement a new transmission system for 3G networks implies taking into account existing factors into the new factors such as wider band applications and larger base of subscribers, as well as existing 2G traffic if any. In the following, before illustrating the dimensioning of a transmission network for our field study we outline some of the major issues that concern building a 3G transmission network.

7.10.1 Building 3G Transport Systems

Some of the factors which will have non-negligible impact on the design of a 3G transmission network include: new services with higher average traffic per subscriber, which will increase the traffic density in urban areas, the latter resulting in a denser BS network, new dedicated sites, new transport links. In addition, service evolution will also play an important role because a large number of 3G networks will also continue to serve 2G subscribers. These subscribers will continue utilizing evolving applications such as: SMS, emails, ftp, Internet, interactive games, video calls, WAP, e-commerce, and video telephony, etc.

Due to the increase in multimedia applications, traffic can easily increase from today's 0.10 Mbyte/user per month to about 35 Mbyte/user per month by 2005. Real-time (RT) services with tighter delay requirements will augment gradually, as will non-real-time services with low packet loss demands.

Thus, a 3G-network transmission will require higher flexibility to increase capacity, incorporate new sites dynamically, and support a broader set of protocols (e.g. ATM, IP, etc.). In addition, QoS will be an inherent requirement, while demanding higher reliability and security. On the other hand, overall transmission operational costs would still need to diminish, despite the increase in functionality. As a result, the challenging tasks to build a 3G transmission network include: access capacity increase to cope with GMS (+GPRS), EDGE¹⁸, and WCDMA sites, increase CN capacity and efficiency from the start of 3G. This will keep the transmission network as a strategic asset, so it can evolve gradually and be optimized for packet traffic all the way from the core.

7.10.1.1 Present State and Future Evolution

Today microwave has become the preferred choice for BS network access. It is cost-efficient and we can deploy it rapidly with significant savings through integrated solutions. In the CN transmission side, i.e. connections from BSCs to MSCS sites, 2 Mbps leased lines, or protected microwave at 16×2 Mbps or own or leased SDH transport is common.

In the future sites will evolve to multi-band multi-standard sites, e.g. plain GSM to GSM-EDGE and/or WCDMA. Macrocell transmission capacity will thus increase from typical 2 Mbps to 10 Mbps as illustrated in Figure 7.13. The level of increase clearly justifies new ways to implement a transmission network to maximize efficiency and to

¹⁸ Although it may not apply to all networks, specially if some choose a direct path to UMTS (e.g. WCDMA).

add flexibility. High capacity SDH (STM4–STM16), e.g. can become the core transport network while complementing the access network with STM1 to enter the backbone, and using $n \times 2\text{M}$ for the access connections. The latter would be mainly microwave radios.

Capacity increases will accumulate primarily around the controller sites, e.g. an area served by a $16 \times 2\text{M}$ radio link will saturate as it takes 3G traffic. Thus, density will be a major challenge with radio links; consequently we will need traffic grooming in the field near the BSs.

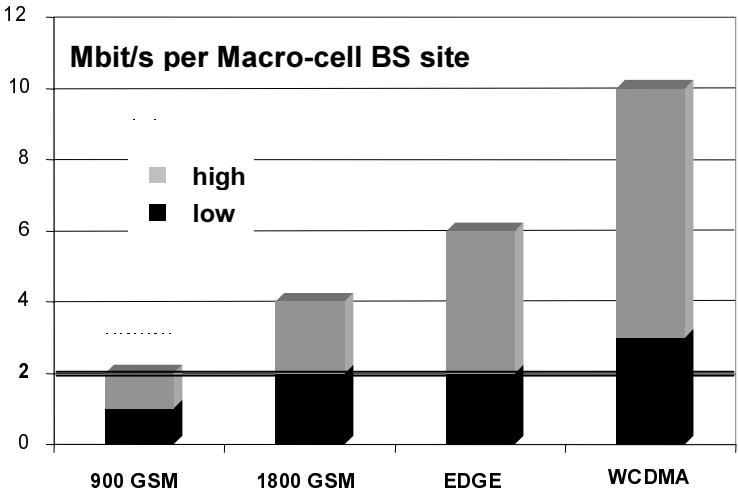


Figure 7.13 Macrocell traffic growth example.

In a mixed environment, e.g. 2G and 3G, each BSC/RNC will concentrate capacity to the NSS and GPRS elements. Thus, even if the total volume is lower than in the access network, an integrated transport network will add efficiency and performance overall as illustrated in Figure 7.14. Consequently, a SDH based transport CN will provide flexible capacity and a base for multi-protocol support as well as basic connectivity.

The access network has more stringent delay requirements, e.g. WCDMA traffic needs ATM transport carried over TDM circuits. Thus, 3GPP technical specifications recommend ATM over semi-permanent point-to-point connection for the RAN network. On the other hand, in the CN transmission, efficiency gets more important. Thus, packet traffic can travel either on the top of ATM, or IP packets get carried directly over TDM infrastructure based on fibre, for example.

7.10.2 Transmission Reference Network

Coming back to our field-study, we make the following assumptions:

- We assume that the transmission network will serve an existing 2G network (e.g. 1800 MHz GSM network).
- About 70% of existing BS sites can house a WCDMA BS. New UTRA sites will replace non-usable GSM sites and/or stand in for coverage where required, e.g. dense areas.
- Figure 7.15 illustrates a typical 2G-access network model, which will serve to project the links required in 2002 and 2005.
- In the model, 30% GSM BTSs connect through leased lines for the last mile access, 35% use MW radio links in cascaded structures, and 35% use MW radio links in hub structures.

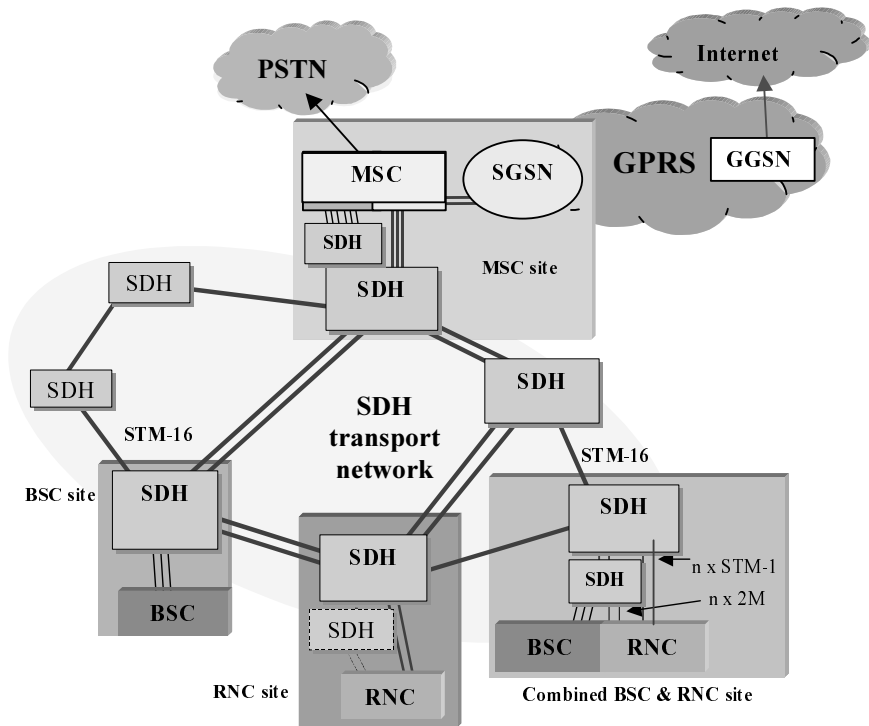


Figure 7.14 An integrated core transport network.

In Figure 7.15 [8] leased lines terminate point-to-point between BTS and BSC, while cascaded microwave structures use three sites connected in a cascade, and the structure hub serves five BTSs including itself. In the example links, about half of the hub-to-BSC links are 4×2 Mb and other half is 8×2 Mb ($4 \times (2 \times 2$ M)).

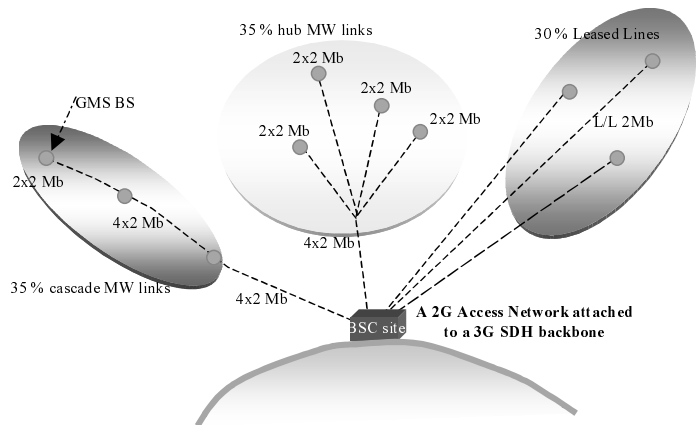


Figure 7.15 Typical 2G access network attached to a SDH backbone.

7.10.2.1 Co-siting Transmission Sites

In a hybrid 2G/3G environment, we would aim to integrate the UMTS network with the existing GSM network infrastructure to re-use the latter to the maximum. Thus, in the macrocell coverage, e.g. we would co-locate WCDMA BSs with existing GSM BSs as much as possible. To achieve this Figure 7.16 [8] illustrates two key options, excluding an initial approach of simply carrying 3G traffic over the existing GSM transmission infrastructure.

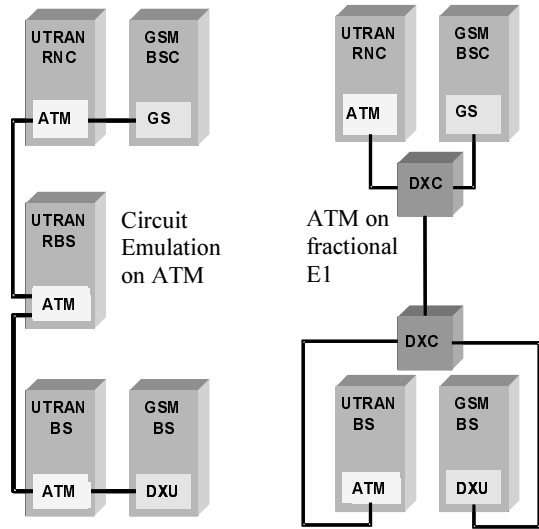


Figure 7.16 2G and 3G transmission site sharing.

In the first option, carrying 3G traffic over the BSS, the ATM transmission to/from the 3G BS gets mapped from/to an E1 2 Mbit/s through 64 kbit/s time slots. We call this

ATM on fractional E1. Here we assume the time slot handling DXC function is included in the GSM BS. At the GSM BSC, we connect the UTRAN time slots to the RNC semi-permanently. This solution may apply while 3G traffic remains low.

In the 2nd option, all radio access network transmission gets carried by a common ATM backbone network composed by UTRAN nodes (e.g. RNCs and BSs). The GSM BSs also get connected to the UTRAN BSs. In the UTRAN BS we map E1s to an emulated ATM based 2 Mbit/s line, and transmit towards the RNC on a common backbone together with ATM based UTRAN traffic from the UTRAN BSs. We call this circuit emulation on ATM. In the RNC E1s from the GSM BS get re-established and sent to/from the GSM BSC node. This option applies when projecting large UTRAN traffic.

7.10.2.2 The Backhaul Network

A SDH regional backhaul infrastructure aims to connect the BSCs with the switch sites in the main cities (i.e. cities A–C in our example). The backhaul network will provide distributed high capacity and high availability infrastructure, from where the last mile access can proceed over microwave links or leased lines. We co-locate an ATM/AAL2 switch with Frame Relay (FR) encapsulation capabilities at each BSC site. The ATM switch and the BSC inter-connection supports fractional ATM flow, thereby allowing AAL1 circuit emulation traffic onto the ATM and TDM domain for a pure separated ATM and TDM domain upstream in the network. GSM voice traffic use TDM transport from the BSC whereas UTRAN traffic and GPRS packet data share common ATM transport resources. We may assume that on average each regional ring terminates 4–6 BSCs to the switch sites. We can also assume that for the size of our example network, six regional rings can terminate in switch site of city A, and four rings can terminate in the switch sites of city B and city C, respectively.

Capacity projections in our example shows an average of 15 E1s required per ring in 2000 based on GSM traffic, and about 80 E1s in 2005 including 3G traffic. Thus, any rings with STM-1 capacity will need an upgrade before 2005. Consequently, a mix of STM-4 rings for larger traffic and STM-1 rings for lower traffic areas will result in the ideal backhaul network.

7.10.2.3 The National Backbone Network

To take the traffic flow from STM1–4 rings, a high capacity STM-16 ring will inter-connect the main BSC sites of each region with the switching sites in cities A–C. In addition, we would expect a high-capacity backbone ATM switch in each switch site.

At the switch sites BSC-to-MSC traffic would terminate to the MSC from the SDH DXC. The UTRAN traffic would terminate over the regional rings to the ATM switch located in the switch site, while aggregating traffic to the RNCs. As noted earlier, RNCs and CN nodes reside co-located at the switch sites hard-wired without requiring any transmission resources. Inter-site MSC traffic flow would use STM transport over the SDH backbone, and UMTS inter-switch site traffic would get routed over the ATM switching layer.

Capacity projections show a higher bandwidth need in 2000 than in 2005 despite the higher number of subscriber in the latter. The reason lies in the better bandwidth utilization for UMTS traffic, e.g. codec at the edge for voice traffic means lower capacity requirement. For example, in 2000 we would need 160 EIs, about 210 EIs in 2002 and 118 EIs 2005. This means that national ring capacity requirements, can drop to STM-4 capacity rather than STM-16.

7.10.3 Transmission Dimensioning Results

In the following for completeness, we outline some dimensioning exercise results assuming 750K UMTS subscribers and 900K GSM subscribers around 2002. We also assume about 1000K UMTS subscribers and 300K GMS subscribers by 2005. These assumptions are mainly to create enough margin and do not necessarily reflect the actual evolution of subscriber population. We do still use the traffic patterns shown in Table 7.5 and subscriber population in Table 7.2. The dimensioning exercise does include the margins for unexpected subscriber growth [8].

7.10.3.1 Traffic per Site

Table 7.27 illustrates the reference number of sites for this transmission analysis, and the values in Table 7.28 indicate the average amount of bandwidth required per site on the Abis and Iub interfaces for the different service areas. The values include frame OH and load factor, ready for mapping onto the payload on the physical PDH or SDH links.

Table 7.27 Reference Number of Sites for 2002 and 2005

	DU		U/IND		SU		FO/OP	
Year	2002	2005	2002	2005	2002	2005	2002	2005
Co-located 2G/3G sites	5	8	230	230	100	100	490	490
Stand alone 3G sites	4	6	60	160	30	75	100	350
Stand alone 2G sites	2	4	80	80	35	35	165	165

In Table 7.27 the number of sites are cumulative. The totals can be summarized as follows: 2002 → co-located = 825, stand alone 3G = 194, stand alone 2G = 282. Total 2002 = 1301. 2005 → co-located = 828, stand alone 3G = 591, stand alone 2G = 284. Total 2005 = 1703. Therefore, the 2002 sites grow to 1703 by 2005 in this example.

Table 7.28 Projected Traffic per Site 2002 and 2005 (Mbps)

	DU		U/IND		SU		FO/OP	
Year	2002	2005	2002	2005	2002	2005	2002	2005
Co-located 2G/3G sites	3	2.0	1.9	1.3	4.7	4.0	2.5	2.0
Stand alone 3G sites	1.5	1.4	1.0	.9	3.8	3.4	1.6	1.6
Stand alone 2G sites	1.0	.5	1.0	.5	1.0	.5	1.0	.5

7.10.3.2 Microwave Network

Based on the assumptions of the site traffic in Table 7.28, Table 7.29 illustrates the number of microwave links projected for 2002 and 2005. Values are absolute figures.

Table 7.29 Microwave Links for 2002 and 2005

	DU		U/IND		SU		FO/OP	
Year	2002	2005	2002	2005	2002	2005	2002	2005
2x2Mbps	5	5	165	235	35	100	330	500
4x2Mbps	3	2	70	95	40	0	75	155
8x2Mbps	0	0	20	0	15	35	120	45
16x2Mbps	0	0	0	0	25	10	0	0
Total	8	7	255	330	115	145	525	700

The MW frequencies will vary from region and/or country to country. For example in Switzerland, the last mile access links in dense urban, urban and industrial areas will most likely use 38 GHz. In suburban, forest and open areas links will use 23 GHz. However, these frequencies are subject to change depending on the radio frequency availability.

7.10.3.3 Leased Line Network

Table 7.30 illustrates the number of projected leased lines for 2002 and 2005. Values for each site category are shown, and are absolute figures:

Table 7.30 Leased Line Projections (E1s) for 2002 and 2005

	DU		U/IND		SU		FO/OP	
Year	2002	2005	2002	2005	2002	2005	2002	2005
Co-located 2G/3G sites	3	2	70	70	90	60	290	295
Stand alone 3G sites	1	1	20	45	20	45	35	100
Stand alone 2G sites	1	1	20	25	10	10	50	50

Taking into account the information of Table 7.29 and Table 7.30, we illustrate the projected capacities for the regional rings in Table 7.31.

Table 7.31 Projected Capacity Regional Rings (E1s) for 2002 and 2005

	City A		City B		City C	
	2002	2005	2002	2005	2002	2005
A1	70	75				
A2	60	70				
A3	80	85				
A4	60	70				
A5	80	85				
A6	80	90				
B1			85	90		
B2			70	80		
B3			60	55		
B4			60	70		
C1					80	90
C2					50	60
C3					70	75
C4					65	65
Total	430	475	272	295	265	290

7.10.3.4 SDH Backbone Network

Table 7.32 projects the number of E1s between the switch sites city A, city B and city C. MSC-MSC values indicate GSM voice traffic over the TDM domain. MGW-MGW values indicate UMTS voice and CS data figures over the ATM domain. All packet data gets terminated locally; thus, packet data transfer between switch sites does not exist .

Table 7.32 Inter Switch Site Traffic (E1s) for 2002 and 2005

Year	MSC - MSC		MGW - MGW	
	2002	2005	2002	2005
City A-city B	75	40	20	1 STM-1
City B-city C	50	20	10	1 STM-1
City C-city A	70	35	20	1 STM-1
Total	195	95	50	3 STM-1

7.10.3.5 Local Switch Site Traffic

The traffic flow in the local switch sites corresponds to the interface load between UMTS nodes at the switch sites. Values in Table 7.33 include all overheads as well as load factors, ready to get mapped onto the payload of the physical PDH/SDH/Ethernet links. Values in Table 7.33 reflect total figures.

Table 7.33 Local Switch Traffic for 2002 and 2005

Local switch traffic (Mbps)	Inter-face	City A		City B		City B	
		2002	2005	2002	2005	2002	2005
UTRAN - RNC	Iub	500	660	300	430	320	405
RNC- MGW	Iu	425	570	260	370	275	350
MGW-GGSN	Gn	235	310	145	200	155	191
MGW-PSTN (external network)		312	420	190	270	200	255
GGSN-ISP (external network)	Gi	174	230	105	150	115	141
MGW-other switch sites	G	95	125	60	80	60	80
MGW-MSC-S (sign)	GCP	8	10	5	7	5	7
MGW-SGSN-S (sign)	GCP	16	20	10	15	9	14
SGSN-S-GGSN (sign)	Gn	16	20	10	15	9	14

7.11 CO-LOCATING AND SHARING SITES

With the widespread use of the 2G mobile network (e.g. GSM), by the time we start deploying UMTS most countries or regions will have already at least one 2G mobile system in operation. While in some places restrictions for new sites will not affect 3G network deployment, in others this will become a key factor to maximize re-use of sites and push site sharing to the limits. In Switzerland for example, regulations on power levels, cost of sites and environmental protection regulations do make it difficult to acquire new sites. Thus, co-locating 2G and 3G equipment becomes imperative. To address these issues in the following we illustrate co-location primarily with UMTS and GSM-1800 MHz.

Site sharing depends on the relative coverage of the existing network when compared to UTRA-FDD¹⁹. Here we compare at the relative UL coverage of GSM1800 with full-rate speech service and WCDMA speech and 144 kbps data service. Table 7.34 illustrates the basic assumptions and reference results.

Table 7.34 Common Path Losses GSM1800 MHz and UTRA-FDD

	GSM1800/ speech	WCDMA/ speech	WCDMA/ 144 kbps
Mobile transmission power (dBm)	30	21	24 (21)
Receiver sensitivity (dBm)	−108	−123	−116
Interference margin (dB)	0.0	2.5	2.5
Fast fade margin (dB)	2.0	2.0	2.0
Antenna gain (dBi)	18.0	18.0	18.0
Body loss (dB)	3.0	3.0	–
Mobile antenna gain (dBi)	0.0	0.0	2.0
Maximum path loss (dB)	153.0	155.0	154.0

In Table 7.34, we assume 2.0 dBi gain for UTRA-FDD data terminals, and 3 dB body loss for the speech terminals. The interference margin in FDD may apply to 50% loading. The FDD sensitivity assumes 5.0 dB BS noise figure with $E_b/N_0 = 5.0$ dB for 12.2 kbps speech (AMR) and 1.5 dB for 144 kbps data. GSM sensitivity = −108 dBm. The FDD fast fade margin includes the macro diversity gain against fast fading. Co-siting assumes three-sector antenna configuration for both GSM and FDD.

From the reference values in Table 7.34, we can extrapolate that the coverage of 144 kbps data service with 21 dBm mobiles may correspond to the GSM1800 speech coverage. Thus, we can provide 144 kbps data with WCDMA when re-using or co-siting GSM1800 sites with the same coverage as GSM1800 speech. In the DL we have less power restrictions; hence, higher 3G data rates may be possible with the GSM1800 coverage.

On the other hand, we should be aware that coverage comparison and site GSM1800 re-utilization for WCDMA will depend on the receiver sensitivity values and other system parameters, e.g. HO, FH, power control, etc. In the following, we will expand the interference levels and criteria for co-siting, as well as de-coupling issues.

7.11.1 Interference Levels and De-coupling

Radio site sharing solutions implies either utilization of multi-band antennas and co-locating 2G/3G HW/SW. To assess these needs we first list the interfering levels given by the standard specifications.

¹⁹ The UTRA FDD uses the WCDMA technology as indicated in earlier chapters.

7.11.1.1 GSM/UMTS Interference

Considering the GSM recommendations (05.05 and 11.21) and the UMTS recommendation (25.104) outlined in Chapter 5, we can calculate the necessary decoupling between:

- two UMTS systems;
- one GSM900 system and one UMTS system;
- one GSM1800 system and one UMTS system.

The analysis of co-location includes: wideband noise, spurious emissions, blocking and intermodulation. The solutions may come from: antenna isolation, use of filters (e.g. diplexer or triplexer). We assume the following reference output powers: 43 dBm output power at the antenna port of GSM900/1800 BTS, and 45 dBm output power at the antenna port of UMTS BS.

Wide band noise: the frequency separation between the GSM900 and the UMTS, as well as between the GSM1800 and the UMTS bands is sufficiently large. Thus, it is frequently assumed that the transmit part of GSM900/1800 does not generate wideband noise in the receive part of UMTS.

Spurious emissions: GSM recommendation 05.05 specifies a maximum level of spurious emissions in the frequency band 1–12.5 GHz, including the UMTS receiving band. This maximum power should not be greater than -30 dBm at the base station RF output port, power measured in 3 MHz bandwidth, which is equivalent to -95 dBm/Hz.

Looking at the maximum tolerable interference level in the Rx channel of -118 dBm, (i.e. -185 dBm/Hz, over 5 MHz worst case), we can calculate the required isolation to avoid interference for the Tx GSM/DCS Rx UMTS. Thus, assuming specified spurious emissions over 3 MHz of -30 dBm, specified spurious emissions per Hz of -95 dBm and maximum level of unwanted signal per Hz -185 dBm, the necessary isolation between antenna connectors is 90 dB.

GSM and UMTS band analysis indicates that a 2nd and 3rd order intermodulation product of GSM900 and 1800 transmission frequencies, respectively, fall into the UMTS uplink band. Thus, GSM 05.05 recommendation specifies that when the intermodulation power level of the interfering signal gets injected into the antenna connector at a level of 30 dB lower than that of the wanted signal, the spurious emissions requirements shall not be exceeded, i.e. -30 dBm in the UMTS receiving band at the BS RF output port measured in 3 MHz bandwidth, which is equivalent to -95 dBm/Hz. Since this is at the same level as that specified for the spurious emissions, we need the decoupling between antenna connectors.

Blocking: UMTS 3GPP TS 25.104 recommendation specifies that out of band, the maximum level of interfering signals (CW carrier) for blocking is equal to -15 dBm. The GSM power is assumed to be 43 dBm at the antenna connector. This power is over 200 kHz, rather than a CW interfering source as defined in TS 25.104. Comparing this value with the UMTS blocking point, we can calculate the necessary isolation between the two antennas. Then, assuming that the jamming system is Tx GSM/DCS and the

victim the Rx UMTS, with a GSM/DCS transmit output power of 43 dBm the blocking point out of the receiving band is -15 dBm. Thus, the necessary isolation between antenna connectors is 58 dB.

In conclusion, from the preceding analysis, we see that we need an isolation of 37 dB and 45 dB to protect GSM900 and 1800 from UMTS interference, respectively. We can obtain these decoupling values by physical separation between the GSM900 and/or 1800 antenna and the UMTS antenna.

Table 7.35 summarizes the required isolation, in case of co-location:

From the preceding analysis, we see that we need an isolation of 37 dB and 45 dB to protect GSM900 and 1800 from UMTS interference, respectively. We can obtain these decoupling values by physical separation between the GSM900 and/or 1800 antenna and the UMTS antenna.

Table 7.35 UMTS GSM Isolation Requirements

System specification requirements	UMTS Tx to GSM900 Rx (dB)	UMTS Tx to GSM1800 Rx (dB)	GSM900/1800 Tx to UMTS Rx (dB)
Spurious emissions and intermodulation products	34	34	90 (39 for BTS R99 or later)
Blocking	37	45	58

To obtain the isolation of 90 dB to protect UMTS from GSM900 or/and 1800 interference gets more difficult through only space separation. We would need specific filters at the GSM side to reduce the spurious emissions levels, and on the UMTS side to protect the UMTS BTS from GSM emissions. The latter will be imperative when using:

- dual band antenna or triband for which the isolation is only about 30 dB between GSM900 or/and 1800 and UMTS bands;
- wideband antenna for which there is no isolation between GSM900 or/1800 and UMTS bands.

Other decoupling needs: for the GSM1800 and UTRA-FDD frequencies, i.e. 1710–1785 and 1920–1980 MHz UL, respectively, and 1805–1880 and 2110–2170 MHz DL, respectively, there is a need for 68 dB air decoupling.

7.11.1.2 Power Level Limitations

Maximum power level requirements also contribute to the limitations for site co-location. Although the limits may vary from region to region, there are limits everywhere. Thus, unnecessary power rise in co-located sites will minimize power emission concerns. On the other hand, with the advent of WCDMA power calculations should not follow the same strictness as for GSM, because a WCDMA carrier will not be transmitting at full power all the time.

7.12 CO-LOCATING ANTENNA SYSTEMS

Another key issue with co-siting or site sharing is the integration of 2G/3G antenna solutions. Thus, to complete the analysis of co-siting in the following we summarize the main aspects implementing antennas for UMTS and 2G systems like GSM [7].

While single band antennas can meet co-siting needs, multi-band antennas are by far the ideal solution for integrated antenna systems in co-siting spaces for 2G/3G. Multi-band antennas (i.e. dual-band or triple-band), paired with an additional diplexer or triplexer minimize the amount of antennas and feeder cables per site. On the other hand, adding diplexers to an existing system will increase losses thereby decreasing coverage. Thus, the trade has to be taken into account when completing radio planning.

7.12.1 Co-siting GSM1800 and UMTS

7.12.1.1 Single Band

Due to demands for separate tilting, e.g. single-band antennas may be ideal. Figure 7.17 [7] illustrates the air-decoupling configuration for single band antennas, where separate feeder cables connect the Node B and BTS, respectively.

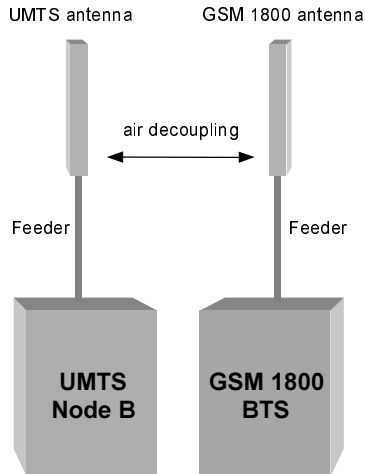


Figure 7.17 Air decoupling in single-band antennas.

The single band antennas get separated either by a vertical distance or by a horizontal distance. Side by side antennas, e.g. GSM1800 and UMTS may decouple already at 40 dB. Thus, to apply to the most relevant frequencies (with better than 42 dB) recommended coupling distance (including security/error margin) will lie between 0.9 and 1.0 m horizontal separation.

Likewise in the vertical case, to reach a decoupling of more than 42 dB in the UMTS frequency range, a coupling distance of 1.8–2.0 m (including security/error margin) of vertical separation is recommended.

7.12.1.2 Wideband Antenna

Figure 7.18 [7] illustrates a wideband solution²⁰ for GSM1800 and UMTS using a diplexer, one feeder cable. This solution appears advantageous because it needs only one feeder cable and one antenna panel. However, it has limitations because it imposes the same antenna characteristics on the GSM1800 and the UMTS bands. Such limitation, for example, prevents different electrical down tilt for the two systems.

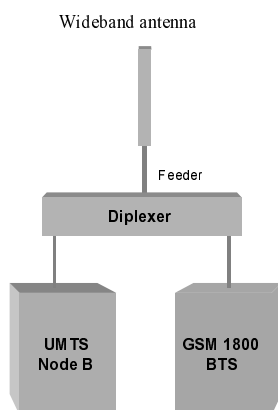


Figure 7.18 Wideband antenna configuration with one diplexer.

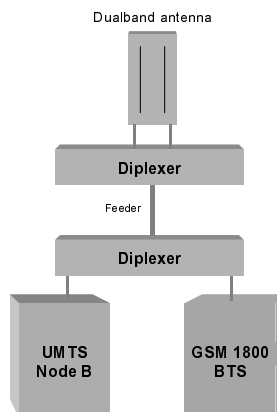


Figure 7.19 Dual band antenna configuration with two diplexers.

7.12.1.3 GSM 1800/UMTS Dual Band Antenna with Double Diplexers

Double diplexer dual band antennas as illustrated in Figure 7.19 [7] include one BTS-side diplexer, one feeder cable, one antenna side and a GSM 1800/UMTS dual band antenna²¹ consisting of two antennas within one panel. The BTS-side diplexer will generally afford 45–48 dB of decoupling from GSM 1800 transmit port to UMTS receive port. While the antenna side diplexer provides a decoupling value of 30–32 dB, which is generally sufficient.

This dual band antenna solution with two diplexers does allow different gain and electrical tilt selection for GSM 1800 and UMTS. On the other hand, it demands the implementation of two diplexers.

7.12.1.4 Feeder Cables

As noted in the preceding sections we can realize dual band systems, either with separated single band antennas or dual band antennas. On the other hand, when an antenna system supports diversity, we require at least two additional antenna branches per BTS sector and mobile system. The latter implies four antenna branches for a dual band BTS sector²². Hence, to prevent the usage of four feeder cables, e.g. we can apply feeder

²⁰ This combination gets doubled for the second antenna branch.

²¹ As in the wide antenna, this combination has to be doubled for the second antenna branch.

²² This does not apply to the solution with broadband antennas for GSM 1800 and UMTS.

sharing with additional diplexers and use two feeder only. Figure 7.20 [7] illustrates this logic with a cross-polarized dual band antenna.

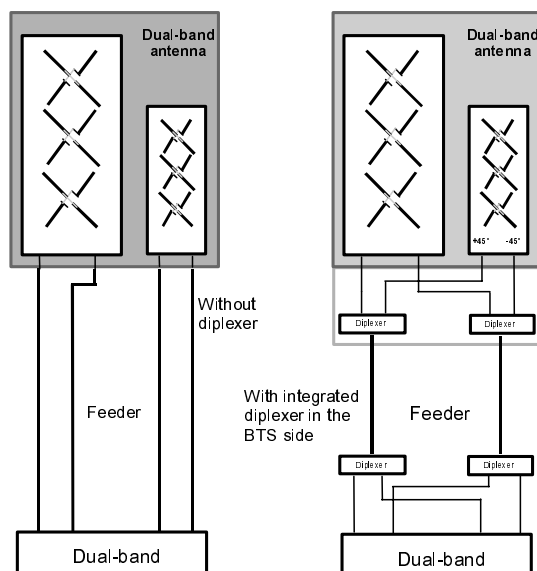


Figure 7.20 Dual band antenna with and without diplexer.

Dual band antennas are characterized by being suitable for both frequency ranges with separate input connectors. Although dual band antennas suit ideally both GSM 1800 and UMTS frequency ranges, they lead to a double number of antenna connectors when compared to corresponding single band antennas if sufficient diplexers are not used. Thus, it seems more efficient to upgrade dual band antennas with additional diplexers in order to decrease the number of antenna connectors by a factor of two. In this case the required feeder system will remain the same as for a single band antenna system.

The trade-off of extra investment for additional diplexers and expense on the feeder system will depend in the end on the losses due to feeder attenuation. In any case, when doing a transition from single band to a dual band system, the existing feeder system can be used, ensuring a fast installation during retrofit if the necessary diplexers are in place [7].

7.13 CONCLUSIONS

In this chapter we have covered essential aspects for 3G network deployment. We have brought together the main issues of network dimensioning on the radio, core network, as well as transmission. We have used a field study to illustrate key parameters and introduce some relevant assumptions when trying to size a network for a given population within a particular region.

The data examples correspond to harmonized values making a case for the method used. However, they do not necessarily reflect the actual growth of network or exact projec-

tion of network size. Nevertheless, do make realistic reference to the given concepts. Thus, they can serve as examples.

The analytical process provided for dimensioning the radio side, aimed at setting the basis to focus on key WCDMA issues that will need optimization as networks get implemented for mixed services. The factors involved in WCDMA appear larger than those in current CDMA systems today. On the core network side, we have also introduced new factors not typically seen in 2G networks. On the transmission side, we have seen the traffic increase will demand larger bandwidths throughout the whole network and not only in the core, but also the radio side.

References

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8

RESOURCE AND NETWORK MANAGEMENT

8.1 INTRODUCTION

Operating a 3G network involves managing resources and Network Elements (NE). This chapter covers these two aspects to complete the deployment issues started in Chapter 7. Resources here refer primarily to the radio resources and NE refers to the 3G building blocks, i.e. elements in the CS, PS and radio access networks.

8.2 RADIO RESOURCE MANAGEMENT AND SIGNALLING

Power control constitutes one of the major tasks of Radio Resource Management (RRM). Other tasks such as admission control, load control and packet scheduling also correspond to RRM; however, we will not emphasize them in this section. Power control aims to minimize interference levels in order to maintain an expected transmission quality in the air-interface. The UTRA FDD mode depends on soft blocking to efficiently manage multi-rate services. This takes place according to appropriate RRM algorithms covered in Chapter 4.

8.2.1 Managing Power

Power control becomes more critical in the FDD than in the TDD mode. Thus, this section concentrates primarily on managing power in WCDMA. The impacts on handover are also presented.

In WCDMA all users share the same RF band separated by spreading codes. As a result, each user appears as a random noise to other users. Non-controlled individual power can therefore interfere unnecessarily with those sharing the same frequency band. To illustrate the need for power control Figure 8.1 shows two MSs in the UL. MS₁ gets closer to the BS than MS₂, now if there was no power control both MSs would transmit at their fixed power P_T . But since MS₁ is closer, it would have higher power than that of MS₂ if we assume that the distance of the latter is three times greater than that of MS₁. Thus, if the required SNR (S/N_{required}) is (1/3), then $S/N_1 = 3$ and $S/N_2 = 1$. Thus, MS₂ will suffer the classical near-far effect and may not satisfy the quality of service required in the link. Furthermore, any 3rd MS coming into the cell will not get the its required S/N either, and may even cause MS₂ to drop its S/N even lower. Power control will thus aim to overcome near-far effects and thereby increase capacity with acceptable link quality.

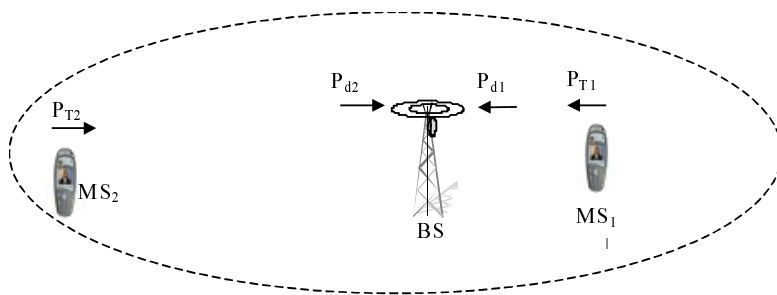


Figure 8.1 Power control to prevent near-far effect.

8.2.1.1 Fast Power Control (FPC)

The FDD mode uses fast power control with 1.5 kHz frequency (i.e. 1500 times/s) in both UL and DL. It operates at a faster rate than any path loss change. The FPC uses the closed-loop option as noted in Chapter 4. We see higher gains of FPC in low mobile speeds than for high mobile speeds, and in received powers than in transmitted powers. At speeds above 50 km/h, e.g. FPC does not contribute much due to the higher multi-path gains. We can find more information about fast power control in [1].

Other gains of FPC depend on diversity, e.g. multi-path diversity, receive, transmit antenna diversity, and macro-diversity. Less diversity implies more variations in the transmitted power. Thus, we get smaller *power rise*¹ in the presence of more multi-path diversity.

In DL macrocell coverage with WCDMA, *power rise* gets critical because it directly intervenes in the required transmission power, which determines the transmitted interference. Hence, to maximize the DL capacity, we should select the quantity of diversity, such that it minimizes the transmission power required by a link, since the received power level does not affect the capacity in the DL.

In the UL, the level of transmission power from the different MSs does have direct impact on the interference to the adjacent cells, and the received power determines the level of interference to other users in the same cell. Diversity in this case does not have much impact, which means that UL capacity of a cell would be maximized by minimizing the required received powers, and the amount of diversity would not affect the UL capacity.

When MSs move at high velocities, the FPC does not follow fast fading; we would require higher received power level to obtain the expected quality. Thus, in this scenario diversity does help to maintain the received power level constant, thereby allowing a lower average received power level to provide the required quality of service.

8.2.1.2 Power Control in Handover (HO)

Before we discuss power control in HO, we briefly review the HO types. The two types of HO in our FDD mode include Softer and Soft HO.

¹ If we define *power rise* as the relative average transmission power in a fading channel compared to the non-fading, while the received power level is the same both in fading and in non-fading channels with ideal power control.

8.2.1.2.1 Softer Handover

As illustrated in Figure 8.2 softer HO occurs when a MS passes through the overlapping coverage of two adjacent sectors of a BS. Communications between the BS and MS take place concurrently through two channels (i.e. one to each sector or cell). The concurrent links use 2 separate DL codes so the signals are perceived by the rake receiver and processed as in multi-path reception, but with the rake fingers generating the corresponding code for each sector.

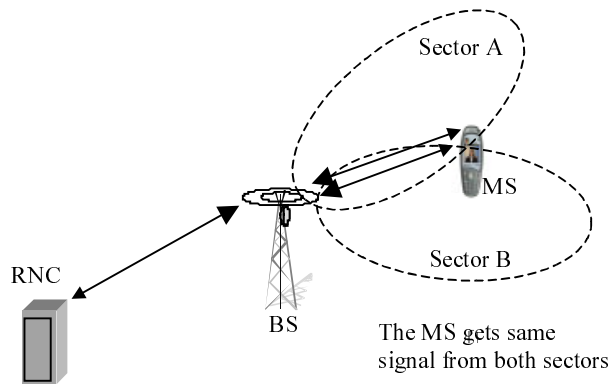


Figure 8.2 Softer handover event.

A similar process occurs in the UL, each BS sector receives the MS code, which gets routed to the same rake receiver for maximal ratio combining. In softer HO we have only one power control loop active per connection.

Softer HO events do not exceed 16% of established links, and in the UL we do not use additional resources except for the extra rake fingers. Neither does the BS need to provide additional DL transmission power to complete the softer HO process.

8.2.1.2.2 Soft Handover

In soft handover, a MS passes through the overlapping cell coverage area of two sectors, which correspond to different BSs, e.g. BS-a and BS-b as illustrated in Figure 8.3. Communications between the MS and BS occurs concurrently through two different channels, i.e. one from each BS. The MS receives both signals by maximal ratio combining Rake processing.

While in the DL *softer* and *soft* HO behave basically in the same way² and the MS does not see any difference between them; in the UL *soft* HO behaves differently. For example, the MS receives the code channel from both BSs. This information then gets routed to the RNC for macro-diversity combining thereby to obtain the same frame reliability indicator provided for outer loop PC, i.e. to select the best frame after each interleaving period within 10–80 ms.

² Thus, soft and softer HO can also take place in combination with each other.

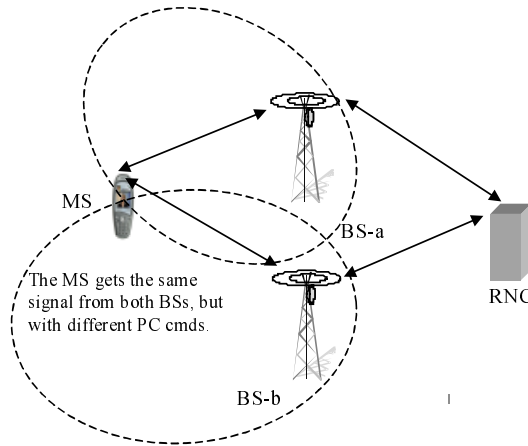


Figure 8.3 Soft handover event.

In general, *soft* HO will not exceed 40% of the links. However, it will not go below 20% either. Thus, we cannot neglect soft HO overhead when dimensioning. For example, we must allocate: extra transmission power in the BS, extra BS rake receiver channels and extra rake fingers in the MS, and extra transmission links between the BSs and the RNCs.

An appropriate provision and/or an efficient FPC management in WCDMA will maintain most of its total capacity³ during HO. In FPC we need to deal effectively with the BS power drifting and the accurate detection of UL power control commands from the MS.

Inaccurate reception of power control commands in the BS due to propagation impacts, such as delay or shadowing will trigger undesired power events from the BSs, e.g. increasing power when expecting power decrease. This power drifting will degrade *soft* HO. On the other hand, the RNC can control such drifting by limiting the power control dynamics or by obtaining DL reference transmission power levels from the BSs. Then send this reference value for the DL transmission powers to the BSs.

In the UL all BSs send independent power control commands to the MS to control its transmission power. The MS can then decrease its power if one BS demands so, and apply maximal ratio combining to the data bits in soft HO since the same data is sent from all soft HO BSs.

8.2.1.3 Outer Loop Power Control

We use outer loop power control to keep the quality of the FPC communication at the required level. An excessive high FPC quality will waste capacity. Outer loop power control applies to both UL and DL, since FPC also applies to both⁴. While FPC has a frequency of 1.5 kHz, the outer loop power control has a frequency range of 10–100 Hz.

³ Otherwise up to 40% of the total capacity can decrease.

⁴ In IS-95 outer loop power control applies only to the UL because there is no fast power control in DL.

8.2.1.4 Conclusions

In the preceding sections, we have highlighted power control and handover aspects primarily to indicate their importance when planning for capacity and coverage. Other source such as [2–6,15,16] cover more in depth power control issues.

Other related areas of radio resources for the FDD mode, e.g. admission control are found in [7–10]. Sources that apply to the resource management of the TDD mode, are found in [11–14].

8.3 NETWORK MANAGEMENT

8.3.1 Introduction

Forthcoming 3G systems such as UMTS will serve as multi-technology platforms⁵ for new and innovative services. These services will appear within a highly competitive market demanding uniqueness at the best price. To meet the demands, it will be imperative to maintain efficient operational costs through an appropriate NE management system. We will obtain the ideal NMS only through the right combination of NE element control techniques. On the other hand, because of the wide spread 2G networks evolving into 3G, managing UMTS NE will not be the only challenge. We also need integrated 2G/3G systems.

8.3.2 Network Management Characteristics

Considering the items on the preceding section, a NMS will have at least the following characteristics:

- Capabilities to integrate and manage 2G NE besides 3G building blocks.
- Support advanced functions and techniques to cope with the multifarious UMTS technology, and maintain diverse service functionality, as well as quality of service provision.
- Have an inherent easy to use man–machine interface to minimize personnel training requirements.
- Support a multiple set of protocols and open interfaces to interact with multi-vendor equipment.

In the context of GSM as a 2G system, a basic set of capabilities will include network management applications in combination with technology specific features to appropriately deploy and operate all components of a complex GSM/GPRS/UMTS network.

8.3.3 A Generic Functional View of a 3G NMS System

Figure 8.4 illustrates a reference architecture of an integrated NMS system capable of managing a combined 2G/3G network. A layered approach allows us to address the complex hybrid system to monitor, i.e. GSM and UMTS NEs and performance.

⁵ For example, IP, ATM; WCDMA, etc.

At the network management level the essential functions would include:

- *Fault control* – control and monitor the function and performance of allocated network resources
- *Ticketing and reporting* – trouble reporting and service assignment to the operations team
- *Set up and configuration* – assist in complex system parameter configuration
- *Resource management* – data and inventory tracking to provide visibility of available physical resources in the network.

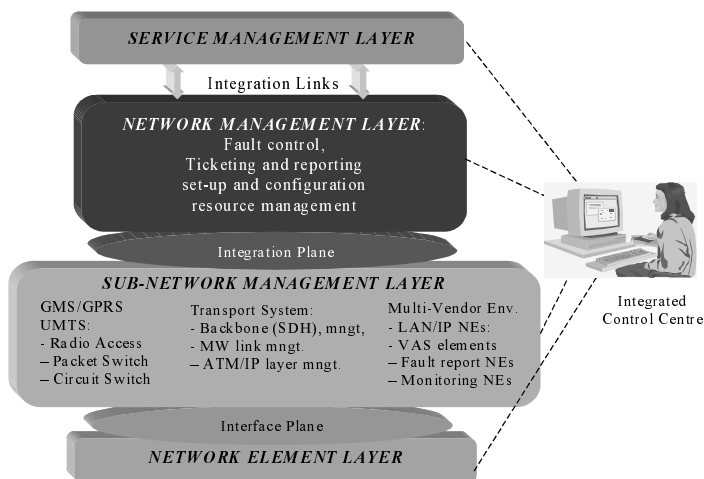


Figure 8.4 A layered NMS architecture reference.

At the sub-network management layer, the integrated architecture will aim to gather different sub-domains into one domain. This blending of different control technologies will provide a unified management process. The result will afford a consolidated view of alarm surveillance; performance and configuration access to all related nodes of the integrated domain. The sub-domains include but are not limited to:

- *GMS/GPRS and UMTS Sub-domains* – incorporating radio access, packet and circuit switching network elements.
- *The Transport system* – has to do primarily with the core transport network incorporating, e.g. a SDH backbone, a set of microwave links, and overlay ATM/IP network running on the SDH ring.
- *The multi-vendor environment* set up stands to support NE from different vendors, which will continue as part of a common element to 2G/3G or evolve through upgrade from 2G to 3G. The setup may incorporate LAN or IP, VAS, and fault report or monitoring NEs.

8.3.4 Main 3G Network Elements for Management

In the following, we describe the components of the network element layer illustrated in Figure 8.4. We start by outlining the elements corresponding to the radio access network. However, because our interest lies primarily with the 3G elements, we describe mainly the elements corresponding to the UTRAN.

8.3.4.1 The UTRAN Building Blocks

The main components of UTRAN (illustrated in Figure 8.5), which would be managed by the integrated management system proposed in the preceding section include:

- 3G Base Stations (BS, in 3GPP called Node B)
- Site solution products, e.g. antennas and Power systems
- Radio Network Controllers (RNC)
- UTRAN Functions (Software for RNC and BS)
- Radio Access Network management

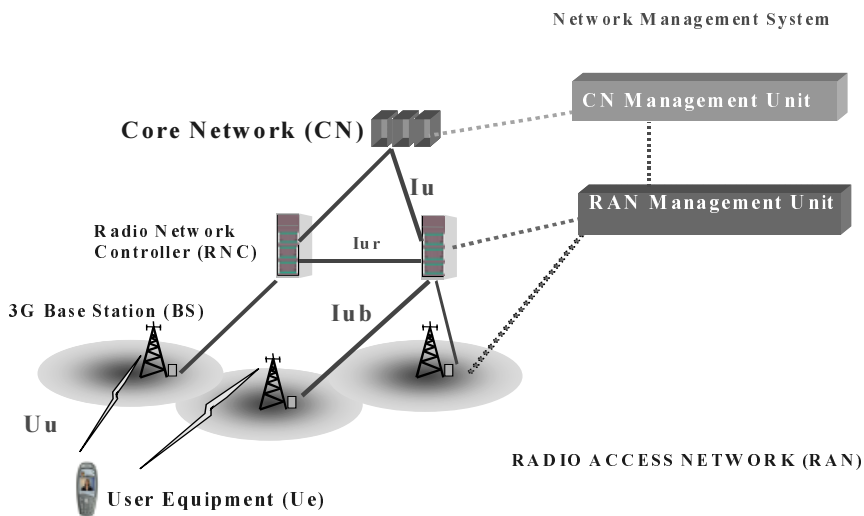


Figure 8.5 Essential UTRAN network elements.

Briefly reviewing from Chapter 3, the RNC takes care of the radio access bearers for user data, the radio network and mobility. The 3G BS provides the radio resources. The main interfaces are: Iu interface between RNC and CN and Uu between User Equipment (UE) and NodeB or 3G BS. Within UTRAN, the RNCs communicate with each other over Iur and with 3G-BSs over Iub. The key functions to manage are thus:

- *The Radio Access Bearer (RAB)* functionality provides the CN with a set of services between the core network and the UE. It offers RABs appropriate for voice, CS data and PS data, including required information processing and signalling. It also supports multiple RAB connections to one UE, e.g. both voice and packet switched services concurrently to one MS.

- *Link control functions*, i.e. paging, signalling channel management, RAB services, allocation and control of radio and other RAB resources.
- *Mobility functions* include: handover, cell re-selection, macro-diversity combining and location update management.
- *Capacity management functions*, i.e. control the trade⁶ off between capacity, quality and coverage. The essential tasks are:
 - *Capacity control* handling allocation of the radio resources, which depends upon resource information from involved cells and neighbouring cells.
 - *Admission control* managing access of new users into the network based, it depends on network load status, subscriber priorities and resource availability.
 - *Congestion control* reducing load in high load situations, e.g. by queuing or delaying packet or best effort traffic.
 - *Quality control* based on power control features.
- *Transmission and Interface control* will aim to manage the logical interfaces, Iu, Iur, Iub, which can flexibly be mixed on the physical transport. For example, we can use the same links for access to the CN to carry Iur, or concentration of traffic to several 3G BSs on one physical link.

8.3.4.2 The Core Network (CN) Building Blocks

The management of the CN components in this example take into account the horizontal integration of functional elements. As illustrated in Figure 8.6, the architecture has a total separation of the payload transport and traffic control into the user plane and the control plane, respectively. Where the media gateways constitute the centre components in the 1st plane, and switching servers (e.g. MSC; SGSN servers) in the data base platforms (e.g. HLR) in the second the 2nd plane. In the user plane, we aim to manage the traffic flow; and in the control plane, we will make sure that the traffic intensity does not overwhelm system boundaries.

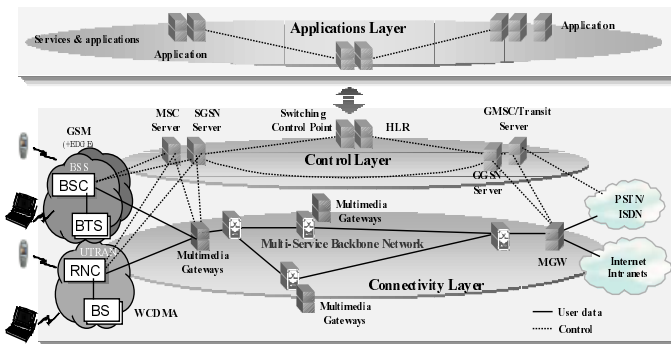


Figure 8.6 3G-CN elements for integrated management.

⁶ A precise management and control of the trade is critical to for the FDD mode or WCDMA.

8.3.4.2.1 *Media Gateway Nodes (MGW)*

The MGW nodes as constituents of the user plane handle CS and PS information and connect to the fixed network for CS traffic (ISUP) and PS traffic (internet/corporate LANs etc.), and to the RAN through the RNC. Various traffic control nodes connecting through H.248 links (Chapter 6) manage the MGW.

8.3.4.2.2 *Traffic Control Servers*

The traffic control servers include CS and PS servers.

The MSC Server Nodes

The MSC server controls the CS traffic in the MGW, including traffic transported on an IP/ATM backbone. The NMS will need thus to capture MSC server functions such as typical MSC functions, GMSC, VLR and signalling functions.

Packet Traffic Control Nodes

The two PS servers include a Serving GPRS Support Node (SGSN Server) and a Gateway GPRS Support Node (GGSN). These server nodes maintain and update contexts for all attached users of packet data services. In the case of the SGSN server, the contexts focuses primarily on macro-mobility, while in the GGSN the contexts deal with the type of network connections.

8.3.4.2.3 *The Subscriber Data Base (HLR)*

The HLR serves as common platform for CS traffic servers (i.e. MSC servers) and the PS traffic servers (SGSN servers and GGSN nodes). It stores subscriber data downloaded to the nodes, from a domain where a subscriber presently roams.

8.3.4.3 *Conclusions*

In the preceding sections, we have outlined mainly the types of 3G functions that an integrated NMS will have to capture. Thus, we assume that a new 3G NMS will incorporate the typical 2G functions from GSM systems, for example, and seamlessly integrate them into its control mechanism. Many of the 3G logical functions will have the same operation principle as that of the 2G. However, the separation of the control and users planes will bring a new dimension to managing a network.

8.4 UMTS NETWORK OPTIMIZATION

Network optimization will depend on the operating environment, the loads for which we design the network, and the appropriated allocation of resources.

The operating environment cannot neglect interference from adjacent networks, assuming the internal network interference is under control. Thus, in the following before we address or review capacity or load enhancing options, and efficient ways to allocate resources, we deal briefly with multi-operator interference issues.

To maximize the performance of the FDD (i.e. WCDMA) system, we need a minimum spectrum mask for a transmitter and highest selectivity for a receiver in the MS and BS, in order to minimize adjacent channel interference. In this context, we define the Adja-

cent Channel Interference Power Ratio (ACIR) as the ratio of the transmission power to the power measured after a receiver filter in the adjacent channel(s). Where we measure both the transmitted and the received power with a root-raised cosine filter response with roll-off 0.22 and a bandwidth equal to the chip rate as described in Chapter 4. ACIR occurs due to imperfect receiver filtering and a non-ideal transmitter. In the UL we get ACIR from the non-linearities of the MS power amplifier, where intermodulation originates adjacent to channel leakage power. In the DL, the receiver selectivity of the FDD terminal will have great impact on ACIR. Technical specifications in Ref. [17] recommend for both UP/DL are 33 dB for adjacent carriers with 5 MHz separation, and 43 dB for the 2nd adjacent carrier with 10 MHz separation.

8.4.1 ACIR Impacts in a Multi-Operator Environment

Non-located BSs of two different operators can originate near-far effects; in particular, when a MS closer to another operator's BS stays far from its own BS. Despite the usage of different carriers, total interfering signal suppression will not be possible. Thus, the BS receiving the interference cannot control the output power of the interfering MS because it belongs to another operator.

As a result there exists a need for Adjacent⁷ Channel Protection (ACP), which is the ratio of the transmitted power and the power measured after a receiver filter in the adjacent channel. The ACP results from the combination of out-of-band emission and receiver selectivity, where these two quantities need balance, to prevent over-specification.

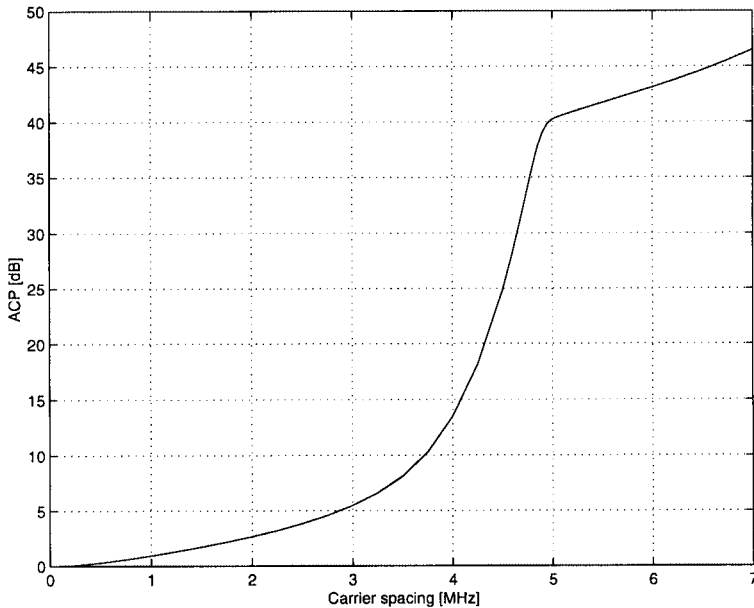


Figure 8.7 ACP as a function of carrier spacing [18].

⁷ Adjacent channel may refer to the channel closest to the assigned channel, and the 2nd adjacent channel.

In [18] measurements have been made on a commercial PA, and a model has been derived. Integrating the power spectrum over a receiver filter gives the ACP. For different offsets of the filter, the ACP as a function of carrier spacing was obtained. Assuming a receiver filter matched to the transmitter pulse, Figure 8.7 illustrates the ACP, where the curve gets steep just below 5 MHz, and where it becomes flatter thereafter. This implies that the adjacent channel interference (ACI) increases when the carrier spacing falls below 5 MHz and that the ACI marginally decreases for carrier spacing larger than 5 MHz.

It also implies that capacity loss depends on ACP, cell radius, and relative location of the BSs. Thus, co-location of BSs will ease the near-far problem, and that location of a base station on another operator's cell border gives the worst case. Consequently, site sharing with other operators in WCDMA will be imperative. In conclusion, about 35–40 dB ACP gives a worst case capacity loss of 5–10% for the worst case located base stations.

In UTRA, e.g. FDD the nominal carrier spacing of 5.0 MHz can get adjusted in steps of 200 kHz according to the needs of the adjacent channel interference. Figure 8.8 illustrates this possibility.

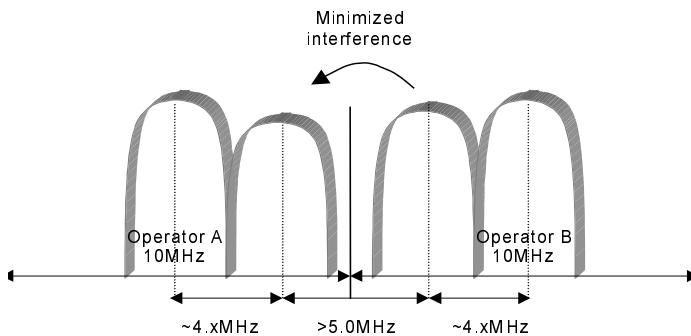


Figure 8.8 Carrier spacing example between operators.

Therefore, the process of optimizing the network to minimize ACIR may involve maximum site sharing between operators, a dynamic adjustment of the carrier spacings within a network as well as inter-networks, de-sensitization of the BS receiver, inter-frequency handovers when excess ACIR occurs, and ideal antenna location or co-location.

8.4.2 Enhancing and Managing Capacity

Efficient network dimensioning and optimized BS deployment will complement a very good admission control system to manage and maximize capacity successfully.

Admission control has to do much with the RNC to maintain the requested quality of service of the radio links. It monitors the load (i.e. downlink power and UL interference level) on the cell carrier. RNC functions desiring to allocate radio resources on given

cell carrier request to the admission control for the target cell resources. The functions triggering the admission control tasks may include:

- signalling connection setup (allocating for radio resources administration)
- soft/softer HO
- intersystem HO (e.g. GSM to UMTS or vice verse)
- radio link co-ordination (i.e. setting up radio resources)
- channel type switching/selection (e.g. common to dedicated channel)
- paging
- code allocation/re-allocation.

The admission control also prioritizes depending on the type of calls, e.g. emergency calls will precede a normal (voice or data) call. System control tasks such as handovers will also have priority over normal data calls. The admission control will thus discriminate between administrative tasks such as soft/softer HO, service calls and/ or emergency calls.

Traffic levels representing the load of service request to a network will start the operation of the admission control for management of congestion control and link quality management. Thus, in the following we briefly discuss the factors affecting load control.

8.4.2.1 Load Control Analysis

We measure traffic intensity generating a certain load in Erlangs. We can define the latter as the average number of simultaneous calls in a given period. For example, we can measure a series of calls lasting x seconds within 1 h in Erlangs, i.e.:

$$\frac{\sum \text{calls} \times x \text{ s}}{3600 \text{ s}} = \text{Erlangs} \quad (8.1)$$

If what we can measure of the generated traffic in a network is the carried load in Erlangs, then the offered load can be expressed as:

$$\text{offered\_load} = \frac{\text{carried\_load}}{(1 - \text{blocking\_rate})}, \quad (8.2)$$

where the blocking rate refers to the measured blocking probability measure in a given BS. The blocking probability, or grade of service can be quantified through the Erlang-B formula, i.e.

$$P_{\text{blocking}} = \frac{\frac{\rho^C / C!}{\sum_{i=0}^C \rho^i / i!}}, \quad (8.3)$$

where the C is the number of channels and ρ is the offered load. Based on the typical assumptions [19] for the offered traffic we can formulate the offered load as

$$\rho = \frac{\lambda}{\mu}, \quad (8.4)$$

where λ is the Poisson arrival rate of λ calls/s, and $1/\mu$ is the exponential call service time of $(1/\mu)$ s/call. Notice that the logic applies mainly to CS calls; for data calls we need to use other distributions described in Chapter 2. Applying the blocking principle to the WCDMA we can now define soft blocking and hard blocking.

Soft blocking assuming UL limitation can be defined from the following assumptions: perfect power control, each subscriber requiring the same E/No, constant number of users in the cell. However, none of the three assumptions will hold in practice specially with mixed services. Nevertheless, we use them here. Then, we can say that soft blocking occurs when the total interference level exceeds the background noise level by a given quantity. Then from the logic for capacity started in Chapter 7 we define the total interference as:

$$I_{\text{total}} = ME_b R(1 + \eta) + N, \quad (8.5)$$

where M is the number of users in the same cell, E_b equals the energy per bit of the signal, R is the baseband data rate, N is the thermal noise power, and η is the loading factor defined in Chapter 7 [20], i.e. the interference ratio brought in by the MSs served by other cells to interference generated by MSs served by the home cell. Then, if we get soft blocking when I_{total} exceeds the background noise level by a given quantity $1/x$, we can prevent soft blocking if

$$I_{\text{total}} \geq ME_b R(1 + \eta) + N. \quad (8.6)$$

If we define $x = N/I_{\text{total}}$, we can define the soft capacity as a function of the maximum allowable interference level for our simple model, which would not necessary apply in practice, but can serve to understand the concept of load control,

$$M \leq \left(\frac{W/R}{E_b/I_o} \right) \left(\frac{1-x}{1+\eta} \right). \quad (8.7)$$

To conclude the load control, we next define hard blocking. Here we assume that the actual ratio of I_{total} to background noise is negligible, i.e. there exists very low probability for soft blocking.

To define hard blocking we first define the actual load and the conditional load. The first refers to the real load passing through a network in action, while the 2nd refers to the actual load plus the administrative traffic such as soft handovers. Then we introduce a traffic factor as

$$\text{traffic\_factor} = \frac{\text{actual\_load}}{\text{conditional\_load}}. \quad (8.8)$$

Then, the actual load is

$$\text{actual\_load} = \text{traffic\_factor} \times \text{conditional\_load}. \quad (8.9)$$

Thus, we have a simple way to quickly determine the number of voice channels, e.g. we can use the actual load in conjunction with Erlang's C formula for the blocking rate. This formula implies that a subscriber who is blocked will re-originate calls until he/she goes through. This blocking probability is equivalent to the probability that the call has been delayed, and it can be expressed as:

$$P_{\text{delayed}} = \frac{\rho^C / C!}{\frac{\rho^C}{C!} + \left(1 - \frac{\rho}{C}\right) \sum_{i=0}^{C-1} \frac{\rho^i}{i!}}. \quad (8.10)$$

8.4.2.2 Conclusions

Enhancing and managing capacity has a lot to do with the condition of the projected traffic and the state of the network. The admission control and traffic congestion control in UTRA has more than voice traffic to deal with. Thus, the preceding formulation needs to be extrapolated to the mixed traffic before it can be applied directly. The traffic factors need to take into account background traffic.

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9 TOWARDS IP BASED NETWORKS

9.1 BACKGROUND

In the preceding chapters we covered UMTS in the context of the 3GPP Release 99 specifications. This chapter covers the forthcoming releases of UMTS, primarily Release 4 and 5 formerly Release 00. However, before we describe the reference architecture we outline the vision of the UMTS technical specification evolution from Ref. [1].

9.1.1 The UMTS Release 99 and Medium Term Architecture

9.1.1.1 Release 99

Figure 9.1 illustrates the service drivers of the UMTS architecture for R99 and future releases starting with R00. The latter has now been broken into Release 4 and 5.

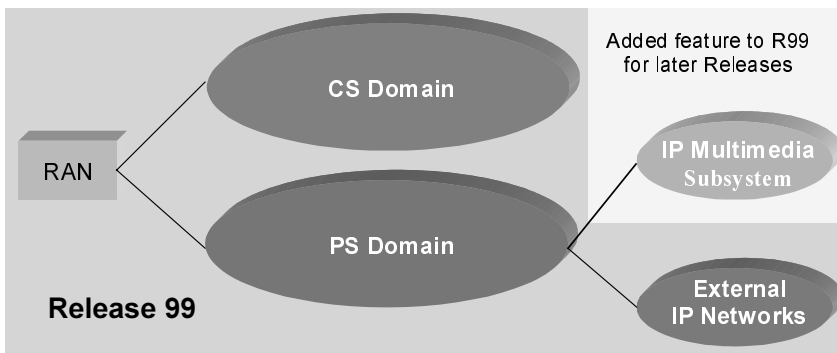


Figure 9.1 Release 99 and medium term architecture.

The service drivers for R99 based on Ref. [1] include: compatibility with GSM, access to high-speed data services, and managed QoS. The CS domain provides circuit-oriented services based on nodal MSCs (an evolved GSM), while the PS domain provides IP-connectivity between the mobiles and IP networks (an evolved GPRS).

9.1.1.2 Release R4 and R5

The medium term vision (starting R4 and R5) has the added feature of IP multimedia as illustrated in Figure 9.1. The service drivers include: compatibility with Release 99, addition of IP-based multimedia services, and an efficient support of voice-over-IP-over-radio for the multimedia service, but not necessarily fully compatible with the telephony service and its supplementary services.

- The CS domain retains and provides 100% backwards compatibility for R99 CS domain services. We can implement this domain through the evolution of MSCs, or MSC servers and a packet backbone.
- The PS domain also retains and provides IP connectivity. It gets upgraded to support QoS for IP-multimedia services.

The added IP-multimedia subsystem provides new IP multimedia services that complement the services provided by the CS domain. These services will not necessarily align with the CS domain in the medium term.

9.1.2 The Long Term UMTS Architecture Vision

After the evolution of R99 culminating with R00 (R4 and R5) we aim to have an integrated platform based entirely on a packet switched system. The service drivers for the long term include: migration of many users to IP-multimedia services and wide-spread adoption of IP-multimedia outside UMTS.

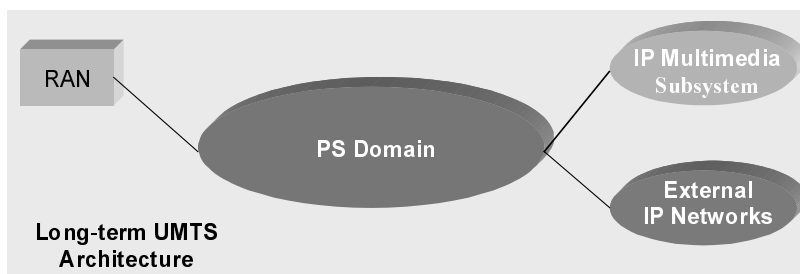


Figure 9.2 Long term UMTS architecture.

By this time, we assume that the IP multimedia subsystem has evolved to the degree that it can practically stand as a substitute for all services previously provided by the CS domain. Here we retain the PS domain but phase out the CS domain. Whether the latter can be achieved in its integrity including all security aspects remains to be seen, since it is still under standardization or technical specification.

9.1.3 The All IP and Service Evolution

As noted in Chapter 1, the widespread usage of Internet and IP's ability to communicate between different networks has made IP a convergence layer to evolve from a simple data platform to larger structure for services. By aiming to reach further than the circuit switch, IP now leads mobile communications to new dimensions.

The IP protocol has opened up a whole range of wireless applications, which will allow service providers and operators to develop totally new and innovative services while enhancing their existing infrastructures. Thus, the main drivers for IP services include a full range of new multimedia applications beside IP telephony.

9.1.3.1 Transition to All IP Services

Passing to ALL IP multimedia services will take some time; therefore both classical CS mobile services and IP multimedia services will co-exist concurrently. As a result, net-

works will have to support traditional CS services and new PS services such as multimedia with the variety of terminals these services will bring in order to offer seamless roaming between evolving 2G networks and optimized 3G networks. This means that Release 2000 (now broken up into R4 and R5) will need to support service offerings while remaining independent from transport technology. The R00 platform will have to support at least the following [2]:

- hybrid architecture
- network evolution path
- new capabilities
- IP based call control
- real-time (including voice) services over IP with end-to-end QoS
- GERAN (support for GSM/EDGE Radio Access Network)
- services provided using toolkits (e.g. CAMEL, MExE, SAT, VHE/OSA)
- backwards compatibility with Release 99 services
- no degradation in QoS, security, authentication, privacy
- support for inter domain roaming and service continuity

The future UMTS releases will have new and improved enabling mechanisms to offer services without using circuit switched network capabilities, as shown in Figure 9.3. Here, we assume that the set of services available to the user, and the quality of the services offered will match those available in networks that use CS enablers.

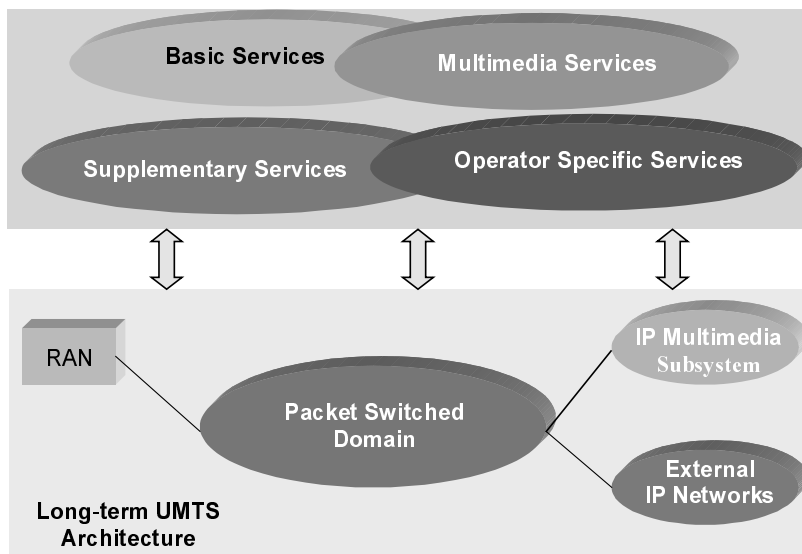


Figure 9.3 Services in the forthcoming UMTS network architecture.

9.1.4 **Classifying Releases 4 and 5 Services**

Following the suggested classification in [2], we can divide basic services into circuit tele-services [3] and bearer services [4], where both can utilize standardized supplementary services [5]. These basic services have not changed much in 2G networks like GSM. GPRS [6] provides IP bearer services, and SMS, USSD and UUS can also be considered as a bearer service for some applications.

IP multimedia services (including IP telephony) using GPRS as a bearer, correspond to the new services in R4 and R5. Supplementary services for IP multimedia services do not get standardized but they can get implemented using the toolkits or at the call control level.

Value added non-call related services (not necessarily standardized) correspond to a large variety of different operator specific services. These services may use proprietary protocols or standardized protocols outside 3GPP.

To create or modify the above services (both call and non-call related services), service providers or operators may utilize standardized 3GPP toolkits (e.g. CAMEL or LCS) or external solutions (e.g. IP toolkit mechanisms). Pre-payment can serve as an example of an application created with toolkits that may apply to all of the above service categories. Additional information and details on general and IP multimedia requirements can be found in Ref. [2].

In the following we introduce the reference architecture which will realize the type of services presented above and illustrated in Figure 9.4.

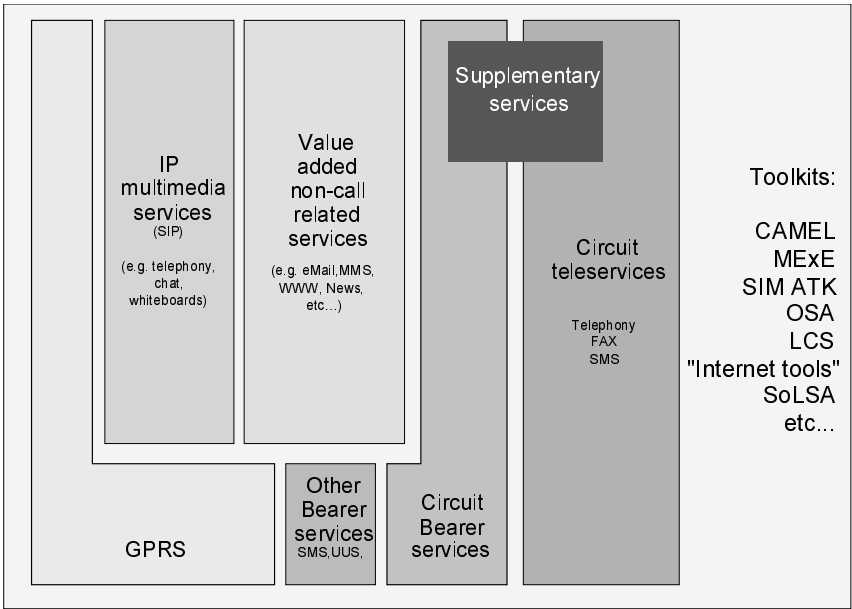


Figure 9.4 Service classification [2].

9.2 RELEASE 00 REFERENCE ARCHITECTURE

While the standardization groups have split R00 in R4 and R5 in order to achieve its specification pragmatically in phases, here all for practical purposes we will refer to them as R00. Hence, all forthcoming notation based on Ref. [7] is addressed as R00.

To achieve access independence and to maintain a smooth interoperation with wireline terminals across the Internet, R00 aims conformance as far as possible with IETF Internet standards for cases where an IETF protocol has been selected, e.g. SIP. To support VoIP, the architecture assumes that the standard includes a minimum set of mandatory codecs and minimum set of mandatory protocol options. Specifications in Ref. [7] outline the principles of the reference architecture; thus, here we provide mainly an overview of the architecture and its components.

9.2.1 Overview of Release 00 Architecture

Figure 9.5 provides a generic view of the R00 architecture. Notice that the following interfaces correspond also to the R00 reference architecture: E interface – between MSCs (including MSC server/MGW); G interface – between VLRs, G interface; Gn interface between SGSNs, Gm interface – between CSCF and UE; Gs interface (optional) between MSC (or MSC server) and SGSN.

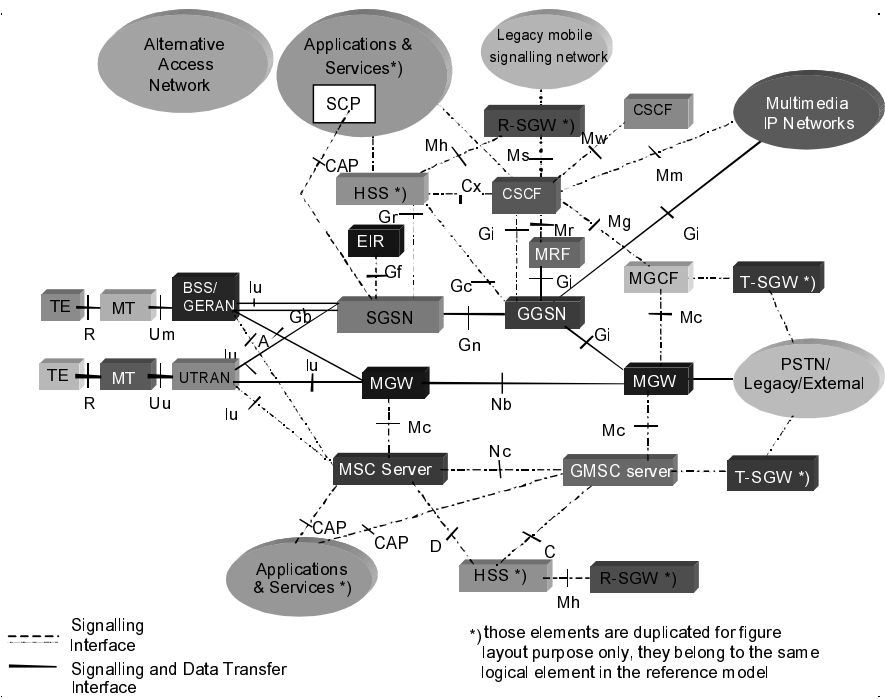


Figure 9.5 Reference Architecture for Release 00 (R4 and R5) after Ref. [7].

9.3 FUNCTIONAL ELEMENTS

The presentation of the R00 functional components in the following corresponds to a direct extract from Ref. [7] in order to keep the vocabulary and the context of the recommendations consistent. However, we also describe its prospective implementation and some key applications as illustrated Figure 9.6.

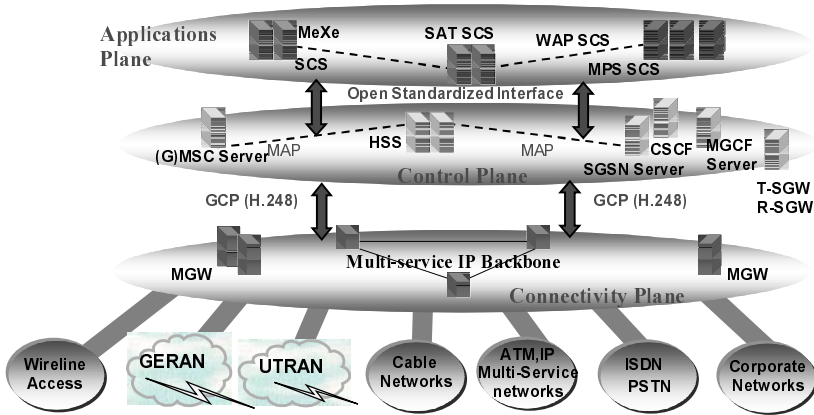


Figure 9.6 The Release 00 integrated architecture.

9.3.1 Call State Control Function (CSCF)

Logically, the CSCF can be divided into three sub-components: the *serving* CSCF (S-CSCF), the *proxy* CSCF (P-CSCF); and the *interrogating* CSCF (I-CSCF).

We use the first to support mobile originated/terminated communications. It provides the Serving Profile Database (SPD) and Address Handling (AH) functionality. The serving CSCF supports the signalling interactions with the UE through the Gm interface. The HSS sends the subscriber data to the serving CSCF for storage. It also gets updated through the latter.

The CSCF acts as the central point of the IP multimedia control system; as well as general call control (setup, supervision, and release). It triggers user controlled supplementary services and call leg handling controlled by user call control supplementary services, e.g. three party call using Multimedia Resource Function (MRF). In addition, it handles user charging and security.

We use the *Interrogating* CSCF (I-CSCF) for Mobile Terminated (MT) communications and to determine routing for mobile terminated calls. With its function always located at the entrance to the home network, we can compare this (I-CSCF) to the GMSC in a GSM network. The I-CSCF interrogates the HSS to get information to enable calls going to the serving CSCF. The interrogating CSCF provides the Incoming Call Gateway (ICGW) and AH functionality.

The *proxy* CSCF, which we may compare to the visited MSC in a GSM network, manages address translation/mapping and handles call control for certain types of calls like emergency calls, legally intercepted calls, etc.

MT communications can use both serving CSCF and interrogating CSCF functionality, while MO communications do not require the interrogating CSCF functionality. Both serving CSCF and interrogating CSCF components may come in a single CSCF when needed. We can summarize the CSCF functions from Ref. [7] as follows:

ICGW (Incoming Call Gateway)

- acts as a first entry point and performs routing of incoming calls;
- incoming call service triggering (e.g. call screening/call forwarding unconditional) may need to reside for optimisation purposes;
- query address handling (implies administrative dependency with other entities);
- communicates with HSS.

CCF (Call Control Function)

- call set-up/termination and state/event management;
- interact with the Multimedia Resource Functions (MRF) in order to support multi-party and other services;
- reports call events for billing, auditing, intercept or other purpose;
- receives and process application level registration;
- query address handling (implies administrative dependency);
- can provide service trigger mechanisms (service capabilities features) towards application and services network (VHE/OSA);
- can invoke location based services relevant to the serving network;
- can check whether the requested outgoing communication is allowed given the current subscription.

SPD (Serving Profile Database)

- interacts with HSS in the home domain to receive profile information for the R00 all-IP network user and may store them depending on the SLA with the home domain;
- notifies the home domain of initial user's access (includes, e.g. CSCF signalling transport address, user ID, etc.; needs further study);
- may cache access related information (e.g. terminal IP address(es) where the user may be reached, etc.).

AH (Address Handling)

- analysis, translation, modification if required, address portability, mapping of alias addresses;
- may do temporary address handling for inter-network routing.

9.3.2 Home Subscriber Server (HSS)

The Home Subscriber Server (HSS) serves as the master database for a given user. It contains the subscription related information, to support the network entities actually handling calls/sessions, e.g. it could provide support for the call control servers to complete routing/roaming procedures by solving authentication, authorization, naming/addressing resolution, location dependencies, etc.

The HSS holds the following user related information:

- user identification, numbering, addressing and security information (i.e. network access control information for authentication and authorisation);
- user location information at inter-system level; the HSS handles the user registration, and stores inter-system location information, etc.;
- the user profile (services, service specific information. etc.).

Based on the above information, the HSS also supports the CC/SM entities of the different control systems (CS domain control, PS domain control, IP multimedia control, etc.) offered by a service provider or an operator.

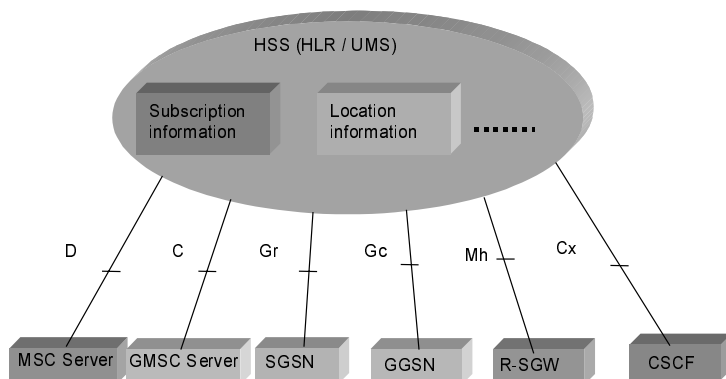


Figure 9.7 A generic HSS structure and basic interfaces.

The HSS can integrate heterogeneous information, and enable enhanced features in the CN for offering to the application and services domain while hiding the heterogeneity, (see Figure 9.7). The main HSS functionality includes:

- user control functions required by the IM CN subsystem;
- the subset of the HLR functionality required by the PS domain;
- and the CS part of the HLR, if it is desired to enable subscriber access to the CS domain or to support roaming to legacy GSM/UMTS CS domain networks.

As illustrated in Figure 9.8, the HSS structure has the following interfaces:

MAP termination: HSS terminates the MAP protocol as described in MAP specifications:

- user location management procedures;

- user authentication management procedures;
- subscriber profile management procedures;
- call handling support procedures (routing information handling);
- SS related procedures, etc.

Addressing protocol termination: the HSS terminates a protocol to solve addressing according to appropriate standards, i.e.:

- procedures for user names/numbers/addresses resolution;
- DNS+ protocol resolution, under definition within the ENUM group in IETF (currently looking into URL/E.164 naming translation, etc.).

Authentication, authorization protocol termination: the HSS terminates authentication and authorization protocols according to appropriate standards, i.e.:

- user authentication and authorization procedures for IP based multimedia services;
- protocol candidate resolution, as it is being defined within IETF.

IP MM control termination: the HSS terminates the IP based MM call control protocol, according to appropriate standards, e.g.:

- user location management procedures for IP based multimedia services;
- IP based multimedia call handling support procedures (routing information handling);
- SIP protocol (or parts related with location procedures).

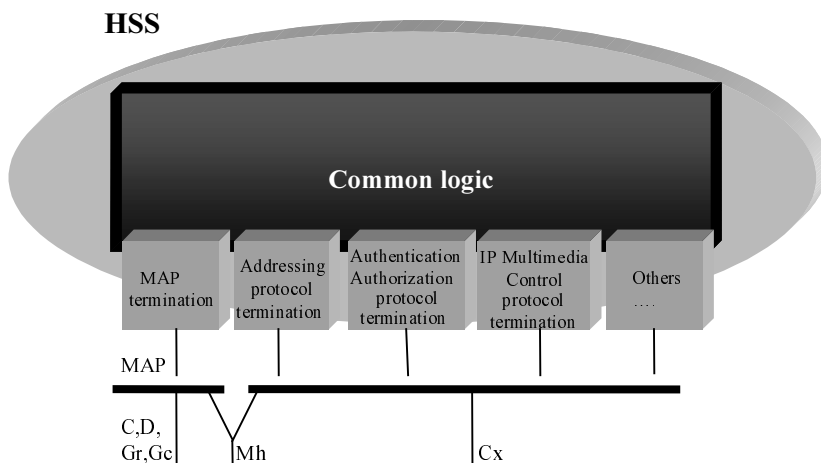


Figure 9.8 A generic HSS structure with protocols over the basic interfaces.

9.3.3 Transport Signalling Gateway Function (T-SGW)

This component serves as the PSTN/PLMN termination point for a defined network. Terminates, e.g. the call control signalling from GSTN mobile networks (typically ISDN) and maps the information onto IP (SIGTRAN) towards the Media Gateway Con-

trol Function (MGCF). The functionality defined within T-SGW should be consistent with existing/ongoing industry protocols/interfaces that will satisfy the requirements:

- maps call related signalling from/to PSTN/PLMN on an IP bearer and sends it to/from the MGCF;
- needs to provide PSTN/PLMN $\leftrightarrow$ IP transport level address mapping.

9.3.4 Roaming Signalling Gateway Function (R-SGW)

The role of the R-SGW concerns only roaming to/from 2G/R99 CS and the GPRS domain to/from the R00 UMTS teleservices domain and the UMTS GPRS domain and does not involve the multimedia domain. According to Ref. [7] the main functions are:

- to ensure proper roaming, the R-SGW performs the signalling conversion at transport level (conversion: Sigtran SCTP/IP versus SS7 MTP) between the legacy SS7 based transport of signalling and the IP based transport of signalling. The R-SGW does not interpret the MAP/CAP messages but may have to interpret the underlying SCCP layer to ensure proper routing of the signalling.
- to support 2G/R99 CS terminals: we use R_SGW services to ensure transport interworking between the SS7 and the IP transport of MAP_E and MAP_G signalling interfaces with a 2G/R99 MSC/VLR.

9.3.5 Media Gateway Control Function (MGCF)

The MGCF serves as the PSTN/PLMN termination point for a defined network. Its defined functionality will satisfy the standard protocols/interfaces to:

- control parts of the call state that pertain to connection control for media channels in a MGW;
- communicate with CSCF;
- select the CSCF depending on the routing number for incoming calls from legacy networks;
- perform protocol conversion between the legacy (e.g. ISUP, R1/R2 etc.) and the R00 network call control protocols;
- assume reception out of band information for forwarding to the CSCF/MGW.

9.3.6 Media Gateway Function (MGW)

The MGW serves as the PSTN/PLMN transport termination point for a defined network and UTRAN interfaces with the CN over Iu. It may terminate bearer channels from a switched circuit network (i.e. DSOs) and media streams from a packet network (e.g. RTP streams in an IP network). Over Iu, the MGW may support media conversion, bearer control and payload processing (e.g. codec, echo canceller, conference bridge) for support of different Iu options for CS services, AAL2/ATM based as well as RTP/UDP/IP based. The main functions include:

- interaction with MGCF, MSC server and GMSC server for resource control;
- ownership and resources handling, e.g. echo cancellers etc.;
- ownership of codecs.

The MGW will have the necessary resources to support UMTS/GSM transport media. It will also have customized H.248 packages to support additional codecs and framing protocols, etc. from other networks besides GSM and UMTS. The MGW bearer control and payload processing capabilities will also support mobile specific functions, e.g. SRNS relocation/handover and anchoring through H.248 protocol enabling. The following principles apply to the CS-MGW resources:

- it shall not be necessary to have the CS-MGW co-located with the MSC server;
- the CS-MGW resources need not be associated with any particular MSC server¹;
- it shall be possible for any MSC server to request resources of any CS-MGW in the network¹;
- it shall be possible for an RNC to connect to the CS-MGW indicated by the MSC msrver.

9.3.7 Multimedia Resource Function (MRF)

The MRF performs:

- multiparty call and multimedia conferencing functions, i.e. would have the same functions as a MCU in an H.323 network;
- performs bearer control (with GGSN and MGW) in cases of multiparty/multimedia conferencing;
- communication with the CSCF for service validation and for multiparty/multimedia sessions.

9.3.8 MSC and Gateway MSC Server

The MSC server includes mainly the call control and mobility control parts of a GSM/UMTS MSC. It has responsibility for the control of MO and MT 04.08CC CS domain calls. It terminates the user-network signalling (04.08 + CC + MM) and translates it into the relevant network-network signalling. The MSC server also contains a VLR to hold the mobile subscriber's service data and CAMEL related data, controls the parts of the call state that pertain to connection control for media channels in a MGW [7].

The GMSC server comprises primarily the call control and mobility control parts of a GSM/UMTS GMSC. A MSC server and a MGW make up the full functionality of a MSC, while the Gateway MSC and a GMSC server and a MGW make up the full functionality of a GMSC.

9.4 REFERENCE POINTS

9.4.1 Cx Reference Point (HSS-CSCF)

The Cx reference point supports information transfer between CSCF and HSS, where the main procedures requiring information transfer between CSCF and HSS include:

- procedures related to serving CSCF assignment;
- procedures related to routing information retrieval from HSS to CSCF;

¹ Extensions to H.248 may be required.

- procedures related to UE-HSS information tunnelling via CSCF.

Details on these procedures can be found in Ref. [7].

9.4.2 Gf Reference Point (SGSN–EIR)

The SGSN server supports the standard Gf interface towards the EIR server. MAP signalling is used over this interface in order to support identity (IMEI) check procedures. For more details refer to TS 23.060.

9.4.3 Gi (GGSN–Multimedia IP Network)

The GGSN supports the Gi interface. It is used for transportation of all end user IP data between the UMTS core network and external IP networks. The interface is implemented according to TS 23.060, the Internet Protocol according to RFC791 and RFC792 (ICMP). Finally, the IPSec according to the following RFCs: 2401, 2402, 2403, 2404, 2405, 2406, 2410 and 2451. IP packets get transported over AAL5 according to RFC 2225 and RFC 1483.

9.4.4 Gn Reference Point (GGSN–SGSN)

We use the Gn interface both for control signalling (i.e. mobility and session management) between SGSN servers and GGSN, as well as for tunnelling of end user data payload within the backbone network.

The GTP-C protocol (running over UDP/IP) used for control signalling can also be included here. The interface is implemented according to TS 23.060 and TS 29.060.

9.4.5 Gm Reference Point (CSCF–UE)

This interface allows the UE to communicate with the CSCF, e.g. register with a CSCF, call origination and termination and supplementary services control.

The Gm reference point supports information transfer between UE and serving CSCF. The main procedures that require information transfer between UE and serving CSCF are:

- procedures related to serving CSCF registration;
- procedures related to user service requests to the serving CSCF;
- procedures related to the authentication of the application/service;
- procedures related to the CSCF's request for core network resources in the visited network.

9.4.6 Mc Reference Point (MGCF–MGW)

The Mc reference point describes the interfaces between the MGCF and MGW, between the MSC server and MGW, and between the GMSC server and MGW. It has the following features [7]:

- full compliance with the H.248 standard, baseline work of which is currently being carried out by ITU-T Study Group 16, in conjunction with IETF MEGACO WG;

- flexible connection handling which allows support of different call models and different media processing purposes not restricted to H.323 usage;
- open architecture where extensions/packages definition work on the interface may be carried out;
- dynamic sharing of MGW physical node resources; a physical MGW can be partitioned into logically separate virtual MGWs/domains consisting of a set of statically allocated terminations;
- dynamic sharing of transmission resources between the domains as the MGW controls bearers and manage resources according to the H.248 protocols.

The functionality across the Mc reference point will require to support mobile specific functions, e.g. SRNS relocation/handover and anchoring. The current H.248/IETF Megaco standard mechanisms will enable these features.

9.4.7 Mg Reference Point (MGCF–CSCF)

The SIP based Mg reference point allows the transfer of session related information between the CSCF and the MGCF. We use this interface to communicate between the IP multimedia networks and the legacy PSTN/ISDN/GSM networks.

9.4.8 Mh Reference Point (HSS–R-SGW)

This interface supports the exchange of mobility management and subscription data information between HSS and R99 and 2G networks. We need this interface to support Release 2000 (R4 and R5) network users who are roaming in R99 and 2G networks, and we implement it with MAP/IP using SCTP and other adaptation protocols developed by the IETF SIGTRAN working group.

9.4.9 Mm Reference Point (CSCF–Multimedia IP networks)

The Mm SIP based reference point stands as an IP interface between CSCF and IP networks. We use the interface, e.g. to receive a call request from another VoIP call control server or terminal. A network in principle will support SIP/SDP between the CSCF and other multimedia networks, with SIP signalling compliant with RFC 2543 and subsequent SIP releases, and with SDP compliant with RFC 2327 and also with its subsequent releases. The interworking between SIP and other protocols, e.g. H.323, occurs at the edge of the IP multimedia network.

9.4.10 Mr Reference Point (CSCF–MRF)

The Mr affords the CSCF to control the resources within the MRF, thus allowing a network to support communication between the CSCF-MRF with either SIP or H.248 depending on the selection by standards. There is interest in the acceptance of IETF protocols such as SIP, e.g. for Mr.

9.4.11 Ms Reference Point (CSCF–R-SGW)

The Ms corresponds to the interface between the CSCF and R-SGW. It will most likely be implemented using M3UA/SCTP.

9.4.12 Mw Reference Point (CSCF–CSCF)

This interface enables the interrogating CSCF to direct mobile terminated calls to the serving CSCF. The protocol supported is SIP according to RFC 2543. However, some additions to SIP beyond what is defined in RFC 2543bis might be required to cope, e.g. with accounting, security or supplementary services requirements.

9.4.13 Nc Reference Point (MSC Server–GMSC Server)

We perform the network–network based call control over the Nc reference point. Examples of this include ISUP or an evolution of ISUP for bearer independent call control (BICC). In the R00 architecture we aim to have different options (including IP) for signalling transport on Nc.

9.4.14 Nb Reference Point (MGW-MGW)

We perform bearer control and transport over the Nb reference point. We may use RTP/UDP/IP or AAL2 to transport user data. In the R00 architecture we aim for different options to transport user data and bearer control, e.g. AAL2/Q.AAL2, STM/none, RTP/H.245.

9.4.15 CAP Based Interfaces

This corresponds to the interfaces from the SGSN to the SCP, from the serving CSCF (and possibly the interrogating CSCF) to the SCP, from the MSC server to the SCP, and the GMSC server to the SCP.

From Ref. [7], the interface from the SGSN to the SCP in the applications and services domain corresponds to the interface defined for UMTS GPRS to support charging application interworking. We require the interface from the CSCF to the SCP to allow the support of existing CAMEL based services. The interface from the MSC server to the SCP, and the GMSC server to the SCP corresponds to the standard interface defined for the CAMEL feature, which provides the mechanisms to support non-standard UMTS/GSM services of operators even when roaming outside the home PLMN.

We can implement the CAP based interfaces by using CAP over IP, or CAP over SS7 as illustrated in Table 9.1.

Table 9.1 Protocol Stack for CAP [7]

CAP		
TCAP		
SCCP		
M3UA	MTP-3B	Narrow-band SS7
SCTP ¹	SAAL	
IP ²	ATM ²	STM ²

¹ In IETF work is ongoing (e.g. SCTP/UDP/IP or directly SCTP/IP).

² The protocols do not correspond to the same OSI layer.

The above includes the interfaces from the GGSN to the HSS (i.e. Gc reference point), from the SGSN to the HSS (i.e. Gr reference point), from the GMSC server to the HSS (i.e. C reference point), and the MSC server to the HSS (i.e. D reference point). We can implement the MAP based interfaces using MAP transported over IP, or MAP over SS7, and we can transport it on the same protocol CAP stacks as illustrated in Table 9.1.

9.4.16 Iu Reference Point

The Iu remains as the reference point between UTRAN and the R00 core network. We realize this reference point by one or more of the following interfaces [7]:

- transport of user data between UTRAN and SGSN takes place based on IP;
- transport of signalling between UTRAN and SGSN takes place based on IP or SS#7;
- transport of user data between UTRAN and MGW takes place based on different technologies (e.g. IP, AAL2), and includes the relevant bearer control protocol in the interface;
- transport of signalling between UTRAN and MSC server takes place based on IP or SS#7.

When we base the Iu_cs on ATM, then we can apply R99 protocols or an evolving version, and when we base the Iu_cs on IP, we need to add new IP transport related protocols as part of the Iu protocols. On the other hand, it will be possible to have a R99 Iu interface with MSCs compliant with R99 specifications in a R00 network.

9.5 MOBILITY MANAGEMENT

9.5.1 Address Management

We can implement a UMTS network as a number of logically separated IP networks, which contain different parts of the overall system. Here we refer to these elements as an IP Addressing Domain. In an IP addressing domain we expect to have nodes with non-overlapping IP address space and be able to route IP packets from any node in the domain to any other node in the domain by conventional IP routing. An IP addressing domain implementation can take place through a physically separate IP network or an IP VPN.

We can interconnect the IP addressing domains at various points where gateways, firewalls or NATs may be present. However, we do not guarantee that IP packets from one IP addressing domain can be directly routed to any interconnected IP addressing domain. Instead inter domain traffic will most likely be handled via firewalls or tunnels. Therefore, different IP addressing domains can have different (and possibly overlapping) address spaces [7]. Figure 9.9 illustrates the IP addressing domains involved in PS domain and IP subsystem services.

UMTS allows usage of different IP addressing domains as shown in Figure 9.9; nonetheless, it is possible that several different IP addressing domains come under a common management. Hence, we can physically implement the different IP addressing domains as a single domain.

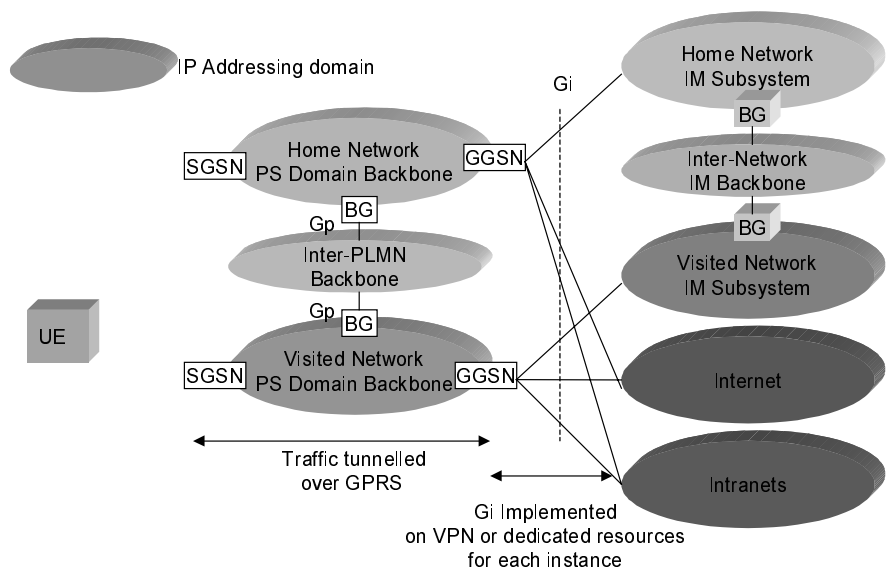


Figure 9.9 IP addressing domains involved in PS domain and IM services [7].

9.5.2 Addressing and Routing to Access IM-Subsystem Services

When a UE gets access to IM subsystem services, an IP address is required, which is logically part of the visited network IM subsystem IP addressing domain. We established this address using an appropriate PDP context, and for routing efficiency this context gets connected though a GGSN in the visited network. Figure 9.10 illustrates the connection between the UE and the visited network IM subsystem.

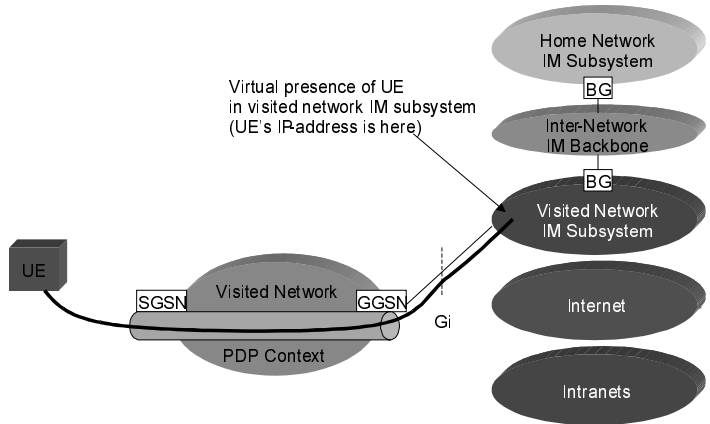


Figure 9.10 UE accessing IM subsystem services in the visited network.

9.5.3 Context Activation and Registration

An IP address allocated to a UE either by GPRS or some other means, e.g. by DHCP, can get used for (but not limited to) the following [7]:

- the exchange application level signalling (e.g. registration, CC) with the serving CSCF from the access network currently used;
- application level registration to an IP MM CN subsystem as an address used to reach the UE;
- an address used to reach the UE for multimedia calls.

In GPRS, we associate the terminal with an IP address when we activate the primary PDP context. This IP address used for the purpose described above can be:

- the IP address obtained by the UE during the activation of a primary PDP context (e.g. if the UE does not have any existing PDP context active or desires to use a different IP address);
- the IP address of one of the already active PDP contexts.

Figure 9.11 illustrates the order in which we execute the registration procedure and how the IP address gets allocated.

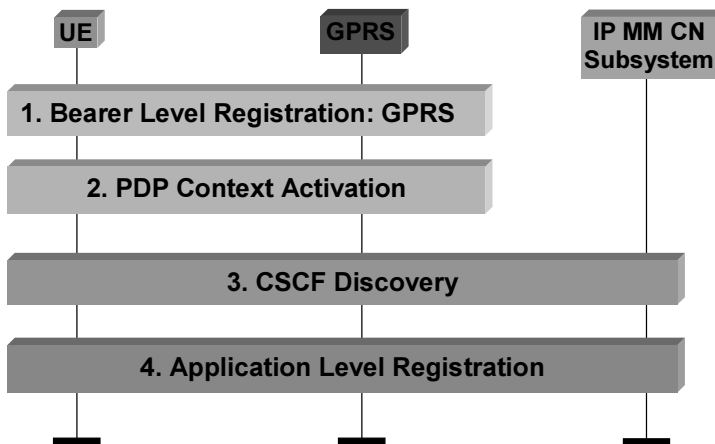


Figure 9.11 Registration of the IP address.

The steps performed include:

1. bearer level registration (e.g. after a MS gets switched on or upon explicit user demand);
2. when the PDP context gets activated, the UE has two options:
 - activate a primary PDP context and obtain a new IP address (e.g. if the UE does not have any existing PDP context active or desires to use a different IP address),
 - activate a secondary PDP context and re-use the IP address of one of the already active PDP contexts;
3. UE performs the CSCF discovery procedure to select the CSCF to register with.

The procedures can have time gaps between them, e.g. the UE may perform PDP context activation and the CSCF discovery, but not the application level registration. The UE may use the activated PDP context for other types of signalling, e.g. for CSCF discovery [7].

- 4. the UE performs application level registration by providing the IP address obtained at step 2 to the CSCF selected at step 3.

In the last step, the signalling IP address gets allocated in association with PDP context activation and not on an incoming call basis, then the selected CSCF becomes the serving CSCF². From the point of view of the latter, the IP address provided by the UE corresponds to the address where the UE is reachable for MT call control signalling and/or any other type of MT signalling. Whether a procedure gets activated individually by the UE or automatically depends on the implementation of the terminal and on the UE's configuration. For example, a UE multimedia application may start the application level registration and steps 2–4 would need to follow in response to support the operation initiated by the application.

9.5.4 Location Management

Figure 9.12 illustrates the registration concept for a R00 subscriber roaming into a UMTS/GSM CN domain.

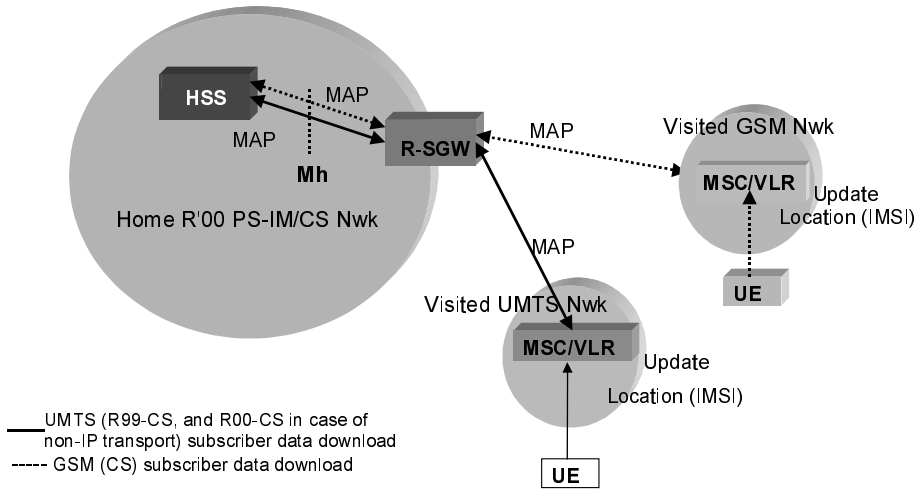


Figure 9.12 A roaming model for registration in a CN domain.

From Ref. [7] Figure 9.13 illustrates the detailed message sequence chart for a UMTS R00 subscriber roaming into a CN domain. The sequence can be summarized as follows:

- 1. The UE initiates the UMTS R99/GSM location update (LU) procedure with the MSC/VLR of the visited network, where the LU message contains the IMSI of the subscriber.

² Note that the S-CSCF can be either in the home or a visited network.

2. The UMTS/GSM authentication gets performed as per the existing UMTS R99/ GSM specifications.
3. The MSC/VLR initiates the MAP location update procedure towards the HSS of the user via R-SGW. The HSS stores the VLR address etc. The message contains IMSI and other parameters as defined in UMTS R99/GSM specifications. The message is passed through the R-SGW transparently while the SS7 to/from IP conversion is performed in the R-SGW.
4. The HSS provides the subscriber data for the roaming user to VLR by sending MAP Insert Subscriber Data message via R-SGW. The message contains IMSI and other necessary parameters as defined in the UMTS/GSM specification. The message is passed through the R-SGW transparently while the SS7 to/from IP conversion is performed in R-SGW.
5. The serving VLR then acknowledges the receipt of the subscriber data to the HSS via R-SGW.
6. The HSS acknowledges the completion of location updating procedure to the MSC/VLR via R-SGW.
7. The MSC/VLR acknowledges the completion of location updating procedure to the UE.
8. The HSS sends the MAP cancel location message to the old MSC/VLR (optional procedure).
9. Location cancellation is acknowledged to the HSS by the old MSC/VLR [7].

The steps 8 and 9 above assume that the UE was previously registered to a CN domain. The MAP messages between the MSC/VLR and HSS get passed transparently via the R-SGW. The R-SGW does not interpret the MAP messages in anyway, but performs only the lower level conversion between SS7 and IP.

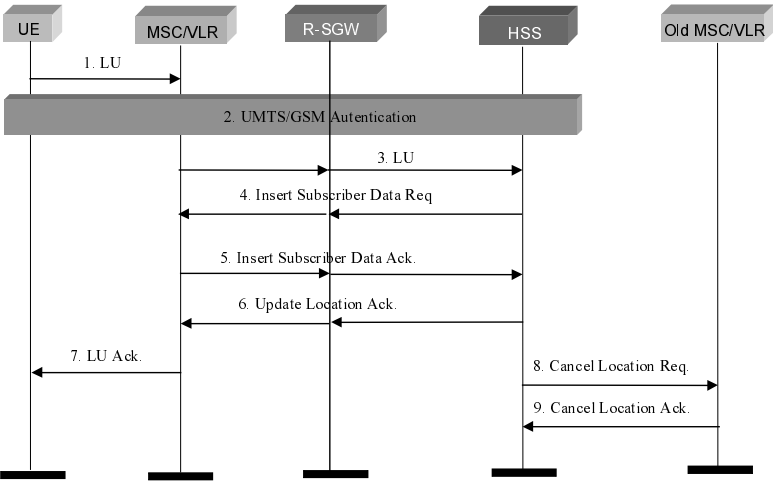


Figure 9.13 Message sequence for roaming into a CN domain [7].

9.5.5 Handover (HO)

For HO of CS services involving the change of CN equipment (only CS-MGW or CS-MGW and MSC-server) the anchor principle applies, i.e.

- The first MSC server involved in a call will become the anchor MSC server for this call during and after HO, and will remain in the call until the call gets released. Every subsequent HO (intra and inter) will then be controlled by this MSC server [7].
- The first CS-MGW involved in a call will become the anchor CS-MGW for this call during and after HO, and will remain in the call until the call is released. The Nc interface gets anchored in the CS-MGW, the correlation between MGW to PSTN and the MGW to UTRAN remain fixed until the call is released [7].

9.6 REGISTRATION ASPECTS

While the final steps for registration are still in process of consolidation at this writing, we can still outline the generic procedures from [7] as follows:

- The R00 architecture will allow the serving CSCFs to have different capabilities and/or access to different capabilities; e.g. a VPN CSCF or CSCFs as a network gets upgraded.
- Service providers or network operators will not need to reveal the internal network structure to another network. Which means that association of the node names of the same entity type and their capabilities, as well as the number of nodes will be kept within the operator's or service provider network. Nevertheless, disclosure of internal network architectures may occur on a per agreement basis.
- A network will not have to expose explicit IP addresses of its nodes, except those belonging to interconnection and security tasks, e.g. firewalls and border gateways.
- For practical purposes and operational simplicity it is desired that the UE will use the same registration procedure(s) within its home and visited networks.
- Likewise, it is desirable that the procedures within the network(s) are transparent to the UE, when it registers with the IM CN subsystem with either its home or visited CSCF.
- Finally, the serving CSCF will understand a service profile and the address of the functionality of the proxy CSCF.

9.6.1 Registration Flows

A preliminary process for use in the architecture definition is described next.

9.6.1.1 Requirements and Assumptions

1. A serving CSCF gets assigned at registration, without precluding additional serving CSCFs (for FFS) or change of CSCF at a later date.

1. We assume that radio bearers are already established for signalling and a mechanism exists for the first REGISTER message to be forwarded to the proxy.
2. The I-CSCF will use the same mechanism in visited and home networks to determine the serving CSCF address based on the required capabilities.

9.6.1.2 Registration Procedures

The registration procedure may be as follows:

- information flow A corresponds to the common initiation of the registration information flows;
- information flow B occurs in where the serving CSCF) is selected to be in the home network;
- information flow C occurs where the serving CSCF is selected to be in the visited network.

Note: the prefix ‘h’ indicates that the functional element is located in the home network, the prefix ‘v’ indicates that the functional element is located in the visited network. When there is no prefix to the functional element the functional element could be in either the visited network or the home network [7].

9.6.1.2.1 Registration Information Flow A : Start of Registration

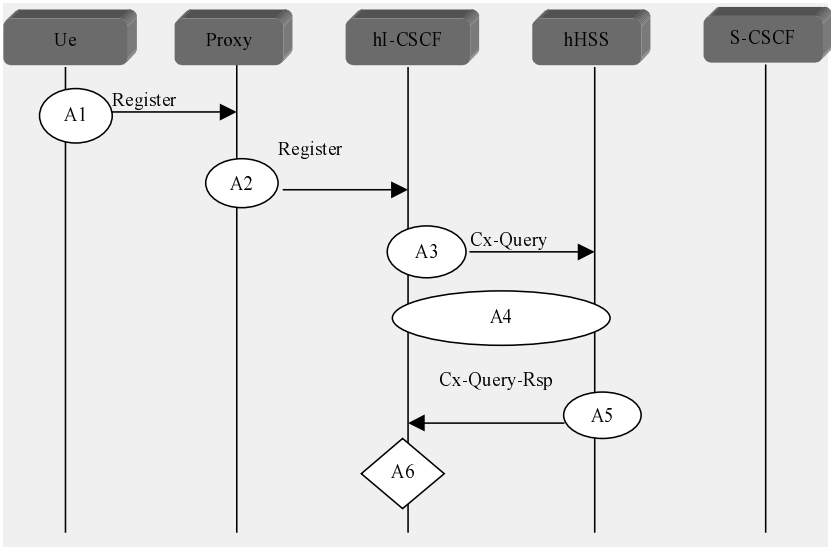


Figure 9.14 Initiating registration.

A1 Register Ue to Proxy

- *Information control:* Ue address, subscriber identity, and home domain name.
- *Initiation of information flow:* Ue secures a signalling channel in the 'PS' domain, and decides to initiate the IPMM subsystem registration.
- *Processing upon receipt:* when the proxy receives the registration request, it examines the home domain name to identify the entry point to the home network. The proxy will then insert its own name in the contact header, as well as insert the local network capabilities and the local network entry point into the register information flow. Then the information flow (A2) initiates. We utilize a name-address resolution mechanism to determine the destination of information flow (A2), with the home domain name as the input.

A2 Register Proxy to h-Interrogating CSCF

- *Information control:* proxy name, subscriber identity, home domain name, local network contact name (the name that the registration would be returned to in the case of visited serving CSCF), and local network capabilities.
- *Initiation of information flow:* receipt of information flow (A1).
- *Processing upon receipt:* when the interrogating CSCF receives the register information flow (1), it queries the HSS with information flow (A3). Corresponds to FFS whether the terminal name, or proxy name, or both is included within this and subsequent register messages.

A3 Cx-Query h-Interrogating CSCF to h-HSS

- *Information control:* proxy name, subscriber identity, local network contact name, and local network capabilities.
- *Initiation of information flow:* receipt of information flow (A2).
- *Processing upon receipt:* information flows (A3) (A4) and (A5) are treated together.

A4 between h-Interrogating CSCF and h-HSS

- *Information control:* subscriber identity, local network capabilities, other information.
- *Initiation of information flow:* receipt of information flow (A2) and (A3).
- *Processing upon receipt:* The h-HSS and the h-interrogating CSCF determine whether the serving CSCF corresponds to the home network or the local network (when the local network is not the home network). When the serving CSCF is in the home network, the name of the serving CSCF gets determined, otherwise it is agreed that the register information flow should be forwarded to the local network to select the serving CSCF.

A5 Cx-Query-Response h-HSS to h-Interrogating CSCF

- *Information control*: required capabilities, other FFS.
- *Initiation of information flow*: receipt of information flow (A3) and procedure (A4).
- *Processing upon receipt*: we assume that after processing information flow (A5) the authentication of the user has been completed and is successful³. The h-interrogating CSCF and the hHSS have determined the serving CSCF, or that the serving CSCF is located in the local network. The behaviour of the h-interrogating CSCF depends on whether the serving CSCF belongs to the home network or in the visiting network. When the serving CSCF corresponds to the home network, we follow information flow B; otherwise we follow information flow C. Figure 9.14 illustrates this sequence. The h-interrogating CSCF has received the name of the HSS and will forward this to serving CSCF for the purpose of downloading the subscriber profile.

A6 h I-CSCF to h S-CSCF or v I-CSCF

- *Information control*
- *Initiation of information flow*: receipt of information flow (A5).
- *Processing upon receipt*: the behaviour of the h-interrogating CSCF depends on whether the serving CSCF is determined to be in the home network of the visited network. When the serving CSCF corresponds to the home network, then the h-interrogating CSCF has the name of the serving CSCF, and information flow (H1) is initiated. When the serving CSCF corresponds to the visited network, the h-interrogating CSCF has the name of the v-interrogating CSCF, which was provided to the home network in information flow (A2), and information flow (V1) is initiated. A name-address resolution mechanism is utilized in order to determine the destination of information flow (H1) or (V1), with the serving CSCF name or v-interrogating CSCF name as the input. See details of information flows A, B and C still under consolidation in [7].

9.7 MULTIMEDIA SIGNALLING

Although multimedia signalling also remains under consolidation within the technical specification bodies of the 3GPP, we can still describe the initial principles based on [7] and extrapolated for R4 and R5 as follows:

- A single call control between the UE and CSCF → For multimedia type services delivered via the PS domain within R4 and R4 architectures, we aim to use a single call control protocol between the user equipment UE and the CSCF (over the Gm reference point).
- Protocols over the Gm reference point → The single protocol applied between the UE and CSCF (over the Gm reference point) within the R4 and R4 architectures

³ Although it may have occurred at an earlier point in the message flows.

will be based on SIP (as defined by RFC 2543, other relevant RFCs, and additional enhancements required to support 3GPP's needs).

- A single call control on the Mw, Mm, Mg → We aim to use single call control protocol on the call control interfaces between MGCF and CSCF, between CSCFs within one operator's network and between CSCFs in different operators' networks.
- Protocols for the Mw, Mm, Mg → We aim to apply SIP⁴ for the single call control protocol applied to the interfaces between MGCF and CSCF, between CSCFs within one operator's network and between CSCFs in different operator's networks.
- UNI versus NNI call control → We may assume that the SIP based signalling interactions between CN elements may be different than SIP based signalling between the UE and the CSCF in some cases if not all.

9.7.1 Support of Roaming Subscribers

Here we assume that R4 and R5 architectures will be based on the principle that the service control for a roaming subscriber is designated by the home network. The serving CSCF can then be located either in the home network or in the visited network as illustrated in Figure 9.15. This assignment of the serving CSCF takes place in the home network during the registration of the UE at the visited network.

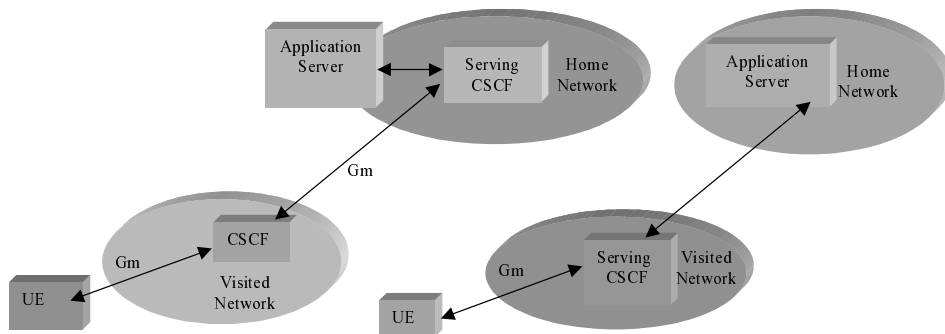


Figure 9.15 Support of the UE via serving CSCF in the home and visited networks.

The R4 and R5 standards will thus support roaming for IP multimedia (IM) users between operators. As noted earlier, to achieve roaming the required serving-CSCF can be located in the visited network or in the home network. The decision as to whether the UE is served by a serving CSCF in the home network or a serving CSCF in the visited network occurs at the home network. According to [7] the following takes place:

⁴ As defined by RFC 2543, other relevant RFCs, and additional enhancements required to support 3GPP's needs.

- When subscribers roam to networks where a serving CSCF does not exist, the roamed to (visited) network will support a proxy CSCF. The proxy CSCF will enable the call control to be passed to the home network based serving CSCF, which will provide service control.
- When subscribers roam to networks where a serving CSCF exists but the home network decides to use a home network-based serving CSCF, the roamed to (visited) network will support a proxy CSCF. The proxy CSCF will enable the call control to be passed to the home network based serving CSCF, which will provide service control.
- When subscribers roam to networks where a serving CSCF exists and the home network decides to use the visited network based serving CSCF solution, the visited network serving CSCF will be used to provide service control to the roamed subscriber.

While the visited network may support a serving CSCF for inbound roamers, it will usually support proxy CSCF for inbound roamers. Thus, if a visited network decides not to offer serving CSCF capability for inbound roamers, then the home network will provide a serving CSCF to support IM roaming.

The home network may provide a serving CSCF for outbound roamers even when a visited network offers the support of a serving CSCF, if so, the visited network provides the proxy CSCF. On the one hand, when users are within their home network, a home network-based serving CSCF provides service control. On the other hand, if the home operator wishes to use home service control for outbound roamers, then a home network-based serving CSCF will be used for outbound roamers' service control [7].

9.7.2 Assignment of Serving CSCF

The home network designates the serving CSCF in the home network or with the help of the visited network, requests a serving CSCF in the visited network. This selection occurs on a per subscriber basis at registration time based on consideration of at least the following factors [7]:

- a. the service capabilities and toolkits supported by the visited network and the home network;
- b. the subscription profile of the subscriber.

9.8 SERVICE PLATFORMS

Services platforms correspond, for example, to VHE/OSA, SAT, CAMEL, etc. However, we will illustrate mainly VHE and SAT.

9.8.1 VHE/OSA

To allow the support of existing CAMEL based services, and to allow the development of new services independent of network domain, the home CSCF will support a CAP

interface. Thus, service providers or operators can choose to register their subscribers to the v-CSCF in networks that also support CAMEL, just as in GSM today. However, when the visited network does not support CAMEL or the capabilities to support a given service, then it will be possible to provide call control in the h-CSCF. This would mean that either the user is registered with the h-CSCF directly, or if the user is registered in the v-CSCF, then the v-CSCF will be capable of forwarding the call set up request to the h-CSCF. The latter will also apply to service providers or operators that choose to have all of their subscribers (including roamers) supported by the h-CSCF.

9.8.1.1 Implementation Options of the VHE/OSA

Figure 9.16 illustrates the main components to realize the Virtual Home Environment (VHE). Notice that the applications servers need a set of enablers and management.

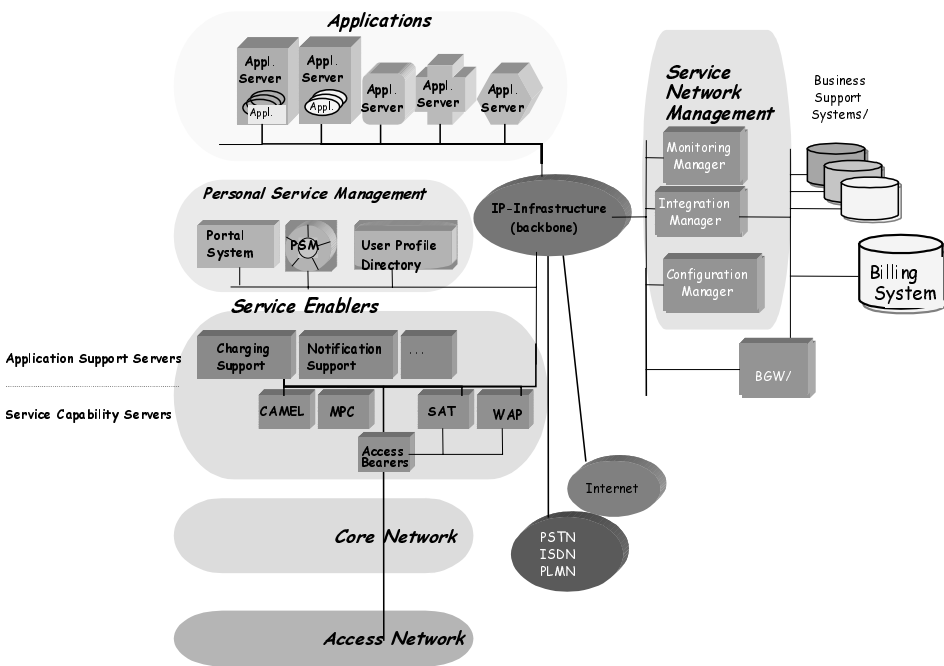


Figure 9.16 IP based service network overview and its relation to core and access networks.

The two key options to follow include:

- Personalization → Subscribers can customize their own service set and its corresponding interaction.
- Transparency → Subscribers can see seamlessly the same service features across network boundaries and between terminals regardless of their location.

To further illustrate the implementation, we can divide the IP-infrastructure service network into functional areas as illustrated in Figure 9.17. Clearly we have the service enablers as shown in Figure 9.16, then personal service management, service network management, and applications.

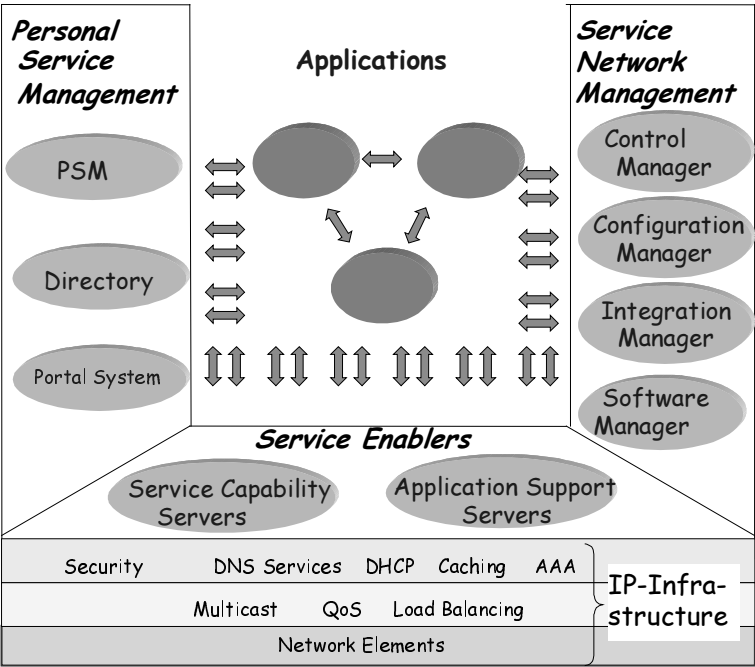


Figure 9.17 Service network functional areas.

The service platform consists of: service network access products, IP infrastructure servers, service enablers consisting of service capability servers, application support servers, and other gateways. It also includes: service network management, which in turn consists of the personal service management, application servers and applications, hardware/service platforms for the servers.

Thus, a launch service network will incorporate the VHE concept as a scenario where one can consistently access telecom and datacom services, through personalized user interface, from any network, on any terminal, anywhere.

9.8.2 The SIM Application Toolkit (SAT)

The SAT browser on the (U) SIM card and the SIM Application Toolkit Service Capability Server (SAT SCS) allow subscribers to access standard Web applications on the Internet, exploiting the widespread Web technology and the inherent security of smart cards, and opening up a whole new range of applications, such as wireless electronic commerce.

A SAT Service Capability Server (SAT-SCS) solution affords service providers or network operators, as well as content providers to supply advanced services using standard tools and either HTML or the Wireless Mark-up Language (WML), where the Wireless Application Protocol (WAP) defines the latter. Thus, 3G terminals complying with the SIM application toolkit can co-exist with WAP terminals accessing the same services. The SAT SCS does not depend on the SIM card supplier.

Coupling the SIM's inherent potential to solve security issues and Over-The-Air (OTA) provisioning for device management, SAT enables an Internet interface for an application provider to create services based on SMS, OTA SIM File management, and SIM application toolkit. On the wireless network side, the SAT SCS uses the SIM toolkit messaging⁵, i.e. for secure transport of the web pages. On the Internet side the SAT SCS connects to one or more applications through HTTP. On the terminal side, the SIM card includes a browser that receives web pages from the applications and converts them into SIM application toolkit commands, thus interacting with the user.

9.8.2.1 SAT Service Capability Server

Further realization of the SAT will include, e.g. a Service Capability Server (SCS), which we can logically divide into two parts, i.e. the request and the push modules. The SCS receives web pages from the application, converts them into byte code and sends them to the browser using GSM 03.48 for the transport. The difference lies in how the web page is fetched from the application. The request module waits for requests from the browser for a certain web page, fetches the page and returns it to the browser. The push module waits for the application to send a web page to a certain browser, thus the user or the browser does not have to take the initiative but it rather comes from the application. A web page could either be written in HTML or WML.

9.8.2.1.1 Browser and Menu

The browser resides on the SIM card of the terminal. It runs byte code strings and converts them into SAT commands. The byte code could be fetched from two places, i.e. they can reside on the SIM card or they could be sent from the SAT SCS. When the browser receives byte code it starts to interpret and convert it into SAT commands. The byte code was originally written in HTML/WML and thus the interpretation can be seen as conversion from HTML/WML to SAT commands.

A key part of SIM application toolkit constitutes the menu inserted into the standard menu structure of the terminal. The menu becomes the starting point for the user to access the web through the terminal. Each menu item belonging to the browser points to a byte code string that is executed when the user selects that menu item.

9.9 QoS ASPECTS

9.9.1 End-to-End QoS Negotiation and Policy Requirements

The main QoS negotiation and policy requirements can be summarized as follows (see other requirements in [7]):

- The UMTS R4 and R5 QoS negotiation mechanisms used for providing end-to-end QoS will be backward compatible with UMTS Release 99. It will not make any assumptions about the situation in external networks, which are not within the scope

⁵ GSM 03.48.

of 3GPP specifications, or about application layer signalling protocols and applications, which may be used on terminal equipment attached to mobile terminals.

- The UMTS network shall be able to negotiate end-to-end QoS also for mobile terminals and applications, which are not able to use QoS negotiation mechanisms other than the ones provided by UMTS. Thus, no changes to non-UMTS specific QoS negotiation mechanisms will occur.
- The UMTS policy mechanisms described in TS 23.060 will be used for control of the UMTS bearers, and the interaction between UMTS bearer services and IP bearer services will only occur at the translation function in the UE and GGSN.

9.9.2 QoS End-to-End Functional Architecture

In the following we describe the base line QoS architecture outlined in [7]; hence, we should also note here that QoS for R4 and R5 remain also under consolidation in the 3GPP specifications. Nonetheless, we shall provide implementation examples.

To provide QoS end-to-end, we manage the QoS within each domain. Thus we use an IP BS manager to control the external IP bearer service. Due to the different techniques applied within the IP network, this communicates to the UMTS BS manager through the translation function.

To enable coordination between events in the application layer and resource management in the IP bearer layer, there exists an element called IP policy control as a logical policy decision element. It is also possible to implement a policy decision element internal to the IP BS manager in the GGSN. The IP policy architecture does not mandate the policy decision point to be external to the GGSN. Whenever resources not owned or controlled by the UMTS network need to provide QoS, inter-work with an external resource manager that controls those resources takes place.

9.9.2.1 IP BS Manager

The IP BS manager utilizes standard IP mechanisms to manage the IP bearer service. These mechanisms may be different from mechanisms used within the UMTS, and may have different parameters controlling the service. The translation/mapping function supports the inter-working between the mechanisms and parameters used within the UMTS and the external IP bearer service, and interacts with the IP BS manager. When an IP BS manager exists both in the UE and the gateway node, these IP BS managers may communicate directly with each other by using relevant signalling protocols. Table 9.2 illustrates the minimum functionality that may be supported by equipment, in order to allow multiple network operators to provide inter-working between their networks for end-to-end QoS. The IP BS managers in the UE and GGSN provide the set of capabilities for the IP bearer level. As noted, provision of the IP BS manager is optional in the UE, and required in the GGSN.

Table 9.2 IP BS manager capability in the UE and GGSN [7].

Capability	UE	GGSN
DiffServ edge function	Optional	Required
RSVP/IntServ	Optional	Optional
IP policy enforcement point	Optional	Required*

*Although the capability of IP policy enforcement is required within the GGSN, the control of IP policy through the GGSN is a network operator choice.

9.9.2.2 IP Policy Control

The IP policy control stands as a logical policy decision element, which applies standard IP mechanisms to implement policy in the IP bearer layer. These mechanisms may conform, e.g. to the framework defined in IETF [RFC2573] *A Framework for Policy-based Admission Control*, where the IP policy control is effectively a Policy Decision Point (PDP). The IP policy control makes decisions regarding the network, based on IP policy rules, and communicates these decisions to the IP BS manager in the GGSN, which is the IP Policy Enforcement Point (PEP). A protocol interface between the IP policy control and application servers/proxies (e.g. local SIP proxy) supports the transfer of policy related information from the application layer to the policy decision point now under consolidation. A protocol interface (e.g. COPS protocol RFC2748) between the IP policy control and GGSN supports the transfer of information and policy decisions between the policy decision point and the IP BS manager in the GGSN.

The IP policy control bases policy decisions only on information obtained from nodes/elements within its domain or from nodes with which it has a trust relationship. The IP policy control needs to be in the same domain⁶ as the GGSN or have a trust relationship with the GGSN [7].

9.9.2.3 Resource Manager

In the UMTS network various nodes perform resource management in the admission control decision, usually under the direct control of the UMTS network. Likewise, in IP networks, there also exists resource management to ensure that resources required for a service are available. Because resources for the IP bearer service to be managed are not necessarily owned by the UMTS network. The resource management of those resources would be performed through an external resource management function for the IP network.

In addition, where the UMTS network is also using external IP network resources as part of the UMTS bearer service, e.g. for the backbone bearer service, it may also be necessary to inter-work with an external IP resource manager.

Figure 9.18 illustrates the scenario for control of an IP service using IP Bearer Service (BS) managers in both possible locations, i.e. in the UE and gateway node and an external resource manager. It also indicates the optional communication path between the IP BS managers in the UE and the gateway node.

⁶ Currently in IETF, inter-domain policy interactions are not defined.

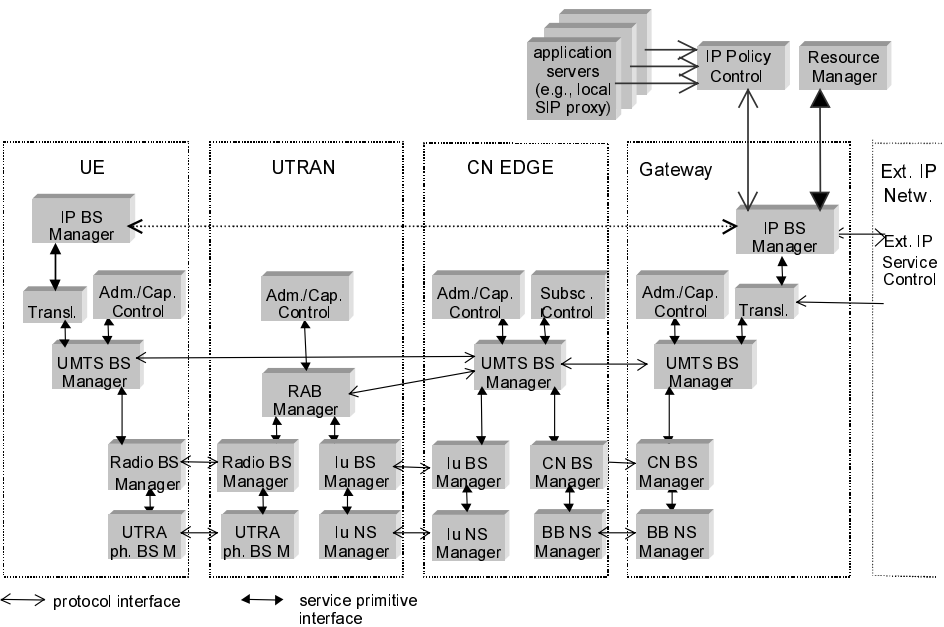


Figure 9.18 UMTS QoS management functions for control plane bearer service of an external IP server [7].

9.9.2.4 Implementing QoS

A practical solution would allow a UMTS operator or service provider to use standardized Internet QoS mechanisms across the IP core backbone network. GGSN/edge router function would connect to the IP backbone. These mechanisms would then afford the use of Service Level Agreements (SLAs) as a management tool to control resources of the IP core backbone network. The SLA would contain specifications on technical issues and administrative contractual information.

The connection from the User Equipment (UE) to the remote terminal/server can be established over different network paths. The UE accesses UTRAN over Uu radio interface. Over the Iu PS interface UTRAN can, e.g. connect to the SGSN and multimedia gateway. The core IP backbone would transport the IP traffic between the SGSN and the GGSN (Gn interface) and between the GGSN and application servers in the operator's service network or a border router to external Internet.

For example, a subscriber flow for real time IP applications will use dedicated radio channels over Uu, and that the QoS through UMTS would be controlled per user flow in RNC, SGSN and GGSN nodes, respectively. The nodes would use the IP core backbone for IP based transport over Iu and Gn. For this transport QoS would be controlled on aggregated flows using DiffServ for classification and conditioning and MPLS for the actual QoS implementation. From the GGSN to remote terminal/server the QoS would be controlled on the aggregated flows. The GGSN would support DiffServ for classification and conditioning and MPLS for the actual QoS implementation for the traffic on the Gi interface.

9.10 TRANSPORT ISSUES

The R00 architecture or more specifically R4 and R5 will support Ipv4/Ipv6 taking into account the following [7]:

- IP transport between network elements of the IP connectivity services (i.e. RNC, SGSN and GGSN) and IP transport for the CS domain.

At this writing the implementation of R99 does already aim to support both Ipv4/Ipv6 for IP connectivity; thus, for R00 or R4 and R5 will be mainly a consolidation.

- IP multimedia CN subsystem elements (UE to CSCF and the other elements e.g. MRF):
 - the architecture shall make optimum use of Ipv6
 - the R00 (R4 and R5) IM CN subsystem shall exclusively support Ipv6
- The R00 (R4 and R5) UE shall exclusively support Ipv6 for the connection to R00 (R4 and R5) IM services
- Access to existing data services (Intranet, Internet,...)
- The UE shall be able to access Ipv4 and Ipv6 based services.

Clearly, the IP multimedia sub-network connectivity will emphasize Ipv6. However, for access to data services it will need to support both (Ipv4 and Ipv6) to comply with backwards compatibility requirements.

9.10.1 Principles of Mobile Ipv4

In this section we highlight some Ipv4 characteristics in the context of the mobile IP concept based on [8].

Mobile IP allows a Mobile Station (MS) to maintain connectivity to the Internet or to a corporate network, while using a single and unchanging address (e.g. its home address) despite changes in the link layer point of attachment. Thus, when a MS moves from a home network to a foreign network it registers with its Home Agent (HA) an IP address that the HA can use to tunnel packets to the MS (i.e. a Care Of Address (COA)). The HA intercepts packets addressed to the MS's home address and tunnels these packets to the COA. Here we do not require interaction with UMTS location registers.

A COA may be, for example a dedicated address each MS gets in the visited network (co-located-COA); if so, the MS is the tunnel endpoint. Otherwise, the COA can be an address advertised (or retrieved) by a Foreign Agent (FA); if so, it is a FA-COA and the FA is the tunnel endpoint. The FA extracts packets from the tunnel and forwards them to the correct RAN logical link in order to deliver them to the appropriate MS.

9.10.2 Differences between IPv4 and IPv6

The key differences between protocols MIPv4 [9] and MIPv6 [10] can be summarized as follows [8]:

- Mobile IPv4 allows the use of Foreign Agents (FAs) to forward traffic thus requiring one care of address for multiple mobile stations, or the use of co-located care-of addresses (COA). In contrast MIPv6 supports co-located COAs only.
- MIPv4 has route optimization as an add-on, whereas it is an integral part of the MIPv6 specification.
- MIPv4 route optimization still requires traffic to be tunnelled between the correspondent host (CH) and the mobile station. In MIPv6 packets can be forwarded without tunnelling, i.e. only with the addition of a routing header.
- In MIPv4 the Home Agent (HA) must get involved in the set up of optimized routes. In MIPv6 the mobile station can initiate an optimized route to a CH directly (without involving the HA), and therefore more quickly and efficiently.
- In MIPv4 we obtain a COA from a FA or via DHCPv4. In MIPv6 we may obtain a COA via IPv6 stateless or state-full address auto-configuration mechanisms.
- In MIPv4 we require separate mobile IP specific messages to communicate with the FA, HA and CHs (when employing route optimization). In MIPv6, we can piggyback mobile IP specific information onto data packets.
- MIPv4 has the ability to provide smoother handover as an add-on feature that forms part of the route optimization protocol. In contrast support for smoother handover is an integral part of the MIPv6 specification.
- In MIPv4 we require reverse tunnelling to avoid ingress filtering problems (where firewalls drop the mobile's outgoing packets) since packets are sent with the home address as the source. In MIPv6 packets may be sent with the COA as the source address, hence there should not be any problems with ingress filtering.
- MIPv4 provides its own security mechanisms whereas MIPv6 employs the IPsec protocol suite.

To adequately assess the evolution and compatibility issues between MIPv4 and MIPv6 when applying to UMTS networks, we have to address each of above differences. We have to address additional issues when preparing the deployment or migration between IPv4 and IPv6 networks in general [8].

9.10.2.1 Reverse Tunnels

In IPv4 we need reverse tunnels (that is tunnels from the FA to the HA), both for remote network secure access and to avoid packet drops due to ingress filtering. Ingress filtering allows tracking of malicious users attempting denial of service attacks based on topologically inconsistent source address spoofing

In mobile IPv6, we do not need reverse tunnels to avoid problems with ingress filters. However they may still be beneficial when the ME is concerned about location privacy. The MN may use the care-of address as sender address but that is not required.

9.10.2.2 Use of Route Optimization

Route optimization reduces delays between the CH and ME, and it also reduces the load placed on HAs. Nonetheless, in MIPv4 it adds to the complexity of the HA and requires security associations between the HA and all CHs. Furthermore it still requires packets to be tunnelled from the CH to the FA-COA. In contrast, route optimization in MIPv6 removes the need to tunnel packets, instead we add a routing header to each packet. The ME also has more control to decide when to optimize routes, since it creates the optimized route rather than the HA; thus resulting in simpler MIPv6 HA. When migrating from MIPv4 to MIPv6, we need to make changes to CHs to employ route optimization. In contrast, all IPv6 CHs will support route optimization automatically.

9.11 CONCLUSIONS

In this chapter we have provided an overview of the R00 (R4 and R5) architecture. We have also described its main functional elements and its key interfaces. In addition, we cover the essential items of the mobility management, application level registration, location services, multimedia, QoS, Transport issues, etc.

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