W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

W-CDMA MOBILE COMMUNICATIONS SYSTEM

W-CDMA Mobile Communications System

Supervising Editor: Keiji Tachikawa

NTT DoCoMo became the first in the world to launch a next-generation mobile phone service that enables large-capacity communications. The W-CDMA mobile communications technology, known as one of the third-generation standard, was adopted to realize this high-speed, high-quality service. This volume, the fruit of collective efforts made by engineers engaged in R&D at NTT DoCoMo, is a standard technical documentation describing the basic technologies that constitute the W-CDMA mobile communications system in detail and individual systems that are expected to play an important role in future implementations.

W-CDMA MOBILE COMMUNICATIONS SYSTEM

Edited by

Keiji Tachikawa

NTT DoCoMo, Inc., Japan



Copyright © 2002

John Wiley & Sons Ltd, The Atrium, Southern Gate, Chichester, West Sussex PO19 8SO, England

Telephone (+44) 1243 779777

Email (for orders and customer service enquiries): cs-books@wiley.co.uk Visit our Home Page on www.wileyeurope.com or www.wiley.com

All Rights Reserved. No part of this publication may be reproduced, stored in a retrieval system or transmitted in any form or by any means, electronic, mechanical, photocopying, recording, scanning or otherwise, except under the terms of the Copyright, Designs and Patents Act 1988 or under the terms of a licence issued by the Copyright Licensing Agency Ltd, 90 Tottenham Court Road, London W1T 4LP, UK, without the permission in writing of the Publisher. Requests to the Publisher should be addressed to the Permissions Department, John Wiley & Sons Ltd, The Atrium, Southern Gate, Chichester, West Sussex PO19 8SQ, England, or emailed to permreq@wiley.co.uk, or faxed to (+44) 1243 770571.

This publication is designed to provide accurate and authoritative information in regard to the subject matter covered. It is sold on the understanding that the Publisher is not engaged in rendering professional services. If professional advice or other expert assistance is required, the services of a competent professional should be sought.

Other Wiley Editorial Offices

John Wiley & Sons Inc., 111 River Street, Hoboken, NJ 07030, USA

Jossey-Bass, 989 Market Street, San Francisco, CA 94103-1741, USA

Wiley-VCH Verlag GmbH, Boschstr. 12, D-69469 Weinheim, Germany

John Wiley & Sons Australia Ltd, 33 Park Road, Milton, Queensland 4064, Australia

John Wiley & Sons (Asia) Pte Ltd, 2 Clementi Loop #02-01, Jin Xing Distripark, Singapore 129809

John Wiley & Sons Canada Ltd, 22 Worcester Road, Etobicoke, Ontario, Canada M9W 1L1

British Library Cataloguing in Publication Data

A catalogue record for this book is available from the British Library

ISBN 0-470-84761-1

Typeset in 10/12pt Times by Laserwords Private Limited, Chennai, India Printed and bound in Great Britain by TJ International, Padstow, Cornwall This book is printed on acid-free paper responsibly manufactured from sustainable forestry in which at least two trees are planted for each one used for paper production.

Contents

_ _ _

Edi	itorial Board	Xi
Suj	pervisor's Note	xiii
Pre	face	xv
1	Overview	1
	Keisuke Suwa, Yoshiyuki Yasuda and Hitoshi Yoshino	
1.1	Generation Change in Cellular Systems	1
	1.1.1 Analog Cellular Systems	1
	1.1.2 Digital Cellular Systems	3
	1.1.3 Mobile Internet Services	7
1.2	Overview of IMT-2000	10
	1.2.1 Objectives of IMT-2000	10
	1.2.2 IMT-2000 Standardization	11
	1.2.3 IMT-2000 Frequency Band	18
	References	19
2	Radio Transmission Systems	21
	Mamoru Sawahashi	
2.1	Direct Sequence Code Division Multiple Access (DS-CDMA)	21
	2.1.1 Principles of DS-CDMA	21
	2.1.2 Spreading Code and Spreading Code Synchronization	24
	2.1.3 Configuration of Radio Transmitter and Receiver	26
	2.1.4 Application of DS-CDMA to Cellular Systems	27
2.2	Basic W-CDMA Transmission Technologies	28
	2.2.1 Two-Layer Spreading Code Assignment and Spreading Modulation	28
	2.2.2 Cell Search	31
	2.2.3 Random Access	41
	2.2.4 Technologies that Satisfy Various Quality Requirements in Multirate	
	Transmissions	42
	2.2.5 Diversity	49
2.3	Link Capacity Expansion Technologies in W-CDMA	66
	2.3.1 Interference Canceller	66
	2.3.2 Adaptive Antenna Array Diversity	71
	References	77

3	Radio System	81
	Seizo Onoe, Takehiro Nakamura, Yoshihiro Ishikawa, Koji Ohno,	
	Yoshiyuki Yasuda, Nobuhiro Ohta, Yoshio Ebine, Atsushi Murase and Akihiro Hata	
3.1	Radio System Requirements and Design Objectives	81
	W-CDMA and System Architecture	82
	3.2.1 Characteristics of W-CDMA	82
	3.2.2 Basic Specifications of W-CDMA	84
	3.2.3 Architecture of Radio Access Network	85
	3.2.4 Key W-CDMA Technologies	87
	3.2.5 Time Division Duplex (TDD) and Frequency Division Duplex (FDD)	92
3.3	Radio Access Interface Standard	92
	3.3.1 Physical Layer	92
	3.3.2 Media Access Control (MAC) Sublayer	126
	3.3.3 Radio Link Control (RLC) Sublayer	131
	3.3.4 Packet Data Convergence Protocol (PDCP) Sublayer	142
	3.3.5 Radio Resource Control (RRC)	145
	3.3.6 Control Sequence	159
3.4	Radio System Design	169
	3.4.1 W-CDMA Radio System Design	169
	3.4.2 Concept of W-CDMA Capacity	170
	3.4.3 Radio Link Design	175
	3.4.4 Cell/Sector Configuration	180
35	Radio Access Network Equipment	182
0.0	3.5.1 Overview of System Configuration of Radio Access Equipment	182
	3.5.2 BTS	182
	3.5.3 RNC	187
	3.5.4 MPE	188
	3.5.5 BS Antenna	189
36	Mobile Terminals	194
5.0	3.6.1 Implementation of Mobile Terminals	194
	3.6.2 Radio Access Specifications and Hardware Configuration Technologies	195
	3.6.3 UIM	202
	3.6.4 Terminal Display Technologies	202
	3.6.5 External Interface	204
	3.6.6 Future Prospects of Mobile Terminals	210
	References	210
	References	211
4	Network Technologies	215
•	Makoto Furukawa, Hiroshi Kawakami, Mutsumaru Miki, Daisuke Igarashi,	-10
	Yukichi Saito, Toyota Nishi, Mayuko Shimokawa, Katsumi Kobayashi,	
	Yasuhiko Kokubun and Masayuki Nakanishi	
4.1	Overview	215
	ATM Technology	213
	4.2.1 Switching Scheme for Multimedia Communications	217
	4.2.2 Basic Configuration of ATM	217
	4.2.3 ATM Adaptation Layer (AAL)	219
	4.2.4 Quality of Service (QoS) and ATM Traffic Management	21)

4.3	Network Control and Signaling Scheme	224
	4.3.1 CN Signaling Systems in IMT-2000	224
	4.3.2 Control Scheme	227
4.4	Packet Communication Scheme	245
	4.4.1 Overview of Mobile Packet Communications	245
	4.4.2 Service Target	246
	4.4.3 Network Architecture	246
	4.4.4 Mobile Packet Communications Technologies	247
	4.4.5 Connection Scheme	250
4.5	Intelligent Network (IN) Scheme	254
	4.5.1 Overview of IN Scheme	254
	4.5.2 Comparison with Conventional Systems	255
	4.5.3 Merits of the IN Scheme	258
	4.5.4 Standardization Trends	258
	4.5.5 Future Prospects	259
4.6	Short Message Scheme	259
	4.6.1 Overview of Scheme	259
	4.6.2 Network Configuration	261
	4.6.3 Routing Scheme	261
	4.6.4 Main Extended Functions of SMS	261
	4.6.5 Example of SMS Applications	264
4.7	Gateway Scheme	265
	4.7.1 Protocol Conversion Gateway	266
	4.7.2 TCP Gateway	267
	4.7.3 Tunneling Gateway	269
		270
	4.7.4 Multimedia Service Platform	270 274
		270 274
5	4.7.4 Multimedia Service Platform References	
5	4.7.4 Multimedia Service Platform References Operation System	274
5	4.7.4 Multimedia Service Platform References	274
	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, 	274
	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 	274 277 277
	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 	274 277
5.1	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration 	274 277 277 277
5.1	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 	274 277 277 277 280
5.1	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 	274 277 277 280 283
5.1	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 	274 277 277 280 283 283
5.1 5.2	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 	274 277 277 277 280 283 283 283 284
5.1 5.2	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 	274 277 277 280 283 283 283 284 287 289
5.1 5.2	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 	274 277 277 277 280 283 283 283 284 287 289 291
5.1 5.2	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 	274 277 277 280 283 283 283 284 287 289
5.1 5.2	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 5.3.3 Network Control Functions 	274 277 277 280 283 283 284 287 289 291 292
5.1 5.2	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 5.3.3 Network Control Functions 5.3.4 Congestion Control During Packet Communications 	274 277 277 280 283 283 284 287 289 291 292 294 295
5.1 5.2 5.3	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 5.3.3 Network Control Functions 5.3.4 Congestion Control During Packet Communications 5.3.5 Achieving High-Speed Restriction Process 	274 277 277 280 283 283 284 287 289 291 292 294 295 295
5.1 5.2 5.3	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 5.3.3 Network Control Functions 5.3.4 Congestion Control During Packet Communications 5.3.5 Achieving High-Speed Restriction Process NE Monitoring 	274 277 277 280 283 283 283 284 287 289 291 292 294 295 295 296
5.1 5.2 5.3	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 5.3.3 Network Control Functions 5.3.4 Congestion Control During Packet Communications 5.3.5 Achieving High-Speed Restriction Process NE Monitoring 5.4.1 NEs in a Multivendor Environment 	274 277 277 280 283 283 283 284 287 289 291 292 294 295 295 296 296 296
5.1 5.2 5.3	 4.7.4 Multimedia Service Platform References Operation System Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma Overview 5.1.1 Positioning of OpS 5.1.2 System Configuration Network Monitoring 5.2.1 Configuration of Network Monitoring Functions 5.2.2 Characteristics of Network Monitoring 5.2.3 Building a Network Monitoring System Network Control 5.3.1 Positioning of Network Control System 5.3.2 Coordination between Systems in Different Types of Networks 5.3.3 Network Control Functions 5.3.4 Congestion Control During Packet Communications 5.3.5 Achieving High-Speed Restriction Process NE Monitoring 	274 277 277 280 283 283 283 284 287 289 291 292 294 295 295 296

5.5	Network Element Management	301
	5.5.1 Network Element Management	301
	5.5.2 Network Quality Management	302
	5.5.3 Remote File Updating	304
	References	305
6	Multimedia Processing Scheme	307
	Minoru Eto, Hiroyuki Yamaguchi, Tomoyuki Oya, Toshiro Kawahara,	
	Hiroshi Uehara, Teruhiro Kubota, Masayuki Tsuda, Seishi Tsukada, Wataru Takita,	
	Kimihiko Sekino and Nobuyuki Miura	
	Overview	307
6.2	Multimedia Signal Processing Scheme	308
	6.2.1 Image Processing	308
	6.2.2 Speech and Audio Processing	313
	6.2.3 Multimedia Signal Processing Methods	320
6.3	Mobile Information Service Provision Methods	325
	6.3.1 Mobile ISP Services	325
	6.3.2 Multimedia Information Distribution Methods	329
	6.3.3 Contents Markup Languages	333
	6.3.4 Mobile Internet Standardization (WAP)	338
6.4	Multimedia Messaging Methods	342
	6.4.1 Overview	342
	6.4.2 Trends of Standardization	343
	6.4.3 Conceptual Model	343
	6.4.4 Implementation Model	344
	6.4.5 Push Technology	345
6.5	Location Information Processing Methods	345
	6.5.1 Location Information Use Overview	346
	6.5.2 Structure of the Location Information Processing System	347
	6.5.3 Transmission System Outside the Mobile Communications Network	348
	6.5.4 Location Information Distribution Methods	349
	6.5.5 Location Information Distribution Platform	350
6.6	Mobile Electronic Authentication Methods	356
	6.6.1 Electronic Authentication	356
	6.6.2 WAP Authentication Model	357
	6.6.3 Electronic Certificate for Mobile Communications	359
	6.6.4 Transport Layer Security (TLS)	359
	6.6.5 Short-Lived Certificate	360
	6.6.6 Future Challenges	361
	References	361
7	Future Prospects	365
	Yoshiyuki Yasuda, Takchiro Nakamura, Shinji Uebayashi, Hiroshi Fujiya and	
	Tomoyuki Oya	
	Overview	365
7.2	Prospects of Radio Technologies	366
	7.2.1 TDD Scheme	366
	7.2.2 High-Speed Downlink Packet Access (HSPDA)	368

7.3	Prospects of Network Technologies	370
	7.3.1 IP Packet Communications in Mobile Communication Networks	370
	7.3.2 Technology Trends in IP Networks	371
	7.3.3 All IP Network Configuration and Deployment	373
7.4	Prospects of Signal Processing Technologies	374
	7.4.1 Tandem Connection Avoidance Technologies	374
	7.4.2 Adaptive MultiRate-WideBand (AMR-WB)	376
	7.4.3 Packet-Transmitted Multimedia	377
	References	378
Ap	pendix–Interface Specifications	381

Index

409

Editorial Board

Editor-in-Chief	Norioki Morinaga
Editors	Kota Kinoshita, Hideaki Yumiba, Takanori Utano, Masafumi Onuki, Shoichiro Ishigaki, Kazuaki Murota, Masaharu Hata, Keisuke Suwa

Supervisor's Note

The progress of the IT revolution is about to change not only the ways in which business is done but also people's lifestyles. The *mobile*, *wireless* and *personal* features of mobile communications will have unprecedented importance in building a mobile multimedia society for the future. Mobile communications is expected to undergo dramatic progress through the development of a wide range of terminals, the advancement of network and gateway functions and the supply of various content and applications. An example is *i-mode*, the world's first wireless Internet access service on cellular phones. Since its commercial launch in February 1999, *i-mode* has acquired more than 21.5 million subscribers as of the end of March 2001. As demonstrated by this example, mobile communication is expected to form the core of information and communications networks in the twenty-first century, in line with the progress of the IT revolution.

Mobile multimedia services in the twenty-first century are expected to move on from "person-to-person" communications (as was the case in the twentieth century) to "person-to-machine" communications (as in *i-mode*, in which mobile terminals are used to access servers over the Internet) and "machine-to-machine" communications (aka machine communications using mobile terminals, which is a form of communications in a broader sense that targets all objects in motion). While progress in this area hitherto has largely been due to technologies that helped digitize mobile networks, Internet protocols will have to be incorporated into mobile communications in the future so as to further integrate mobile communications with the Internet. This should enable the provision of cheaper and more efficient services.

In Japan, a digital mobile phone system referred to as the *second-generation mobile communications system* and built in compliance with Japan's domestic standard was put to practical use in 1993. Today's progress is attributable to this system, which increased subscriber capacity through highly efficient frequency usage and led to the development of new services and various types of terminals. By the end of May 2001, the world's first service based on the third-generation mobile communications system (IMT-2000) using W-CDMA was launched under a service brand *FOMA*. This new system is expected to further facilitate the market penetration of mobile multimedia, as various types of content can be transmitted at speeds faster than the existing system by more than a digit and processed smoothly without sacrificing their high quality.

This volume consists of detailed articles written by leading engineers for readers who wish to learn about the basic technologies, systems, networks, services and operations of W-CDMA in a systematic manner. We hope that it will help deepen your interest in, and understanding of, mobile communication technologies.

Keiji Tachikawa, Doctor of Engineering President and CEO NTT DoCoMo, Inc.

Preface

The remarkable progress in information technology (IT) since the late 1990s continues to facilitate faster communications, broadband access and lower communication costs in the information and communications sector. Consequently, communications has penetrated not only the business scene but also every aspect in personal life, to the extent of dramatically changing people's lifestyles. The widespread use of the Internet, which appeared in the 1990s, is also contributing to the advent of a wide range of multimedia services that undermine the barriers of time and place.

In Japan, the automobile phone service based on cellular technology was commercially launched in 1979, followed by the portable mobile phone system in 1987. Since 1994, the number of subscribers has skyrocketed at a rate of 10 million per year, owing to improved and enhanced network coverage and quality, liberation of terminal sales and continuous tariff reductions. As of March 2000, the number of mobile phone subscribers reached 56.8 million, accounting for approximately 50% of the Japanese population. In February 1999, the commercial service of *i-mode*, a mobile communications service enabling Internet access, was started. As of the end of March 2001, *i-mode* subscribers totaled about 21.5 million in number. i-mode, which enables subscribers to access the Internet by using a packet-switched network overlaid on the existing mobile phone network, has been successful in winning the hearts of mobile Internet users by lowering communication costs through data-volume-based billing, developing easy-to-use handsets, and establishing new business models including the bill collection service on behalf of the content providers. The evolution of cellular-based mobile communication systems from the first-generation (analog) to the second-generation (digital), as described above, has been made possible by solving many technical issues along the way. Efforts to develop a global standard for providing high-speed, high-quality multimedia services have crystallized in the form of the third-generation (3G) systems, under the IMT-2000 standard. The world's first 3G system was implemented by Japan in 2001 on the basis of the latest research results, and other countries are expected to follow suit. 3G systems are expected to bring about radical socioeconomic and cultural changes that would affect people around the world.

As explained above, recent mobile communication systems are based on the wealth of an extremely wide range of advanced technologies, including radio transmission technologies, radio link control technologies, network technologies, operation technologies, terminal equipment technologies and other multimedia processing technologies. The cellular phone system together with the Personal Handyphone System (PHS) and other information infrastructure provide a vital means for communication in our everyday life. In light of these facts, this volume reviews in detail the basic technologies applied to W-CDMA, a standard 3G mobile communications technology. The focus is to explain the technologies that will play an important part in future developments, with reference to the latest research results.

Chapter 1 "Overview" briefly reviews various cellular systems, ranging from analog to digital, describes their characteristics and discusses the objectives of IMT-2000 and the status of standardization. Chapter 2 "Radio Transmission Systems" explains, in an easyto-understand manner, the mechanism and the characteristics of CDMA as discussed in this volume with respect to radio access systems, a basic technology that is vital for mobile communications. It also describes basic transmission technologies such as cell search technologies, transmission power control technologies and diversity technologies, in addition to capacity-enhancement technologies based on adaptive array antennas. Chapter 3 "Radio Systems" provides a detailed explanation of radio access interfaces and radio system designs that form the basis of W-CDMA technology, as well as an introduction to mobile terminals. Chapter 4 "Network Technologies" reviews in detail ATM technologies, packet communication systems and other types of network systems. Chapter 5 "Operation System" gives an outline of network monitoring/control and equipment monitoring/administration. Chapter 6 "Multimedia Processing Methods" describes in detail the processing schemes for multimedia signals including audio and video adopted in radio systems, information distribution schemes, location information processing and electronic authentication systems. Chapter 7 "Future Prospects" provides an outlook on the future directions of radio technologies, network technologies and signal processing technologies.

This volume was written by NTT DoCoMo's engineers working at the forefront of research and development of W-CDMA. Much consideration was given to ensure that the descriptions are sufficiently covered and consistent. It was written to enable a wide range of readers to gain a general understanding of W-CDMA, with researchers, developers and operators in the mobile communications sector in mind, as well as students and end users.

The editors are immensely grateful to Professor Fumiyuki Adachi at Tohoku University, for his pioneering research findings, and Teruaki Kuwabara at Maruzen Co., Ltd, for his cooperation in planning and publishing this work.

Editors

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

Overview

Keisuke Suwa, Yoshiyuki Yasuda and Hitoshi Yoshino

1.1 Generation Change in Cellular Systems

In Japan, mobile communications systems based on cellular technology have evolved, as illustrated in Figure 1.1. The first-generation analog car phones were first introduced in 1979, followed by the commercialization of the second-generation digital phones in 1993. Mobile phone subscribers have rapidly increased in number since then, owing to the liberation of terminal sales and continuous price reductions. In March 2000, the number of mobile phone subscribers outnumbered those of fixed telephones. Meanwhile, the expansion of data communications on a global scale – spearheaded by the Internet – is promoting the introduction of Packet-Switched (PS) communication systems that are suitable for data communications in a mobile environment.

The standardization and system development of the next-generation mobile communications system, known as the Third-Generation (3G) International Mobile Telecommunications-2000 (IMT-2000), began in response to the rising need in recent years to achieve high-speed data communications capable of supporting mobile multimedia services and developing a common platform that would enable mobile phone subscribers to use their mobile terminals in any country across the world. From 2001 onwards, IMT-2000 systems using Wideband Code Division Multiple Access (W-CDMA) technology are due to be introduced.

The following is a rundown of mobile phone and car phone systems that have been commercialized to date.

1.1.1 Analog Cellular Systems

Analog cellular systems were studied by Bell Laboratories in the United States and the Nippon Telegraph and Telephone Public Corporation (predecessor of NTT) in Japan. The American and Japanese systems are referred to as the Advanced Mobile Phone Service (AMPS) and the NTT system, respectively. Both systems are called *cellular systems* because they subdivide the service area into multiple "cells".

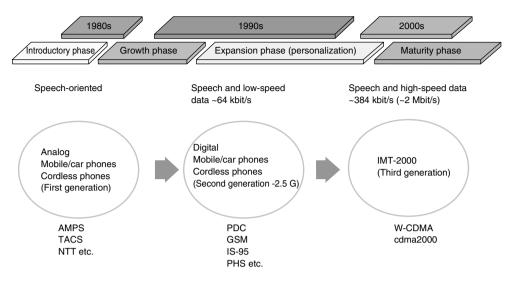


Figure 1.1 Progress in networks

The NTT system embraced the following cellular system element technologies:

- 1. Use of the new 800-MHz frequency band,
- 2. small-zone configuration (radius: several kilometers) and iterative use of the same frequency,
- 3. allocation of a radio channel for control signal transmission separate from speech transmission,
- 4. development of a mobile terminal that can switch hundreds of radio channels by a frequency synthesizer, and
- 5. establishment of new mobile-switching technologies to track and access mobile terminals.

The NTT system became commercially available as the Large-Capacity Land Mobile Telephone System in 1979, initially targeting the Tokyo metropolitan area. Later, the service area was gradually expanded to accommodate other major cities nationwide [1].

Moreover, on the basis of this system, efforts were made to improve the adaptability to small and medium-sized cities and to make smaller, more economical mobile terminals. This led to the development of the Medium-Capacity Land Mobile Telephone System, which was rolled out on a nationwide scale.

Subsequently, the further increase in demand for the NTT system prompted the development of a car phone system that would allow the continuous use of legacy mobile phones aimed at dealing with the increasing number of subscribers, improving service quality and miniaturizing the terminals. This resulted in the so-called *large-capacity system*, characterized by one of the narrowest frequency spacings among analog cellular systems worldwide. The system achieved a radical increase in capacity, smaller radio base station (BSs), advanced functions and a wider range of services [2]. Table 1.1 shows the basic specifications of the NTT system.

		NTT system	
		Large city system	Large-capacity system
Frequency band	Base station transmission	$870 \sim 885 \text{ MHz}$	8
1 2			$70\sim 885~\mathrm{MHz}$
			$860 \sim 870 \ \mathrm{MHz^a}$
	Base station reception	$925 \sim 940 \ \text{MHz}$	$925 \sim 940 \text{ MHz}$
	*		$915 \sim 925 \ \mathrm{MHz^a}$
Transmission/Reception (TX/RX) frequency spacing		55 MHz	55 MHz
Channel spacing interleave		25 kHz	12.5 kHz
			6.25 kHz
Number of channels		600	1199
			800

Table 1.1 Specifications of the NTT system

^aUsed by IDO Corporation (predecessor of au Corporation).

On the basis of the American analog cellular standard AMPS, Motorola, Inc. developed a system customized for Britain called the *Total Access Communication System* (TACS). A version of TACS with a frequency allocation adapted to Japan is called *J-TACS*. Another version that achieves greater subscriber capacity by halving the bandwidth of radio channels is called *N-TACS*. Table 1.2 shows the basic specifications of TACS. TACS is characterized by increasing the subscriber capacity, by securing a wider frequency carrier spacing for voice channels to improve the tolerance against radio interference and by subdividing each zone into a maximum of six sectors to shorten the distance for frequency reuse.

1.1.2 Digital Cellular Systems

Digital cellular systems have many features, such as improved communication quality due to various digital signal processing technologies, new services (e.g. nontelephony services), improved ciphering, greater conformity with digital networks and efficient utilization of the radio spectrum.

The development of digital cellular systems was triggered by standardization efforts in Europe, which was home to many competing analog systems. In Europe, analog cellular systems in each country used different frequency bands and schemes, which made interconnection impossible across national borders. In 1982, the European Conference of Postal and Telecommunications Administrations (CEPT) established the Group Special Mobile (GSM), and development efforts were carried out under the leadership of the European Telecommunications Standards Institute (ETSI). GSM-based services were launched in 1992.

In the United States, the IS-54 standard was developed under the Electronic Industries Association (EIA) and the Telecommunications Industry Association (TIA). IS-54 services, launched in 1993, were required to satisfy dual-mode (both analog and digital cellular) operations and adopted Time-Division Multiple Access (TDMA). Studies on

System	TACS (Britain)	J-TACS	N-TACS
Base station frequency band	$890 \sim 915 \text{ MHz}$	$860 \sim 870 \ \mathrm{MHz}$	$860 \sim 870 \text{ MHz}^{a}$ $843 \sim 846 \text{ MHz}$
Mobile station frequency band	$935 \sim 960 \text{ MHz}$	$915 \sim 925 \ \mathrm{MHz}$	$\begin{array}{c} 915 \sim 925 \ \text{MHz}^a \\ 898 \sim 901 \ \text{MHz} \end{array}$
Channel spacing	Speech: 25 kHz interleave	Speech: 25 kHz interleave	Speech: 12.5 kHz interleave
	Data: 25 kHz interleave	Data: 25 kHz interleave	Data: 25 kHz interleave
Modulation scheme	PM	PM	PM
Maximum frequency	Speech: 9.5 kHz	Speech: 9.5 kHz	Speech: 9.5 kHz
shift	Data: 6.4 kHz	Data: 6.4 kHz	Data: 6.4 kHz
Control signal data speed	8 kbit/s	8 kbit/s	8 kbit/s
Control channel	Transmission by	Transmission by	Transmission by
configuration	zone	zone	zone

 Table 1.2
 Specifications of the TACS system

 $^{\rm a}\text{IDO}$ Corporation (predecessor of au Corporation) applied the system, sharing the frequency band with the NTT system;

Note: PM: Pulse Modulation.

CDMA inclusive of field tests had been carried out in a vigorous manner from 1989 onwards, and consequently, the IS-95 standard-based CDMA technology was adopted in 1993.

Japan was no exception in that it needed to standardize the radio interface between BSs and MSs in order to promote the use of mobile and car phone services and enable subscribers to access all local mobile communication networks across the nation. In 1989, studies on technical requirements for digital systems began under the request from the Ministry of Posts and Telecommunications (predecessor of the Ministry of Public Management, Home Affairs, Posts and Telecommunications), which crystallized in the form of a recommendation to adopt TDMA in 1990. In parallel, Research and Development Center for Radio System [RCR: predecessor of the Association of Radio Industries and Businesses (ARIB)] studied the radio interface specifications in detail, which led to the establishment of a digital car phone system standard called Japan Digital Cellular (JDC) in 1991. The JDC was subsequently renamed Personal Digital Cellular Telecommunication System (PDC) for the purpose of spreading and promoting the standard [3]. In Japan, the evolution from an analog mobile system to the PDC system required the installation of separate radio access equipment (radio BS and control equipment), as their configurations were totally different between analog and digital. However, the transit switch and the backbone network were shared by the analog and digital systems - this network configuration was possible because a common transmission system could be applied to the transit network.

Table 1.3 shows the basic specifications of the European, American and Japanese digital cellular standards. Other than IS-95, all standards are based on TDMA. Multiplexing, in terms of full rate/half rate, is 3/6 in the American and Japanese standards and 8/16 in the European standard. The modulation and demodulation scheme adopted by the American

	PDC (Japan)	North America		Europe GSM
		IS-54	IS-95	
Frequency band	800 MHz/ 1.5 GHz	800 M	IHz band	800 MHz band
Carrier frequency spacing	50 kHz (25 kHz interleave)	50 kHz (25 kHz interleave)	1.25 MHz	400 kHz (200 kHz interleave)
Access scheme	TDMA/FDD	TDMA/FDD	DS-CDMA/FDD	TDMA/FDD
Multiplexing	3/6	3/6	_	8/16
Transmission speed	42 kbit/s	48.6 kbit/s	1.2288 M chips/s	270 kbit/s
Speech encoding scheme	11.2 kbit/s VSELP 5.6 kbit/s PSI-CELP	13 kbit/s VSELP	8.5 kbit/s QCELP (4-step variable rate)	22.8 kbit/s RPE-LTP-LPC 11.4 kbit/s EVSELP
Modulation scheme	$\pi/4$ -shift QPSK	π/4-shift QPSK QPSK	Downlink: QPSK Uplink: OQPSK	GMSK

 Table 1.3
 Basic specifications of digital cellular systems

Note: RPE: Regular Pulse Excited Predictive Coding;

LTP: Long-Term Predictive Coding;

LPC: Linear Predictive Coder; FDD: Frequency Division Duplex; and PSI-CELP: Pitch Synchronous Innovation-Code Excited Linear Prediction.

and Japanese standards is $\pi/4$ -shift Quadrature Phase Shift Keying (QPSK), which not only has a higher efficiency of frequency usage than the Gaussian Minimum Shift Keying (GMSK) applied in Europe but also allows a simpler configuration of linear amplifiers than QPSK. IS-95 has a wider carrier bandwidth of 1.25 MHz, and identifies users by spreading codes. The American standard shares the same frequency band with the analog system, whereas the Japanese and European standards use the 800 MHz band. Japan uses the 1.5 GHz band as well.

Figure 1.2 shows the configuration of the Japanese standard PDC [The Telecommunications Technology Committee (TTC) Standard JJ-70.10] [9].

(1) Visited Mobile Switching Center (V-MSC)

V-MSC has call connection control functions for the mobile terminals located inside the area under its control and mobility support functions including service control, radio BS control, location registration and so on.

(2) Gateway Mobile Switching Center (G-MSC)

G-MSC is the switching center that receives incoming calls from another network directed to subscribers within its own network and incoming calls directed to subscribers who are roaming in its own network. It has the function of routing calls to V-MSC or the roaming network in which the mobile terminal is located by identifying the terminal's Home Location Register (HLR) and Gateway Location Register (GLR) and sending queries.

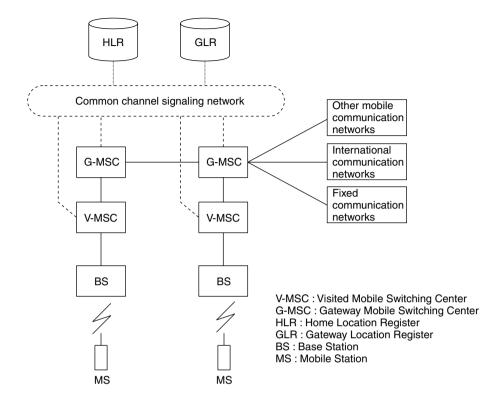


Figure 1.2 PDC system configuration model

(3) Home Location Register (HLR)

HLR is a database that administers information required for assuring the mobility of mobile terminals and providing services (e.g. routing information to mobile terminals, service contract information).

(4) Gateway Location Register (GLR)

GLR is a database that administers information required for providing services to mobile terminals roaming from another network. It has the function to acquire information on the roaming mobile terminal from the HLR of the terminal's home network. GLR is temporarily established when there are mobile terminals roaming from other networks.

(5) Base Station (BS)

BS has the function to traffic and control channels between V-MSC and BS, as well as those between BS and the Mobile Station (MS).

(6) Mobile Station (MS)

MS is the termination of the radio link from the mobile subscriber's point of view. It has the function to provide various communication services to mobile subscribers.

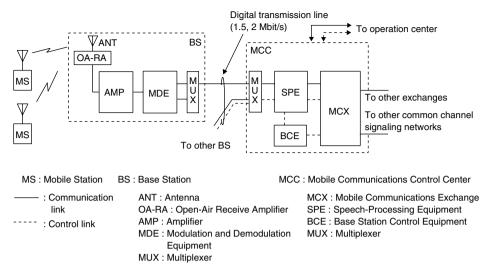


Figure 1.3 Configuration of the digital mobile communications system

Figure 1.3 shows the configuration of NTT's digital mobile communications system, which consists of the Mobile Communications Control Center (MCC), BS and MS.

MCC consists of a mobile communication switch based on the improved D60 digital switch, Speech-Processing Equipment (SPE), which harnesses a speech CODEC for the radio interface, and Base station Control Equipment (BCE), which handles the control of BSs. The SPE can accommodate three traffic channels in a 64 kbit/s channel, as it executes low bit rate speech coding (11.2 kbit/s).

BS consists of Modulation and Demodulation Equipment (MDE), AMPlifier (AMP), Open-Air Receive Amplifier (OA-RA), ANTenna (ANT) and so on. MDE is composed of a π /4-shift QPSK modem and a TDMA circuit for each carrier. The MDE can accommodate 96 carriers (equivalent to 288 channels) in a cabinet. AMP amplifies numerous radio carriers from MDE *en bloc* and sends them to ANT. In order to suppress the distortion from intermodulation due to nonlinear properties of AMP, it adopts a feed-forward compensation circuit. OA-RA uses a low-noise AMP. ANT is the same as its analog counterpart in terms of structure.

In order to achieve miniaturization and lower power consumption, NTT developed a power AMP that controls the voltage of the power supply according to the signal envelope level and thereby secured the same conversion efficiency as in analog systems. NTT also developed and implemented a digital synthesizer that enables high-speed frequency switching.

1.1.3 Mobile Internet Services

The rapid diffusion of the Internet over fixed communication networks was accompanied by an increase in demand for data communications for both business and personal purposes in mobile environments as well. To meet this demand, a mobile PS communications system was developed, adapted to the properties of data communications. In Japan, NTT DoCoMo launched the PDC-based Personal Digital Cellular-Packet (PDC-P) system in 1997. NTT DoCoMo built a mobile network dedicated to PS communications – independent of the PDC network – with the aim to minimize the impact to the PDC system (voice service), which had been widely used at the time, and to render PS data communication services as soon as possible. In February 1999, NTT DoCoMo became the world's first mobile Internet Service Provider (ISP) through the launch of *i-mode*, which enabled Internet access from mobile phones via PDC-P [4]. *i-mode*, which is a commodity developed under the concept "cellular phone-to-talk into cellular phone-to-use", is a convenient service that enables users to enjoy mobile banking, booking of tickets, reading the news, checking weather forecasts, playing games and even indulging in fortune-telling. *i-mode* service is composed of four major components (Figure 1.4).

The first component is the *i-mode* mobile phone, which supports 9.6 kbit/s PS communications and is equipped with a browser (browsing software), in addition to basic voice telephony functions. The browser can read text in Hyper Text Markup Language (HTML), which is the Internet standard accounting for 99% of all digital content worldwide. The screen of the *i-mode* mobile phone is similar to conventional mobile phones in size: 8 to 10 double-byte characters horizontally, and 6 to 10 lines vertically.

The second component is the PS network. *i-mode* uses the same network as NTT DoCoMo's PS communication service (DoPa). NTT DoCoMo decided to adopt the single-slot-type (9.6 kbit/s) network, as its slow transmission speed had been deemed acceptable for making *i-mode* mobile phones smaller, lighter and text-centric.

The adoption of the PS communications system accelerates the response from the accessed Web server, enabling users to transmit and receive information far more smoothly than by circuit-switched (CS) systems.

The use of *i-mode* service incurs a monthly basic fee of \$300 and a packet communications charge. The charge is billed according to the transferred data volume [\$0.3 per packet (128 bytes)] rather than by connection time. This billing scheme is suitable for those who are not used to operating the *i-mode* mobile phone, as they can spend as

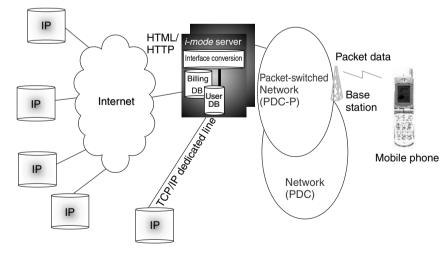


Figure 1.4 *i-mode* network configuration

much time as they want without worrying about the operation time (which translates into communication tariff in a CS system).

The third component is the *i-mode* server, which functions as the gateway between NTT DoCoMo's network and the Internet. Specifically, its functions include distribution of information; transmission, reception and storage of e-mail; *i-mode* subscriber management; Information Provider (IP) management and billing according to data volume.

The fourth component is content. Figure 1.5 shows the services available from *i-mode*.

For the *i-mode* business to be viable, online services must be used by many users (they must be attractive enough to lure users), digital content owners must be able to offer their existing resources at low cost, and parties contributing to the business must be rewarded according to their respective efforts. To meet these requirements, NTT DoCoMo decided to adopt HTML as the description language for information service providers (companies), so that the digital content they had already been providing over the Internet could be used in *i-mode* more or less in its original form.

Functions of *i-mode* include normal phone calls, as well as the phone-to-function, which enables users to directly call a phone number acquired from a Web site. It also supports simple mail that allows users to transmit and receive short messages using the addressee's mobile phone number as the address, in addition to the e-mail. Furthermore, *i-mode* users can access the Web by URL (Uniform Resource Locator) entry and enjoy online services.

On the basis of development concepts as such, *i-mode* has spread rapidly since the launch of the service. As of early January 2002, the number of subscribers totaled 30.3 million and voluntary sites exceeded 52,400. *i-mode* is expected to develop further, especially in the area of mobile commerce applications among others, as program downloading has been enabled with the introduction of Java technology in January 2001, and higher security measures are planned to be implemented.

As for other PS systems, a PS service called *PacketOne* was commercially launched in 1999, based on the cdmaOne system compliant to IS-95. Overseas, Cellular Digital Packet Data (CDPD) has been implemented over the analog AMPS system in North America, and General Packet Radio Service (GPRS) over GSM in Europe.

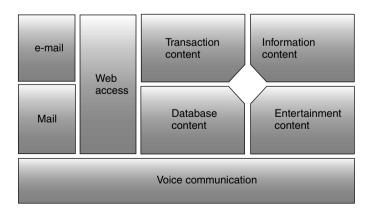


Figure 1.5 Services available from *i-mode*

1.2 Overview of IMT-2000

1.2.1 Objectives of IMT-2000

Research and development efforts have been made for IMT-2000, with the aim to offer high-speed, high-quality multimedia services that harness a wide range of content including voice, data and video in a mobile environment [5, 6]. The IMT-2000 system aims to achieve the following.

(1) Personal Communication Services through Improved Spectrum Efficiency (Personalization)

Further improvements in the efficiency of frequency utilization and the miniaturization of terminals will enable "person-to-machine" and "machine-to-machine" communications.

(2) Global, Seamless Communication Services (Globalization)

Users will be able to communicate and receive uniform services anywhere in the world with a single terminal.

(3) Multimedia Services through High-Speed, High-Quality Transmission (Multimedia)

Use of a wider bandwidth enables high-speed, high-quality transmission of data in large volume, still pictures and video, in addition to voice connections.

The International Telecommunication Union (ITU) specifies the requirements for the IMT-2000 radio transmission system to provide multimedia services in various environments as shown in Table 1.4. The required transmission speed is 144 kbit/s in a high-speed moving environment, 384 kbit/s when traveling at low speeds and 2 Mbit/s in an indoor environment.

Figure 1.6 shows the mobile multimedia services presumed under IMT-2000 in business, public and private domains.

(1) Business Domain

Mobile communications services have been used by numerous business users since its early days of services. In the business domain, IMT-2000 is believed to be used for image communications in addition to text data. There are high expectations for services that would enable users to acquire large volumes of various business data in a timely manner and communicate their thoughts smoothly, regardless of place and time.

(2) Public Domain

A typical example of applications to be used in the public domain is the emergency communications service taking advantage of the merit of mobile systems that is highly tolerant against disaster situations. Remote monitoring applications realizing "machineto-machine" communications are also considered to be widely used in the public domain.

 Table 1.4
 Requirements of the IMT-2000 radio transmission system

	Indoor	Pedestrian	Inside car
Transmission speed (kbit/s)	2048	384	144

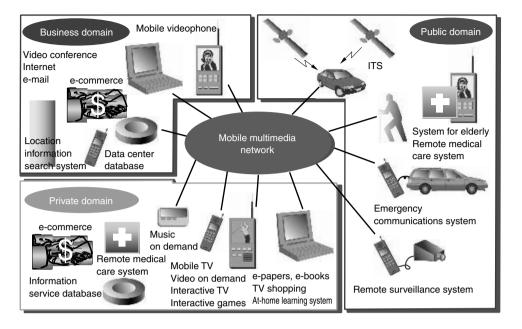


Figure 1.6 Mobile multimedia services

Other potential services include the adoption of mobile systems as part of Intelligent Transport Systems (ITS), the use of *i-mode* for safe driving, car-navigation systems based on communications networks and pedestrian-navigation systems.

(3) Private Domain

The private domain has been the driving force behind mobile communications in recent years. With the introduction of IMT-2000, advanced forms of mobile Internet services such as *i-mode* are expected to become available as part of private applications. In video communications, videophones are likely to appear, whereas on the mail front, multimedia mail is expected to become available, enabling users to attach video and voice messages to an e-mail. As for information distribution services, it is hoped that music distribution and video distribution will be taken up widely in the market.

1.2.2 IMT-2000 Standardization

Research on IMT-2000 started in 1985, originally in the name of Future Public Land Mobile Telecommunications System (FPLMTS) under the ITU-Radio communication sector (ITU-R) with an aim to achieve the aforementioned objectives. In conjunction with this, the ITU-Telecommunication standardization sector (ITU-T) took up the research of IMT-2000 as an important task and conducted studies on high-layer signaling of protocols, identifiers, services, speech/video encoding and so on. This was followed by studies on detailed specifications under the Third-Generation Partnership Project (3GPP), and efforts to build a consensus among the organizations toward the development of a standardized radio interface. This section describes the key activities.

1.2.2.1 ITU Activities

ITU-R's Efforts

IMT-2000 standardization activities in ITU-R were launched in 1985, originally in the name of FPLMTS. ITU-R started out the studies by clarifying the system concept of IMT-2000, consisting of both terrestrial and satellite systems. As part of such efforts, ITU-R [7, 8] agreed on recommendations relating to the basic concept and principles, followed by recommendations on the general framework and requirements of IMT-2000. ITU-R then started to prepare a radio interface recommendation to meet the requirements set forth in those recommendations, which followed the procedures as shown in Figure 1.7.

First, ITU-R clarified the minimum requirements of the radio interface of IMT-2000. Table 1.4 shows the minimum performance requirements. In response, nations and organizations were required to propose a radio interface that would satisfy those requirements by June 1998. Nations, regions and organizations conducted studies at consortiums other than ITU, such as Japan's ARIB and the ETSI. As a result, 10 terrestrial systems and 6 satellite systems were proposed to ITU-R, all of which were then assessed by evaluation groups of various countries and organizations. Following the confirmation that all systems had satisfied the requirements of IMT-2000, the key characteristics of the radio interface were refined in consideration of the Radio Frequency (RF) characteristics and key base band characteristics. Efforts were made simultaneously to build a consensus among the competing advocates to develop a standard radio interface, which crystallized in the agreement on the recommendation for the basic specifications in March 1999. At its last meeting in November 1999, ITU TG8/1 reached an agreement on the recommendation for the detailed specifications of the radio interface, including the specifications relating to higher layers. These draft recommendations were officially approved as an ITU recommendation at the RA-2000 meeting in May 2000. As shown in Figures 1.8 and 1.9, the recommendations suggest the following with respect to the IMT-2000 radio interface:

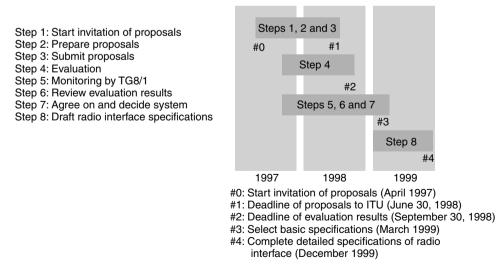
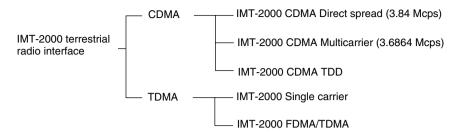
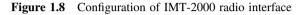


Figure 1.7 ITU-R standardization schedule





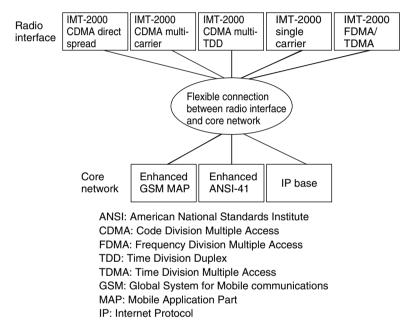


Figure 1.9 Connection between radio interfaces and core networks

- 1. The radio interface standard consists of CDMA and TDMA technologies.
- The CDMA includes Frequency Division Duplex (FDD) direct spread mode, FDD multicarrier mode and Time-Division Duplex (TDD) mode. The chip rate of FDD direct spread mode and FDD multicarrier mode should be 3.84 Mcps and 3.6864 Mcps, respectively.
- 3. The TDMA group consists of FDD single-carrier mode and FDD Frequency Division Multiple Access (FDMA)/TDMA mode.
- 4. Each of these radio technologies must be operable on the two major 3G core networks [e.g. evolved versions of GSM and ANSI-41 (American National Standards Institute)].

The recommendations state the detailed specifications of each mode; among them, direct spread mode is the so-called *W-CDMA*.

From the proposal of the radio interface up to the formulation of basic specifications, a consensus was reached largely due to coordination and harmonization activities by and among the standardization bodies of the countries and regions concerned, including the ITU.

ITU-T's Efforts

ITU-T started working on the IMT-2000 signaling scheme in 1993. Consequently, Q.1701 (Framework for IMT-2000 Networks) and Q.1711 (Network Functional Model for IMT-2000), which specify the framework and architecture of IMT-2000 networks, were officially adopted as recommendations in March 1999 [10, 11].

The IMT-2000 system can be divided into the Radio Access Network (RAN), which controls and terminates radio signals, and the CN, which handles location control, Call Control (CC) and service control. Figure 1.10 shows the logical functional model for IMT-2000 referred to in ITU-T Recommendation Q.1711. RAN includes the BS and the Radio Network Controller (RNC), whereas CN consists of the exchange, the HLR, the Service Control Point (SCP) and so on. The functions inside CN are the same as the logical functions of PDC shown in Figure 1.2, apart from the exchange, which has a packet-switching function Packet Data Serving Node/Packet Data Gateway Node (PDSN/PDGN) and a circuit-switching function [MSC/Gateway MSC (G-MSC)].

ITU-T Recommendation Q.1701 defines a "family concept" that enables global provision of services across multiple IMT-2000 systems, even if they are based on different schemes. The aim is to meet the market demand for utilizing the existing facilities and resources to the greatest extent possible in IMT-2000. The family concept specifies "family members", which are groups of systems that have the IMT-2000 capabilities. ITU-T

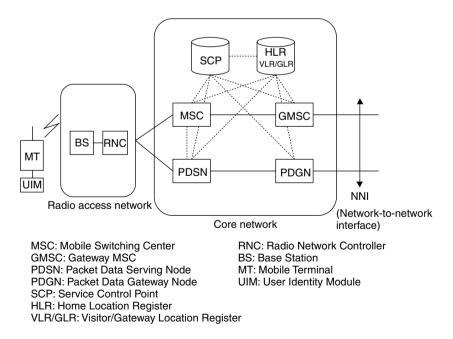


Figure 1.10 Configuration of IMT-2000 logical system (ITU-T)

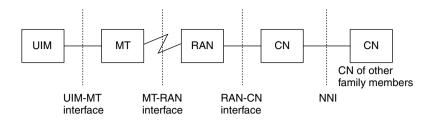


Figure 1.11 Interface supported by IMT-2000 family member

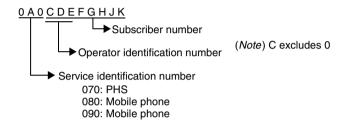


Figure 1.12 Numbering plan for mobile communications in Japan

concentrates on standardizing the interface's signaling scheme required to enable roaming among family members. Each family member is allowed to have specifications unique to its system (Figure 1.11).

As specifications within each family member had to be prepared by the respective regional standardization bodies, two organizations were established between December 1998 and January 1999 with the aim to let the regional standardization bodies develop common specifications: the 3GPP and the Third-Generation Partnership Project 2 (3GPP2). 3GPP adopts W-CDMA for RAN and an evolved-GSM CNs for CN. On the other hand, 3GPP2 has prepared standard specifications for a family system that adopts cdma2000 for RAN and an evolved ANSI-41 CN. This volume elaborates on mobile communication systems that use W-CDMA, which is standardized by 3GPP.

The numbering plan for IMT-2000 mobile communications must comply with ITU-T Recommendation E.164 (The International Public Telecommunication Numbering Plan), and enable mobile users to communicate with users of fixed telephone networks and vice versa [12]. Interconnectivity with other networks is achieved by making the identification number of IMT-2000 mobile phones comply with the domestic numbering plan in each country. In Japan, a numbering plan as described in Figure 1.12 is defined. The numbering system for IMT-2000 is the same as PDC (service identification number: 090/080 mobile phone).

1.2.2.2 Regional Standardization Bodies' Activities Relating to Radio Transmission Systems

In order to submit proposals on radio transmission technologies to ITU-R by June 1998, standardization bodies in each country and region carried out activities to draft proposals.

ARIB

In Japan, ARIB established the IMT-2000 Study Committee (originally the FPLMTS Study Committee), under which the Radio Transmission Technology Special Group conducted studies. There were 24 proposals as of October 1994; later, they were consolidated into three proposals for CDMA FDD, one proposal for CDMA TDD and two proposals for TDMA. As shown in Figure 1.13, the group decided to merge two of the CDMA FDD proposals (B and C) into the core proposal A, and then included TDD as well, in order to integrate them into a single proposal and carry it forward to the detailed study stage. This ultimately became the W-CDMA proposal from ARIB. The decision was approved by the IMT-2000 Study Committee in January 1997. While studies on the other two TDMA proposals were to be sustained at this point, W-CDMA eventually became the sole proposal from Japan to ITU-R as it was subsequently decided that the TDMA proposals would be dropped.

ARIB restructured its organization to conduct detailed studies on W-CDMA. Under the new structure, its Air Interface Working Group (WG) propelled the detailed studies and prepared the specifications, and at the same time, drafted the Radio Transmission Technology (RTT) proposal documentation and evaluation reports for ITU-R.

After the submission of the RTT proposal in June 1998, ARIB continued technical studies and actively engaged in coordination activities with other regions.

ETSI

In Europe, studies were conducted by ETSI. While there had been research projects on Wideband CDMA, Wideband TDMA technologies and so forth, ETSI created five concept groups in 1997, as shown in Figure 1.14, in order to make a decision on the system to be proposed to ITU-R. In the final stage, W-CDMA and TD-CDMA survived as strong candidates and were subject to deliberation. The split between the W-CDMA and TD-CDMA camps continued until the voting at the ETSI Special Mobile Group (SMG), which ultimately resulted in the decision to adopt W-CDMA and TD-CDMA

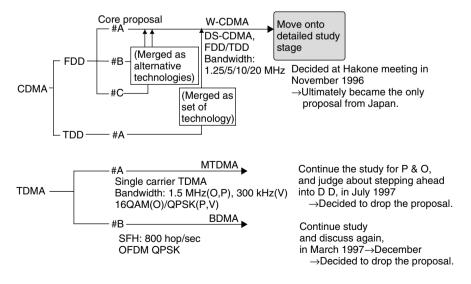


Figure 1.13 Radio transmission technology proposals studied by ARIB

α : W-CDMA
 β : OFDMA
 γ : W-TDMA
 δ : TD-CDMA
 ε : ODMA

Figure 1.14 Radio access concept under ETSI

for paired band and unpaired band, respectively, in January 1998. In Europe, the 3G mobile communication system is called the *Universal Mobile Telecommunications System* (UMTS), whereas the terrestrial radio access system is referred to as the *UMTS Terrestrial Radio Access* (UTRA), which is why W-CDMA is called *UTRA FDD* and TD-CDMA is called *UTRA TDD* in Europe.

Other Standardization Bodies

Standardization bodies that submitted proposals similar to W-CDMA to ITU-R include Telecommunications and Technology Association (TTA) (South Korea), T1P1 and TIA TR46.1 (USA). The proposals made by T1P1 and TR46.1 were later merged into one proposal. China Wireless Telecommunication Standard (CWTS) (China), whose proposal was limited to the TDD system, advocated Time-Division Synchronous CDMA (TD-SCDMA), which is similar to UTRA TDD.

1.2.2.3 3GPP: Specifications Development Group

The radio transmission technology proposals from ARIB and ETSI were harmonized to a great extent by the time they were submitted to ITU-R, with matching basic parameters. This was achieved partly because ARIB members and ETSI members had come together at informal discussions and official conferences on various occasions. There were concerns, however, that specifications developed by region would not result in a genuinely global standard, as compatibility cannot be assured unless the specifications comply with each other in every detail. Consequently, a proposal was made to create a joint forum for developing specifications, and in December 1998, major regional standardization bodies agreed to establish the 3GPP. According to the procedures agreed upon, 3GPP develops the technical specifications, and the completed specifications are approved as technical standard in each country or region by the authorities in charge. The Organizational Partners of 3GPP include ARIB and TTC (Japan), ETSI (Europe), T1P1 (USA), TTA (South Korea) and CWTS (China). In 3GPP, radio access is referred to as UTRA and W-CDMA is called UTRA FDD. 3GPP has developed a single set of detailed specifications centering on the proposals made by ARIB and ETSI, incorporating other individual technologies such as the proposals made by T1P1 (USA) and TTA (South Korea). The specifications also absorbed the proposal made by China as far as TDD is concerned. Since the effective completion of Release '99 (R99) in December 1999, 3GPP has continued to work on the maintenance of R99 and drafting of the next release.

As stated before, specifications developed at 3GPP become the standard of regional standardization bodies. As the specifications of regional standardization bodies are referred to by ITU's recommendations for IMT-2000 (documentation of external organizations need to be referred to for detailed specifications), 3GPP's specifications are ultimately

reflected in ITU's recommendations through this process, even though 3GPP is not a legal entity.

1.2.2.4 Harmonization Activities

As mentioned in the preceding text, efforts to harmonize proposals similar to W-CDMA were carried out through the coordination activities between ARIB and ETSI, and through the establishment of 3GPP complete uniformity was guaranteed. Cdma2000, which is an alternative CDMA proposal made by TIA TR45.5, was ultimately approved as IMT-2000 CDMA multicarrier at ITU-R, and efforts to harmonize the proposal with W-CDMA were continued until the final stages. In addition to official activities, discussion were held at unofficial venues, including those launched by the Operators Harmonization Group (OHG) in January 1999. In May 1999, OHG ultimately decided to make the parameters in both systems similar by partially modifying some key parameters such as the chip rate and to develop specifications that would enable flexible interconnection between the CNs. This was immediately reflected in the 3GPP specifications, as well as the radio transmission technology proposals that had already been submitted to ITU-R.

1.2.2.5 Ministerial Ordinances in Japan

In September 1999, the Telecommunications Technology Council (then) issued a report to the Ministry of Posts and Telecommunications (then) on The Technical Conditions for Next-Generation Mobile Communication Systems [5], which summarized the findings of studies on the technical requirements for introducing IMT-2000 (in the process of standardization by ITU at the time) into Japan. Both Direct Sequence Code Division Multiple Access (DS-CDMA) and Multicarrier Code Division Multiple Access (MC-CDMA) were included as transmission technologies, which correspond to IMT-2000 CDMA direct spread and IMT-2000 CDMA multicarrier, respectively, of the five modes advocated by ITU-R.

In conjunction with the council report, a ministerial ordinance bill was submitted to the Radio Regulatory Council (then) in December 1999, for the purpose of partially revising the enforcement regulations of the Radio Law, the radio equipment regulations and so on. The ordinance was enforced from April 2000.

1.2.3 IMT-2000 Frequency Band

The frequency band for IMT-2000 was assigned at the World Administrative Radio Conference-92 (WARC-92) held in 1992. A total of 230 MHz of spectrum in the 2 GHz band (1885–2025 MHz, 2110–2200 MHz) was allocated presuming that it would be put to use in each country according to market trends and domestic circumstances. However, the subsequent surge in demand for mobile communications and the trends in mobile multimedia led the ITU-R to predict, between 1999 and 2000, that the IMT-2000 frequency band would become insufficient in the near future [13]. Specifically, ITU-R projected that the number of IMT-2000 subscribers would reach 200 million worldwide by 2010 and acknowledged the need to secure a globally common frequency band while achieving lower pricing through the cross-border usage of IMT-2000 terminals on a global scale

and development of common terminal specifications. ITU-R estimated that the shortage of bandwidth in 2010 would amount to 160 MHz in terrestrial systems worldwide, and 2×67 MHz in satellite systems across the globe. In response, the decision was made to deliberate on prospective extra bands to be allocated to IMT-2000 in concrete terms at the World Radiocommunication Conference-2000 (WRC-2000) held between May to June, 2000.

As a consequence, WRC-2000 approved the preservation of the 800 MHz band (806– 960 MHz), the 1.7 GHz (1710–1885 MHz) and the 2.5 GHz band (2500–2690 MHz) for future IMT-2000 use worldwide, and the allocation of adequate frequencies from these bands by each country according to domestic demand and in consideration of other business applications and so on.

References

- Special Articles on Car Phones', *Electrical Communication Laboratories Technical Journal*, 26(7), 1977, 1813–2174.
- [2] Kuramoto, M., 'Large Capacity Car Phone Systems', The Journal of the Institute of Electronics, Information and Communication Engineers, 71(10), 1988, 1011–1022.
- [3] Kuwahara, M., editor, Digital Mobile Communications, Kagaku Shimbun-Sha, Tokyo, 1992.
- [4] Special Article on i-mode Services, NTT DoCoMo Technical Journal, 7(2), 6-32, Jul. 1999.
- [5] 'Technical Requirements of Radio Equipment using Frequency Division Multiple Access based on Code Division Multiple Access', Telecommunications Council Report, Ministry of Posts and Telecommunications, September 1999.
- [6] 'Wideband Coherent DS-CDMA', Special Article on Radio Access, NTT DoCoMo Technical Journal, 4(3), 6–24, Oct. 1996.
- [7] ITU-R Recommendation M.1455, Key Characteristics for The International Mobile Telecommunications-2000 (IMT-2000) Radio Interfaces, May 2000.
- [8] ITU-R Recommendation M.1457, Detailed Specifications of the Radio Interfaces of International Mobile Telecommunications-2000 (IMT-2000), May 2000.
- (9) 'Mobile Application Part (MAP) Signaling System of Digital Mobile Communications Network Inter-Node Interface (DMNI) for PDC', Vol. 7, The Telecommunications Technology Committee JJ-70.10, April 2000.
- [10] ITU-T Recommendation E.164, The International Public Telecommunication Numbering Plan, May 1997.
- [11] ITU-T Recommendation Q.1701, Framework for IMT-2000 Networks, March 1999.
- [12] ITU-T Recommendation Q.1711, Network Functional Model for IMT-2000, March 1999.
- [13] ITU-R Report M.2023, Spectrum Requirements for International Mobile Telecommunications-2000 (IMT-2000), May 2000.

2

Radio Transmission Systems

Mamoru Sawahashi

2.1 Direct Sequence Code Division Multiple Access (DS-CDMA)

2.1.1 Principles of DS-CDMA

DS-CDMA is a radio-access technology that enables multiple access based on a spread spectrum system. Figure 2.1a shows how DS-CDMA works [1–3]. The transmitted data sequence is spread across the spectrum after being encoded by spreading codes, each of which is assigned uniquely to each user at a higher rate than the symbol rate of the information data. [Wideband Code Division Multiple Access (W-CDMA) spreads the information data over a 5 MHz band per carrier.] The spread high-speed data sequence is referred to as *chip* and the rate at which the spread data varies is called *chip rate*. The ratio of chip rate to symbol rate is called the *Spreading Factor* (SF). The destination mobile phone uses the same spreading code as the one used for spreading at the transmission point to perform correlation detection (a process called *despreading*), in order to recover the transmitted data sequence. As signals received by other users carry different spreading codes, the signal power is reduced evenly to 1/SF. In DS-CDMA, all users share the same frequency band and time frame to communicate, and each user is identified by a spreading code uniquely assigned to the user.

In contrast, as shown in Figure 2.1b, Frequency Division Multiple Access (FDMA) assigns to each user a different carrier frequency, depending on the frequency generated in the frequency synthesizer, and Time Division Multiple Access (TDMA) assigns to each user not only a carrier frequency but also a time slot (hereinafter referred to as *slot*) to engage in communications. At the reception point, the frequency generated by the frequency synthesizer is set in such a manner that the signals in the assigned carrier frequency can be down-converted in the destination mobile phone and the transmitted data sequence is extracted from specific slots with reference to the demodulated signals. In DS-CDMA, there is basically no need to assign carrier frequencies or time slots as such to the users.

Figure 2.2 shows a sample waveform of spreading signals, assuming SF = 8. The information data sequence transmitted by Users 1 and 2 is spread with the spreading code assigned uniquely to each user, and a spreading data sequence is generated at a chip rate equivalent to the symbol rate of the information data multiplied by SF. In the

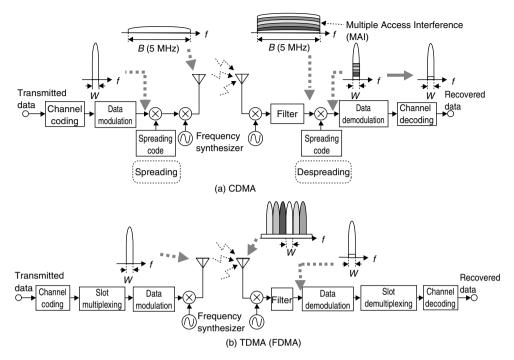


Figure 2.1 Principles of DS-CDMA

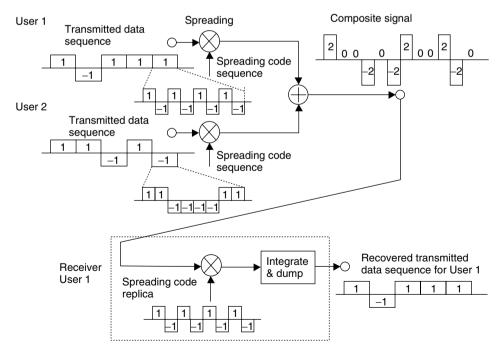


Figure 2.2 Waveform of spreading codes in DS-CDMA

case of Figure 2.2, the spreading data sequences of Users 1 and 2 are added together to generate multiplex signals for transmission over the radio channel. The mobile phone at the receiving end synchronizes the spreading code (same as the one used for spreading) with the code sequence of the received signals and multiplies it by the multiplexed spreading data sequence. After multiplication, signals are subject to integration over the symbol length (which is a process called *despreading* or *integrate and dump*) to recover the transmitted information data sequence.

Assuming that $d_k(t)$ and $c_k(t)$ are User k's data modulation waveform and spreading signal waveform, respectively, $d_k(t)$ and $c_k(t)$ are represented by the following equation:

$$d_k(t) = \sum_{i=-\infty}^{\infty} b_k(i) \cdot u\left(\frac{t}{T_s} - i\right) = \sum_{i=-\infty}^{\infty} \exp[j\phi_k(i)] \cdot u\left(\frac{t}{T_s} - i\right)$$
(1)

$$c_k(t) = \sum_{i=-\infty}^{\infty} p(i) \cdot u\left(\frac{t}{T_c} - i\right)$$
⁽²⁾

In the above equations, T_s and T_c represent the symbol length and the chip length, respectively, in which $SF = T_s/T_c$. u(t) is a step function in which u(t) = 1(0) when $0 \le t < 1$ (otherwise). $p_k(i)$ is a binary spreading code sequence in which $|p_k(i)| = 1$, whereas $b_k(i)$ is an encoding information data sequence. Assuming that the data modulation phase is Quadrature Phase Shift Keying (QPSK), $\phi(i) \in \{j\pi/2 + \pi/4; j = 0, 1, 2, 3\}$.

In a mobile communications environment, multiple paths (multipath) are generated because of variations in transmission time caused by buildings and constructions between the Base Station [BS; referred to as *Node B* under the Third-Generation Partnership Project (3GPP)] and the Mobile Station (MS; referred to as *User Equipment* (UE) under 3GPP). Moreover, the reflection and dispersion of waves due to buildings and so on in the vicinity of MS give rise to random standing waves (referred to as *fading*), as many waves coming from different directions interfere with each other. Multiple paths, marred by variations in delay time and fading unique to each path, lead to *multipath fading*, that is, variation in signal strength within the frequency band. Reception signal r(t) is represented by the following equation, assuming that K is the number of uplink communication users and L_k is the number of paths by which the signals transmitted by User k(k = 0, 1, ..., k - 1) are received via a propagation path affected by multipath fading, in which the delay time varies with each path:

$$r(t) = \sum_{k=0}^{K-1} \sqrt{2S_k} \sum_{l=0}^{L_k-1} \xi_{k,l}(t) c_k(t - \tau_{k,l}) d_k(t - \tau_{k,l}) + w(t)$$
(3)

In Equation (3), S_k represents the transmission power of User k, and $\xi_{k,l}$ and $\tau_{k,l}$ stand for the complex channel gain (fading complex envelope) of user k's path $l(l = 0, ..., L_k - 1)$ and delay time, respectively. It is assumed that $E\left[\sum_{l=0}^{L_k-1} |\xi_{k,l}(t)|^2\right] = 1$, in which $E(\cdot)$ represents the ensemble mean. w(t) is the Gaussian noise portion of the power spectrum density on one side $N_0/2$. With respect to path 0 of User 0, reception signal r(t) is despread by a code Matched Filter (MF) in synchronization with the reception time of path 0 using the spreading code replica of User 0. For the sake of simplicity, it is assumed that $0 \le \tau_{0,0} \le \tau_{k,l} (k \ne 0, l \ne 0) \le T_s$. The despread signal of symbol m in path 0 of User 0 is represented by the equation below:

$$\begin{aligned} z_{0,0}(t) &= \frac{1}{T_s} \int_{mT_s + \tau_{0,0}}^{(m+1)T_s + \tau_{0,0}} r(t) c_0^*(t - \tau_{0,0}) dt \\ &= \sqrt{2S_0} \,\xi_{0,0}(m) b_o(m) \\ &+ \sqrt{\frac{2S_0}{T_s}} \sum_{l=1}^{L_0 - 1} \left[\begin{array}{c} \xi_{0,l}(m-1) b_0(m-1) \int_{mT_s + \tau_{0,0}}^{mT_s + \tau_{0,l}} c_0(t - \tau_{0,l}) c_0^*(t - \tau_{0,0}) dt \\ &+ \xi_{0,l}(m) b_0(m) \int_{mT_s + \tau_{0,l}}^{(m+1)T_s + \tau_{0,0}} c_0(t - \tau_{0,l}) c_0^*(t - \tau_{0,0}) dt \end{array} \right] \\ &+ \sum_{k=1}^{K-1} \sqrt{\frac{2S_k}{T_s}} \sum_{l=0}^{L_k - 1} \left[\begin{array}{c} \xi_{k,l}(m-1) b_k(m-1) \int_{mT_s + \tau_{0,0}}^{mT_s + \tau_{k,l}} c_k(t - \tau_{k,l}) c_0^*(t - \tau_{0,0}) dt \\ &+ \xi_{k,l}(m) b_k(m) \int_{mT_s + \tau_{k,l}}^{(m+1)T_s + \tau_{0,0}} c_k(t - \tau_{k,l}) c_0^*(t - \tau_{0,0}) dt \end{array} \right] \\ &+ \frac{1}{T_s} \int_{mT_s + \tau_{0,0}}^{(m+1)T_s + \tau_{0,0}} w(t) c_0^*(t - \tau_{0,0}) dt \end{aligned}$$

In Equation (4), * represents a complex conjugate. The method of estimating $\xi_{k,l}$ (i.e. the channel estimation method) is described in Section 2.2.5. The first term on the righthand side of Equation (4) is the sequence of information data to be transmitted, the second term is the MultiPath Interference (MPI) of the user's channel, the third term is the Multiple Access Interference (MAI) and the fourth term is the background noise component. In a multipath-fading environment, it is generally difficult to prevent the spreading codes assigned to the respective users from affecting each other, that is, it is hard to achieve perfect orthogonality along the code axis. (In downlink, it is possible to achieve orthogonality between the same propagation channels when the orthogonal coding scheme is used, as has been explained later.) Hence, as shown in Equation (4), the despreading process is marred by interference from multipaths within the user's channel (second term) and interference from other users (third term). As more users communicate at the same time over the same frequency band, the power of the interference increases. The maximum interference power is determined by the Signal-to-Interference Power Ratio (SIR) that meets the prescribed Bit Error Rate (BER) or the BLock Error Rate (BLER), meaning that the number of users that can be accommodated by the system depends on the same.

2.1.2 Spreading Code and Spreading Code Synchronization

There are certain requirements for spreading codes: the autocorrelation peak must be acute upon synchronization (time shift = 0), autocorrelation must be minimal in terms of absolute value when time shift $\neq 0$ and autocorrelation must be minimal in absolute value between different codes at all timings. A code that meets these requirements is the Gold sequence, which is acquired through addition by bit, of the two outputs of alternative maximum period shift register sequences (M-sequences) with the same periods generated by specifying a default value other than 0 for the linear feedback shift register

with a feedback tap as shown in Figure 2.3 (modular 2 adder) [3]. Figure 2.3 shows the scrambling encoder used in downlink W-CDMA. Code sequences with a period of the power of $2^n (n > 3)$ plus "0" at the end of the Gold sequence (which alternatively may be represented as (-1) are called *orthogonal Gold codes*, which achieve orthogonality when time shift = 0 [4]. The Walsh code generated through Walsh-Hadamard Transform is also an orthogonal code with a period of the power of $2^n (n \ge 1)$ [2, 3]. The respective number of Walsh codes and orthogonal Gold codes with a code length of SF is equal to SF. The application of these codes in a cellular system requires spreading code cell iteration, as in the case of frequency reuse that is essential to the TDMA system. As a result, the number of spreading codes that can be used in one cell will be limited, and therefore the system capacity cannot be expanded. To make it possible to use the same orthogonal code sequences repeatedly in each cell, two layers of spreading codes are assigned by multiplying the orthogonal code sequence by scrambling codes with an iteration period that is substantially longer than the information symbol rate [2]. The iteration period of the scrambling code is one-radio-frame long (= 10 msec), that is, 38,400 chips long. It is assigned uniquely to each cell in downlink and to each user in uplink.

In order to extract the information data components, the destination mobile phone needs to execute the spreading code synchronization, which consists of two processes, namely, *acquisition* and *tracking*, in which tracking maintains the synchronization timing within ± 1 chip of acquisition [1, 3]. The despreader may be a sliding correlator or an MF with high-speed synchronization capabilities equivalent to an array of multiple sliding correlators. In W-CDMA, a sliding correlator is generally applied, while MF is often used in the first step of the three-step cell search referred to in Section 2.2.2. For tracking, Delay Locked Loop (DLL) and Tau Dither Loop (TDL) are generally well known [3]. Both of them determine the timing error (*S* curve) with reference to the correlation peak calculated by shifting the synchronization timing of spreading codes by $\pm \Delta$ (in general, $\Delta = 1/2$ chip length) and adjust the timing of the spreading code replica so as to minimize the timing error. In a multipath mobile communications environment, the reception power and the delay time vary dynamically in each path. In such an environment, path search is normally executed on the basis of the power delay profile referred to in

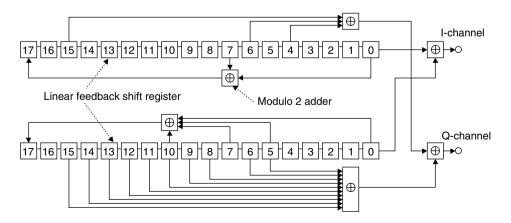


Figure 2.3 Configuration of Gold code encoder

Section 2.2.5.1; DLL and TDL are rarely used owing to their poor ability to track the number of paths with substantial reception power and rapid fluctuations of the delay time in each path.

2.1.3 Configuration of Radio Transmitter and Receiver

Figure 2.4 shows a generic block configuration of radio transmitter and receiver in W-CDMA (DS-CDMA). Layer 1 (physical layer) adds a Cyclic Redundancy Check (CRC) code, for detecting block errors, to each Transport Block (TB), which is the basic unit of data that is subject to processing [unit of data forwarded from Medium Access Control (MAC) layer to Layer 1]. This is followed by channel encoding [Forward Error Correction (FEC)] and interleaving. The interleaved bit sequence is subject to overhead additions (e.g. pilot bits for channel estimation), followed by data modulation. In-phase and quadrature components in the phase plane mapped following data modulation are spread across the spectrum by two layers of spreading code sequences. The resulting chip data sequence is restricted to the 5 MHz band by a square root–raised cosine Nyquist filter (roll-off factor = 0.22) and then converted into analog signals through a D/A converter so as to undergo orthogonal modulation. The orthogonally modulated Intermediate Frequency (IF) signals are further converted into Radio Frequency (RF) signals in the 2 GHz band and are subject to power amplification thereafter.

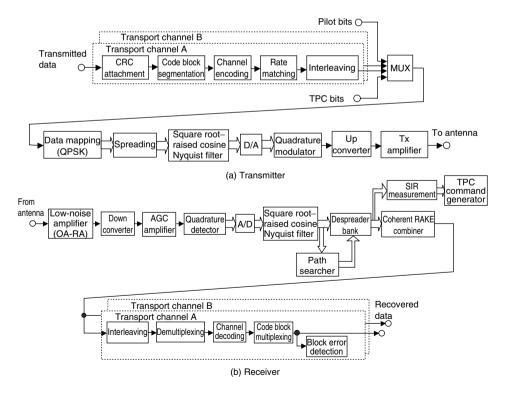


Figure 2.4 Configuration of W-CDMA transmitter and receiver

The signals received by the destination mobile phone are amplified by a low-noise AMPlifier (AMP) and converted into IF signals, to further undergo linear amplification by an Automatic Gain Control (AGC) AMP. The amplified signals are subject to quadrature detection to generate in-phase and quadrature components. The analog signals of these components are converted into digital signals through an A/D converter. The digitized in-phase and quadrature components are bound within the specified band by a square root–raised cosine Nyquist filter and are time-divided into a number of multipath components with different propagation delay times through a despreading process that uses the same spreading code as the one used for spreading the reception signals. The time-divide paths are combined through a coherent RAKE combiner, after which the resulting data sequences are deinterleaved and subject to channel decoding (error-correction decoding). The transmitted data sequence is recovered by binary data decision, which is then divided into transport channels and is subject to block error detection, to be forwarded to the higher layer.

2.1.4 Application of DS-CDMA to Cellular Systems

The following characteristics of the DS-CDMA radio access scheme should be noted when it is applied to cellular systems:

(i) Uplink Requires Transmit Power Control (TPC)

In DS-CDMA, multiple users scattered within the same cell share the same frequency band in order to communicate. Therefore, in uplink, if multiple MSs execute transmission with the same transmission power, damping of the reception signal generally worsens as the distance from BS increases owing to propagation losses. As a result, signals received from an MS located far away from the BS (i.e. around the edge of the cell) are masked by signals received from other MSs that are closer to the BS – the so-called near–far problem. (The power of interference signals entering the destination mobile phone can be reduced to 1/SF on average in the despreading process, but if the power of interference signals is larger than the power of the target signals to the extent of undermining the spreading gain, SIR will be less than 1 after despreading.) Thus, TPC is required for controlling the transmission power of MS so that the power of signals from all users received by BS would be the same [5].

(ii) One-Cell Frequency Reuse Capability

In DS-CDMA, the same frequency band can be applied to adjacent cells (sectors) because each user is identified with reference to a uniquely assigned spreading code (one-cell frequency reuse). Compared to TDMA, the system can thereby expand its capacity in a multicell configuration such as a cellular system. Also, one-cell frequency reuse brings about greater increases in the capacity of systems based on a sector configuration than TDMA.

(iii) Efficient Reception of Multipath Signals by RAKE Reception

In DS-CDMA, data is transmitted through spreading, on the basis of a sequence of highspeed spreading codes. This allows paths with a delay accounting for more than 1 chip length (multipath) to be time-divided and combined in-phase (RAKE combining), which enables the efficient use of multipath signal power and the achievement of higher reception quality.

(iv) Flexible Implementation of Variable Rate Services

Assuming that the spreading frequency band (i.e. chip rate) remains constant, the channel's symbol rate is inversely proportional to SF. Therefore, the symbol rate (i.e. information rate) can be changed in a flexible manner by varying SF without changing the frequency band.

(v) Soft Handover (Site Diversity)

Owing to one-cell frequency reuse, it is relatively easy to implement soft (referred to as *softer* in the case of intersector) handover (also referred to as *site diversity* in terms of establishing radio links with multiple cell sites) [2], which involves the reception and transmission of signals across multiple cells overlapping in time. This enables high-quality reception at the edge of cells free from interruption.

2.2 Basic W-CDMA Transmission Technologies

W-CDMA secures a wider bandwidth of 5 MHz by applying the DS-CDMA radio-access technology with the aforementioned characteristics. The wider band makes it possible to divide and combine reception signals propagated through multipath-fading channels into more multipath components, which helps improve the reception quality through RAKE time diversity. (As the chip rate is 3.84 Mchip/s (cps) and the length of one chip is 0.26 μ s, multipath division can be performed at this resolution.) Its merits include the ability to accommodate a greater number of users who communicate at high speed – for example, at 64 and 384 kbit/s (bps). (It has also been verified in experiments that high-quality data transmission at 2 Mbit/s can be implemented using the 5 MHz bandwidth.) In addition to the fruits of wideband as such, W-CDMA harnesses the distinguishable radio-access technologies explained hereunder.

2.2.1 Two-Layer Spreading Code Assignment and Spreading Modulation

An asynchronous cell configuration allows the system to expand in a seamless, flexible manner from outdoors to indoors, as it does not require a Global Positioning System (GPS)

Figure 2.5 OVSF code–generation method

or any other external time synchronization system. To build an intercell asynchronous system as such, W-CDMA resorts to two-layer spreading code assignment [6, 7]. In short, double-spreading is performed using a short code with an iteration period equivalent to the symbol length (which is referred to as *channelization code* under 3GPP, as the short code is used for identifying each physical channel in downlink) and the scrambling code with an iteration period far longer than the symbol length. When applied to the channelization code, an orthogonal code such as the Walsh code and the orthogonal Gold code enables orthogonality to be achieved between multiplexed code channels where time shift = 0. A method of assigning the Orthogonal Variable Spreading Factor (OVSF) code has also been advocated to secure orthogonality between channels with a different SF (i.e. symbol rate) [8]. Figure 2.5 illustrates how OVSF codes are assigned. Starting at $C_{ch,1,0} = (1)$ (*SF* = 1), OVSF codes can be sequentially generated in the next layer (i.e. double SF) on the basis of the rule represented by Equation (5),

$$\begin{bmatrix} C_{ch,2^{(n+1)},0} \\ C_{ch,2^{(n+1)},1} \\ C_{ch,2^{(n+1)},2} \\ \vdots \\ C_{ch,2^{(n+1)},2^{(n+1)}-2} \\ \vdots \\ C_{ch,2^{(n+1)},2^{(n+1)}-1} \end{bmatrix} = \begin{bmatrix} C_{ch,2^n,0} & C_{ch,2^n,0} \\ C_{ch,2^n,0} & -C_{ch,2^n,0} \\ C_{ch,2^n,1} & C_{ch,2^n,1} \\ C_{ch,2^n,1} & -C_{ch,2^n,1} \\ \vdots \\ C_{ch,2^n,2^n-1} & C_{ch,2^n,2^n-1} \\ C_{ch,2^n,2^n-1} & -C_{ch,2^n,2^n-1} \end{bmatrix}$$
(5)

In the SF = k layer, the number of OVSF codes generated is k, and orthogonality is maintained between the codes totaling k in number. Moreover, orthogonality can be secured even between two OVSF codes in different layers only when *neither* code is derived from the other code (i.e. they are in a hierarchical relationship in the code tree). For example, orthogonality is always maintained between $C_{ch,2,0}$ and $C_{ch,4,2}$, regardless of the symbol pattern of the information data. When the $C_{ch,2,0}$ code is assigned, no code generated from the lower strata of the $C_{ch,2,0}$ code tree can be applied (restriction to OVSF code assignment). In downlink, signals transmitted over multiple channels from BS are received as multipath signals at MS, owing to differences in the duration of propagation resulting from reflection against various buildings, constructions and so forth over different propagation paths. Multiple physical channels that share the same propagation path have the same amplitude and phase shift keying. Hence, the application of OVSF codes between multiple channels (physical channels) that share the same propagation path makes it possible to secure orthogonality between channels even if they do not have the same SF (i.e. symbol rate), as long as they have the same propagation path. This is an extremely effective way to achieve high-quality reception properties.

Figure 2.6 shows the average BER characteristics of MS in downlink when OVSF codes generated according to Equation (5) are used as channelization codes [8]. The figure shows the average BER properties of one channel in which SF = 8 (symbol rate = 512 ksps) and a low-rate (SF = 64) channel in a variable SF transmission that consists of eight channels, in which SF = 64 (symbol rate = 64 ksps) in each channel. The propagation model is a two-path model with equal average power subject to independent Rayleigh fading fluctuations, in which the maximum Doppler frequency $f_D = 80$ Hz. The figure also illustrates the properties of orthogonal multicode transmission over 16 channels, in which SF = 64, and the interference power is the same for each channel

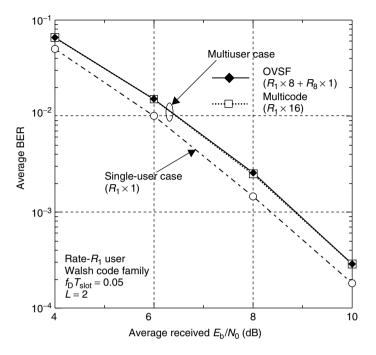


Figure 2.6 Average BER characteristics in downlink using OVSF codes

in which SF = 64 in variable SF transmission. In the case of variable SF and multicode transmissions shown in the figure, as multipath interference increases, the required average reception E_b/N_0 to achieve an average BER = 10^{-3} increases by approximately 0.5 dB compared to a single channel (E_b/N_0 is the abbreviation of signal energy per bit-to-background noise power spectrum density ratio). However, the characteristics of variable SF transmission is extremely similar to those of multicode transmission, and the figure shows that orthogonality is secured in the same propagation path as the channel transmitting eight times faster (SF = 8).

Preference to apply variable SF helps to achieve a lower peak-to-average power ratio at the transmission side than multicode transmission that involves the multiplexing of multiple code channels, and also makes it possible to build a one-sequence RAKE receiver configuration at the receiving end. In the case of high-rate data that cannot be realized even if SF is reduced to 4 or 8, multicode transmission that uses multiple code channels of this SF is applied. Variable SF and multicode transmissions make it possible to transmit information in a flexible manner, ranging widely from low-rate (speech-band) to high-rate communications.

Figure 2.7 shows the spreading modulation process of the Dedicated Physical CHannel (DPCH) in W-CDMA uplink [9]. DPCH consists of the Dedicated Physical Data CHannel (DPDCH), which is mapped into in-phase (I) components, and the Dedicated Physical Control CHannel (DPCCH), which is mapped into quadrature (Q) components. DPDCH is composed of channel-encoding information bits and DPCCH comprises pilot bits for channel estimation, downlink TPC bits, Transport Format Combination Indicator (TFCI)

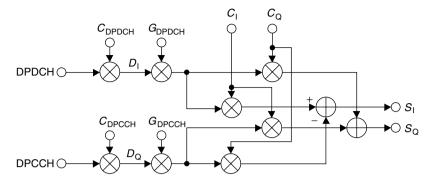


Figure 2.7 Conceptual diagram of complex spreading process

bits and FeedBack Information (FBI) bits used for controlling transmission diversity in downlink. (Refer to Section 2.2.2 onwards for information on each DPCCH bit.) Spreading of channelization codes is performed by using a different OVSF code for each data sequence mapped on the I/Q phase plane. Complex spreading is performed on the spreading data sequence in the I/Q channel with the use of the two scrambling codes generated by time-shifting, according to Equation (6),

$$S_{\rm I} = D_{\rm I}C_{\rm I} - D_{\rm Q}C_{\rm Q}$$

$$S_{\rm Q} = D_{\rm I}C_{\rm Q} + D_{\rm Q}C_{\rm I}$$
(6)

In Equation (6), $D_{I(Q)}$ is the I(Q) component of the data sequence spread by the channelization codes, whereas $C_{I(Q)}$ is the I(Q) component of the scrambling code. G_{DPDCH} and G_{DPCCH} represent the gain of DPDCH and DPCCH, respectively. The merit of complex spreading is that when the amplitude of DPCCH is different from that of DPDCH (i.e. $G_{DPCCH} \neq G_{DPDCH}$), it can substantially reduce the incidences of peak power in comparison to the method of executing spreading over I and Q channels independently of each other, while the ratio of peak power to average power remains the same. In QPSK spreading modulation, the phase shift in the chip after spreading over the I/Q phase plane (i.e. ultimately the shift in the carrier phase subsequent to carrier modulation) might change by 180° to intersect with the origin. In the event of such a phase shift, the impact of nonlinear distortion in the power AMP increases. 3GPP specifications adopt Hybrid Phase Shift Keying (HPSK) [9], which reduces the probability of such 180° phase shift in uplink to decrease the effects of nonlinear distortion in the power AMP.

2.2.2 Cell Search

In W-CDMA, upon the establishment of a radio link between BS and MS, the MS first establishes spreading code synchronization in downlink and then decodes the Broadcast CHannel (BCH) information of the Primary-Common Control Physical CHannel (P-CCPCH) in downlink. The signals are transmitted over a Random Access CHannel (RACH) in uplink according to a predetermined transmission timing. The BS then establishes spreading code synchronization in uplink and decodes the RACH information, to establish the radio link in both uplink and downlink.

Immediately after turning on the power, or before entering soft handover, or when in intermittent reception mode for standby, the MS needs to detect the cell with the smallest path loss (the cell with the second smallest path loss when entering soft handover mode) caused by long-zone fluctuations and shadowing fluctuations in which instantaneous fading fluctuations are averaged. This process involves the detection of a cell with a scrambling code in the Common PIlot CHannel (CPICH) that has the largest reception power (correlated peak power after despreading) in downlink. The process is referred to as *cell search*, as it involves the search of cells required for establishing the radio link. Once the radio link is established by securing spreading code synchronization in downlink, MS transmits RACH at a predetermined timing with reference to the timing in downlink, so that BS can quickly establish spreading code synchronization regardless of the spreading code length, simply by detecting the timing of spreading code synchronization within the scope of uncertainty (the scope of the uplink search window) determined by the propagation delay time. There are three modes of cell search: initial cell search, which involves the search of cells required for establishing the radio link when MS's power is switched on; search of handover-destination cell before executing soft handover and search of cells required for establishing the radio link in the event of intermittent reception during standby mode.

In general, synchronization of spreading codes requires correlation detection on each timing accounting for the length (number of chips) of each and every spreading code that needs to be searched and the detection of the synchronization points. In downlink, the number of scrambling codes is set at a sufficiently large value, 512, to enable flexible scrambling code assignment. Accordingly, in initial cell search, MS needs to conduct a search on 512 types of scrambling codes in a sequential manner to find the scrambling code of the cell with the smallest path loss required for establishing the radio link, which is normally an extremely time-consuming process. In contrast, a synchronous inter-BS system is able to perform quick cell search by applying one type of scrambling code to each cell by time-shifting it at certain intervals. With this in mind, the three-step cell search method has been proposed to enable quick cell search in asynchronous inter-BS systems [10]. In 3GPP, modifications have been made to the Synchronization Code (SC) generation method, cell search radio parameters and so forth on the basis of the cell search scheme advocated in Ref. [9, 11].

2.2.2.1 Three-Step Cell Search

Figure 2.8 shows the configuration of transmission frames in CPICH and Synchronization CHannel (SCH) used for three-step cell search. In the 256-chips-long zone in the header of each slot, the Primary Synchronization CHannel (Primary-SCH) and the Secondary Synchronization CHannel (Secondary-SCH) are code-multiplexed with CPICH for transmission. (P-CCPCH is transmitted to parts excluding the first 256-chips-long part in each slot.) SC is a spreading code used for spreading SCH. There are two types of SC, both of which have a code length of 256: Primary Synchronization Code (PSC), which is used for spreading Primary-SCH, and Secondary Synchronization Code (SSC), for spreading Secondary-SCH[9]. As described later, an MF is used to detect Primary-SCH. As the circuit would become bulkier if a 256-tap MF is used for the direct detection of PSC correlations, 256-chip code sequences are generated through the iteration of 16 modulation patterns with minimal autocorrelation peaks based on time-shifted, 16-chips-long

in

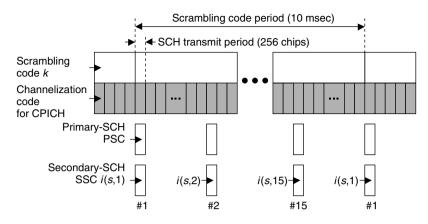


Figure 2.8 Configuration of CPICH and SCH transmission frames

orthogonal code sequences. Assuming that C_{PSC} represents PSC, C_{PSC} is defined by the equation below as a complex code sequence in which the real part and the imaginary part are equal.

$$= \{1, 1, 1, 1, 1, 1, -1, -1, 1, -1, 1, -1, 1, -1, 1, -1, 1\}.$$

There are 16 types of SSC; assuming that this is represented by $C_{\text{SSC},k}(k = 0, 1, 2, ..., 15)$, 1 code in C_{SSC} is equivalent to 256 components generated by multiplying the *j* component of vector **Z** in the 256-chips-long common sequence ($0 \le j \le 225$) and the *j* component of the *n*th row in the Hadamard matrix **H**₈. As it would be extremely time consuming to perform correlation detection on all 512 scrambling codes, the 512 codes are divided into 64 groups in advance. Once the group is identified, cell search is executed on the 8 scrambling codes belonging to that group, thereby shortening the time consumed in cell search. As represented in $n = 16 \times (k - 1)$, every 16th row is selected from the 256 rows in the Hadamard matrix (i.e. 16 rows are selected in total) to generate 16 units of C_{SSC} . Assuming that the *j*th symbol in the *n*th row of the Hadamard matrix is $h_n(j)$ and the *j*th symbol of common sequence *Z* is z(j), $C_{\text{SSC},k}$ is represented by the following equation.

$$-x_{10}, -x_{11}, -x_{12}, -x_{13}, -x_{14}, -x_{15}, -x_{16}$$
(8)

The following equation represents the reception signal assuming that CPICH, Primary-CCPCH (BCH), SCH and C-channel DPCH are transmitted from K cells. For the sake of simplicity, the equation is based on a one-path model. On the right hand side, the

first term is CPICH ($S_{k,0}$, $c_{k,0}$ and $d_{k,0}$ represent the transmission power, spreading code waveform and data modulation signal waveform of CPICH, respectively), the second term is Primary-CCPCH ($S_{k,1}$, $c_{k,1}$ and $d_{k,1}$ represent the transmission power, spreading code waveform and data modulation signal waveform of Primary-CCPCH, respectively), the third term is DPCH, the fourth term is SCH and the fifth term is the background noise component.

$$r(t) = \sum_{k=0}^{K-1} \sqrt{2S_{k,0}} \,\xi_k(t) c_{k,o}(t-\tau_k) d_{k,0}(t-\tau_k) + \sum_{k=0}^{K-1} \sqrt{2S_{k,1}} \,u(t) \xi_k(t) c_{k,1}(t-\tau_k) d_{k,1}(t-\tau_k) + \sum_{k=0}^{K-1} \xi_k(t) \sum_{i=2}^{C+1} \sqrt{2S_{k,i}} \,c_{k,i}(t-\tau_k) d_{k,i}(t-\tau_k) + \sum_{k=0}^{K-1} \sqrt{2S_{k,1}} \,[1-u(t)] \xi_k(t) [c_{\text{psc}}(t-\tau_k) + c_{\text{ssc},i(s,m)}(t-\tau_k)] + w(t)$$
(9)

Only the 256-chips-long zone in the header of each slot is transmitted over Primary-SCH and Secondary-SCH. Hence, u(t) = 0 in the range of $nT_{\text{Frame}} + mT_{\text{Slot}} \le t \le nT_{\text{Frame}} + mT_{\text{Slot}} + 256T_{\text{c}}$ (in which $T_{\text{Frame}} = 38,400T_{\text{c}}, T_{\text{Slot}} = T_{\text{Frame}}/15, n = \text{integer representing}$ the frame number and $m = \text{Value representing the slot number}, 0 \le m \le 14$). i(s, m) shows the SSC transmission patterns unique to scrambling code group s, which may be between 1 and 16. The pattern of scrambling codes used in SSC alternate every 15 slots, depending on the scrambling code group. By detecting the code patterns, MS can identify the scrambling code group used for spreading the reception signal and determine the reception timing (frame timing) of the scrambling codes.

In Steps 1 and 2, the correlation value of Primary-SCH and Secondary-SCH are time-averaged with T_1 and T_2 , respectively, to calculate the maximum correlation peak excluding the impact of instantaneous fading fluctuations. In a low-speed fading environment, the incidence of erroneous detection increases in the event of the failure to fully remove the impact of fading fluctuations within the average time in Steps 1 and 2, especially when the reception power is small. To reduce such erroneous detection in these steps, Time Switched Transmit Diversity (TSTD) is applied, in which the Primary-SCH and Secondary-SCH of the same slot are transmitted as a pair alternately from two transmit ANTennas ANTs of the BS [11, 12]. The application of TSTD helps to reduce variations in the reception level caused by fading fluctuations, through the isolation of fading fluctuations when the fading correlation between two ANTs is small.

Figure 2.9 shows the operation flow of three-step cell search, which detects the cell required for establishing the radio link in three steps as described below [10].

Step 1: Detection of Primary-SC Reception Timing

MS detects the correlation between the reception signal and the PSC using MF and detects the correlation peak in the Primary-SCH reception location. The instantaneous correlation

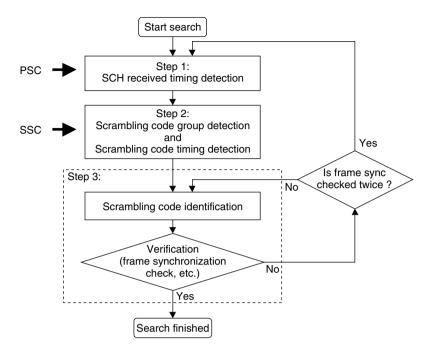


Figure 2.9 Three-step cell search flow

output of MF at time t is described by the following equation.

$$\psi_1(t) = \frac{1}{256T_c} \int_0^{256T_c} r(t-\mu) \cdot c_{\rm psc}(256T_c-\mu) \, d\mu \tag{10}$$

The value of $\psi_1(t)$ when $t = \tau + mT_{\text{Slot}} + nT_{\text{Frame}}$ is represented by $\psi_1(\tau, m, n)$. The value is averaged over time $T_1(=N_1T_{\text{Frame}}; N_1 = \text{natural number})$ in order to reduce the impact of fading fluctuations as well as the impact of noise and interference on the instantaneous correlation power calculated by $\psi_1(\tau, m, n)$. Signal $(\overline{\Psi})_1(\tau)$ subsequent to averaging is represented by the following equation, assuming that cell search in Step 1 begins from frame number n_1 .

$$\overline{\Psi}_{1}(\tau) = \frac{1}{15N_{1}} \sum_{n=n_{1}}^{n_{1}+N_{1}-1} \sum_{m=0}^{14} |\psi_{1}(\tau,m,n)|^{2}$$
(11)

It is assumed that the time at which $(\overline{\Psi})_1(\tau)$ is maximized is the reception timing of SCH of the cell that needs to be searched. In other words, $\frac{\max}{\tau}\overline{\Psi}_1(\tau) \Rightarrow \hat{\tau}$. As frame synchronization is yet to be done at this stage, *m* is subject to 15 patterns of uncertainty.

Step 2: Identification of Scrambling Code Group and Detection of Frame Timing

In Step 2, the reception timing of Primary-SCH detected in Step 1 (i.e. reception timing of Secondary-SCH), $\hat{\tau}$, is used to calculate the correlation between the reception signal r(t) and 15 SSC patterns in 64 scrambling code groups in $\hat{t} = \hat{\tau} + mT_{\text{Slot}} + nT_{\text{Frame}}$, according

to the equation below.

$$\psi_2(x,\hat{\tau},m,n) = \frac{1}{256T_c} \int_0^{256T_c} r(\hat{t}-\mu) \cdot c_{\mathrm{ssc},x}(256T_c-\mu) \, d\mu \tag{12}$$

In Equation (12), x refers to any value that may be applicable to i(s, m), that is, an integer of $0 \le x \le 15$. $\psi_2(x, (\hat{\tau}), m, n)$ is averaged in the same manner as in Step 1 over time $T_2(=N_2T_{\text{Frame}}; N_2 = \text{natural number})$. As described in the following equation, one of the averaging methods is to add power to the correlation output amplitude component of SSC in each slot (assuming that averaging starts at frame number n_2).

$$\overline{\Psi}_2(s,m) = \frac{1}{15N_2} \sum_{n=n_2}^{n_2+N_2-1} \sum_{k=0}^{14} |\psi_2(i(s,k),\hat{\tau},(m+k) \bmod 15,n)|^2$$
(13)

As PSC has already been detected in Step 1, phase addition can be performed as shown in the following equation, by assuming that the reference phase is the PSC correlation output for compensating phase variations caused by fading. The process of in-phase addition and averaging helps reduce background noise and interference components, which leads to greater detection accuracy in Step 2 than the power addition and averaging method shown in Equation (13).

$$\overline{\Psi}_{2}(s,m) = \frac{1}{15N_{2}} \left| \sum_{n=n_{2}}^{n_{2}+N_{2}-1} \sum_{k=0}^{14} \psi_{2}[i(s,k), \hat{\tau}, (m+k) \mod 15, n] \right|^{2} \times \xi^{*}[\hat{\tau}, (m+k) \mod 15, n] \right|^{2}$$
(14)

In Equation (14), $\xi(\hat{\tau}, (m+k) \mod 15, n)$ is the value of $\overline{\Psi}_1(t)$ in $t = \hat{\tau} + (m+k)T_{\text{Slot}} + nT_{\text{Frame}}$. The scrambling code group *s* and frame timing are calculated with reference to the SSC set and the timing that maximize the correlation output power, according to Equation (13) or (14). In other words, $\max_{s,m} \overline{\Psi}_2(s, m) \Rightarrow \hat{s}, \hat{m}$.

Step 3: Identification of Scrambling Codes

MS identifies the scrambling code by detecting the correlation between the reception signal and the candidate scrambling codes in the scrambling code group detected with reference to the frame timing detected in Step 2 and by determining the threshold level. Assuming that $L(\hat{s}, i)$ stands for the index of the *i*th scrambling code in group \hat{s} , correlation detection is performed on scrambling code $L(\hat{s}, i)$ using CPICH on the basis of H_3 symbol length for each scrambling code. The correlation output power is represented by the following equation.

$$\Psi_{3}(L(\hat{s},i),\hat{\tau},\hat{m},n) = \frac{1}{H_{3}} \sum_{j=1}^{H_{3}} \left| \frac{1}{256T_{c}} \int_{256jT_{c}+\hat{\tau}}^{256(j+1)T_{c}+\hat{\tau}} r(t+\eta) \cdot c_{L(\hat{s},i),0} \right|^{2} \times (\eta + \hat{m}T_{\text{slot}} + nT_{\text{Frame}} - \hat{\tau}) d\eta \right|^{2}$$
(15)

If the correlation peak power in Equation (15) is larger than the predetermined threshold power, the scrambling code is regarded as the one for the cell that needs to be searched. In other words, $c_{L(\hat{s},i),0} \Rightarrow c_{\hat{k},0}$ if $\Psi_3(L(\hat{s},i), \hat{m}, \hat{\tau}) \ge \varepsilon \overline{\Psi}_1(\hat{\tau})$ (in which ε is a variable. In this case, the correlation peak in Step 1 multiplied by ε is used as the threshold value to identify the scrambling code).

As PSC is common to all cells, the probability of MS receiving Primary-SCH from multiple cells at the same timing is not zero. (Transmission offset is assigned to prevent the SCH transmission timing from overlapping in adjacent cells or sectors.) Although it is extremely unlikely, even if the reception timing of multiple SCHs turns out to be the same at the MS, it is virtually improbable for the scrambling code and the scrambling code group to match. Thus, the cell with the largest reception power is detected in Steps 2 and 3. When TSTD is used, the detection of Primary-SCH is performed through power addition, and Secondary-SCH is averaged through in-phase addition, using the Primary-SCH reception phase as the reference phase. As a result, the process in the receiver is the same as in the case of transmission from one ANT, while erroneous detection can be reduced especially in a low-speed fading environment due to the diversity effect of transmitting SCH alternately from two ANTs.

The characteristics of detection probability of cell search time obtained from field tests of three-step cell search, performed in the Funabashi region near Tokyo, are described in this section. At a chip rate of 4.096 Mcps, the spread bandwidth was 5 MHz. One frame was 10 msec, consisting of 16 slots (slot length = 0.625 msec). BS executed transmission over CPICH, Primary-SCH and Secondary-SCH, in addition to 10-channel code-multiplexed DPCH with a symbol rate of 64 ksps (SF = 64) as a load. Signals were transmitted from a 60° sector ANT in one of the six sectors in the direction of the measurement course. The spreading modulation method for Primary-SCH and Secondary-SCH was Binary Phase Shift Keying (BPSK) and for CPICH and DPCH was QPSK. The transmission power of Primary-SCH and Secondary-SCH was set at the CPICH transmission power -3 dB. The ANT of BS was 59 m high, and MS was loaded on a measurement vehicle, with its ANT being 2.9 m high. The test was conducted along measurement courses 1 and 2 referred to in Ref. [8], at an average speed of 30 km/h. Figure 2.10 shows an example of the measured power delay profile along measurement course 1. The peripheral environment of the measurement course will not be explained here as it is described in Ref. [13]. In course 1, two paths were observed in the beginning of the course, more or less one path of signals in the middle and two to three paths of signals of unequal average power in the end. In course 2, two to three paths were observed in the beginning, one path of signals in the middle and latter parts owing to the elevation in the course and three paths of limited power in the end of the course. The maximum delay time in the measurement course was approximately 1 µsec.

Figure 2.11 shows the characteristics of detection probability against the cell search time in cases where Primary-SCH, Secondary-SCH, CPICH and 10-channel DPCH were transmitted on the basis of a single cell model [13]. In this assessment, the 512 scrambling codes were divided into 32 groups. In order to reduce erroneous synchronization, the identification of scrambling codes in Step 3 was followed by the confirmation of synchronization by pattern-matching 128 pilot bits per frame. If errors in the 128 pilot bits per frame were 25 bits or less, cell search was deemed to have been completed. If errors accounted for 26 bits or more, Step 3 was repeated; when pilot-aided confirmation of

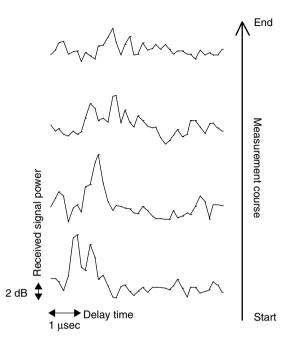


Figure 2.10 Example of power delay profile in field test

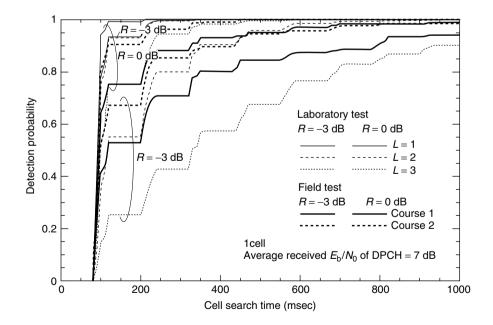


Figure 2.11 Detection probability in cell search time using three-step cell search method

synchronization failed twice in a row, the search was performed again from Step 1. When the scrambling code detected by MS matched that of BS, cell detection was regarded to have ended normally. When scrambling codes other than the ones from the BS were detected, it was treated as erroneous detection and the cell search time was regarded infinite. The figure shows the properties when average reception E_b/N_0 of DPCH = 7 dB in the measurement course. The assessment was made when the transmission power ratio of CPICH to DPCH was R = 0, -3 dB. The properties of course 1 (full line) and course 2 (broken line) are shown in the figure, as well as the properties of the equal average power L = 1, 2, 3 path model (maximum Doppler frequency $f_D = 80$ Hz) in a laboratory test, for the purpose of comparison. As shown in the power delay profile, course 1 has two to three paths and course 2 has more or less one path for propagation. The properties of the cell search time of course 1 are almost identical to the two-path properties determined in the laboratory test, as those of course 2 are to three-path properties. As shown in the figure, an increase in R brings about a reduction in cell search time owing to improvements in the received SIR of CPICH. When DPCH average reception $E_{\rm b}/N_0 = 7$ dB, assuming R = 0, -3 dB, cell search time that can achieve a detection probability of 90% in course 1 is approximately 200 and 450 msec and in course 2, approximately 130 and 250 msec, which shows that quick cell search properties can be attained. As described before, the number of scrambling code groups under the 3GPP specifications is 64 (there are 8 scrambling codes in each group). However, since the number of codes in SSC in Step 2 are the same as in this assessment, the time consumed in calculating the correlation peak power due to the doubling of the number of groups is relatively shorter than the time consumed in SSC correlation detection and averaging. In Step 3, 16 correlators are used to carry out the correlation detection process on 16 scrambling codes that are parallel to each other, so that there is no major difference in search time in Step 3 even if the number of scrambling codes is reduced to 8. Hence, it is believed that more or less the same cell search characteristics as those observed in the test in Funabashi could be demonstrated when 3GPP parameters are applied.

2.2.2.2 Peripheral Cell Search During Communications in Active Mode

Peripheral cell search during communications in active mode, which takes place before executing soft handover, is different from initial cell search since the scrambling codes of peripheral cells are notified over BCH from the handover-source cell site that is already connected to DPCH, so that the cell search only has to be done on about 20 notified scrambling codes. As in the case of initial cell search, three-step cell search can be applied in this case. Since the reception timing and the reception power of the CPICH path of the handover-source cell with DPCH connection are already known on the basis of the measurements of the power delay profile, the path from the handover-source cell is excluded from the power delay profile generated during peripheral cell search, in order to detect the cell that transmits CPICH with the second largest reception power and the scrambling code of that cell. If no such cell can be detected after repeating this process over a predetermined number of times, three-step cell search is performed without excluding the path from the handover-source cell may be the same as that of the path from the cell that needs to be searched, that is, the one with the second largest reception power.

In peripheral cell search in active mode, although the number of candidate cells is much smaller (about 20) than in initial cell search, the interference from the common channel and DPCH from the handover-source cell has an extremely large impact on the search of the cell with the second largest reception power. Reportedly, it takes longer to execute cell search than in initial cell search, as more time is consumed in the averaging process in each step in an effort to reduce the impact of such interference [14].

2.2.2.3 Peripheral Cell Search During Intermittent Reception (Idle Mode)

In intermittent reception mode during communication standby (idle mode), an algorithm has been advocated to achieve a faster cell search than the three-step cell search method [15]. Figure 2.12 shows an example of the relative transmission timing phase of scrambling codes. $Cell^{(k)}$ is the cell through which the radio link is currently established, and cells surrounding $\operatorname{Cell}^{(k)}$ are represented by $\operatorname{Cell}_1^{(k)}$, $\operatorname{Cell}_2^{(k)}$ and so on. The difference in transmission timing of the CPICH scrambling codes between Cell^(k) and the peripheral cells is indicated by Δ_{k1}, Δ_{k2} and so on. Before switching to soft handover mode, MS measures the difference in the timing of scrambling code transmission by CPICH between the handover-source cell and the handover-destination cell and notifies the findings to the handover-source cell. Normally, the location at which MS measures the difference in CPICH scrambling code timing among multiple cells depends on the MS-it should be noted that this is the location where the difference between the CPICH reception level of the cell that establishes the radio link and the peripheral cells falls below the handover threshold. Therefore, owing to the differences in propagation delay time, the reception timing of scrambling code between specific cells measured by each MS varies. To tackle this, $\operatorname{Cell}^{(k)}$ averages the difference with peripheral cell $\operatorname{Cell}_i^{(k)}$ in CPICH scrambling code timing notified from many MSs, to determine the average scrambling code timing difference between $\operatorname{Cell}^{(k)}_{i}$ and $\operatorname{Cell}^{(k)}_{i}$.

Figure 2.13 shows the operation flow of high-speed cell search in MS during intermittent reception. During communication standby, MS executes cell search on the cell carrying CPICH with the largest reception level on a regular basis and receives the Paging CHannel (PCH) from that cell intermittently. Through PCH, MS receives information relating to the type of scrambling code of Cell^(k) that established the radio link or received and demodulated PCH in the course of intermittent reception and of Cell^(k) ($1 \le i \le n$,

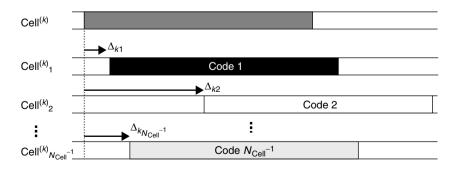


Figure 2.12 Relative transmission timing of scrambling codes in downlink

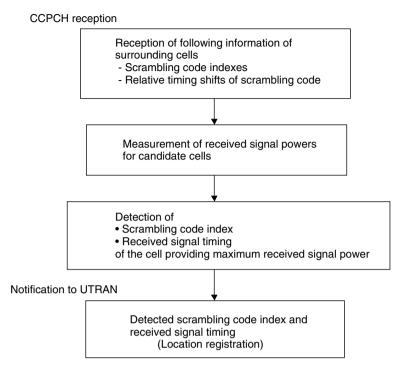


Figure 2.13 High-speed cell search algorithm during intermittent reception

in which n = approximately 20 in number) that needs to be searched in the periphery of Cell^(k), as well as information concerning the difference in CPICH scrambling code timing between Cell^(k) and Cell^(k). As the type of the scrambling code of the peripheral cell that needs to be searched and the CPICH average reception timing at MS are already known, peripheral cell search can be performed in a short time. (This corresponds to the situation in which the code phase that needs to be searched is already known by the inter-BS synchronization system.) Also, highly accurate cell detection is made possible through the detection of cells using the average correlation value calculated by averaging the power more than once with reference to correlation profiles generated upon intermittent reception in the past, which reduces the impact of reception level fluctuations caused by fading.

2.2.3 Random Access

Upon the establishment of a radio link, MS establishes the radio link in downlink through cell search, and then transmits RACH of the uplink [the corresponding physical channel is the Physical Random Access CHannel (PRACH)] [11]. The transmission of PRACH involves the use of slotted ALOHA: MS starts transmitting PRACH from a predetermined number of time-offsets, 15 of which are set at an interval of 5120 chips in 2 radio frames, called *access slots*. In random access control, the upper layer selects the subchannel group from the random access service groups that can be used by the corresponding

Access Service Class (ASC) and uses one signature randomly selected from access slots and signatures that can be used in the chosen random access subchannel group.

Figure 2.14 shows the configuration of PRACH [16], which consists of at least one preamble and message section. The preamble (chip length = 4096) is a short signal transmitted for the purpose of detecting spreading code synchronization before transmitting the message section. The preamble is spread by two layers of spreading codes: short codes based on the repetition of the signature 256 times and scrambling codes assigned from the upper layer. On the other hand, the message section is spread by short codes of OVSF codes that are uniquely defined by the preamble signature and the same two layers of spreading codes as the ones used for the preamble signature. This means that the spreading codes and the reception timing of the subsequent message section can be detected by the detection of the preamble. When the transmission power of the preamble is extremely large, other users' signals would suffer substantial interference. To tackle this, a technique called *power ramping* is applied, in which the transmission power of the preamble is gradually increased in steps that are determined from a small default value specified by the upper layer. MS repeats transmission by increasing the transmission power exactly by the step-width of power ramping several times, until the Acquisition Indicator (AI) is received over the Acquisition Indicator CHannel (AICH), which indicates that the preamble has been detected from BS. Reportedly, as there is hardly any variation in the propagation path of the preamble section and the subsequent message section, highly accurate path detection is possible on the basis of the detection of the RAKE-combined path using the pilot symbol of the message section in addition to the preamble section [17].

2.2.4 Technologies that Satisfy Various Quality Requirements in Multirate Transmissions

2.2.4.1 Error Control

There are two ways to control errors: channel encoding [Forward Error Correction (FEC)] and Automatic Repeat reQuest (ARQ). In W-CDMA, the use of channel encoding (FEC) in parts of the bandwidth expanded through spreading by random codes can increase the gain due to channel encoding, in addition to the spreading gain. Turbo codes have been advocated for channel encoding, which outperform convolutional encoding [18, 19]. Figure 2.15 shows an example of the configuration of a turbo encoder and decoder. The turbo encoder consists of two Recursive Systematic Convolutional (RSC) encoders RSC1

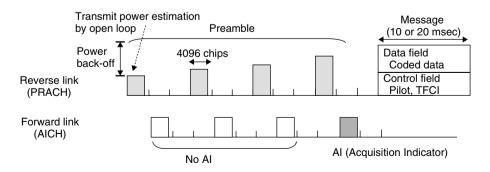


Figure 2.14 Operations in random access

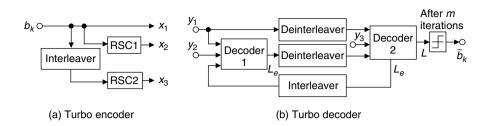


Figure 2.15 Configuration of turbo encoder and decoder

and RSC2 and a turbo interleaver inside the turbo encoder. The receiver enters the channelinterleaved, soft-decision RAKE output $\{y_1, y_2, y_3\}$ into the turbo (replication) decoder. In the iterative decoding algorithm of the turbo decoder, the soft-input, soft-output Decoder 1 calculates external information L_e with reference to y_1 and y_2 . Next, the soft-input, soft-output Decoder 2 updates L_e with reference to y_1, y_3 and L_e and feeds back L_e to Decoder 1 to repeat the aforementioned process. After *m* iterations, the transmission data sequence is recovered by the hard-decision of the Log Likelihood Ratio (LLR) $L(b_k)$. LLR for decoded bit $b_k, L(b_k)$, is represented by the following equation [20].

$$L(b_k) = \ln\left(\frac{P(b_k = +1)}{P(b_k = -1)}\right)$$
(16)

In Equation (16), $P(b_k = +1)$ and $P(b_k = -1)$ refer to the probability of $b_k = +1$ and $b_k = -1$, respectively. A soft-input, soft-output decoder such as the Max-log-Mobile Application Part (MAP) decoder is used. In W-CDMA, channel encoding involves the use of convolutional encoding for speech and low-speed data transmissions and turbo encoding for high-speed data transmissions at 64 kbit/s, 384 kbit/s and so on. Channel interleavers and turbo interleavers inside turbo encoders that have been advocated include the Multistage Interleaver (MIL) [21], which randomizes the orderly parts of the block interleaver and the Prime Interleaver (PIL) [22], which reduces the processing volume of MIL.

In packet-switched (PS) transmission of data traffic, error control especially by ARQ is a prerequisite because of the need to assure error-free transmission. In addition, it must be used in conjunction with hybrid ARQ with an FEC function (error-correction decoding by FEC before ARQ error detection) when applying adaptive modulation, demodulation and error correction that improves throughput by selecting the optimal modulation and encoding scheme depending on the status of the propagation path as described in Chapter 7, because the measurement errors, control delays and so forth inevitably cause packet errors. Figure 2.16 shows the mechanism of hybrid ARQ. ARQ used in Radio Link Control (RLC) under 3GPP is a Selective Repeat (SR) Type-I (a retransmission technique in which the data of the retransmitted packet is the same as the original packet) Hybrid ARQ (hereinafter referred to as *Basic Type-I*) [23]. At the point of transmission, Basic Type-I applies error detection coding and FEC to the information signal sequence for transmission. At the point of reception, the received packet is subject to error-correction decoding, after which errors are detected by error detection codes. If any errors are found, the packet including the error is disposed of and a retransmission request is fed back to

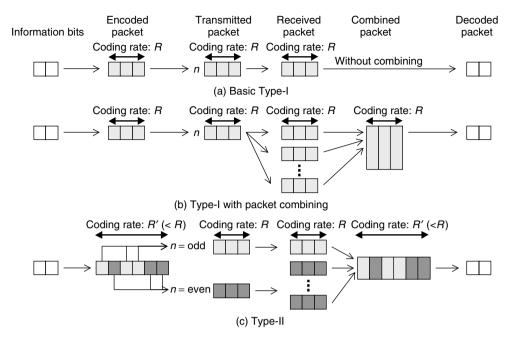


Figure 2.16 Mechanism of hybrid ARQ

the transmitter, where the packet is encoded by the same code in response to the retransmission request and retransmitted. This process is repeated until no errors are detected, which makes error-free transmission possible. In this manner, Basic Type-I uses FEC in combination with ARQ to perform error-correction decoding prior to error detection, which helps reduce the packet error rate and improve throughput characteristics.

Reception characteristics can be further enhanced by storing, in the reception buffer, the soft-decision information of the packet in which an error was detected and by combining it with the retransmission packet for each symbol, which improves the Signal to Interference and Noise power Ratio (SINR). This process, which is called *Packet* Combining (PC) or Chase Combining, can be applied together with Type-I Hybrid ARQ (hereinafter referred to as Type-I with PC) [24, 25]. An alternative Hybrid ARQ scheme is the Type-II (a retransmission method in which the packet to be retransmitted consists of data different from the original packet) Hybrid ARQ (hereinafter referred to as Type-II) [26]. This scheme was originally designed to transmit information bits first, and in the event of retransmission, send FEC parity bits for error correction. The scheme can also be implemented to convolutional encoding and turbo encoding based on punctured encoding [26]. After encoding the information signal sequence at encoding rate R, the sequence undergoes punctured encoding to be transmitted on the basis of a deletion rule depending on the number of times it is to be transmitted [e.g. encoding rate R(=2/3) > R'(=1/3)]. As the retransmitted coding sequence is different from the sequence transmitted first, the receiver can execute decoding at a rate lower than the post-punctured encoding rate R by combining the packet initially received and stored in the reception buffer with the retransmitted packet (code combining). This way, Type-II improves the reception characteristics by improving the encoding gain. As Type-II requires a reception buffer proportionate to the size of the packet encoded at the encoding rate prior to punctured encoding, its reception buffer must be larger than Type-I with PC, assuming that the encoding rate at the time of transmission is the same. Reportedly, Type-II is somewhat superior to Type-I with PC in terms of throughput characteristics over multipath-fading channels [27].

2.2.4.2 Rate Matching

The transport channel is a channel defined for the transmission of different types of data and is offered to the MAC layer. As shown in Figure 2.17, multiple transport channels with various information rates and Quality of Service (QoS) are mapped and transmitted over one physical channel. As previously explained, the TB is the basic unit of data transfer in the MAC layer and Layer 1 [e.g. 1 (1) represents the first block in transport channel 1] [28]. By adaptively changing the transmission power and modulation scheme (adaptive modulation and demodulation) depending on fading fluctuations, it ensures that the quality of the physical channel remains unchanged according to QoS (BLER or BER). Generally, this can be achieved by changing the target SIR value and altering the average reception (transmission) power based on adaptive fast (TPC) [2] outer-loop control according to QoS. However, the average reception power is constant because the target SIR is uniform in 1 radio frame (=10 msec) in the physical channel. Therefore, in order to transmit transport channels with different QoS over one physical channel, the number of bits in the encoded data sequence to be mapped over the physical channel is changed so that the transport channels simultaneously satisfy various OoS requirements based on the same average reception signal power (rate matching). In short, the reception quality after decoding improves when bits are repeatedly inserted into the encoded data sequence at a fixed cycle ("repetition"), while the reception quality deteriorates after decoding when the bit sequence is punctured from the FEC bit sequence at a fixed cycle ("puncture") [28]. Thereby, the number of bits (rate) of each transport channel mapped over the physical channel can be flexibly changed by radio frame. The following technical terms shall be

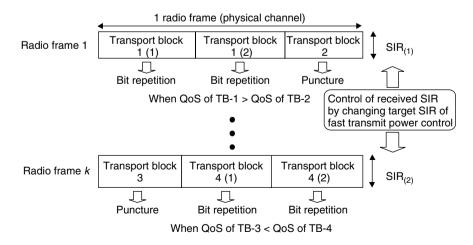


Figure 2.17 Operations in rate matching

noted here. A set of TBs for each transport channel is referred to as a TB set, and the time interval in which the TB set is transferred between the MAC layer and Layer 1 is called the *Transmission Time Interval* (TTI). TTI is equivalent to the interleave length of channel encoding, which may be 10, 20, 40 or 80 msec. The format in which the TB set is transferred between the MAC layer and Layer 1 at each TTI over the transport channel is referred to as the *Transport Format* (TF). The combination of TFs that can be transferred simultaneously in Layer 1, that is, the combination of TFs of transport channels to be mapped over one physical channel, is called the *Transport Format Combination* (TFC) and the set of all TFCs that can be transferred over this physical channel is known as the *Transport Format Combination Set* (TFCS).

In uplink, rate matching is performed on the encoded data sequence after the initial interleaving of the transport channels [28]. Each transport channel calculates the number of bits subject to bit iteration or puncturing based on the rate-matching attribute specified by the upper layer. In uplink, Discontinuous Transmission (DTX), which is the mode in which transmission is not carried out if there are no transmission bits in the transport channel to be mapped over the physical channel, is not defined. As such, SF of the physical channel (i.e. symbol rate) is first determined according to the sum of the number of bits per radio frame in the transport channels to be mapped over one physical channel. Then, the rate matching is performed in such a manner that the sum of the number of bits per radio frame of all transport channels subsequent to rate matching would be the same as the number of bits per radio frame of the physical channel with the assigned SF. $N_{i,j}$ is the number of bits of the pre-rate-matched encoded bit sequence in the radio frame of transport channel i in TFC j and $\Delta N_{i,i}$ is the number of bits subject to bit iteration or puncturing per radio frame. (Such number of bits is subject to bit iteration and puncturing when the value of Δ is positive and negative, respectively.) $Z_{i,j}$, which is necessary for determining the value of $\Delta N_{i,j}$, is calculated sequentially with reference to the value of rate-matching attribute RM_i , as shown in the equation below.

$$Z_{0,j} = 0$$

$$Z_{i,j} = \left\lfloor \frac{\left\{ \left(\sum_{m=1}^{i} RM_m x N_{m,j} \right) x N_{\text{data},j} \right\}}{\sum_{m=1}^{i} RM_m x N_{m,j}} \right\rfloor \text{ for all } i = 1, \dots, I \quad (17)$$

In Equation (17), $N_{\text{data},j}$ is the total number of bits that can be assigned to the encoding bits in all transport channels to be multiplexed in the radio frame of TFC j. $\lfloor x \rfloor$ is an integer defined by $x - 1 \leq \lfloor x \rfloor \leq x$. $\Delta N_{i,j}$ can be calculated by the equation below using $Z_{i,j}$, which is calculated sequentially on the basis of Equation (17).

$$\Delta N_{i,j} = Z_{i,j} - Z_{i-1,j} - N_{i,j} \text{ for all } i = 1, \dots, I$$
(18)

In uplink, rate matching is performed for each radio frame according to Equation (18). In contrast, in downlink, there is no need to update the rate-matching pattern for each radio frame because the execution of DTX is specified if there are no transmission bits in the transport channel, unlike in the case of uplink. Specifically, the number of bits per TTI before rate matching in transport channel i, represented by $N_{i,j}^{\text{TTI}}$, is calculated for each TFC h belonging to TFCS. Then, with reference to this value and the number of radio frames F_i in TTI of transport channel *i*, the number of bits per radio frame is determined. The value is calculated for all TFCs belonging to TFCS. This is followed by rate matching, in a manner such that the total number of bits per radio frame when TFC h_{Max} (the sum of the number of bits in transport channels per radio frame is maximized) is equal to the number of bits per radio frame that can be used over the physical channel. In short, this involves the calculation of the number of bits subject to bit iteration or puncturing per radio frame for each transport channel, that is, TTI. According to the ratematching pattern acquired thereby, each transport channel updates the number of bits per radio frame at each TTI. If the length of TTI of each transport channel to be multiplexed is not the same, the total number of bits in the radio frame with the shortest TTI will vary as a result. If the number of bits per radio frame of transport channel *i* after rate matching is less than the maximum number of bits per radio frame assigned to transport channel i, the mode switches to transmission-off mode for the duration in which the number of bits is insufficient.

The receiver needs to detect the TF of transport channels transmitted over one physical channel. Methods specified to achieve this include the transmission of control information indicating the TF of the multiplexed transport channel (referred to as the TFCI) over DPCCH and blind rate detection using the CRC results from a predetermined rate pattern (downlink only) [28].

2.2.4.3 Fast TPC Based on SIR Measurement

When applied to DPCH, both uplink and downlink, fast TPC based on SIR measurement [29, 30] helps constantly minimize the transmission power relative to the required reception quality, thereby increasing the system capacity. Especially in uplink, fast TPC is essential as it exercises control so that the SIR received by BS from each MS would be constant to solve the so-called near-far problem. In downlink, although SIR is constant regardless of the location of MS as long as it is in the same propagation path, multipath signals within the user's cell suffer from independent fading fluctuations and the impact of interference from other cells becomes heavier near the edge of the cell. Fast TPC is therefore applied to downlink as well, to exercise control so that the SIR would be constant as required, despite multipath interference and interference from other cells.

Figure 2.18 shows the configuration of fast TPC loop based on SIR measurement. Fast TPC consists of two loops, namely, the inner loop and the outer loop [30]. The inner loop measures the SIR of signals after RAKE combining in each slot, generates a binary TPC command bit that controls fluctuations in the transmission power so that the measured SIR value would be equal to the target SIR value and transmits it over DPCCH of the opposite link (e.g. this refers to the downlink when the uplink is controlled). It is known that when fast TPC is applied, the distribution of the reception power is extremely similar to log-normal distribution [31]. In order to achieve highly accurate TPC, a high-precision SIR measurement method is required. One method proposed for the measurement of SIR after RAKE combining is to measure the SIR of each path and add them together to determine the equivalent of SIR after RAKE combining [32]. In contrast with measuring the SIR of signals directly after RAKE combining, this method minimizes the impact of

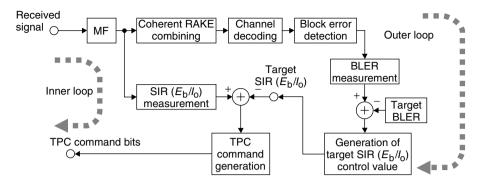


Figure 2.18 Mechanism of two-loop TPC

the channel-estimation error to achieve high-precision SIR measurement. The following equation is used to determine signal power $\tilde{S}_l(k)$ of path 1 ($1 \leq l \leq L$) in slot k using N_p pilot symbols assuming that $r_l(n, k)$ is the despread signal in the *l*th path for the *n*th symbol in slot k.

$$\tilde{S}_l(k) = |\bar{r}_l(k)|^2 \tag{19}$$

Here,

$$\bar{r}_l(k) = \frac{1}{N_p} \sum_{l=0}^{L-1} r_l(n,k) \exp\left(-\frac{j\pi}{4}\right)$$
(20)

(The equation assumes that the data modulation phase of the pilot symbol is π /4.) Then, the interference power $\tilde{I}_l(k)$ including instantaneous background noise components of path *l* in slot *k* is calculated according to the equation below using N_p pilot symbols in number.

$$\tilde{I}_{l}(k) = \frac{1}{N_{\rm p}} \sum_{n=0}^{N_{\rm p}-1} |r_{l}(n,k) \exp(-j\pi/4) - \bar{r}_{l}(k)|^{2}$$
(21)

Especially in uplink, the combining of interference signals from many users that suffer from independent fading fluctuations result in smaller variations in the interference power relative to the slot cycle. Thus, the average value of interference and background noise power can be determined at high accuracy by further averaging the interference power $\tilde{I}_l(k)$ in each slot calculated by Equation (21) through a primary filter using forgetful factor μ (<1) over multiple slots.

$$\bar{I}_l(k) = \mu \bar{I}_l(k-1) + (1-\mu)\tilde{I}_l(k)$$
(22)

Thus, SIR $\tilde{\lambda}_l(k)$ of path *l* in slot *k* is represented by the equation below.

$$\bar{I}_l(k) = \bar{S}_l(k) / \bar{I}_l(k) \tag{23}$$

Ultimately, SIR $(\tilde{\lambda})(k)$ in slot k can be determined by the following equation.

$$\bar{\lambda}(k) = \sum_{l=1}^{L} \tilde{\lambda}_l(k) \tag{24}$$

In the inner loop, transmission power is updated in each slot (=0.667 msec), meaning that it is updated 1500 times per second. When the size of each step of updating the transmission power is larger, it is easier to track violent fluctuations in the propagation path. However, when it is excessively large, the variation (dispersion) of reception power may be so significant in a steady state that the performance will deteriorate. Reportedly, the characteristics are optimized when the step size is 1 dB [14].

On the other hand, the same reception quality (BLER or BER) cannot necessarily be assured even if the target SIR value is the same, due to the difference in the number of propagation paths, the speed at which MS is moving (maximum Doppler frequency) and other factors in the propagation environment, as well as the difference in the SIR measurement method. Therefore, the outer loop measures the reception quality over long distances and corrects the target SIR over a slow cycle based on the measured reception quality value. When BLER-based outer loop control is implemented, BLER is measured with reference to the number of TBs in which the same CRC calculation results are obtained in the data sequence after error-correction decoding. Then, the value of the target SIR is corrected so that the measured BLER value would be equal to the required BLER value. In high-quality, high-speed data transmission (e.g. when the average required BER is 10^{-6}), BLER would be a small value as well, which not only makes the measurement extremely time-consuming but also makes it impossible to track fluctuations in the propagation path at high accuracy. Hence, in high-quality, high-speed data transmission, the target SIR value is not corrected by BLER to enhance the outer loop's trackability of fluctuations in the propagation environment; instead, the binary decision data after error-correction decoding is channel-encoded again, the BER of the tentative decision data sequence after RAKE combining is determined with reference to the data sequence subsequent to interleaving and the target SIR value is corrected in such a manner that the measured value would be the same as the target BER value. In practice, the decision data after error-correction decoding, which is used as reference signals, are tainted by bit errors; however, they are so rare that the impact is believed to be negligible.

2.2.5 Diversity

2.2.5.1 Coherent RAKE Reception (RAKE Time Diversity)

The DS-CDMA receiver despreads the reception signal using the spreading code replica synchronized with the spreading code of the reception signal in order to time-divide it into a number of multipath components, each having a different propagation delay time. This requires despreading based on a spreading code replica in sync with the reception timing of the desired signals and the detection of the reception timing of each path. Accordingly, the receiver shifts the timing of the spreading code replica over one information symbol by one chip to despread it over one symbol zone and generate a power delay profile (refer to Figure 2.19). With reference to the generated power delay profile, paths for RAKE combining are chosen in descending order of path reception power, accounting for the number of correlators, channel estimators and phase variation compensators (hereinafter collectively referred to as *RAKE fingers*), that is, the number of RAKE fingers. In the event of implementing space diversity (ANT diversity) reception or intersector diversity, the power delay profile is generated for each branch and paths are chosen in descending order of reception power delay profile of all branches. In

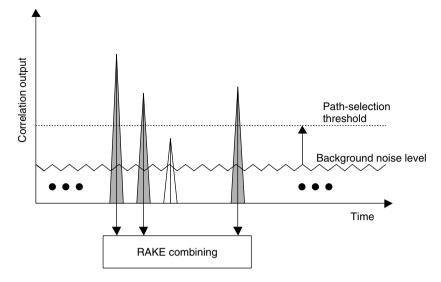


Figure 2.19 RAKE combining path selection method

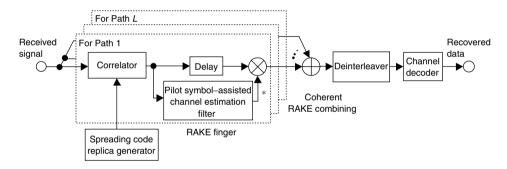


Figure 2.20 Configuration of coherent RAKE combining unit

practice, as despread signals are marred by interference from signals of other users and multipath signals over the user's own channel, a threshold value is set with reference to the level of background noise power and paths with an effective SIR (with reception power exceeding the threshold) are chosen. As the location of the path (delay time) subject to RAKE combining frequently changes in line with the movement of MS (or owing to changes in the surrounding propagation environment even if the location of MS is fixed), the receiver updates the power delay profile periodically and updates the RAKE combining path based on the updated profile (which is referred to as *path search* because it involves the search of paths required for RAKE combining) [33].

As the separated path is received via an independent propagation path, it is subject to different fading fluctuations. Figure 2.20 shows the configuration of a coherent RAKE receiver. W-CDMA adopts highly efficient absolute coherent detection demodulation in both uplink and downlink. Absolute coherent detection requires the estimation of variations in the phase and the amplitude of the reception signal due to fading fluctuations

in each path, that is, the fading complex envelope (the estimation of the fading complex envelope is hereinafter referred to as *channel estimation*). If the receiver travels at 100 km/h on average relative, the maximum Doppler frequency would be as high as 200 Hz when a carrier frequency band of 2 GHz is used. In order to track such rapid channel fluctuations, W-CDMA resorts to channel estimation using pilot symbols (channels) [34, 35]. A pilot symbol is a symbol for which the transmission data modulation phase is already known by the receiver. The channel estimation value is determined for each slot with reference to the reception phase and the amplitude of the pilot symbol. Figure 2.21 describes compensation for phase variation due to fading fluctuations in the pilot symbol in uplink and downlink. As mentioned in Section 2.2.1, pilot symbols over DPCH in uplink are mapped over channel Q as part of DPDCH and DPDCH–which consists of an encoded data sequence-is mapped over channel I. DPCCH and DPDCH are subject to data modulation by BPSK. On the other hand, for downlink DPCH, DPCCH and DPDCH are time-multiplexed and subject to data modulation by QPSK. Assuming that $\xi_{l}(k)$ is the channel fluctuation $\xi_{l}(k)$ is the complex of amplitude and phase components) due to fading in slot k in the lth path $(1 \le l \le L: L = number of RAKE$ -combined paths) and $\tilde{\xi}_l(k)$ is the estimate of $\xi_l(k)$, the amplitude and phase variations due to fading fluctuations $\tilde{\xi}_l(k)$ can be determined from the reception phase of the pilot symbols as shown in Figure 2.21. Assuming that the despread signal in the *l*th path for the *n*th symbol on DPDCH in slot k is represented by $r_l(n, k)$, the complex conjugate of the channel estimation value based on the estimation of each information symbol is multiplied to compensate for the phase variation arising from fading. As shown in the equation below, the signal in each path compensated for phase variations is added together in-phase (coherent RAKE combining). Each path is subject to weighted combining according to the reception power, that is, Maximum Ratio Combining (MRC). [Paths with large reception power (SIR) are highly reliable, so they are combined with a lot of weight.] [36].

$$\tilde{d}(n,k) = \sum_{l=1}^{L} r_l(n,k) \tilde{\xi}_l^*(k)$$
(25)

The RAKE-combined data sequence $\tilde{d}(n, k)$ is subject to deinterleaving followed by error-correction decoding for the purpose of recovering the transmission data sequence. While the data modulation scheme is QPSK in downlink, the RAKE combining process is basically the same as in uplink, as shown in Figure 2.21. Generally, however, CPICH is used for channel estimation. As described by Equation (25), the performance of coherent RAKE reception depends on the accuracy of the generated channel estimation value $\xi_l(k)$. As the despread signal in each path is marred by interference from other users, multipath interference and background noise, the impact of these factors need to be reduced to determine the channel estimation value at high accuracy.

The Weighted MultiSlot Averaging (WMSA) channel estimation filter has been advocated to achieve high-precision channel estimation [35, 37]. Figure 2.22 shows the configuration of a WMSA channel estimation filter. The channel estimation value in slot k is determined as follows. Firstly, pilot symbols of all slots are added together in-phase (amplitude components of I/Q are independently added together) to determine the instantaneous channel estimation value $\hat{\xi}_l(k+i)(i = -J + 1, ..., 0, 1, ..., J)$ in each slot. The channel estimation value in 2J slots before and after the target slot is subject to weighted

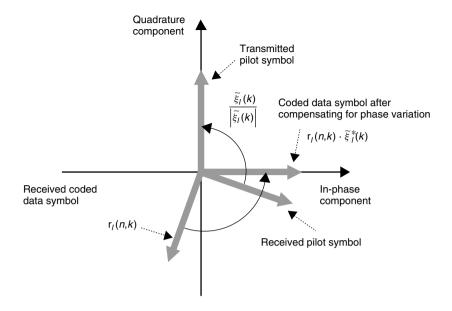


Figure 2.21 Compensation for phase variation

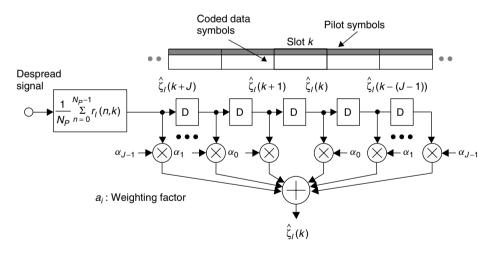


Figure 2.22 Configuration of WMSA channel estimation filter

averaging based on Equation (26), so as to determine the ultimate channel estimation value $\tilde{\xi}_l(k)$.

$$\tilde{\xi}_{l}(k) = \sum_{i=0}^{J-1} \alpha_{i} \hat{\xi}_{l}(k-i) + \sum_{i=1}^{J} \alpha_{i-1} \hat{\xi}_{l}(k+i)$$
(26)

In Equation (26), α_i is a real weighted coefficient. By applying a WMSA channel estimation filter, as the channel estimation value in slots with a large (small) fading correlation

are combined with large (small) weighted coefficients, a greater number of pilot symbols can be used, which results in high-precision channel estimation. One of the advocated methods involves the determination of the fading correlation between slots based on the inner product of signals for which pilot signals in all slots have been added together and averaged in-phase and the updating of the weighted coefficient of each slot in an adaptive manner according to the determined fading correlation value [38].

Figure 2.23 shows the average BLER, BER, target E_b/I_0 and average transmission $E_{\rm b}/N_0$ characteristics of MS against $f_{\rm D}$ when fast TPC with outer-loop control is applied to uplink. (In this case, transmission $E_{\rm b}/N_0$ is defined as the ratio of transmission power per information bit for which the path loss and shadowing fluctuation have been compensated to the background noise per hertz at the point of reception.) The target $E_{\rm b}/I_0$ is corrected so that the average BLER after soft-decision Viterbi decoding would be 10^{-2} , assuming that the block length is equivalent to one radio frame. As shown in the figure, high-precision control is achieved that assures a measurement of $BLER = 10^{-2}$ over a wide range, from $f_{\rm D} = 5$ to 640 Hz, and the average BER is more or less constant over this range. On the other hand, the target $E_{\rm b}/I_0$ and the MS transmission power fluctuate within the range of approximately 3 dB. The transmission $E_{\rm b}/N_0$ characteristics of MS is more or less constant between $f_{\rm D} = 5$ and 300 Hz. This is attributable to the fact that fast TPC and channel encoding complement each other in low-speed and high-speed fading environments: fast TPC is effective when $f_{\rm D}$ is small (around 5 Hz), but the ability of fast TPC to track fading fluctuations deteriorates as f_D increases and as a result the TPC error becomes larger – channel encoding offsets this, as its error-correction effect grows as

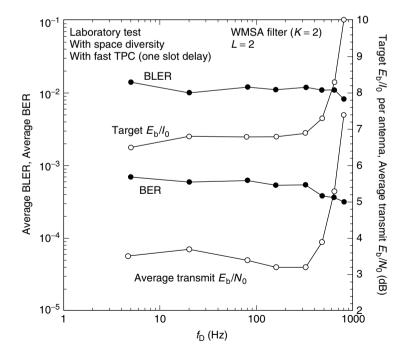


Figure 2.23 Characteristics of coherent RAKE reception using two-loop fast TPC

 $f_{\rm D}$ increases, owing to error pattern randomization associated with the time-interleaving effect. When $f_{\rm D} = 300$ Hz or higher, the required transmission $E_{\rm b}/N_0$ increases as high-speed fading cannot be tracked by channel estimation using pilot symbols.

Figure 2.24 shows the average BER characteristics against the target E_b/I_0 of fast TPC (TPC delay per slot) in a field test [37]. The test was conducted using the same courses 1 and 2 as in the three-step cell search test referred to in Section 2.2.2. The figure shows the characteristics of single-user reception at an information rate of 32 kbit/s (no interference signals) and the characteristics when there is one interfering user at an information rate of 64 kbps engaged in fast TPC independently of the desired signals (the transmission power is double the desired signals as the reception quality was set to be the same as the desired signals, and accordingly, the same target E_b/I_0). The figure also shows the results of the indoor laboratory test assuming the *L*-path model of equal average power. The target E_b/I_0 that achieves the required BER characteristics when there are interference signals is more or less identical to that when there are no interference signals, showing that fast TPC is working properly in an outdoor environment in practice. The average number of RAKE fingers in measurement courses 1 and 2 were 2.0 and 1.6, respectively. On

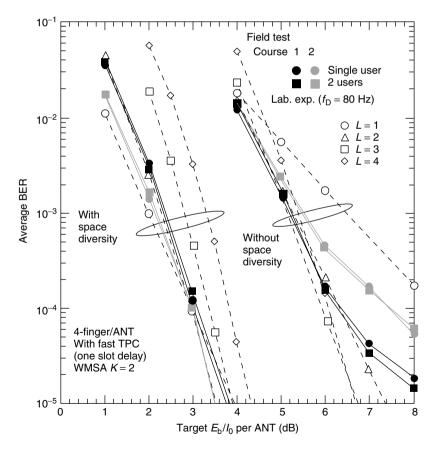


Figure 2.24 Characteristics of coherent RAKE reception using fast TPC

the other hand, Figure 2.24 shows that the BER characteristics in measurement course 1 is almost identical to that when L = 2 in a laboratory test, and the characteristics in measurement course 2 lies somewhere between L = 1 and 2. In turn, this shows that the average BER characteristics found in the field test are the same as the results of the laboratory test based on the number of paths observed in the power delay profile. The figure also shows that the required target E_b/I_0 that complies with average BER = 10^{-3} can be reduced by approximately 3 dB by using two-branch space diversity reception. The use of space diversity reception also makes it possible to achieve reception quality of average BER = 10^{-3} based on the required target $E_b/I_0 = \sim 3$ dB per ANT.

Figure 2.25 shows the average BER characteristics against the average transmission power of MS when the information rate is 64 kbps, assuming the implementation of turbo encoding in a field test (measurement course 2 is used) [39]. Turbo encoding was performed at rate R = 1/3 and constraint length of K = 4 bits, and PIL and MIL were applied as the turbo interleaver and the channel interleaver, respectively (interleave length = 40 msec). For comparison, the figure also shows the characteristics when convolutional encoding at R = 1/3 and K = 9 bits was applied. Max-log-Map decoding and soft-decision Viterbi decoding were applied for decoding turbo codes and convolutional codes, respectively, both iterated eight times. Figure 2.25 shows that turbo encoding can reduce the required average transmission power of MS in compliance with average BER = 10^{-6} by approximately 0.6 dB without space diversity reception or by approximately 0.3 dB with it, compared to convolutional encoding. However, while the effects

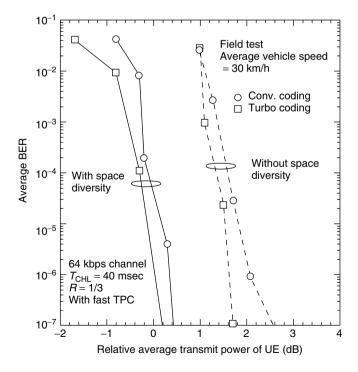


Figure 2.25 Average BER characteristics and MS transmit power using turbo encoding

of reducing the required transmission power by turbo encoding relative to convolutional encoding were verified in this manner in a real multipath-fading environment, the effects of improving the characteristics compared to the laboratory test using a fading simulator is somewhat weaker. (In the laboratory test, the required average transmission power of MS compliant with average BER = 10^{-6} was approximately 1.0 dB lower in the case of turbo encoding than convolutional encoding.) This is believed to be attributable to the larger impact of SIR measurement errors in path search and TPC due to the lower reception power of each path, caused by a higher encoding gain brought about by turbo encoding in a real propagation environment, in which the delay time of each path varies. The effects of improving the encoding gain are believed to be suppressed as a result of this.

2.2.5.2 Site Diversity (Soft/Softer Handover)

W-CDMA can improve the reception quality near the edge of cells and assure communications free of interruption by soft/softer handover (site diversity) [40, 41], in which MS establishes radio links with multiple cells (sectors) simultaneously for communication. MS connects DPCH to multiple cell sites or sectors when the difference between the CPICH reception power of a cell that is currently in connection with DPCH and the CPICH reception power of adjacent cells falls below a threshold predetermined by the system. Figure 2.26 shows the configuration of uplink and downlink site diversity when MS connects DPCH with cell sites or sectors of station N. It should be noted that N is determined by the system. MS measures the downlink CPICH reception power of adjacent cells (sectors) and executes soft/softer handover (site diversity), selecting the higher-ranking station N that has a reception power below a certain threshold as the

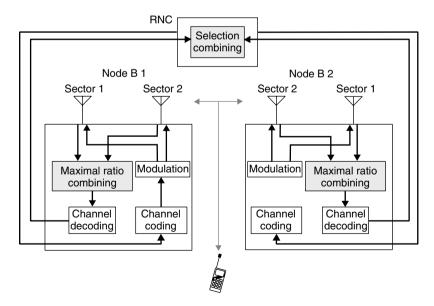


Figure 2.26 Mechanism of site diversity

candidate for site diversity. While the optimal values of the soft handover threshold are different between uplink and downlink, the same threshold value is normally used.

In downlink, the Radio Network Controller (RNC) transmits the information data sequence over a fixed transmission line to the cell sites or sectors of station N, and each cell site or sector transmits downlink DPCH. MS receives the said DPCH, selects L_{RAKE} paths in number with a larger reception level than the threshold decided with reference to the power delay profile of station N after despreading. After performing channel estimation and MRC combining (RAKE combining) by symbol on each selected path, MS executes deinterleaving and error-correction decoding. On the other hand, in uplink, signals transmitted from MS are received at the cell sites or sectors of station N to be combined. Intersector diversity involves the reception of DPCH in sectors within the same cell site, the selection of L_{RAKE} paths in number in descending order of reception power-which must exceed the predetermined threshold-with reference to the combined power delay profile of the target sector and channel estimation and RAKE combining by symbol as in the case of downlink site diversity, followed by deinterleaving and error-correction decoding. In contrast, intercell site diversity involves the transmission of hard-decision data sequence subsequent to error-correction decoding to RNC over a fixed transmission line, together with reliability information, and the selection combining of the decoded data sequence transmitted from the target cells by RNC according to the reliability information [42]. Therefore, the quality of selected and combined signals depend on the types of reliability information used. As examples of reliability information, explanations are provided below for (1) CRC results by each branch selection cycle $T_{\rm SFL}$ and (2) the mechanism of intercell selection switching site diversity when average SIR is applied in channel coding interleave period $T_{\rm ILV}$ [43]. Assuming that T_{FRM} is the frame length, the relationship $T_{\text{FRM}} \leq T_{\text{SEL}}(n \times T_{\text{FRM}}) \leq T_{\text{ILV}}$ is established. Selection switching and combining of the decoded data sequence from the cell site of station N is executed on the basis of a two-step decision with reference to two types of reliability information. As an indicator of the size of the average received SIR referred to in (2), L_{SIR} is used, which refers to the number of slots in which the measured E_b/I_0 value in each slot in $T_{ILV} = m \times T_{SLOT}(T_{SLOT} = \text{slot length})$ zone is larger than the target E_b/I_0 value. In other words, larger L_{SIR} is deemed to bring about higher reliability as the signals received from the concerned BS has a larger SIR.

In uplink, fast TPC is performed as follows. For simplicity, it is assumed that soft handover takes place between two BSs, from BS 1 to BS 2. Upon site diversity, the uplink TPC initial value is set in such a manner that the target E_b/I_0 value of BS 2 (the destination of soft handover) would be equal to the target E_b/I_0 value of BS 1. In uplink fast TPC, BS 1 and BS 2 transmit the TPC command bits independently of each other on the basis of the measured E_b/I_0 value. MS reduces the transmission power, unless the demodulated TPC bits from BS 1 and BS 2 designate otherwise. Hence, fast TPC is controlled according to the command bits from a BS with limited path loss and large reception power.

Figure 2.27 shows the fluctuations in the characteristics over time when intercell site diversity is applied in uplink, according to a field test. The course and the conditions of the field test are described in detail in Ref. [37]. In the test, fast TPC was applied only to uplink, and high reception power was used for reception in downlink. MS was

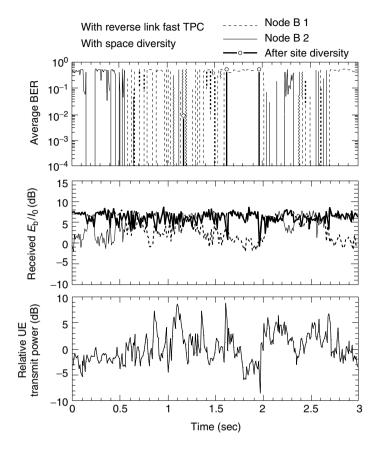


Figure 2.27 Characteristics of intercell site diversity in uplink

located approximately 1200 m away from both BS 1 and BS 2 and moved along the measurement course at an average speed of approximately 30 km/h. The threshold of soft handover was $\Delta_{SHO} = 3$ dB, and the average reception power over CPICH from the two BSs along the measurement course differed by about 1 dB. The course was inside the soft handover areas. In uplink, the power delay profile observed in the first half and the second half of the measurement course for BS 1 was about one path and two paths, respectively, whereas the same for BS 2 was about two paths and one path, respectively. The test assumed that $T_{\text{SEL}} = 10$ ms. Figure 2.27a shows the average BER of BS 1, BS 2 and by frame after selection combining; Figure 2.27b shows the reception E_b/I_0 and Figure 2.27c depicts the transmission power of MS. Target E_b/I_0 was set at 7 dB in each BS, so that the average BER would be approximately 10^{-3} after site diversity combining. As shown in Figure 2.27b, some deterioration in the received $E_{\rm b}/I_0$ from BS 1 and BS 2 was observed along the course, which is reflected in the deterioration of average BER of reception signals at each BS as shown in Figure 2.27a. However, the received $E_{\rm b}/I_0$ after site diversity combining is kept more or less constant (as depicted in Figure 2.27b), and consequently the average BER characteristics improved substantially compared to the values in each BS (Figure 2.27a).

Fast TPC in downlink intercell site diversity mode is executed according to the common TPC bits from MS transmitted over uplink DPCH. Thus, if there are any errors in the TPC bits, the transmission power of the candidate cell site for site diversity would vary. (Assuming that the ultimate power, that is, the required average BER of decision data after error-correction decoding is 10^{-3} , the average BER characteristics of TPC bits would be in the vicinity of 10^{-1} .) Accordingly, any errors in TPC bits would increase the difference in transmission power between the candidate cells for site diversity, reduce the site diversity gain and aggravate the interference to other users. Figure 2.28 shows a double-loop compensation algorithm [42], in which each cell controls the transmission power according to the average path loss difference from MS (including shadowing fluctuations) during downlink intercell site diversity if there are errors in TPC bits.

First Loop

On the basis of the SIR measurement results subsequent to RAKE combining at MS, the correction value of the transmission power of each cell $\Delta P^{(k)}$ (dB) is transmitted by the layer (Layer 3) information in uplink. The reference transmission power $P_{\text{REF}}^{(k)}$ of target cell k is compensated by $\Delta P^{(k)}$ (dB) on the basis of the transmission power correction value.

$$\Delta P^{(k)} = \text{Target}_{E_b}/I_0 - \text{Measured}_{\text{total}}_{E_b}/I_0 \text{ (dB)}$$
(27)

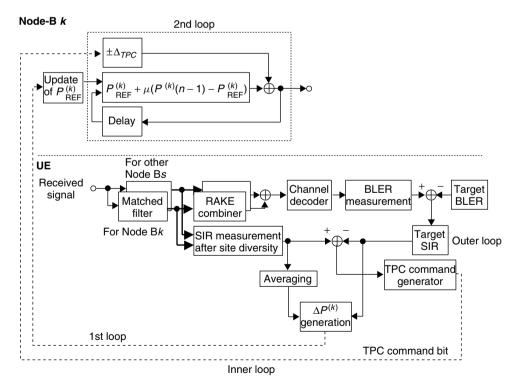


Figure 2.28 Mechanism of two-step TPC correction of BS in downlink cell site diversity

In Equation (27), Measured_total_ E_b/I_0 is the measured E_b/I_0 value after RAKE combining, while Target_ E_b/I_0 is the target E_b/I_0 value of MS. A constant value is used for $P_{\text{REF}}^{(k)}$ for a period of G slots, and the value of $P_{\text{REF}}^{(k)}(n)$ in n(g = G) slot is updated for every G slot period, according to the equation $P_{\text{REF}}^{(k)}(g \times G) = P_{\text{REF}}^{(k)}((g - 1) \times G) + \Delta P^{(k)}$.

Second Loop

In the second loop, forgetting factor μ is inserted in the reference transmission power of each cell corrected in the first loop [44], and the instantaneous transmission power in each slot $P_{\text{CL}}^{(k)}(n)$ is controlled according to TPC bits in each slot (Δ_{TPC}) pursuant to the following equation.

$$P_{\rm CL}^{(k)}(n) = P_{\rm REF}^{(k)} + \mu(P_{\rm CL}^{(k)}(n-1) - P_{\rm REF}^{(k)}) + \Delta_{\rm TPC}$$
(28)

In downlink, soft handover leads to a larger site-diversity gain due to the transmission of DPCH from multiple cells (sectors) to one MS. However, it also brings about larger interference and is less effective in expanding link capacity than in uplink. Figure 2.29 shows the cell selection method in downlink (TPC algorithm). In Conventional TPC (CTPC), as shown in Figure 2.29a, MS measures SIR of the signal after RAKE combining and generates TPC bits based on the results of comparing the measured SIR value with the target SIR value. The TPC bits are subsequently transmitted uplink, and the candidate cell controls the transmit power according to the common TPC bits. An alternative method is Site Selection Diversity Transmission Power Control (SSDT), as shown in Figure 2.29b, which defines a candidate cell that transmits downlink DPCH to MS in soft handover mode as does CTPC, transmits DPCH from only one cell from the candidates at a certain time and switches the transmission cell at high speed in such a manner that the fading fluctuations would be tracked. It has been proved that larger capacity can be secured by reducing the interference power, especially in cases where the difference in path loss between the candidate cell for soft handover and MS is substantial [45]. Another method is Site Independent Diversity Transmit Power Control (SIDTPC), as shown in Figure 2.29c. Here, MS has the function to measure SIR of signals after RAKE combining for each of the soft handover candidate cells. The TPC bits

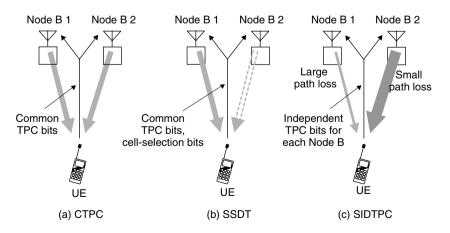


Figure 2.29 Cell selection (TPC) algorithm in downlink intercell site diversity

are generated for each cell and TPC is performed independently by each cell [46]. With SIDTPC, transmission power can be minimized according to the difference in path loss between the candidate cell and MS while maintaining SIR after RAKE combining at a constant level.

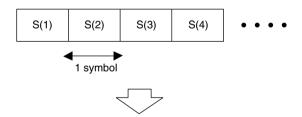
2.2.5.3 Transmission Diversity

In downlink, high-quality reception can be assured by the diversity effects without increasing the complexity of the MS receiver: the diversity effects can be obtained by transmitting signals to the same terminal from two ANTs at BS, which have a fading correlation of less than 1 [47, 48]. Radio interface specifications defined under 3GPP are also open to transmission diversity [11, 16], adopting two types of open-loop models that require no information fed back from MS and two types of closed-loop models that control the transmission carrier phase and amplitude of ANT 2 using the information fed back from MS.

The open-loop TSTD is applied to SCH, whereas Space-Time Transmit Diversity (STTD) [49, 50] is applied to the Common Control Physical CHannel (CCPCH), DPCH and so forth. [11]

Figure 2.30 shows the transmission symbol pattern of two ANTs when STTD is performed. Figure 2.31a,b illustrates the configuration of the transmitter and the receiver, respectively. As shown in Figure 2.30, the STTD encoding unit converts the channelencoded and interleaved data into two data sequences (data sequences based on the original data and its complex, in which the order of two adjacent symbols and the codes are changed). After the pilot symbols and TPC symbols are time-multiplexed with each

Original symbol sequence



Transmit symbol sequences after STTD encoding

ANT 1

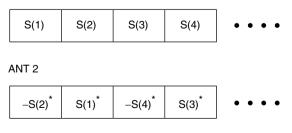


Figure 2.30 STTD-encoding pattern

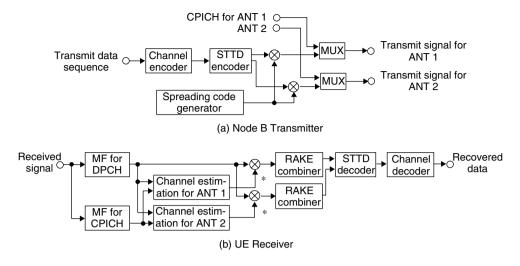


Figure 2.31 Configuration of transmitter and receiver for STTD

data sequence, they are subject to QPSK data modulation mapping. Then, they are spread over channel I and channel Q by different spreading codes (QPSK spreading modulation). (However, the same spreading code can be used for each sequence because by design, when the two sequences generated by the STTD encoder is spread on the basis of the same spreading code, they are equivalent to being spread by an orthogonal spreading code with double SF.) The receiver executes despreading using the spreading code replica of the signals transmitted from each ANT and determines the channel estimation value of each slot from the despread CPICH signals. Phase variations over DPCH from each ANT are compensated by using this channel estimation value and the DPCH data sequences received from the two ANTs are phase-combined (MRC combining). The MRC-combined data sequence is then deinterleaved to undergo error-correction decoding to reproduce the transmitted data sequence.

Figure 2.32a shows the configuration of the transmitter of closed-loop transmission diversity [16] and Figure 2.32b shows the configuration of the receiver of the same. Closed-loop transmission diversity is applicable only to DPCH. The two transmission data sequences are multiplied by complex weight $W_{1,n} = A_{1,n} e^{j\phi_{1,n}}$ and $W_{2,n} = A_{2,n} e^{j\varphi_{2,n}}$ based on the feedback control (FBI) bits from MS, and then spread for transmission. Firstly, CPICH is transmitted in the same carrier phase from the two ANTs. Orthogonality is assured by spreading the CPICH from the two ANTs by the same spreading code and changing the pilot symbol pattern. The receiver refers to the difference in the phase and the amplitude of the received carrier of the separated signals from the two ANTs after CPICH despreading in order to generate FBI bits that control the carrier phase and amplitude of the transmission signals of ANT 2, so that the signal sequences sent from the two ANTs would be in the same phase and amplitude at the point of reception of MS, and sends them over DPCCH of uplink DPCH. The use of FBI bits from the MS to control the phase and amplitude of the transmission carrier of ANT 2 in this manner helps reduce bit errors caused by a drop in received signal power due to fading. At the BS transmitter, the transmission data sequences of the two ANTs are transmitted after being multiplied

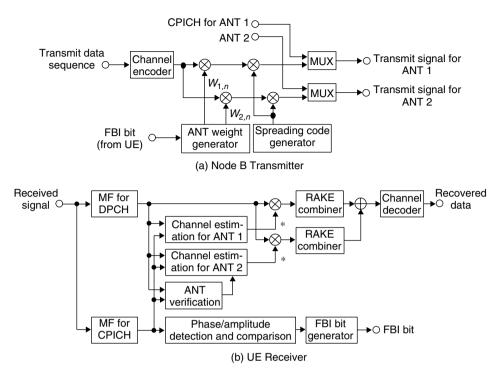


Figure 2.32 Configuration of transmitter and receiver for closed-loop transmission diversity

by transmission ANT weights $W_{1,n}$ and $W_{2,n}$ generated on the basis of FBI bits from the MS. 3GPP specifies two modes of closed-loop transmission diversity. Mode 1 controls the DPCH transmission carrier phase of ANT 2 at a carrier phase resolution of $\pi/4$ so that the signals received from the two ANTs would be more or less in the same phase when received at the MS. Mode 2 controls the DPCH transmission carrier phase of ANT 2 at a carrier phase resolution of $\pi/4$, while selecting and controlling the transmission power ratio of signals transmitted from the two ANTs from two patterns.

The following is an explanation of Mode 1. The transmission amplitude of the two ANTs in slot *n* is $A_{1,n} = A_{2,n} = \sqrt{1/2}$ and the transmission carrier phase is $\phi_{1,n} = 0$, $\phi_{2,n} = \{\pm \pi/4, \pm 3\pi/4\}$. MS generates FBI bits b_n in slot *n* from reception carrier phase of CPICH transmitted from the two ANTs $[\theta_{1,n}^{CP}, \theta_{2,n}^{CP}]$. Assuming that $[\hat{\theta}_{1,n}^{CP}]$ and $[\hat{\theta}_{2,n}^{CP}]$ are the estimated values of $[\theta_{1,n}^{CP}]$ and $[\theta_{2,n}^{CP}]$, respectively,

When slot n is an even number,

if
$$-\pi/2 \le (\hat{\theta}_{1,n}^{\text{CP}} - \hat{\theta}_{2,n}^{\text{CP}}) < \pi/2$$
, then $b_n = 0$, otherwise $b_n = 1$ (29a)

When slot n is an odd number,

if
$$0 \le (\hat{\theta}_{1,n}^{CP} - \hat{\theta}_{2,n}^{CP}) < \pi$$
, then $b_n = 0$, otherwise $b_n = 1$ (29b)

On the basis of the following equation, BS determines the transmission carrier phase in slot (n + 1) of DPCH in ANT $2\varphi_{2,(n+1)}$ according to the decoding results of the FBI bits $\hat{b}_n(\hat{b}_n = b_n)$ if there are no decoding errors in FBI bits).

When *n* is an even number,

if
$$b_n = 0$$
, then $\varphi_{2,(n+1)} = 0$, otherwise $\varphi_{2,(n+1)} = \pi$ (30a)

When *n* is an odd number,

if
$$b_n = 0$$
, then $\varphi_{2,(n+1)} = \pi/2$, otherwise $\varphi_{2,(n+1)} = -\pi/2$ (30b)

The transmission carrier phase of ANT 2 in slot (n + 1), $\phi_{2(n+1)}$, is ultimately determined from the tentative carrier phase of slot n and (n + 1) according to the equation below.

$$\phi_{2,(n+1)} = (\varphi_{2,n} + \varphi_{2,(n+1)})/2 \tag{31}$$

Since fast TPC is applied in uplink, in the case of voice transmission, for instance, with a required BER = 10^{-3} , the average BER of the FBI bits would be in the vicinity of 10^{-1} as in the case of TPC bits. As BS executes transmission in a carrier phase different from the control command from MS. MS estimates the transmission weight in each slot of DPCH (the transmission carrier phase in Mode 1), which is referred to as ANT verification. Mode 1 performs ANT verification on the transmission carrier phase of ANT 2 based on the equation hereunder.

When *n* is an even number,

if
$$2\sum_{l=1}^{L} \frac{1}{\sigma_l^2} \{2\operatorname{Re}(\gamma \hat{\xi}_{2,n,l}^{\mathrm{D}} \hat{\xi}_{2,n,l}^{\mathrm{CP}*})\} > \frac{\ln[(\bar{p}(\varphi_{2,n} = \pi)]]}{\ln[(\bar{p}(\varphi_{2,n} = 0)]]}, \text{ then } \{\hat{\varphi}_{1,n}, \hat{\varphi}_{2,n}\} = \{0, 0\},$$

otherwise $\{\hat{\varphi}_{1,n}, \hat{\varphi}_{2,n}\} = \{0, \pi\}$ (32a)

When *n* is an odd number,

if
$$-2\sum_{l=1}^{L} \frac{1}{\sigma_l^2} \{ 2 \operatorname{Im}(\gamma \hat{\xi}_{2,n,l}^{\mathrm{D}} \hat{\xi}_{2,n,l}^{\mathrm{CP}^*}) \} > \frac{\ln[\bar{p}(\varphi_{2,n} = \pi/2)]}{\ln[\bar{p}(\varphi_{2,n} = -\pi/2)]}, \text{ then } \{\hat{\varphi}_{1,n}, \hat{\varphi}_{2,n}\} = \{0, \pi/2\},$$

otherwise $\{\hat{\varphi}_{1,n}, \hat{\varphi}_{2,n}\} = \{0, -\pi/2\}$ (32b)

otherwise $\{\hat{\varphi}_{1,n}, \hat{\varphi}_{2,n}\} = \{0, -\pi/2\}$

According to the two equations shown above, the transmission carrier phase of ANT 2 in slot n can be determined by the following equation.

$$\hat{\phi}_{1,n} = 0, \, \hat{\phi}_{2,n} = (\hat{\varphi}_{2,n-1} + \hat{\varphi}_{2,n})/2 = \{\pm \pi/4, \pm 3\pi/4\}, \{\hat{A}_{1,n}, \hat{A}_{2,n}\} = \{\sqrt{1/2}, \sqrt{1/2}\}$$
(33)

In Equation (33), $\hat{\xi}_{2,n,l}^{CP}$ is the instantaneous channel estimation value of CPICH of the *l*th path in the *n*th slot of transmission ANT *i* and $\hat{\xi}_{2,n,l}^{\mathrm{D}}$ is the instantaneous channel estimation value of DPCH of the same. γ is the ratio of SIR of DPCH to CPICH, σ_1^2 is the thermal noise and interference power of each path and $\bar{p}(\cdot)$ represents the prior probability. The channel estimation values are determined by the WMSA (K = 1) channel estimation filter using the respective pilot symbols of CPICH and DPCH for the *l*th path in slot n, on the basis of the transmission weights estimated in the manner described above, $(\hat{\phi})_{1,n}$ and $(\hat{\phi})_{2,n}$ and instantaneous channel estimation values determined with reference to the pilot symbol of CPICH and DPCH in slot n, $\hat{\xi}_{i,n,l}^{\text{CP}}$ and $\hat{\xi}_{i,n,l}^{\text{D}}$. The values, which are calculated by equations $\bar{\xi}_{i,n,l}^{\text{CP}} = \hat{\xi}_{i,n,l}^{\text{CP}} e^{-j\phi_{i,n+1,l}} e^{-j\phi_{i,n+1,l}}$ and $\bar{\xi}_{i,n,l}^{\text{D}} = \hat{\xi}_{i,n,l}^{\text{D}} + \hat{\xi}_{i,n+1,l}^{\text{D}}$, are added together in-phase daveraged. The ultimate channel estimation values are calculated by $\bar{\xi}_{i,n,l} = A_i(\bar{\xi}_{i,n,l}^{\text{CP}} + \bar{\xi}_{i,n,l}^{\text{D}})$.

Figure 2.33 shows the average BER characteristics against the average transmission power of all BSs when closed-loop transmission diversity was applied to downlink in a field test [51]. The test was conducted on the basis of the three-step cell search test referred to in Section 2.2.2, in which the MS traveled at approximately 30 km/h on average in measurement course 2. The conditions of the test are described in detail in Ref. [46]. The characteristics were assessed for the aforementioned transmission diversity Mode 1 (hereinafter referred to as *Phase Diversity* (PD)- $\pi/2$), phase diversity with π phase resolution ($\phi_{1,n} = 0, \phi_{2,n} = \{0, \pi\}, A_{1,n} = A_{2,n} = \sqrt{1/2}$) (hereinafter referred to as PD- π) and Selection Transmit Diversity (STD), which controls only the amplitude components $(\phi_{1,n} = \phi_{2,n} = 0, (A_{1,n}, A_{2,n}) = \{(1, 0), (0, 1)\})$. In both uplink and downlink, fast TPC was performed on the basis of SIR measurement with the same target $E_{\rm b}/I_0$ value. The values of the transmission power are indicated in relative values to the transmission power that satisfies an average $BER = 10^{-3}$ with space diversity receiving at the MS and without fast TPC. ANT verification was performed at MS to reduce the impact of decoding errors in FBI bits. The figure shows that when PD- $\pi/2$ and PD- π phase transmission diversity is used, the required average transmission power of all BSs satisfying a BER = 10^{-3} is reduced by approximately 1.0 to 1.5 compared with the case

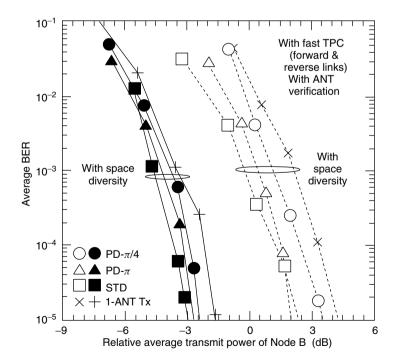


Figure 2.33 Average BER characteristics using closed-loop transmission diversity

without transmission diversity when space diversity reception is not performed by the MS, whereas it is approximately 0.3 to 0.7 lower with space diversity reception by MS. STD is the most effective way to improve the characteristics, as demonstrated by the fact that it reduces the required average transmission power of all BSs by approximately 2.0 dB without space diversity reception by MS and by approximately 1.0 dB with space diversity reception by MS. When space diversity reception is applied to MS, the effects of reducing transmission power are smaller owing to the use of transmission diversity in one-ANT transmission. This is because the use of fast TPC, channel encoding, RAKE time diversity and space diversity reception substantially prevent the deterioration of characteristics due to multipath fading, and improvements in the characteristics brought about by the use of transmission diversity are offset by ANT verification errors due to smaller reception power and other factors of deterioration. (In particular, the characteristics of PD- $\pi/2$ with smaller phase control resolution are somewhat poorer than PD- π .)

2.3 Link Capacity Expansion Technologies in W-CDMA

W-CDMA has a radio interface that can apply interference canceller and adaptive ANT array diversity reception and transmission as technologies that may further expand the link capacity in the future. In uplink, while it is important to expand the link capacity by reducing MAI and MPI, the joint use of fast TPC is effective in reducing the transmission power of MS to extend battery life and expanding the cell coverage. On the other hand, in downlink, expansion of link capacity is more strongly called for in uplink, as high-speed data downloading over the Internet and multicast services are likely to be more widely used. OVSF codes may be used to achieve orthogonality in the same propagation path, but MPI from high-rate users cause extremely large interference to low-rate (e.g. speech communication) users. Further expansion of link capacity in downlink is therefore an important task. Technologies to accomplish this include the use of interference canceller (multipath interference canceller) or adaptive ANT array diversity reception in MS and adaptive ANT array transmission diversity in BS. Among them, adaptive ANT array transmission diversity is an effective and practical technology for further expansion of the downlink capacity, as it requires no substantial changes in the functions of MS and can be implemented through transmission processing at the BS side.

2.3.1 Interference Canceller

2.3.1.1 Multistage Interference Canceller

Effective technologies for the reduction of MAI and MPI in uplink reception at the BS include interference canceller and MultiUser Detection (MUD) [52, 53] and numerous reports on research and development results are available on the subject. Figure 2.34 shows the categories of interference cancellers. Single-user reception requires no spreading codes or received signal information of other users. The orthogonal filter is well known, which executes updating using the Minimum Mean-Squared Error (MMSE) algorithm so that the spreading code replica used for despreading would be orthogonal to the spreading code of signals of other users (including multipath signals) [54]. Although the orthogonal filter has an easier configuration than multiuser reception, which is referred to later, the problem is that it cannot be applied to scrambling codes that have much longer iteration

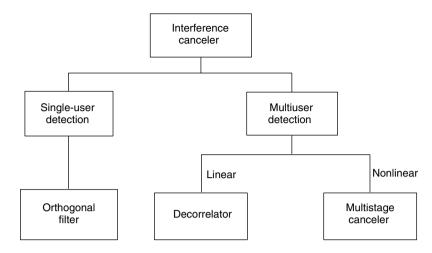


Figure 2.34 Categories of DS-CDMA interference cancellers

period than the symbol length (long codes). In contrast, multiuser reception [47] uses the reception signals and decoding data sequence of users to reduce the interference of other users in a mutually dependent manner, which is a method suitable for reception by BS. The decorrelator calculates the inverse matrix of the cross-correlation matrix of the spreading code of the user and multiplies it by users' signals after despreading. Although this method is not affected by channel estimation, the process of calculating the inverse matrix becomes extremely complex as the number of users increases. One of the proposed methods is MultiStage Interference Canceller (MSIC), which generates the MAI and MPI replica at the receiving side based on the received fading complex envelope estimated by multiuser reception and the decision data, and deducts it from the reception signals to enhance SIR and improve the reception characteristics [55]. MSIC sequentially reduces the interference of other users in multiple stages and is a practical method. The performance of MSIC depends on how accurately the interference signals of users, that is, interference replica, can be generated by the receiver. It is therefore vital to achieve high-precision channel estimation and reduce decision data errors in actual multipath-fading channels.

One solution that has been advocated is the COherent MultiStage Interference Canceller (COMSIC), which updates channel estimation value in a sequential manner at each stage using pilot symbols [56]. COMSIC sequentially updates the channel estimation value for the signal sequence with improved SIR where interference has been removed from each stage using pilot symbols (ideally, the removal of MAI would leave nothing but signals of the user's channel), in order to raise the accuracy of channel estimation and thereby improve the precision of generating interference replicas, which helps enhance the effects of reducing interference and expand link capacity. COMSIC may be divided into two types: serial type (also referred to as successive type); and parallel type. Figure 2.35a,b shows the block configuration of serial-type COMSIC and parallel-type COMSIC [57], respectively. Both types of COMSIC consist of multistage Channel Estimation and Interference Generation Units (CEIGUs), which handle the channel estimation of each user's path, RAKE combining, data decision and generation of interference replicas. Serial-type

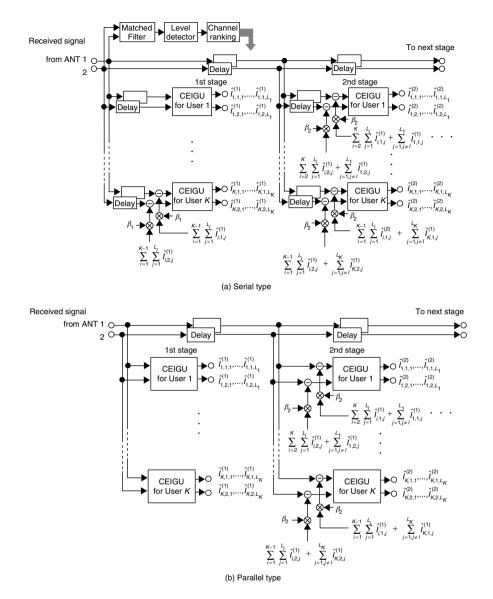


Figure 2.35 Configuration of multistage interference canceller (a) serial type and (b) parallel type

COMSIC firstly measures the received SIR at the MF output and ranks users by the measured SIR in descending order. CEIGU executes demodulation and generates interference replicas according to this order, starting from higher-ranking users with larger received SIR. Assuming that the number of users is K, the number of CEIGUs constituting each stage is K. The MF reception signal of the *l*th path $(1 \le l \le L_k)$ of the *k*th user $(1 \le k \le K)$ in the *p*th stage $(1 \le p \le P)$ is $I_{k,l}^{(p)}$, and the estimated value is $\hat{I}_{k,l}^{(p)}$. In the first stage, the interference replica of users who rank higher than the user's own

channel $\hat{I}_{k,l}^{(p)}$ is deducted from the received signal sequence, leaving the input signal of CEIGU of the user's own channel. The input signal of CEIGU of the lowest-ranking user is what remains after subtracting the interference replicas of all users from the received signal. In the following stages, the interference replicas of other users ranking higher than the user's own channel generated in the same stage are deducted from the received signal sequence, whereas the interference replicas of other users ranking lower than the user's own channel generated in the previous stage are deducted from the same. In such a manner, the interference replicas are generated with reference to the reception timing of each RAKE-combined path estimated by the receiver, the channel estimation value and the decision data. However, interference replicas are tainted with errors, especially when they have been generated with errors in the channel estimation value. Hence, a weighted real value of less than 1, referred to as the Interference Rejection Weight (IRW) β_p , is subtracted from the interference replicas attributable to channel estimation errors and improves the BER characteristics in COMSIC [57].

In contrast, parallel-type COMSIC carries out demodulation and generates interference replicas for all users in a parallel manner (i.e. simultaneously). In the first stage, the reception signal is directly entered in CEIGU as in the case of MF-based RAKE reception. In CEIGU in the following stages, the interference replicas of all other users $\hat{I}_{k,l}^{(p-1)}$ generated in the previous stage with reference to the reception signals are weighted and subtracted by IRW β_p and the resulting signals are entered. As explained before, serialtype COMSIC subtracts the interference replicas of users ranking higher than the user's own channel from the reception signals in the first stage, which enables the generation of interference replicas at higher accuracy than parallel-type COMSIC. Therefore, assuming that the number of stages is the same, serial-type COMSIC can assure superior BER characteristics to its parallel counterpart. The drawback of serial-type COMSIC is that it suffers from substantial increases in the delay time in the demodulation process as the number of users increases, because it carries out demodulation and generates interference replicas in the descending order of reception power. Considering the delay time associated with the demodulation process, parallel-type COMSIC is more practical than its serial counterpart.

2.3.1.2 Iterative Channel Estimation Multistage Interference Canceller

As mentioned above, parallel-type COMSIC has to generate interference replicas based on MF reception without any suppression of interference in the first stage. The accuracy of interference replicas generated by parallel-type COMSIC is therefore poorer than serial-type COMSIC, owing to inferior channel estimation accuracy and more data decision errors. As a solution to make the interference replicas generated by parallel-type COMSIC as accurate as its serial counterpart, parallel-type COMSIC that performs iterative channel estimation and data decision in each stage using pilot symbols and decision feedback data symbols has been proposed [58]. Figure 2.36 shows the configuration of CEIGU performing iterative channel estimation using pilot symbols and decision feedback data before or after error-correction decoding. MF input signals received via the first multipath propagation path of the *k*th user in the *p*th stage are reception signals from which interference replicas of all other users and interference replicas of other

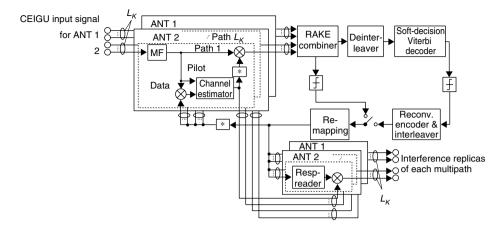


Figure 2.36 Configuration of CEIGU for iterative channel estimation

multipath reception signals in the user's own channel have been subtracted. Firstly, channel estimation is performed on each path using pilot symbols to undergo RAKE combining. The RAKE-combined signals are deinterleaved and then subject to errorcorrection decoding. Secondly, the binary decision data after error-correction decoding are subject to FEC again (channel encoding) and interleaved. The resulting data sequence is used to remove the data modulation components in the MF output signals (demodulation). The information data symbols are added to the pilot symbols in order to perform channel estimation again, to be used in channel estimation for generating interference replicas and RAKE combining. The repetition of channel estimation using pilot symbols and decision feedback data sequence after error-correction decoding with hardly any decision errors improves the accuracy of channel estimation, which helps radically improve the accuracy of generating interference replicas. The use of tentative decision data symbols after RAKE combining results in a shorter process delay time than using decision data after error-correction decoding, but the anti-interference characteristics diminish owing to many data decision errors. The use of iterative channel estimation on parallel-type COMSIC using pilot symbols and decision feedback data after errorcorrection decoding is known to achieve more or less the same anti-interference effects with a shorter delay time associated with demodulation processing compared to its serial counterpart.

Figure 2.37 shows the number of users normalized by processing gain P_g against the transmit E_b/N_0 at an average BER = 10^{-3} using three-stage parallel-type COMSIC with iterative channel estimation. The propagation model assumes the two-path model with equal average power subject to independent Rayleigh fading at $f_D = 80$ Hz. For comparison, the figure also shows the characteristics of MF-based RAKE-combined reception using two-branch space diversity reception. The figure shows that in an environment with substantial noise power, the effects of COMSIC is limited in increasing the capacity, whereas in areas where the interference power is dominant and transmission E_b/N_0 is large, the capacity can be dramatically increased. COMSIC that generates interference replicas based on channel estimation using pilot symbols only can increase capacity by

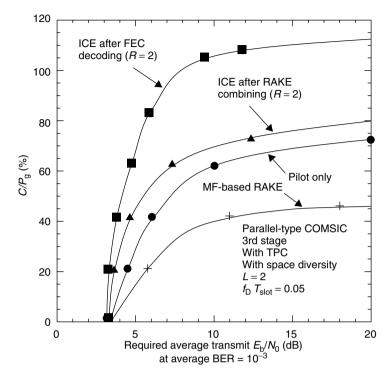


Figure 2.37 Capacity characteristics of isolated cell using parallel-type COMSIC

approximately 1.6 times compared to MF-based RAKE-combined reception. COMSIC with iterative channel estimation (iteration = 2) using decision data after error-correction decoding in addition to pilot symbols can increase capacity by about 2.5 times compared to MF-based RAKE-combined reception.

2.3.2 Adaptive Antenna Array Diversity

As shown in Figure 2.38, adaptive ANT array diversity involves the installation of an adaptive ANT array transceiver in BS and the multiplication of signals received from multiple ANTs in uplink by optimal reception ANT weight, followed by combining. This results in an orientation pattern with a main lobe in the direction from which the desired signals arrive and a beam in the direction from which interference waves arrive aimed at nullifying such interference waves, which enables the maximization of received SIR [59–61]. This consequently helps reduce MAI and increases the uplink system capacity. In downlink, it involves the multiplication of transmission ANT weight for each user generated adaptively at BS by the transmission signals of each user, aimed at generating a main lobe in the direction of the user's desired signals and carrying out transmission in a manner that reduces interference in the direction of other users. This enables the system capacity to expand in downlink.

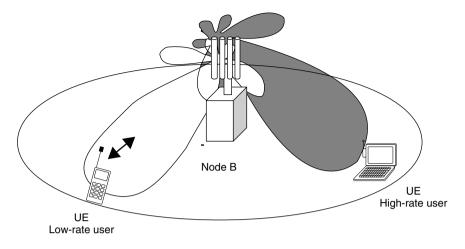


Figure 2.38 Mechanism of adaptive antenna array diversity

2.3.2.1 Configuration of Adaptive Antenna Array Diversity

Adaptive ANT array diversity is a technology that processes adaptive reception (uplink) and adaptive transmission (downlink) at BS and is configured in a way that is applicable to the W-CDMA radio interface specified by 3GPP [61]. Adaptive Antenna Array Transmit Diversity (AAA-TD) generates transmission ANT weight by executing the following on the reception ANT weight generated at the BS reception unit: (1) RF circuit calibration, which compensates for phase and amplitude deviations between branches in the RF transmission and reception circuits and (2) carrier frequency calibration, which compensates for shifts in the direction of the main lobe and nullifying beam in the transmission ANT weight caused by differences in the carrier frequency between uplink and downlink (the focus is on Frequency Division Duplex (FDD), in which the carrier frequency is not the same for uplink and downlink). Figure 2.39 shows the block configuration of BS equipment, which performs adaptive ANT array transmission and reception diversity. Signals received by each ANT in uplink are limited in bandwidth and amplified through an RF reception circuit and then weighted by complex reception ANT weight and combined at the Coherent Adaptive Antenna Array Diversity (CAAAD) reception unit. This is followed by the estimation of phase and amplitude fluctuations caused by fading in each path (channel estimation) and RAKE combining. RAKE-combined signals are then subject to deinterleaving and error-correction decoding, for the recovery of the transmission data sequence. On the other hand, at the BS transmission unit, the channel-encoded (FEC), interleaved and QPSK data-mapped in-phase (I) and quadrature (Q) components are multiplied by complex transmission ANT weights and subject to spreading. Subsequently, their frequency is converted and amplified at the RF transmission unit for transmission.

As shown in Figure 2.39, reception ANT weights generated in the CAAAD reception unit are affected by phase and amplitude fluctuations between branches in the RF reception circuit, in addition to the incident angle and average reception power of desired waves and interference waves at the end of the ANT. Moreover, the transmission ANT weight generated in the baseband digital processing unit is affected by phase and amplitude

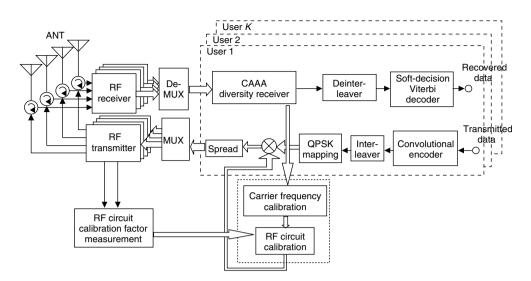


Figure 2.39 General configuration of adaptive antenna array diversity

fluctuations in the RF transmission circuit up to the point of output in the transmitter. Thus, in order to generate reception ANT weights that reflect nothing but the incident angle and average reception power of signals at the end of the ANT, it is necessary to eradicate the phase and amplitude deviations between branches in the RF reception circuits from the reception ANT weights generated by MMSE algorithm. Assuming that the complex transmission function of the RF reception circuit is $x_{RX}^{(j)}$ from the low-noise AMP in the *j*th ($1 \le j \le J$, J = number of ANTs) reception system up to the stage after correcting the control power of the AGC AMP, the ideal complex ANT weight $w_{ideal}^{(j)}$ ("ideal" referring to the weight at $x_{RX}^{(j)} = 1$) is determined by the equation below.

$$w_{\text{ideal}}^{(j)} = w_{\text{R}}^{(j)} x_{\text{RX}}^{(j)}$$
(34)

In Equation (34), $x_{\rm R}^{(j)}$ is the complex reception ANT weight generated in the CAAAD reception unit. The weight determined by Equation (34) requires further compensation in advance for the impact of the transmission function of the RF transmission circuit in the base band transmission ANT weight generation unit. Assuming that $x_{\rm TX}^{(j)}$ is the transmission function of the *j*th RF transmission circuit, the transmission ANT weight $w_{\rm T}^{(j)}$ that needs to be generated by the adaptive ANT array transmission diversity processing unit is expressed in the following equation.

$$w_{\rm T}^{(j)} = w_{\rm ideal}^{(j)} / x_{\rm TX}^{(j)} = w_{\rm R}^{(j)} (x_{\rm RX}^{(j)} / x_{\rm TX}^{(j)})$$
(35)

The calibration of the reception ANT weight generated in the CAAAD reception unit by the transmission function of the RF reception/transmission circuit shown in Equation (35) makes it possible to generate a transmission ANT weight that nullifies interference signals by producing beams in their incident angle at the point of transmission of the ANT [62].

One way to execute carrier frequency calibration is to correct the transmission beam pattern so that the main lobe in the direction of the desired signals and the nullifying beam would be in the same direction within the transmission beam pattern, or that the directions of the main lobe and the nullifying beam within the transmission beam pattern would be the same as their counterparts within the reception beam pattern. If the difference between uplink and downlink carrier frequencies is approximately 185 MHz, provided that the carrier frequencies are in the 2 GHz band, reportedly, the required reception E_b/N_0 in compliance with average BER = 10^{-3} at MS is approximately 0.5 dB lower with carrier frequency calibration (aimed at matching the direction of the main lobe in the transmission beam pattern with its counterpart in the reception beam pattern) than without such calibration [63].

2.3.2.2 Configuration of CAAAD Reception Unit

Figure 2.40 shows the block configuration of the CAAAD reception unit [64, 65]. As the reception ANT weight generated in uplink based on this method is used as the transmission ANT weight in downlink, the adaptive ANT array and the RAKE combining unit have different functions. In short, beam patterns generated by the adaptive ANT array are controlled so that the average received SIR would be maximized, rather than tracking instantaneous channel fluctuations. On the other hand, the RAKE combining unit follows the beam pattern generation unit and maximizes the instantaneous received SIR by executing weighted-combining with the fading reception envelope. The CAAAD reception unit consists of MF, beam pattern generation unit, coherent RAKE combining unit and ANT weight control unit. The received signal sequence of each ANT branch is despread on the basis of a despreading code replica synchronized with the multipath reception timing of each user according to estimation. The despread signal is weighted with reception ANT weight and combined, and channel estimation is performed using pilot symbols. The channel estimation value is used to compensate for the phase variations caused by fading

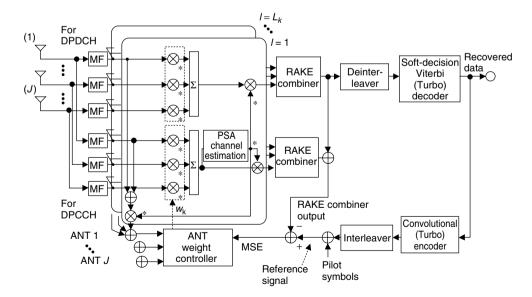


Figure 2.40 Configuration of CAAAD reception unit

of the information data symbol of each path and is subject to Maximal Ratio Combining (MRC). In the ANT weight control unit, the reception ANT weight is updated using an adaptive algorithm so that the Mean-Squared Error (MSE) of the RAKE-combined signals would be minimal. In order to accelerate the convergence of ANT weight, pilot symbols and decision feedback data symbols after error-correction decoding are used as reference signals.

Cellular systems require an ANT weight-tracking function that can track MS in fast motion at high accuracy. As it is generally difficult to track fast-moving MS by merely updating the reception ANT weight using an adaptive algorithm, a two-step, high-speed reception ANT weight-tracking method has been proposed, which uses the SIR measurement value of each slot in combination with the adaptive algorithm. It has been shown that this method is capable of tracking changes in the incident angle of signals of an angular speed up to 34°/sec (speed at a location 100 m away from BS is approximately 220 km/h) [66].

Furthermore, in uplink, short packets such as RACHs need to be received as mentioned earlier. Multibeam reception with a fixed ANT weight (12 beams or so) has been found to be suitable for the task of receiving short packets that are 10 or 20 msec long, in place of CAAAD reception that does not allow the reception ANT weight to converge sufficiently in the packet zone [67].

A practical number of ANT element arrays is extremely small compared to the number of speech transmission (low-rate) users that can be accommodated by each sector. As the number of nullifying beams that can be generated depends on the number of ANT element arrays, it is difficult to generate nullifying beams in the incident angle of signals from each low-rate user. Hence, the beam pattern generated in the CAAAD reception unit gives priority to the creation of beams to nullify signals in the incident angle of signals from high-transmission power, high-rate users. On the other hand, in downlink, the reception quality of low-rate users depends on the extent to which interference from other users (especially high-rate users) could be reduced. Accordingly, in downlink, fast TPC based on SIR measurement is used as well, for the purpose of reducing interference from highrate users. The transmission ANT weight of low-rate users are transmitted in the direction of high-rate users with a transmission weight that has a nullifying beam. This results in less interference from many low-rate users in the direction of high-rate users, and the use of high-speed TPC enables lower transmission power of high-rate users. High-rate users can further reduce their transmission power by executing transmission with a transmission ANT weight oriented toward their own channels. Such a reduction in transmission power of high-rate users helps to reduce interference toward low-rate users and to expand system capacity.

2.3.2.3 Test Results

Figure 2.41 depicts the field test results showing the average BER characteristics against the average received power of the BS for MS 1, taking the average received SIR as parameters when four-ANT CAAAD reception is used in uplink (ANT spacing = $\lambda/2$, where λ is the wavelength of the uplink carrier frequency) [65]. The BS ANT is approximately 50 m high, and a sector ANT with a beam angle of 120° was used in the test. MS 1 (the target mobile station) moved, at 30 km/h on average, along the measurement course in

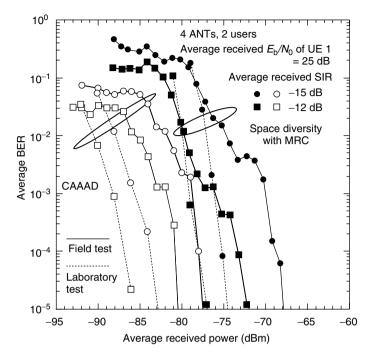


Figure 2.41 Average BER characteristics using closed CAAAD reception in uplink

which the Direction Of Arrival (DOA) of signals at 600 to 850 m away from BS varied by 10° in angle. In the first half of the course, one to two paths were observed, whereas in the latter half two to three paths were detected in which the difference in the average reception power was approximately 3 dB. Meanwhile, the DOA of MS 2 (the interfering mobile station) was fixed at about +40°, approximately 600 m away from BS. MS 2 was almost in the line of sight of BS and signals of one path were observed. For comparison, the figure shows the characteristics resulting from the implementation of four-ANT space diversity reception (ANT spacing = 10 λ) with MRC. As shown by the figure, CAAAD reception effectively suppresses interference, especially when the interference power is large, and the required reception power that satisfies an average BER = 10⁻³ was reduced by approximately 8 to 10 dB compared to MRC space diversity reception.

Figure 2.42 illustrates the field test results showing the average BER characteristics against the average reception power of MS using AAA-TD in downlink [68]. The characteristics were obtained assuming that average received SIR = 0 dB for MS 1 in uplink and that transmission SIR = -5, -10 and -12 dB for MS 1 before multiplication by ANT weight in downlink. The conditions of DOA and other factors associated with MS 1 and MS 2 in the test were the same as in Figure 2.41. For the sake of comparison, the figure also depicts the characteristics obtained by one-ANT transmission. The figure shows that one-ANT transmission gives rise to an error floor owing to the MPI of high-transmission-power interference wave as transmission SIR decreases, whereas the use of AAA-TD radically improves the BER characteristics. Even when MS 1 is traveling at an average speed of 30 km/h, the main lobe in the transmission beam pattern is able to track

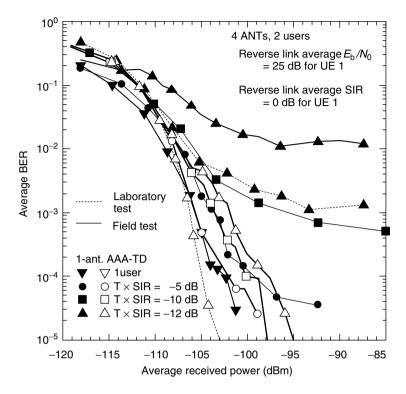


Figure 2.42 Average BER characteristics using AAA-TD in downlink

the movement of MS 1, showing that AAA-TD is effective in reducing interference from high-rate users.

References

- Simon, M.K., Omura, J.K., Scholtz, R.A. and Levitt, B.K., Spread Spectrum Communications, Vol. I, II & III, Computer Science Press, Rockville, MD, 1985.
- [2] Viterbi, A.J., CDMA: Principles of Spread Spectrum Communication, Addison-Wesley, Reading, MA., 1995.
- [3] Marubayashi, G., Nakagawa, M. and Kohno, R. Spread Spectrum Communications and their Applications, Institute of Electronics, Information and Communication Engineers (IEICE), Japan, 1998.
- [4] Su, K. and Zhu, J., 'Performance of SS Communication System by Orthogonal Gold Sequences', Journal of China Institute of Communications, 10(2), 1989.
- [5] Gilhousen, K.S., Jacobs, I.M., Padovani, R., Viterbi, A.J., Weaver, L.A. and Wheatley III, C.E., 'On the Capacity of a Cellular CDMA System', *IEEE Transaction on Vehicular Technology*, VT-40(2), 1991, 303–312.
- [6] Adachi, F., Sawahashi, M. and Suda, H., 'Wideband DS-CDMA for Next Generation Mobile Communication System', *IEEE Communication Magazine*, 36(9), 1998, 56–69.
- [7] Dahlman, E., Gudmundson, B., Nilsson, M. and Skold, J., 'UMTS/IMT-2000 Based on Wideband CDMA', *IEEE Communication Magazine*, 36(9), 1998, 70–80.
- [8] Okawa, K. and Adachi, F., 'Orthogonal Forward Link Using Orthogonal Multi-Spreading Factor Codes for Coherent DS-CDMA Mobile Radio', *IEICE Transactions on Communications*, E81-B(4), 1998, 777–784.
- [9] 3GPP RAN, 3G TS 25.213 V3.3.0, September 2000.

- [10] Higuchi, K., Sawahashi, M. and Adachi, F., 'Fast Cell Search Algorithm in DS-CDMA Mobile Radio using Long Spreading Codes', *IEICE Transactions on Communications*, E81-B(7), 1998, 1527–1534.
- [11] 3GPP RAN, 3G TS 25.211 V3.2.0, March 2000.
- [12] ETSI SMG2 UMTS-L1, 'Proposal for Downlink Time Switched Transmission Diversity', Tdoc 53/98, May 1998.
- [13] Dohi, T., Okumura, Y. and Adachi, F., 'Further Results on Field Experiments of Coherent Wideband DS-CDMA Mobile Radio', *IEICE Transactions on Communications*, E81-B(6), 1998, 1239–1247.
- [14] Higuchi, K., Hanada, Y., Sawahashi, M. and Adachi, F., 'Experimental Evaluation of 3-Step Cell Search Method in W-CDMA Mobile Radio', Proceedings of the 51st IEEE VTC2000, Tokyo, May 2000, pp. 303–307.
- [15] Hanada, Y., Higuchi, K., Sawabashi, M. and Adachi, F., 'Experiments on Target Cell Search Time during Active Mode Using 3-step Cell Search Method in W-CDMA Mobile Radio', Technical Report of IEICE RCS-99–154, November 1999, pp. 91–98.
- [16] Hanada, Y., Higuchi, K., Sawahashi, M. and Adachi, F., 'Fast Cell Search Algorithm in Idle Mode for Inter-Cell Asynchronous W-CDMA Mobile Radio', *IEICE Transactions on Communications*, E83-B(8), 2000, 1610–1618.
- [17] 3GPP RAN, 3G TS 25.214 V3.4.0, September 2000.
- [18] Ishii, M., Nakamura, T. and Onoe, S., 'Performance Evaluation of Path Timing Detection in CDMA Random Access', IEICE General Conference B-5-52, March 2000.
- [19] Berrou, C., Glavieux, A. and Thitimajshima, P., 'Near Shannon Limit Error-Correcting Coding and Decoding: Turbo-Codes', Proceedings of the IEEE ICC '93, Geneva, May 1993, pp. 1064–1070.
- [20] Fujiwara, A., Suda, H. and Adachi, F., 'Turbo Codes Application to DS-CDMA Mobile Radio', *IEICE Transactions on Fundamentals*, E81-A(11), 1998, 2269–2273.
- [21] Woodard, J.P. and Hanzo, L., 'Comparative Study of Turbo Decoding Techniques: An Overview', IEEE Transactions on Vehicular Technology, 49(6), 2000, 2203–2233.
- [22] Shibutani, A., Suda, H. and Adachi, F., 'Multi-Stage Interleaver for Turbo Codes in DS-CDMA Mobile Radio', Proceedings of the IEEE APCC/ICCS '98, November 1998, 391–395.
- [23] Shibutani, A., Suda, H. and Yamao, Y., 'W-CDMA Mobile Radio Performances with Turbo Codes using Prime Interleaver', Technical Report of IEICE RCS99–95, August 1999, pp. 59–66.
- [24] 3GPP RAN, 3G TS 25.322 V.3.2.0, March 2000.
- [25] Chase, D., 'Code Combining a Maximum-Likelihood Decoding Approach for Combining an Arbitrary Number of Noisy Packets', *IEEE Transactions on Communications*, 33(5), 1985, 385–393.
- [26] Souissi, S. and Wicker, S., 'A Diversity Combining DS/CDMA System with Convolutional Encoding and Viterbi Decoding', *IEEE Transactions on Vehicular Technology*, 44(2), 1995, 304–312.
- [27] Hagennauer, J., 'Rate-Compatible Punctured Convolutional Codes (RCPC codes) and Their Applications', *IEEE Transactions on Communications*, 36(4), 1988, 389–400.
- [28] Miki, N., Atarashi, H., Abeta, S. and Sawahashi, M., 'Comparison of Hybrid ARQ Schemes and Optimization of Key Parameters for High-speed Packet Transmission in W-CDMA Forward Link', *IEICE Transactions on Fundamentals*, E84-A(7), 2001, 1681–1690.
- [29] 3GPP RAN, 3G TS 25.212 V3.4.0, September 2000.
- [30] Ariyavisitakul, S., 'Signal and Interference Statistics of a CDMA System with Feedback Power Controlpart II', *IEEE Transactions on Communications*, 42(21314), 1994, 597–605.
- [31] Seo, S., Dohi, T. and Adachi, F., 'SIR-Based Transmit Power Control of Reverse Link for Coherent DS-CDMA Mobile Radio', *IEICE Transactions on Communications*, E81-B(7), 1998, 1508–1516.
- [32] Dohi, T., Sawahashi, M. and Adachi, F., 'Performance of SIR Based Power Control in the Presence of Non-uniform Traffic Distribution', IEEE ICUPC '95, Tokyo, November 1995, pp. 334–338.
- [33] Amezawa, Y. and Sato, S., 'A Study of SIR Measurement Methods Using Signal before Rake Combining', 1998 IEICE General Conference, B-5-110, 1998.
- [34] Fukumoto, S., Okawa, K., Higuchi, K., Sawahashi, M. and Adachi, F., 'Path Search Performance and Its Parameter Optimization of Pilot Symbol-Assisted Coherent Rake Receiver for W-CDMA Mobile Radio', *IEICE Transactions on Fundamentals*, E83-A, 2000, 2110–2119.
- [35] Ling, F., 'Coherent Detection with Reference-Symbol Based Estimation for Direct Sequence CDMA Uplink Communications', Proceedings of the VTC '93, New Jersey, May 1993, pp. 400–403.
- [36] Andoh, H., Sawahashi, M. and Adachi, F., 'Channel Estimation Filter Using Time-Multiplexed Pilot Channel for Coherent Rake Combining in DS-CDMA Mobile Radio', *IEICE Transactions on Communications*, E81-B(7), 1998, 1517–1526.

- [37] Schwartz, M., Bennett, W.R. and Stein, S., Communication Systems and Techniques, McGRAW-HILL, New York, 1996.
- [38] Higuchi, K., Andoh, H., Okawa, K., Sawahashi, M. and Adachi, F., 'Experimental Evaluation of Combined Effect of Coherent Rake Combining and SIR-based Fast Transmit Power Control for Reverse Link of DS-CDMA Mobile Radio', *IEEE Journal on Selected Areas in Communication*, 18(8), 2000, 1526–1535.
- [39] Abeta, S., Sawahashi, M. and Adachi, F., 'Adaptive Channel Estimation for Coherent DS-CDMA Mobile Radio Using Time-Multiplexed Pilot and Parallel Pilot Structures', *IEICE Transactions on Communications*, E82-B(9), 1999, 1505–1513.
- [40] Higuchi, K., Ikeda, T., Fukumoto, S., Sawahashi, M. and Adachi, F., 'Experimental Evaluations of High Rate Data Transmission Using Turbo/Convolutional Coding in W-CDMA Mobile Radio', Proceedings of Wireless 2000, Calgary, July 2000, pp. 687–693.
- [41] Wong, D. and Lim, T.J., 'Soft handoffs in CDMA Mobile Systems', *IEEE Personal Communications*, 4(6), 1997, 6–17.
- [42] TIA/EIA/IS-95, 'Mobile Station-Base Station Compatibility Standard for Dual-Mode Wideband Spread Spectrum Cellular System', Telecommunication Industry Association, July 1993.
- [43] Fukumoto, S., Higuchi, K., Morimoto, A., Sawahashi, M. and Adachi, F., 'Combined Effect of Site Diversity and Fast Transmit Power Control in W-CDMA Mobile Radio', Proceedings of the 51st VTC '2000, Tokyo, May 2000, pp. 1527–1534.
- [44] Morimoto, A., Higuchi, K., Fukumoto, S., Sawahashi, M. and Adachi, F., 'Experiments on Inter-Cell Site Diversity Using Two-step Selection Combining in W-CDMA Reverse Link', *IEICE Transactions on Communications*, E84-B(3), 2001, 435–445.
- [45] NEC, 'Adjustment Loop in Down Link Power Control During Handover', TSG-RAN Working 1 Meeting, October 1999.
- [46] Furukawa, H., Hamabe, K. and Ushirokawa, A., 'SSDT-Site Selection Diversity Transmission Power Control for CDMA Forward Link', *IEEE Journal on Selected Areas in Communication*, 18(8), 2000, 1546–1554.
- [47] Morimoto, A., Higuchi, K. and Sawahashi, M., 'Independent Fast Transmit Power Control for Each Cell Site in W-CDMA Forward Link Inter-cell Site Diversity', Technical Report of IEICE RCS 2000–164, November 2000, pp. 1–7.
- [48] Hottinen, A. and Wichman, R., 'Transmit Diversity by Antenna Selection in CDMA Downlink', Proceedings of the IEEE ISSSTA '98, September 1998, pp. 767–770.
- [49] Fukumoto, S., Sawahashi, M. and Adachi, F., 'Performance Comparison of Forward Link Transmit Diversity Techniques for W-CDMA Mobile Radio', Proceedings of the 10th IEEE PIMRC '99, Osaka, September 1998, pp. 1139–1143.
- [50] ETSI/SMG2, UMTS-L1, 'Space Time Block Coded Transmit Antenna Diversity for WCDMA', Tdoc 662/98, December 1998.
- [51] Alamouti, S.M., 'A Simple Transmit Diversity Technique for Wireless Communications', IEEE Journal on Selected Areas in Communication, 16(8), 1998, 1451–1458.
- [52] Fukumoto, S., Higuchi, K., Sawahashi, M. and Adachi, F., 'Field Experiments on Closed Loop Mode Transmit Diversity in W-CDMA Forward Link', Proceedings of IEEE ISSSTA '2000, New Jersey, September 2000, pp. 433–438.
- [53] Duel-Hallen, A. Holtzman, J. and Zvonar, Z., 'Multiuser Detection for CDMA Systems', *IEEE Personal Communications*, 2(2), 1995, 46–58.
- [54] Moshavi, S., 'Multi-user Detection for DS-CDMA Communications', *IEEE Communication Magazine*, 34(10), 1996, 124–136.
- [55] Yoshida, S., Ushirokawa, A., Yanagi S. and Furuya, Y., 'Delay-detection-type DS-CDMA Adaptive Interference Canceller suitable for High-speed Fading Transmission Paths', *IEICE Transactions on Communications*, J77-B-II(11), 1994, 618–627.
- [56] Varanasi, M.K. and Aazhang, B.A., 'Multistage Detection in Asynchronous Code-Division Multiple-Access Communications', *IEEE Transactions on Communications*, COM-38(4)P, 1990, 509–519.
- [57] Sawahashi, M., Miki, Y. Andoh, H. and Higuchi, K., 'Pilot Symbol-Assisted Coherent Multistage Interference Canceller Using Recursive Channel Estimation for DS-CDMA Mobile Radio', *IEICE Transactions* on Communications, E79-B(9), 1996, 1262–1270.
- [58] Sawahashi, M., Andoh, H. and Higuchi, K., 'Interference Rejection Weight Control for Pilot Symbol-Assisted Coherent Multistage Interference Canceller Using Recursive Channel Estimation in DS-CDMA Mobile Radio', *IEICE Transactions on Fundamentals*, E81-A(5), 1998, 957–972.

- [59] Okawa, K., Higuchi, K. and Sawahashi, M., 'Parallel-type Coherent Multi-stage Interference Canceller with Iterative Channel Estimation Using Both Pilot and Decision-feedback Data Symbols for W-CDMA Mobile Radio', *IEICE Transactions on Communications*, E84-B(3), 2001, 446–456.
- [60] Tsoulos, G.V., Beach, M.A. and McGeehan, J., 'Wireless personal communications for the 21st century: European technological advances in adaptive antennas', *IEEE Communications Magazine*, 35(9), 1997, 102–109.
- [61] Compton Jr., R.T., 'An Adaptive Antenna in a Spread-Spectrum Communication System', Proceedings of the IEEE, 66(3), 1978, 289–295.
- [62] Wang, H., Kohno, R. and Imai, H., 'Adaptive Array Antenna Combined with Tapped Delay Line Using Processing Gain for Direct-Sequence/Spread-Spectrum Multiple Access', *IEICE Transactions on Commu*nications, J75-B-II(11), 1992, 815–825.
- [63] Harada, A., Tanaka, S., Sawahashi, M. and Adachi, F., 'Performance of Adaptive Antenna Array Diversity Transmitter for W-CDMA Forward Link', Proceedings of the 10th IEEE PIMRC '99, Osaka, September, 1999, pp. 1134–1138.
- [64] Taoka, H., Tanaka, S., Nakaminami, N. and Sawahashi, M., 'Experiments on Adaptive Antenna Array Transmit Diversity Using Fast Transmit Power Control in W-CDMA Forward Link', Technical Report of IEICE, RCS2000–169, November 2000, pp. 41–48.
- [65] Tanaka, S., Sawahashi, M. and Adachi, F., 'Pilot Symbol-Assisted Decision-Directed Coherent Adaptive Array Diversity for DS-CDMA Mobile Radio Reverse Link', *IEICE Transactions on Fundamentals*, E80-A(12), 1997, 2445–2454.
- [66] Tanaka, S., Harada, A., Sawahashi, M. and Adachi, F., 'Experiments on Coherent Adaptive Antenna Array Diversity for Wideband DS-CDMA Mobile Radio', *IEEE Journal on Selected Areas in Communication*, 18(8), 2000, 1495–1504.
- [67] Ihara, T., Tanaka, S., Sawahashi, M. and Adachi, F., 'Fast Antenna-Weights Tracking Algorithm of Adaptive Antenna Array Diversity Receiver in W-CDMA Reverse Link', *IEICE Transactions on Communications*, E84-B(7), 2001, 1836–1849.
- [68] Nakaminami, N., Tanaka, S., Ihara, T. and Sawahashi, M., 'Comparison of Coherent Adaptive Antenna Array Diversity and Multi-beam Receiver for Packet Transmission in W-CDMA Reverse Link', *IEICE Transactions on Communications*, E84-B(7), 2001, 1824–1835.
- [69] Harada, A., Tanaka, S., Sawahashi, M. and Adachi, F., 'Experiments on Adaptive Antenna Array Diversity Transmitter in W-CDMA Forward Link', Proceedings of the 5th CDMA International Conference (CIC), Seoul, November 2000, pp. 47–51.
- [70] 3GPP RAN, 3G TS 25.201 V3.2.0, March 2000.
- [71] 3GPP RAN, 3G TS 25.301 V3.5.0, June 2000.

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

3

Radio System

Seizo Onoe, Takehiro Nakamura, Yoshihiro Ishikawa, Koji Ohno, Yoshiyuki Yasuda, Nobuhiro Ohta, Yoshio Ebine, Atsushi Murase and Akihiro Hata

3.1 Radio System Requirements and Design Objectives

As stated in Chapter 1, Section 1.2, requirements for International Mobile Telecommunications-2000 (IMT-2000) include system flexibility, economy and conditions on data transmission speed defined in numerical terms. The minimum performance requirement in terms of transmission speed is 2 Mbit/s in an indoor environment, 384 kbit/s in a pedestrian mode and 144 kbit/s in a vehicle mode. For the radio system, Wideband Code Division Multiple Access (W-CDMA), which outperforms the stated requirements, was proposed as the air interface, which led to efforts in standardization and system development. IMT-2000 is noteworthy for its global nature more than anything else, and strong efforts were made to harmonize multiple competing systems that had been proposed in the standardization process, as it was regarded important to develop a globally common air interface to assure the sharing of terminal hardware. As mentioned in Section 1.2.2.1 in Chapter 1, W-CDMA was approved as one of the interfaces in a recommendation by the International Telecommunication Union (ITU), under which it is referred to as IMT-2000 CDMA Direct Spread. In fact, the technology is expected to spread widely in North America, Europe and Asia.

As for the services, one of the major objectives is to provide full-fledged multimedia in the world of mobile communications. The high-speed transmission capability referred to earlier will make this possible. Under IMT-2000, the air interface and the radio system must be able to accommodate various data speeds, provide multiple services simultaneously and render efficient Packet-Switched (PS) services as well as Circuit-Switched (CS) services. W-CDMA is an effective way to meet these requirements as well.

Regardless of the generation change, the effective use of frequency resources remains as an universal issue for mobile communications. It is important to tackle this issue under IMT-2000 particularly owing to the need to deal with the increasing demand in high-speed data communications.

The frequency band used by IMT-2000 is the 2 GHz band. Because of the higher frequency compared to the Second-Generation (2G) 800 MHz band cellular systems, it is

theoretically more difficult to build cells with a long radius because of the propagation loss. Moreover, link design requirements are stricter as more information needs to be transmitted in volume for the provision of high-speed data services, which increases the required transmission power. Hence, in the development stages, it became an important objective to build an economical system that would assure coverage with more or less the same number of Base Stations (BSs) as in the existing 800 MHz system by applying various types of technologies.

This chapter reviews the characteristics of W-CDMA as a radio system developed with the aforementioned objectives in mind, as well as the system architecture and the key technologies. It also describes the interface specifications of the Radio Access Network (RAN) as a standard, and the configuration of the radio Network Equipment (NE) in actual system development.

3.2 W-CDMA and System Architecture

3.2.1 Characteristics of W-CDMA

W-CDMA has the following technical characteristics.

(i) Highly Efficient Frequency Usage

In principle, the potential capacity of the system should be regarded the same even when multiple access technologies like Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) are applied. While Code Division Multiple Access (CDMA) is often claimed to have a high efficiency of frequency usage, it should be interpreted as referring to how easy it is to improve the efficiency of frequency usage. For example, CDMA can achieve a certain level of efficiency by precise Transmit Power Control (TPC), whereas TDMA would have to resort to an extremely sophisticated dynamic channel assignment to achieve the same level of efficiency. Using the basic technologies of the CDMA system in the right way would lead to a system with highly efficient frequency usage.

(ii) Freedom from Frequency Administration

As CDMA allows adjacent cells to share the same frequency, no frequency allocation plan is required. In contrast, FDMA and TDMA require frequency allocation – in particular, much difficulty is involved in frequency allocation because of the way in which stations are located in practice, as irregular propagation patterns and topographic features need to be considered. It should also be noted that imperfect frequency allocation designs diminish the efficiency of frequency usage. CDMA requires no frequency allocation plan as such.

(iii) Low Mobile Station Transmit Power

CDMA can improve reception performance and reduce the transmission power of Mobile Stations (MSs) by technologies like RAKE reception and so on. In TDMA, transmission is intermittent; the peak power required for the transmission of 1 bit is multiple times the number of TDMA multiplexes compared to continual transmission. On the other hand, the peak power may be small in CDMA, as continual transmission is possible. The additional merit of this feature is that it minimizes the impact to the electromagnetic field.

(iv) Resources Used Independently in Uplink and Downlink

In CDMA, it is easy to support an asymmetric uplink and downlink configuration. For example, in other access systems such as TDMA, it is difficult to assign time slots for uplink and downlink to one user independent of the other. In FDMA, it is difficult to build an asymmetric uplink and downlink configuration because the carrier bandwidth in uplink and downlink would have to be changed. In contrast, in CDMA, the Spreading Factor (SF) can be set independently between uplink and downlink for each user, and thereby set different speeds in uplink and downlink. This allows the efficient use of radio resources even in asymmetric communications, such as Internet access. When there is no transmission, no radio resources are used; therefore, if one user is executing transmission in uplink only, and another user is performing transmission in downlink resources. Generally, TDMA and FDMA would have to assign two pairs of radio resources in such cases.

The wideband properties of W-CDMA allow higher efficiency in the following aspects.

(i) Wide Range of Data Speeds

Wideband enables transmission at high speed. It also enables the efficient provision of services when there is a combination of low-speed services and high-speed services.

For example, in TDMA, various transmission speeds can be offered by varying the settings of the assigned number of time slots, but a low-speed, speech-only mobile phone would still require the same peak power as the peak transmission power required for maximum-speed services.

(ii) Improved Multipath Resolution

RAKE diversity reception technology improves the reception performance by separating multipaths into individual paths for reception and combining. As wideband improves the resolution of the propagation path, the required reception power need not be high because of the path diversity effect brought about by the increased number of paths. This helps reduce transmission power and increase capacity. A typical example of this has been demonstrated in a field test revealing that the required transmission power at approximately 4 Mcps is about 3 dB less than at approximately 1 Mcps.

(iii) Statistical Multiplexing Effect

Wideband increases the number of users to be multiplexed by each carrier. Hence, the capacity increases because of the statistical multiplexing effect. Figure 3.1 shows the characteristics of the statistical multiplexing effect. The figure shows that there is some 30% difference when the number of users per carrier is 25 compared to 100. The characteristics are particularly evident in relatively high-speed data communications: the efficiency decreases in narrowband, as the number of channels that can be accommodated by each carrier is limited, whereas in wideband, the efficiency improves because of the statistical multiplexing effect.

(iv) Reduced Intermittent Reception Rate

Wideband accelerates the bit rate in the control channel, and makes it possible to reduce the rate of intermittent reception, which makes the mobile phone receive limited signals when it is in idle mode for saving power. This extends the standby time of the MS (Mobile Station).

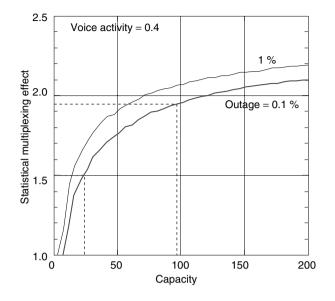


Figure 3.1 Statistical multiplexing effect

Access scheme	Direct sequence CDMA
Duplex scheme	FDD
Bandwidth	5 MHz
Chip rate	3.84 Mcps
Carrier spacing	200 kHz raster
Data speed	\sim 2 Mbit/s
Frame length	10, 20, 40, 80 msec
Forward error correction	Turbo code, convolutional code
Data modulation	Downlink: QPSK, uplink: BPSK
Spreading modulation	Downlink: QPSK, uplink: HPSK
Spreading factor	$4 \sim 512$
Synchronization between base stations	Asynchronous (sync operation also possible)
Speech coding	AMR(1.95 k-12.2 kbit/s)

 Table 3.1
 Basic specifications of W-CDMA

Note: AMR: Adaptive Multi Rate; BPSK: Binary Phase Shift Keying; FDD: Frequency Division Duplex; HPSK: Hybrid Phase Shift Keying; QPSK: Quadrature Phase Shift Keying.

3.2.2 Basic Specifications of W-CDMA

Table 3.1 shows the basic specifications of W-CDMA.

Initially, the Association of Radio Industries and Businesses (ARIB) and the European Telecommunications Standards Institute (ETSI) advocated radio systems centering on a 5-MHz carrier, which also included 10 MHz and 20-MHz carriers. The 3rd Generation

Partnership Project (3GPP) concentrated on completing the specifications for the 5 MHz bandwidth and deleted specifications for other bands. This is attributable to the fact that a 5-MHz-band carrier is enough to achieve 2 Mbit/s transmission even though 20 MHz band is more efficient for transmitting data at 2 Mbit/s, not to mention 3GPP's objective to refine the detailed specifications as quickly as possible. Hence, the current version of specifications by 3GPP and standards by ARIB and ETSI are limited to the 5 MHz bandwidth.

Asynchronous mode between BSs is applied, which requires no strict synchronicity between all the BSs so as to allow for the flexible deployment of the BSs. By design, synchronous mode may also be applied between BSs.

The frame length is basically 10 msec, which may assume values shown in Table 3.1 through interleave.

The data modulation scheme is Quadrature Phase Shift Keying (QPSK) for downlink and Binary Phase Shift Keying (BPSK) for uplink. Hybrid Phase Shift Keying (HPSK) is applied to spreading modulation in uplink. Detection is based on pilot-symbol-aided coherent detection. For downlink, pilot symbols are time-multiplexed, which helps minimize delays in TPC and simplify the reception circuit in the MS. For uplink, pilot symbols are spread by spreading codes different from the data and are I/Q-multiplexed with the data. This ensures continuous transmission even when variable-rate transmission is carried out, and minimizes the peak factor in the transmission waveform. It is also an effective way to reduce electromagnetic effects and relax the requirements of the transmission AMPlifier (AMP) in the mobile phone.

Variable SF is applied to achieve multirate transmission. For downlink, Orthogonal Variable Spreading Factor (OVSF) is applied. Multicode may also be used.

Convolutional codes are used for channel encoding. For high-speed data, turbo codes are applied.

Dedicated pilot symbol scheme is applied, which is effective for fast closed-loop TPC in downlink. In addition, common pilot symbols for the demodulation of common channels are available, which may also be used for the demodulation of dedicated channels. The dedicated pilot symbol scheme has the edge in that it can assure extensibility for applying adaptive ANTennas (ANTs) and other technologies.

3.2.3 Architecture of Radio Access Network

Figure 3.2 illustrates the system architecture of W-CDMA. The RAN consists of the Radio Network Controller (RNC) and Node B, and is connected with the CN (switching system network) via the Iu interface. Under 3GPP, RAN is referred to as UMTS Terrestrial Radio Access Network (UTRAN).

RNC is in charge of the administration of radio resources and the control of Node B; for example, it performs handover control. Node B stands for the logical node in charge of radio transmission and reception, and is specifically called the Base Transceiver Station (BTS). The interface between Node B and RNC is called Iub. The interface between RNCs is also specified, referred to as Iur. This is a logical interface that may establish connection physically between RNCs; however, alternative transmission methods may be applied, such as physical connection via the Core Network (CN).

Node B covers one or more cells. If the BS is sectorized by multiple directional ANTs, each sector is called a cell. Node B is connected with the User Equipment (UE) via the

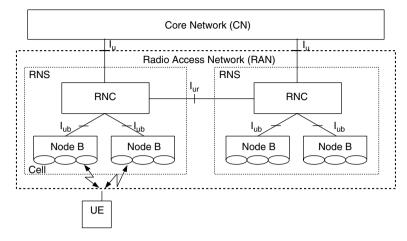


Figure 3.2 Network architecture

radio interface. This section concentrates on the description of standardized specifications; the configuration of equipment will be discussed in detail in Section 3.5.

Figure 3.2 illustrates the protocol architecture of the radio interface for W-CDMA systems, which consists of three layers: the physical layer (Layer 1; L1), the data link layer (Layer 2; L2) and the network layer (Layer 3; L3). Layer 2 can be divided into two sublayers: Medium Access Control (MAC) and Radio Link Control (RLC). RLC is in charge of retransmission control and so on.

The Control-Plane (C-Plane) is engaged in forwarding control signals, whereas the User-Plane (U-Plane) is in charge of forwarding user information. The Packet Data Convergence Protocol (PDCP) and Broadcast/Multicast Control (BMC) of Layer 2 are applicable only to the U-Plane.

Layer 3 consists of Radio Resource Control (RRC) terminated at RAN and higher layers terminated at CN (e.g. Call Control (CC), Mobility Management (MM)). As the focus is on the radio access interface, this chapter describes Layer 3 with reference to RRC only.

In order to deal flexibly with various types of services and multicall capabilities, the radio interface is configured on the basis of three layers of channels: physical channels, transport channels and logical channels.

The ellipse in Figure 3.3 indicates the Service Access Point (SAP) between layers or sublayers. SAP between RLC and MAC offers logical channels, that is, the logical channels are supplied from the MAC sublayer to the RLC sublayer. Logical channels are categorized depending on the function of transmission signals and their logical properties, and are characterized by the content of information transmitted.

SAP between RLC and physical layer L1 offers transport channels, that is, the transport channels are supplied from the physical layer to the MAC sublayer. Transport channels are categorized depending on the transmission format and are characterized depending on how and what kind of information is transmitted through the radio interface.

Physical channels are categorized in consideration of their physical-layer functions, and are identified by the spreading code and frequency carrier, and in the case of uplink the modulation phase (I phase, Q phase).

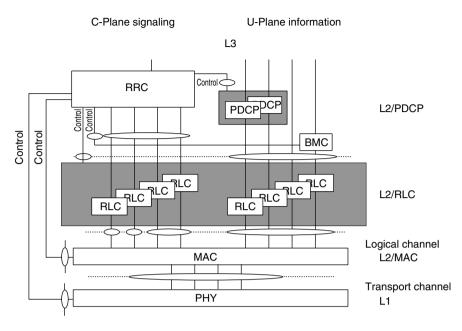


Figure 3.3 Protocol architecture

Multiplexing and transmitting multiple transport channels over these physical channels make it possible to multiplex user data and control information, and multiplex and transmit multiple user data associated with multiaccess. Also, linking multiple logical channels to a single transport channel enables efficient transmission. Mapping of the transport channel to the physical channel takes place in the physical layer, whereas mapping of the logical channel to the transport channel takes place in the MAC sublayer.

Figure 3.4 illustrates how mapping takes place between the principal physical channels, transport channels and logical channels.

Dedicated Physical CHannel (DPCH) consists of the Dedicated Physical Data CHannel (DPDCH) and the Dedicated Physical Control CHannel (DPCCH). DPDCH is a channel for sending data, whereas the DPCCH is attached to DPDCH to execute L1 control such as TPC. Physical channels other than those illustrated in Figure 3.4 include the Synchronization CHannel (SCH), Common PIlot CHannel (CPICH), Acquisition Indicator CHannel (AICH) and Paging Indicator CHannel (PICH). SCH is used for cell search. CPICH is a channel for transmitting pilot symbols to demodulate Common Control Physical CHannel (CCPCH) and is also used to improve the demodulation of dedicated channels as well as common channels. AICH is used for random access. PICH is applied to improve the rate of intermittent reception between UEs upon the transmission of paging signals. The details and the applications of transport channels, physical channels and logical channels are described in Sections 3.3.1.1, 3.3.1.2 and 3.3.2.1, respectively.

3.2.4 Key W-CDMA Technologies

W-CDMA adopts the following distinctive technologies.

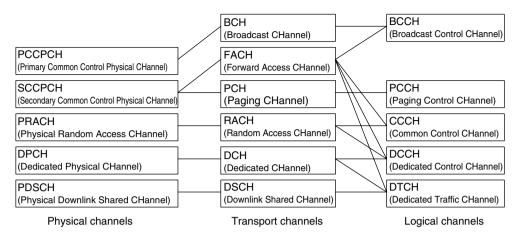


Figure 3.4 Mapping between key physical channels, transport channels and logical channels

3.2.4.1 Inter-BS Asynchronous Mode and Downlink Code Allocation

Asynchronous mode is applied when there is no need to maintain accurate synchronicity among all BSs. It is adopted with the aim to ensure an easy deployment of seamless BS coverage from indoors to outdoors. Figure 3.5 illustrates the downlink spreading code allocation for asynchronous systems. Two sets of spreading codes are used; the scrambling code and the channelization code. A scrambling code is a code assigned to each cell for cell identification purposes, with a frame length of 10 msec (longer than a channelization code) and treats interfering signals from other cells as noise. The channelization code is for identifying each user, and a set of codes that are orthogonal to each other are used in each cell.

Synchronous mode assigns a code corresponding to a scrambling code to each cell at multiple timings, by time-shifting a single code pattern. In contrast, asynchronous mode

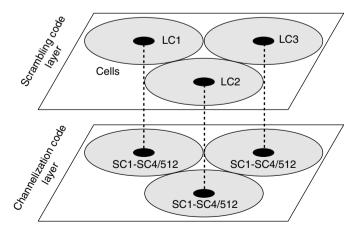


Figure 3.5 Downlink code allocation in inter-BS asynchronous mode

assigns as many patterns as the number of scrambling codes. In this case, some creativity is required to make the UE detect the cell to which it belongs. The system adopts a threestep, high-speed cell search technology that radically reduces the time consumed by the UE in cell searching, which makes asynchronous mode between BSs feasible. Figure 3.6 shows the mechanism of three-step, high-speed cell search.

3.2.4.2 OVSF Transmission

In order to provide multimedia services, the scheme must be efficient even when there is a combination of services at various speeds, ranging from high to low data rates. For downlink, a spreading code that assures OVSF is applied, which generates codes that are orthogonal to each other even if the SF (i.e. code length) is different. This enables the provision of various bit rate services through channels that are orthogonal to each other.

3.2.4.3 Pilot Configuration

Pilot-symbol-aided coherent detection is applied not only to downlink but also to uplink. The pilot symbols in downlink are time-multiplexed with data symbols, which help minimize delays in TPC and simplifies the reception process in UE. The pilot symbol used for time-multiplexing dedicated channels in downlink is also effective in fast downlink TPC.

On the other hand, for uplink, data symbols are I/Q-multiplexed with pilot symbols. In other words, they are subject to BPSK modulation, and are combined at phase zero and $\pi/2$. This makes variable-rate uplink transmissions continual and nonbursty. It also minimizes the peak factor in the transmission waveform and relaxes the requirements of the transmission AMP in the UE. Figure 3.7 is a conceptual diagram of pilot symbols and data multiplexing.

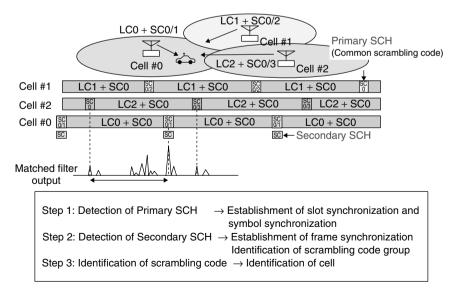


Figure 3.6 Mechanism of three-step fast cell search

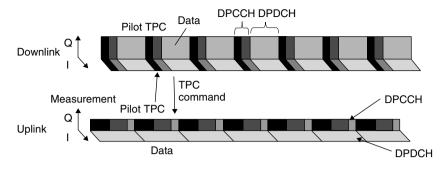


Figure 3.7 Pilot structure

For downlink, CPICH that is used for demodulating the common channel is also applied for the demodulation of dedicated channels.

Dedicated pilot symbols multiplexed over dedicated channels are also an effective solution for assuring extensibility, for the application of applying adaptive ANTs and other technologies for further improvement.

3.2.4.4 Packet Access Method

As packet transmission constitutes the key to third-generation (3G) services, various studies were conducted on the transmission technologies. W-CDMA adopts a system that adaptively switches between common channels and dedicated channels depending on the data traffic, harnessing the characteristics of CDMA in packet transmission.

Figure 3.8 shows the mechanism of packet transmission. When the volume of transmission data is large, it is more efficient to assign DPCH and use minimal power by TPC. On the other hand, when the volume of data is small, and if traffic is bursty, it is more efficient to use a common channel than assigning DPCH. In this scheme, the system adaptively switches between common channels and dedicated channels according to the data traffic [1].

Other schemes are also adopted, including downlink-shared channel, in which the downlink channel is shared by multiple users. Figure 3.9 illustrates the behavior of the downlink-shared channel. Low-speed dedicated channels are attached to the downlink-shared channel. The physical Control CHannels (CCH) on these dedicated channels carry out control and also indicate the information required for decoding the shared channel. This arrangement is required because of the fact that the shared channel is used by multiple users, which makes it necessary to inform as to whether decoding should be executed on the basis of the user's own data. The downlink-shared channel is believed to be effective in downlink high-speed data transmissions.

3.2.4.5 Turbo Codes

As for error-correction codes, studies were conducted on the application of turbo codes to mobile communications, which are claimed to have high error-correction performance for relatively high-speed transmissions. Turbo codes are adopted with an optimized interleaver.

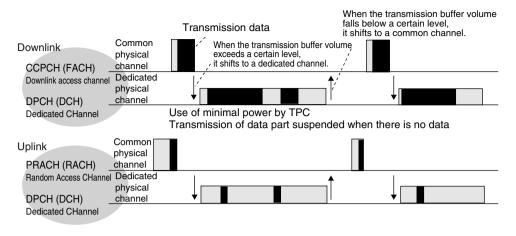
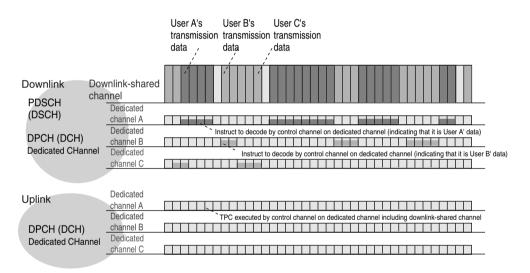
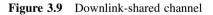


Figure 3.8 Packet transmission adapted to common and dedicated channels





3.2.4.6 TPC

As uplink TPC is a necessary function for avoiding the so-called near-far problem in Direct-Sequence CDMA (DS-CDMA), (Signal to Interference power ratio) SIR-based TPC is applied. For downlink, TPC with the same cycle as uplink is applied, as fast TPC is effective in improving downlink efficiency as well.

3.2.4.7 Transmission Diversity

A number of transmission diversity technologies have been studied and subsequently adopted to boost performance: the open-loop-type Time-Switched Transmit Diversity (TSTD) and Space-time block coding based Transmit ANT Diversity (STTD), which use no feedback loop; and the closed-loop type, which resorts to feedback. TSTD switches the transmission ANT in each slot, whereas STTD improves the error-correcting effects by randomizing the errors at the point of reception by encoding the same data and sending them from two transmission ANTs simultaneously. The closed-loop type, which is applied to dedicated channels, reduces fading by controlling the carrier phase transmitted from two ANTs with reference to feedback from the UE at the point of reception.

3.2.5 Time Division Duplex (TDD) and Frequency Division Duplex (FDD)

The duplex scheme in W-CDMA is FDD. However, 3GPP, which develops specifications of W-CDMA (i.e. UTRA FDD), is not restricted to the FDD mode; it also develops specifications of the TDD mode, UTRA TDD. The TDD mode is developed in such a way that it has many common characteristics with FDD; in fact, the higher-layer protocols are the same in FDD and TDD. The basic parameters in Layer 1 of TDD are also the same as FDD. For example, chip rate, frame length, modulation and demodulation schemes, and other key parameters are the same in both modes. There are two options regarding the chip rate: 3.84 Mcps, and 1.28 Mcps (which is 1/3 of the former). Refer to Section 7.2 for the technical details on TDD mode.

3.3 Radio Access Interface Standard

3.3.1 Physical Layer

3.3.1.1 Transport Channel

The transport channels are the channels supplied from the physical layer [2-5] to the MAC sublayer. There are several types of transport channels to transmit data with different properties and transmission formats over the physical layer.

Table 3.2 is a list of transport channels.

3.3.1.2 Physical Channel

Physical channels are identified by code and frequency in FDD mode.

They are normally based on a layer configuration of radio frames and timeslots (excluding some physical channels). The form of radio frames and timeslots depends on the symbol rate of the physical channel.

Radio Frame: The minimum unit in the decoding process, consisting of 15 time slots.

Time slot: The minimum unit in the Layer 1 bit sequence. Also the minimum unit in TPC and channel estimation process. The number of bits that can be accommodated in one time slot depends on the physical channel.

Table 3.3 shows the types and applications of the physical channels. The structure of key physical channels is described in the following sections.

Name of physical channel	Application		
DCH (Dedicated CHannel)	A bidirectional channel used for transmitting user data. Assigned individually to each UE. Able to vary the rate and control the power at high speed.		
BCH (Broadcast CHannel)	A downlink common channel for transmitting broadcast information (e.g. system information, cell information).		
	BCH is transmitted at a fixed rate.		
FACH (Forward Access CHannel)	A downlink common channel used for transmitting control information and user data. Shared by multiple UEs.		
	Used for low-rate data transmissions from the higher layer.		
PCH (Paging CHannel)	A downlink common channel used for transmitting paging signals.		
RACH (Random Access CHannel)	An uplink common channel used for transmitting control information and user data.		
	Applied in random access, and used for low-rate data transmissions from the higher layer.		
CPCH (Common Packet CHannel)	An uplink common channel used for transmitting user data.		
	Applied in random access, and used primarily for high-rate, bursty data transmissions.		
DSCH (Downlink Shared	A downlink common channel used for transmitting		
CHannel)	packet data.		
	Shared by multiple UEs. Used primarily for high-rate data transmissions.		

Table 3.2 List of transport channels

Table 3.3	List of	physical	channels
-----------	---------	----------	----------

Name of physical channel	Application
DPCH (Dedicated Physical CHannel)	A bidirectional uplink/downlink channel, assigned individually to each UE. Consists of the Dedicated Physical Data CHannel (DPDCH) and the Dedicated Physical Control CHannel (DPCCH). For downlink, DPDCH and DPCCH are time-multiplexed in the time slot; for uplink, they are mapped to I phase and Q phase, respectively.
DPDCH (Dedicated Physical Data CHannel)	At least one DPDCH is assigned to each UE using DPCH. Used for transmitting data from the higher layer.

(continued overleaf)

Name of physical channel	Application
DPCCH (Dedicated Physical Control CHannel)	Only one DPCCH is assigned to each UE using DPCH. Used for controlling the physical layer of DPCH (DPDCH and DPCCH).
PRACH (Physical Random Access CHannel)	An uplink common channel. Used for transmitting data from the higher layer (mainly low-rate).
PCPCH (Physical Common Packet CHannel) CPICH (Common Pilot CHannel)	 An uplink common channel. Used for transmitting packet data (mainly high-rate). A downlink common channel. There are two types of CPICH: Primary CPICH and Secondary CPICH. One Primary CPICH exists in each cell. Primary CPICH is mainly used for downlink channel estimation, for UE cell search, and as the timing reference of other downlink physical channels in the same cell. Secondary CPICH is mainly used when Adaptive Antenna Array (AAA) is applied.
P-CCPCH (Primary Common Control Physical CHannel)	A downlink common channel. One P-CCPCH exists in each cell. Used for transmitting broadcast information.
S-CCPCH (Secondary Common Control Physical CHannel) SCH (Synchronization CHannel)	 A downlink common channel. More than one S-CCPCH may exist in each cell. Used for transmitting paging signals and data from the higher layer (mainly low-rate). A downlink common channel. There are two types of SCH: Primary SCH and Secondary SCH. One Primary SCH and one Secondary SCH exists in each cell. Used for UE cell search.
PDSCH (Physical Downlink Shared CHannel)	A downlink common channel. Each cell can have multiple PDSCHs (or none). Used for transmitting packet data (mainly high-rate).
AICH (Acquisition Indication CHannel)PICH (Page Indication CHannel)	 A downlink common channel, which exists as a pair with PRACH. Used for PRACH random access control. A downlink common channel, which exists as a pair with S-CCPCH (onto which paging signals are mapped). Transmits call-termination information for each group of terminating calls. UE belonging to call termination group #n receives a Paging CHannel (PCH) in the radio frame mapped to S-CCPCH when it is informed of a terminating call to call termination group #n via PICH.
AP-AICH (Access Preamble Acquisition Indicator CHannel)	A downlink common channel, which exists as a pair with PCPCH. Used for PCPCH random access control.

Table 3.3 (continued)

Name of physical channel	Application
CD/CA-ICH (CPCH Collision Detection/Channel Assignment Indicator CHannel)	A downlink common channel, which exists as a pair with PCPCH. Used for PCPCH collision control.
CSICH (CPCH Status Indicator CHannel)	A downlink common channel, which is associated with AP-AICH. It transmits information on the PCPCH communication state.

Table 3.3(continued)

Uplink Dedicated Physical CHannel (Uplink DPCH)

There are two types of uplink DPCHs, the uplink DPDCH and the uplink DPCCH. DPDCH and PDCCH (Packet Dedicated Control Channel) are I/Q multiplexed within each radio frame.

DPDCH is used for transmitting data generated in the higher layer, that is, for transmitting DCH data. Depending on the connection arrangement of Layer 1, there may be 1, several or no DPDCH.

DPCCH transmits control information generated in the physical layer. The control information consists of the known pilot bits used for channel estimation in coherent detection, the TPC command, FeedBack Information (FBI) and the Transport Format Combination (TFC) Indicator (TFCI).

TFCI refers to the information indicating how many transport channels are multiplexed in the uplink DPDCH reception frame, and what kind of transport format (TF) is used in each transport channel.

Regardless of the connection format in Layer 1, there is always at least one DPCCH.

Figure 3.10 shows the frame structure of Uplink DPCH. Each radio frame (10 ms) is split into 15 slots. Each slot consists of 2560 chips.

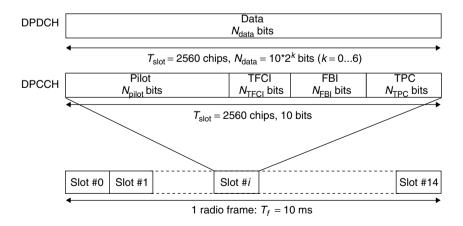


Figure 3.10 Uplink DPCH frame structure

In Figure 3.10, the number of bits per slot in uplink DPDCH/DPCCH is determined by parameter k, which corresponds to the Spreading Factor ($SF = 256/2^k$) of the physical channel. The SF of DPDCH is set in the range from 256 to 4, whereas the SF of DPCCH is always set at 256 (constant).

The FBI field includes information transmitted to the BS from the terminal for closedloop transmission diversity (refer to Section 3.3.1.12) and Site Selection Diversity Transmit power control (SSDT).

In DPCCH, the used slot format is determined by whether TFCI or FBI (the number of bits used) is used, and whether compressed mode is applied (the number of transmission slots). (Refer to Section 3.3.1.13 for compressed mode.)

Physical Random Access CHannel (PRACH)

Random access transmission is based on a slotted ALOHA approach with fast acquisition indication. Specifically, UE transmits the preamble by random access before sending the message part. When it receives an acquisition indication corresponding to the preamble from the network, UE sends the message part.

UE starts the transmission of RACH from a number of predetermined time-offsets, called access slots. There are 15 access slots per 2 frames, which are spaced 5120 chips apart. Figure 3.11 shows the number of access slots and their spacing. Access slots that can be used are specified by the higher layer.

Figure 3.12 illustrates the configuration of PRACH. Random access transmission consists of one or more preambles (4096 chips) and a message (10 ms or 20 ms).

The length of the message part and the arrangement between signature and the access slot are predetermined by the higher layer.

Figure 3.13 shows the radio frame configuration of the random access message part. The message part radio frame of length 10 ms is divided into 15 slots, each consisting of 2560 chips. Each slot consists of a data part that transmits Layer 2 information and a control part that transmits Layer 1 control information (pilot bits and TFCI). The data part and the control part are transmitted in parallel with each other through I/Q multiplexing. The 20-ms-long message part consists of two consecutive message part radio frames.

The data part consists of $10^{*}2^{k}$ bits (k = 0, 1, 2, 3), which corresponds to the Spreading Factor (SF = 256, 128, 64, 32).

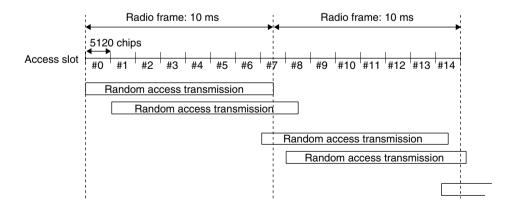


Figure 3.11 Number of RACH access slots and their spacing

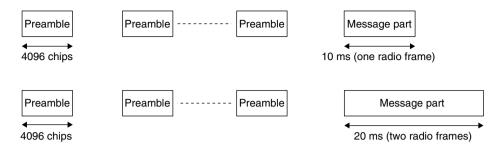


Figure 3.12 Structure of random access transmission

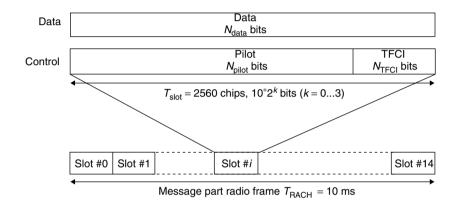


Figure 3.13 Radio frame structure of random access message part

The control part consists of known pilot bits used for channel estimation in coherent detection ($N_{\text{pilot}} = 8$ bits) and TFCI bits ($N_{\text{TFCI}} = 2$ bits).

Downlink Dedicated Physical CHannel (Downlink DPCH)

Downlink DPCH is different from Uplink DPCH in that DPDCH and DPCCH are timemultiplexed.

Figure 3.14 illustrates the frame configuration of downlink DPCH. Each frame is of length 10 ms, which is subdivided into 15 slots. Each slot is 2560 chips, which corresponds to one fast power-control period. The total number of bits in one slot corresponds one-to-one to $SF = 512/2^k$, and SF may take a value between 512 and 4.

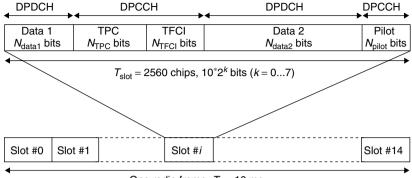
The number of bits in the TPC field (N_{TPC}) may have a value of either 2, 4, 8 or 16, depending on the SF and whether compressed mode is being applied.

The number of bits in the TFCI field (N_{TFCI}) may not be used ($N_{\text{TPC}} = 0$) depending on the method of TF detection in UE. When used, it may have a value of either 2, 4, 8 or 16, depending on the SF and whether compressed mode is being applied.

The number of bits in the Pilot field (N_{pilot}) may have a value of either 2, 4, 8, 16 or 32, depending on SF and whether compressed mode is being applied.

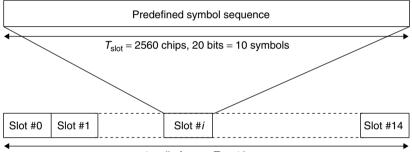
Common Pilot CHannel (CPICH)

CPICH is a fixed rate (30 kbps, SF = 256) channel for transmitting predefined bits and symbol sequences. Figure 3.15 shows the frame configuration of CPICH.



One radio frame, $T_f = 10 \text{ ms}$

Figure 3.14 Downlink DPCH frame structure



1 radio frame: $T_f = 10 \text{ ms}$

Figure 3.15 CPICH frame structure

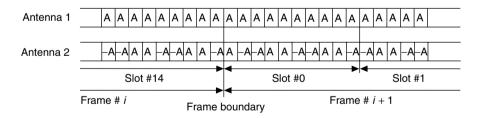


Figure 3.16 CPICH modulation pattern (A = 1 + j)

When transmission diversity (refer to Section 3.3.1.12) is applied (open-loop and closedloop), CPICH shall be transmitted from both ANTs using the same channelization code and scrambling code. In this case, as illustrated in Figure 3.16, the predefined symbol sequences to be transmitted over CPICH are different between ANT 1 and ANT 2. If transmission diversity is not applied, the sequence of ANT 1 in Figure 3.16 is transmitted.

There are two types of CPICH: Primary CPICH (P-CPICH) and Secondary CPICH (S-CPICH).

P-CPICH has the following characteristics:

- 1. The same channelization code is always used for P-CPICH
- 2. Scrambling is performed by a primary scrambling code
- 3. Only one P-CPICH exists in each cell and
- 4. P-CPICH is broadcast over the entire cell.

P-CPICH serves as a phase-reference for channel estimation for SCH, P-CCPCH, AICH and PICH. It can also be used for all other downlink channels.

S-CPICH has the following characteristics:

- 1. An arbitrary channelization code of SF = 256 is used for S-CPICH
- 2. Scrambling can be performed by either a Primary or a Secondary scrambling code
- 3. There may be zero, one or several S-CPICH per cell and
- 4. A S-CPICH may be transmitted over the entire cell or a part of the cell.

S-CPICH can serve as a reference to S-CCPCH and downlink DPCH. One of the main usages of the S-CPICH is as a phase-reference for channel estimation when adaptive ANT array is applied.

Primary Common Control Physical CHannel (P-CCPCH)

P-CCPCH is a fixed rate (30 kbps, SF = 256) downlink physical channel for transmitting BCH.

Figure 3.17 shows the frame configuration of P-CCPCH. It is different from downlink DPCH in that it does not transmit Pilot, TPC or TFCI. P-CCPCH is not transmitted during the first 256 chips of each slot. Instead, the SCH is transmitted during this period.

Secondary Common Control Physical CHannel (S-CCPCH)

S-CCPCH is a physical channel for transmitting FACH and PCH. There are two types of S-CCPCH: with TFCI and without TFCI. Figure 3.18 shows the frame structure of S-CCPCH.

The number of bits inside the downlink S-CCPCH frame is determined by the parameter k. k corresponds to SF of the physical channel: $SF = 256/2^k$. SF may have a value between 256 and 4.

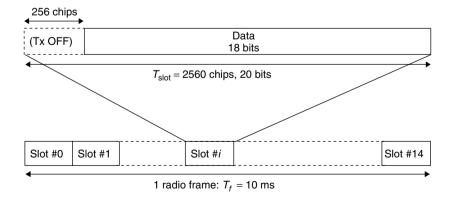


Figure 3.17 P-CCPCH frame structure

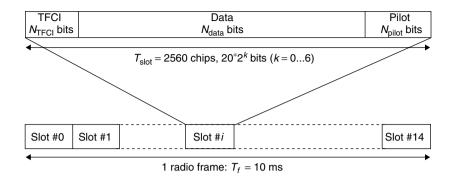


Figure 3.18 Frame structure of S-CCPCH

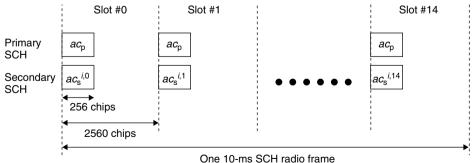
CCPCH is basically different from downlink DPCH in that it does not execute closedloop TPC. S-CCPCH basically differs from P-CCPCH in that the former can support variable bit rates using the TFCI field, whereas the latter is transmitted at a predetermined fixed rate.

Synchronization CHannel (SCH)

SCH is a downlink physical channel used for cell search. SCH consists of two subchannels: Primary SCH (P-SCH) and Secondary SCH (S-SCH). Figure 3.19 shows the frame structure of SCH.

P-SCH is used in step 1 of cell search, so that the UE can establish slot synchronization with the cell. P-SCH is spread by a 256-chip-long code called the Primary Synchronization Code (PSC). PSC referred to as c_p in Figure 3.19 is transmitted once in each slot. PSC is common to all cells in the system.

S-SCH is used in step 2 of cell search, so that the UE can establish frame synchronization with the cell, and find the scrambling code group to which the cell belongs. S-SCH is spread by a 256-chip-long modulated code called the Secondary Synchronization Code (SSC), which changes every 15 slots. There are 64 patterns of the 15-slot-cycle SSC, and the scrambling code group used in the same cell corresponds to the pattern at a one-to-one ratio. In Figure 3.19, SSC is referred to as $c_s^{i,k}$, in which *i* represents the scrambling code



One to-ms Son hadio hame

Figure 3.19 Frame structure of SCH

group number (1-64) and k stands for the slot number (0-14). S-SCH and P-SCH are transmitted simultaneously.

Physical Downlink Shared CHannel (PDSCH)

PDSCH is a physical channel for transmitting the DSCH, and is shared by multiple users. PDSCH is always used together with the associated Downlink DPCH.

Figure 3.20 shows the frame structure of PDSCH.

PDSCH and DPCH do not necessarily have to have the same SF. PDSCH may use a different SF in each frame.

Layer 1 control information of PDSCH is transmitted using the DPCCH part of the associated Downlink DPCH, that is, the PDSCH does not carry the L1 control information. The SF of PDSCH may take a value between 256 and 4.

Acquisition Indicator CHannel (AICH)

AICH is a downlink physical channel used for random access control, and transmits the Acquisition Indicator (AI) in the preamble of PRACH. Refer to Section 3.3.1.11 for random access control.

Acquisition Indicator AI_S corresponds to the signature S of PRACH.

Figure 3.21 shows the frame structure of AICH, which consists of a repeated sequence of 15 consecutive access slots. Each access slot consists of 40 bits. The first 32 bits are the AI part, and the remaining 8 bits are not transmitted.

In Figure 3.21, a_0, a_1, \ldots, a_{31} is determined by the following equation.

$$a_{j} = \sum_{s=0}^{15} AI_{s} b_{s,j}$$
(1)

In the equation, AI_s indicates the response to the preamble reception of Signature (S): ACK = +1, NACK = -1, and nonreception = 0. $b_{s,j}$ is the Signature pattern of AICH corresponding to the Signature(s) received in the preamble, and consists of 32 bits. There are 16 patterns according to the Signature of the preamble.

Paging Indicator CHannel (PICH)

PICH is a channel used for the purpose of reducing the rate of intermittent reception, to save the UE battery. PICH transmits a short PI to inform UE whether there are incoming

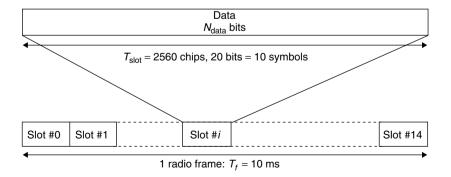


Figure 3.20 Frame structure of PDSCH

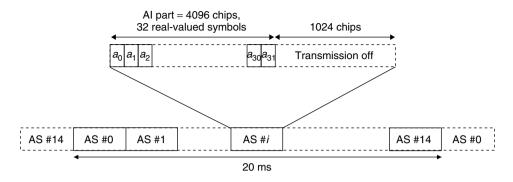


Figure 3.21 Frame structure of AICH

calls or not. UE in idle mode normally receives nothing but the PI. UE receives PCH in the radio frame of the S-CCPCH corresponding to the PI, only when it is informed of an incoming call by the PI. PIs are divided into several groups. As the frequency of call termination in each group can be reduced to an extremely low level, UE in idle mode only has to receive a short PI most of the time, which helps decrease the frequency of receiving long PCH.

PICH is always associated one-to-one with an S-CCPCH to which a PCH is mapped.

Figure 3.22 illustrates the frame structure of PICH. There are 300 bits in the 10 ms frame, of which 288 bits constitute several PI groups. The remaining 12 bits are unused and not transmitted.

The PI { P_0, \ldots, P_{Np-1} } that corresponds to the group number in N frames is transmitted in the respective PICH frames. The value of Np refers to the number of groups in one frame, which may be 18, 36, 72 or 144.

Table 3.4 shows the conversion from $\{P_0, \ldots, P_{Np-1}\}$ into PICH bits $\{b_0, b_1, \ldots, b_{287}\}$.

If a particular PI is set to 1, the UEs associated with this PI must read the corresponding frame of the associated S-CCPCH.

3.3.1.3 Mapping of Transport Channels onto Physical Channels

Figure 3.23 summarizes the mapping of transport channels over physical channels.

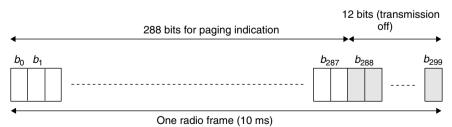


Figure 3.22 Structure of PICH

Number of paging indicators per frame (<i>Np</i>)	$P_q = 1$	$P_q = 0$
Np = 18	$\{b_{16q},\ldots,b_{16q+15}\}$	$\{b_{16q},\ldots,b_{16q+15}\}$
Np = 36	$= \{-1, -1, \dots, -1\}$ $\{b_{8q}, \dots, b_{8q+7}\}$	$= \{+1, +1, \dots, +1\}$ $\{b_{8q}, \dots, b_{8q+7}\}$
Np = 72	$= \{-1, -1, \dots, -1\}$ $\{b_{4q}, \dots, b_{4q+3}\}$	$= \{+1, +1, \dots, +1\}$ $\{b_{4q}, \dots, b_{4q+3}\}$
Np = 144	$= \{-1, -1, \dots, -1\}$ $\{b_{2q}, b_{2q+1}\} = \{-1, -1\}$	$= \{+1, +1, \dots, +1\}$ $\{b_{2q}, b_{2q+1}\} = \{+1, +1\}$

 Table 3.4
 Conversion from PI to PICH bit sequence

Transport channel	Physical channel	UL/DL
DCH ———	-Dedicated Physical Data CHannel•DPDCH•	UL and DL
	Dedicated Physical Control CHannel•DPCCH•	UL and DL
RACH ——	-Physical Random Access CHannel •PRACH•	UL
CPCH	-Physical Common Packet CHannel-PCPCH-	DL
	Common Pilot CHannel•CPICH•	DL
BCH ———	-Primary Common Control Physical CHannel P-CCPCH	DL
FACH	-Secondary Common Control Physical CHannel-S-CCPCH-	DL
PCH —		
	Supphropiostion CHannel SCH	DI
DSCH	Synchronisation CHannel•SCH•	
DSCH	-Physical Downlink Shared CHannel•PDSCH•	
	Acquisition Indication CHannel•AICH•	DL
	Page Indication CHannel•PICH•	DL

Figure 3.23 Possible transport-channel to physical-channel mapping

3.3.1.4 Transport Channel Multiplexing

Requirements of the next-generation mobile communications system include high-quality, multimedia services. Forward Error Correction (FEC) (channel coding) is an essential technology for high-quality transmissions. In particular, it is important to jointly use channel interleaving technology to fully appreciate the effects of FEC in a mobile communications environment in which burst errors often take place. Moreover, for multimedia services, multiple transport channels with various Qualities of Service (QoS) need to be multiplexed and transmitted over one physical channel. In order to meet these requirements, rate matching is applied (refer to Section 3.3.1.6). Also, Multistage InterLeaver (MIL) is used, which is a high-performance interleaver that takes multiplexing of transport channels into account.

Figure 3.24 illustrates FEC, interleaving and multiplexing schemes applied upon multiplexing multiple transport channels in uplink and downlink. FEC (Channel coding) and the 1st interleaving in one or more frames is performed for each transport channel. Then, frame segmentation is executed on each channel, followed by the multiplexing of transport channels. Subsequently, interleaving in frames is performed by the 2nd interleaver. In

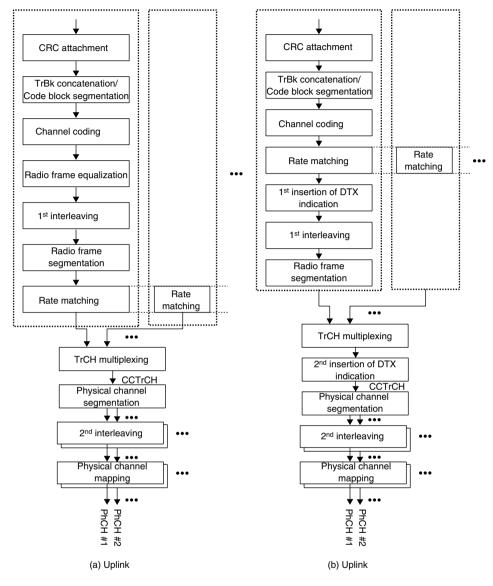


Figure 3.24 Transport channel multiplexing structure

uplink, the rate-matching process takes place after the 1st interleaving and radio frame segmentation, whereas in downlink, it takes place beforehand. This is because SF varies with each frame in the uplink, whereas it is constant in the downlink.

3.3.1.5 FEC (Channel Coding)

There are two types of coding schemes, namely, convolutional encoding and turbo encoding, which can be used according to QoS. It is also possible not to apply FEC. Because of the characteristics of the coding schemes, turbo encoding is effective for video and other high-speed, high-quality data (coding rate = 1/3, constraint length = 4), whereas convolutional encoding is effective for speech and other low-speed data. In convolutional encoding, a coding rate of either 1/2 or 1/3 (constraint length = 9 in both cases) is applied depending on QoS.

Figures 3.25 and 3.26 illustrate the configuration of a convolutional coder and a turbo coder, respectively.

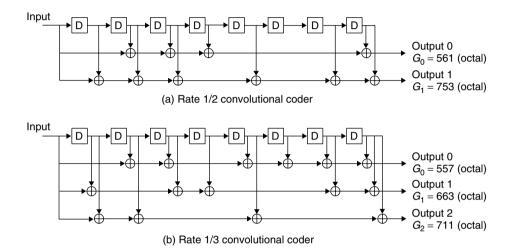


Figure 3.25 Configuration of convolutional coder

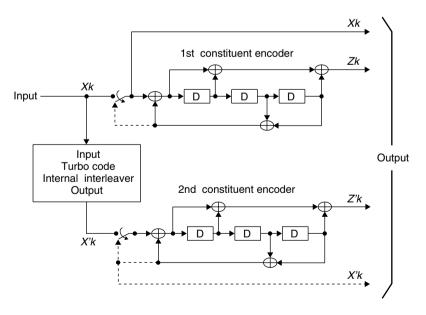


Figure 3.26 Configuration of turbo coder

Type of TrCH	Coding scheme	Coding rate
BCH PCH RACH	Convolutional coding	1/2
CPCH, DCH, DSCH, FACH	Turbo coding No coding	1/3, 1/2 1/3

 Table 3.5
 Transport channels and applicable forward error-correction schemes

Applicable coding schemes are linked to transport channels in consideration of the speed at which the data needs to be transmitted over the transport channels and the required quality. Table 3.5 shows the transport channels and the applicable FEC schemes.

3.3.1.6 Rate Matching

Rate matching is performed on bit sequences after channel coding, according to the number of multiplexed transport channels, and the transmission bit rate and QoS of each transport channel. Through the rate-matching process, more bits in the physical channel are allocated to transport channels with higher transmission bit rate and higher QoS relative to other transport channels.

For rate matching, either *puncturing* or *repetition* is applied. The former involves the removal of bits from the bit sequence at a fixed cycle, whereas the later involves the iterative insertion of bits into the bit sequence at a fixed cycle.

As a result of these operations, transport channels with various QoS can be multiplexed and transmitted over the physical channel as a bit sequence of uniform quality.

3.3.1.7 Interleaving

The interleaving process is divided into two parts: 1st interleaving, which takes place before the multiplexing of transport channels; and 2nd interleaving, which is executed after multiplexing. 1st interleaving is processed on each transport channel, and interleaving is carried out by frame. 2nd interleaving involves interleaving by bit in the frame. This makes it possible to deal flexibly with all sorts of transport channel multiplexing patterns and achieve high error-correction performance. 1st interleaving applies the same interleaving pattern to each interleaving size, whereas 2nd interleaving uses a universally common pattern to minimize the processing load and thereby decrease the scale of hardware and reduce power consumption.

3.3.1.8 Spreading and Modulation

Uplink Spreading and Modulation Process

Figure 3.27 illustrates the basic principles of the spreading process for DPCCH and DPDCH. The binary DPCCH and DPDCH to be spread are represented by a real-valued

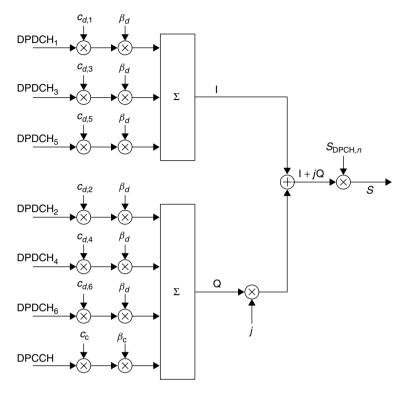


Figure 3.27 Spreading for uplink DPCCH/DPDCH

sequence. (In other words, the binary value "0" is mapped to the real value "+1", while the binary value "1" is mapped to the real value "-1".) DPCCH is spread by the channelization code c_c . In uplink, more than one DPDCH can be set only if SF = 4. Assuming that the *n*th DPDCH is DPDCH_n, DPDCH_n is spread by channelization code $c_{d,n}$. One DPCCH and up to 6 DPDCHs can be transmitted simultaneously (Hence, $0 \le n \le 6$).

In the figure, β is the gain factor, which refers to the weight coefficient corresponding to the ratio of transmission power of DPDCH to DPCCH. $\beta = 1.0$ corresponds to the instantaneous maximum transmission power in the set DPCCH, or one or more DPDCHs. The value of β is specified by 4 bits.

Figure 3.28 illustrates the basic principles of the spreading process of the PRACH message part, which consists of two components, namely, the data part and the control information part. The control data part is spread up to the chip rate by channelization code c_c , whereas the real data part is spread by channelization code c_d . It is weighted according to the transmission power ratio by coefficient β as in the case of DPCCH and DPDCH.

The chip sequence represented by the complex value generated through the spreading process is QPSK modulated as shown in Figure 3.29.

Downlink Spreading and Modulation Process

All downlink physical channels apart from SCH undergo the spreading process based on the circuit shown in Figure 3.30. The symbol of physical channels before spreading

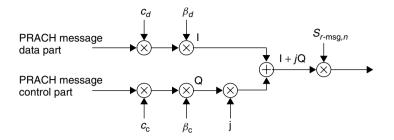


Figure 3.28 Spreading for PRACH message part

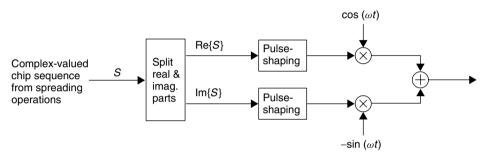


Figure 3.29 Uplink modulation process

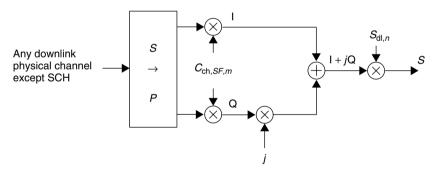


Figure 3.30 Spreading for downlink physical channels other than SCH

is a real-valued sequence. Symbols of all physical channels may have a value of +1, -1 or 0, excluding AICH. [0 stands for Discontinuous Transmission (DTX), that is, transmission-off]. The value of the symbol of AICH depends on the combination of the AIs transmitted.

Two successive symbols are at first converted from serial to parallel, and mapped to I and Q branches. Even-number-sequenced symbols are mapped to I phase, whereas odd-number-sequenced symbols are mapped to Q phase. Subsequently, both I and Q branches are spread up to the chip rate by the same channelization code, $C_{ch,SF,m}$. The two real-valued chip sequences of I and Q branches are treated as one complex-valued sequence, and is randomized by the complex-valued scrambling code $S_{dl,n}$.

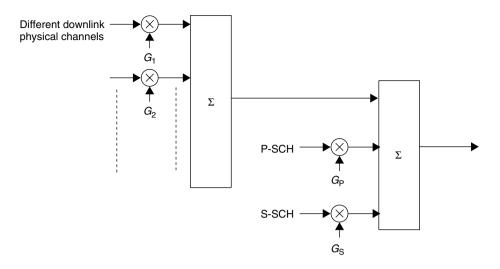


Figure 3.31 Combining of downlink physical channels

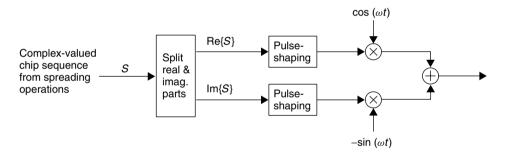


Figure 3.32 Downlink modulation process

Figure 3.31 illustrates the method of combining multiple downlink channels. In Figure 3.31, complex-valued sequences corresponding to *S* are weighted by weight coefficient G_i . Complex-valued P-SCH and S-SCH are weighted by G_p and G_s , respectively. In this manner, all downlink physical channels are combined through the summation of complex-valued chips.

The complex-valued chip sequence generated as a result of the spreading process is subject to QPSK modulation as shown in Figure 3.32.

Spreading Codes

The following is a brief description of the types of spreading codes in the W-CDMA system, and the way in which they are applied. There are two types of codes as such: channelization codes and scrambling codes. Scrambling codes are relatively long spreading codes – they are based on 38,400-chip-long codes (long scrambling codes) or 256-chip-long codes (short scrambling codes). An extremely large number of scrambling codes can be used. Channelization codes are short spreading codes with a chip length between 4 and

512; and 4 to 512 types of codes can be used depending on the length. In the spreading process, transmission data is spread by scrambling codes and channelization codes.

These codes are used in a different manner between uplink and downlink. First, the basic way in which scrambling codes are used must be understood. In uplink, a scrambling code is assigned to each UE, and the BS identifies the UEs according to their scrambling codes. In downlink, a different scrambling code is assigned to each sector, and each UE identifies the sector by executing despreading with the use of the scrambling code used in the visited sector.

On the other hand, channelization codes are basically applied in the following manner. In uplink, each UE uses a channelization code to identify physical channels. Multiple UEs can share the same channelization code, because UEs are identified by BS with their respective scrambling codes as mentioned above. In downlink, channelization codes are used for identifying physical channels in the same sector. Sectors can share the same channelization code, as a different scrambling code is assigned to each sector.

(1) Uplink Channelization Code

The channelization code is OVSF, which is a code that assures orthogonality between codes, regardless of whether they share the same SF or not. The use of this code for spreading the physical channel enables the elimination of interference components arising from multiple physical channels, which helps increase capacity. OVSF codes are defined on the basis of the code tree referred to in Figure 3.34.

In Figure 3.33, channelization code is represented by $C_{ch,SF,k}$. SF refers to the spreading factor, and k represents the code number. The length of the code and SF are determined by the number of rows in the code tree. Figure 3.34 shows how channelization codes are generated.

The following restrictions apply to the assignment of channelization codes to DPCCH and DPDCH:

- 1. DPCCH is always spread by code $c_c = C_{ch,256,0}$.
- 2. When there is only one DPDCH, DPDCH₁ is spread by $c_{d,1} = C_{ch,SF,k}$. SF refers to the spreading factor of DPDCH₁, while k = SF/4.

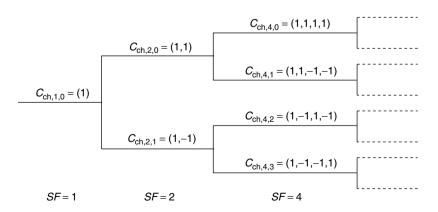


Figure 3.33 OVSF code tree

$$\begin{split} & C_{\mathrm{ch},1,0} = 1, \\ & \begin{bmatrix} C_{\mathrm{ch},2,0} \\ C_{\mathrm{ch},2,1} \end{bmatrix} = \begin{bmatrix} C_{\mathrm{ch},1,0} & C_{\mathrm{ch},1,0} \\ C_{\mathrm{ch},1,0} & -C_{\mathrm{ch},1,0} \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \\ & \begin{bmatrix} C_{\mathrm{ch},2}^{(n+1)}, 0 \\ C_{\mathrm{ch},2}^{(n+1)}, 2 \\ C_{\mathrm{ch},2}^{(n+1)}, 2 \\ C_{\mathrm{ch},2}^{(n+1)}, 2 \\ \vdots \\ C_{\mathrm{ch},2}^{(n+1)}, 2^{(n+1)} - 2 \\ C_{\mathrm{ch},2}^{(n+1)}, 2^{(n+1)} - 1 \end{bmatrix} = \begin{bmatrix} C_{\mathrm{ch},2^n,0} & C_{\mathrm{ch},2^n,0} \\ C_{\mathrm{ch},2^n,0} & -C_{\mathrm{ch},2^n,0} \\ C_{\mathrm{ch},2^n,1} & C_{\mathrm{ch},2^n,1} \\ C_{\mathrm{ch},2^n,1} & -C_{\mathrm{ch},2^n,1} \\ C_{\mathrm{ch},2^n,1} & -C_{\mathrm{ch},2^n,1} \\ C_{\mathrm{ch},2^n,2^n,-1} & C_{\mathrm{ch},2^n,2^n,-1} \\ C_{\mathrm{ch},2^n,2^n,-1} & -C_{\mathrm{ch},2^n,2^n,-1} \\ \end{bmatrix} \end{split}$$

Figure 3.34 Channelization code generation method

3. In uplink, DPDCH multicode transmission is permissible only when SF = 4. Put differently, when more than one DPDCH is to be transmitted, SF of all DPDCHs is 4. DPDCH_n is spread by $C_{d,n.} = C_{ch,4,k}$. If n = 1 or 2, k = 1; if n = 3 or 4, k = 2; and if n = 5 or 6, k = 3.

(2) Downlink Channelization Code

The downlink channelization code is the same OVSF code as the one used in the uplink physical channel.

The channelization codes used for P-CPICH is fixed at $C_{ch,256,0}$. The channelization code used for P-CCPCH is fixed at $C_{ch,256,1}$. Codes used for other physical channels are specified by the higher layer.

In the event of migrating to compressed mode by halving SF (refer to Section 3.3.1.13), the OVSF code used in the compressed frame is compliant with the following rules:

- 1. Code of $C_{ch,SF/2|n/2|}$ if a normal scrambling code is to be used.
- 2. Code of $C_{ch,SF/2,n \mod SF/2}$ if a scrambling code for compressed mode is to be used.

Here, $C_{ch,SF,n}$ refers to a code before the application of compressed mode.

(3) Uplink Scrambling Code

The chip pattern of all uplink physical channels are randomized by a scrambling code. A scrambling code is a complex-valued sequence. There are two type of scrambling codes: long scrambling codes and short scrambling codes. Short scrambling codes are designed to streamline the reception process at the BS upon the application of an uplink interference canceller. There are 2^{24} codes (16,777,216 codes) in both scrambling codes. Long scrambling codes are part of the Gold sequence, which has relatively good cross-correlation and autocorrelation properties. Counting from the beginning of the Gold sequence, it is 38,400-chips long. Short scrambling codes are 256-chips long, complex-valued sequences. UE is informed by the higher layer as to which scrambling code should be used.

The long scrambling code is a complex sequence, and is generated from two Gold sequences that are 2^{24} long, $C_{\log,1,n}$ and $C_{\log,2,n}$. $C_{\log,2,n}$ is generated by shifting

 $C_{\log,1,n}$ by 16,777,232 chips. HPSK is applied to the uplink spreading process. HPSK is a spreading phase shift keying scheme that reduces the incidence of 180° phase changes and reduces nonlinear distortion by repeating QPSK and $\pi/2$ BPSK alternately at each chip timing. In order to achieve this, long scrambling codes – which are complex sequences – are generated on the basis of $C_{\log,1,n}$ and $C_{\log,2,n}$ according to the following equation.

$$C_{\log,n}(i) = c_{\log,1,n}(i)(1+j(-1)^{i}c_{\log,2,n}(2\lfloor i/2 \rfloor))$$
(2)

Either long scrambling codes or short scrambling codes are applied to uplink DPCCH and DPDCH.

Scrambling codes used for the PRACH message part are long scrambling codes that are 10 ms-long (38,400 chips), each of which are unique to each cell. They are set at a one-to-one ratio with the scrambling codes used for the preamble part.

Codes used for the preamble of PRACH are complex sequences. They are generated from $S_{r-pre,n}$ and the preamble signature $C_{sig,s}$ as follows.

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

 $S_{r-\text{pre},n}$ is the sequence starting from the 0th chip and ending at the 4095th chip of $C_{\text{long},1,n}$.

 $C_{\text{sig},s}$ is a sequence corresponding to signature *s* used for random access control. Specifically, it is a 4096-chips-long sequence formed by repeating the 16-chips-long signature pattern $P_s(n)$ 256 times. Signature pattern $P_s(n)$ is a 16 Hadamard code, and is an orthogonal sequence. This enables the accurate determination of signatures upon the detection of preamble at the BS.

(4) Downlink Scrambling Code

Long scrambling codes are complex sequences generated from Gold sequence Z_n with a sequence length of 2^{18} . (*n* is the scrambling code number, linked to the Gold sequence generation method.) Specifically, the real number part is Z_n , and the imaginary number part is a sequence generated by shifting Z_n by 131,072 chips. In total, $2^{18} - 1 = 262,143$ scrambling codes can be generated. However, not all of the scrambling codes are actually used. The scrambling codes are divided into 512 code groups. Each group consists of one primary scrambling code and 15 secondary scrambling codes.

A primary scrambling code is a code of $n = 16 \times i$ (in which $n = 16 \times i$, i = 0 - 511), and a secondary scrambling code is a code of $n = 16 \times i + k$ (in which k = 1 - 15).

Primary scrambling codes and secondary scrambling codes are linked to each other. In other words, the *i*th primary scrambling code corresponds to the *i*th set of secondary scrambling codes.

On the basis of the above explanation, only 8192 scrambling codes are used (k = 0-8191). Each scrambling code is linked to a left alternative scrambling code and a right alternative scrambling code. These alternative codes are scrambling codes used in compressed mode. The left alternative scrambling code number corresponding to scrambling code number k is k + 8192, whereas the right alternative scrambling code is used when n < SF/2, whereas the left alternative scrambling code is used when n < SF/2. $c_{ch,SF,n}$ is the channelization code before the activation of compressed mode.

Primary scrambling codes, which total 512 in number, are divided into 64 scrambling code groups. Each scrambling code group consists of 8 primary scrambling codes. The *j*th scrambling code group is composed of a code with a primary scrambling code number $16 \times 8 \times j + 16 \times k$ (in which 0 < j < 63, 0 < k < 7).

One primary scrambling code is assigned to each cell. P-CCPCH and P-CPICH are always spread by a primary scrambling code. Other physical channels may be transmitted using either secondary scrambling codes – which are paired with a primary scrambling code assigned to each cell – or the primary scrambling code.

(5) Synchronization Code

SCs are used for spreading modulation of SCH. Two types of such codes are used, namely, Primary Synchronization Codes (PSC, c_p referred to in the section on SCH in Section 3.3.1.2) and Secondary Synchronization Codes (SSC, $c_s^{i,x}$ referred to in the section on SCH in Section 3.3.1.2), which are used for spreading P-SCH and S-SCH, respectively. PSC consists of a generalized hierarchical Golay sequence. Codes with superior autocorrelation properties are chosen from the sequence and used.

SSC is based on the Hadamard sequence; 16 types of such sequences are used. The sequence for S-SCH is generated by joining 15 SSCs together, and 64 types of such sequences are used. The 64 types of S-SCH sequences correspond to 64 scrambling code groups at a one-to-one ratio, and this relationship is used to improve the cell search properties.

3.3.1.9 TPC

In DS-CDMA, each channel engaged in communication suffers from Multiple Access Interference (MAI), which is caused by communication channels other than the user's, and multipath interference, which results from the user's own communication channel. In the W-CDMA system, such interference limits the subscriber capacity. This means that the radio link capacity can be increased by minimizing the power for transmitting each channel without sacrificing the required quality. The TPC scheme in the W-CDMA system is designed in view of increasing the radio link capacity, as well as saving the battery. TPC used in W-CDMA can be broadly divided into two groups: open-loop TPC and closed-loop TPC.

Open-Loop TPC

UE estimates the downlink propagation loss and determines the uplink transmission power on the basis of the estimate using the downlink Common Control CHannel (CCCH). In dedicated channels to which closed-loop TPC is applied, the initial transmission power is normally decided by open-loop TPC. In particular, closed-loop TPC cannot be applied to the uplink CCCH because it is not a channel in which uplink and downlink are used in pairs; therefore, open-loop TPC is used.

Closed-Loop TPC

Figure 3.35 depicts the concept of closed-loop fast TPC for uplink and downlink. In closed-loop TPC, the quality of the communication channel is measured at the point of reception, and on the basis of the measurement results, TPC bits are transmitted using the

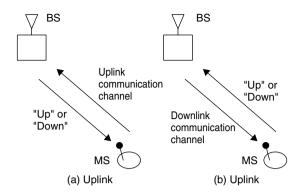


Figure 3.35 Conceptual diagram of closed-loop TPC

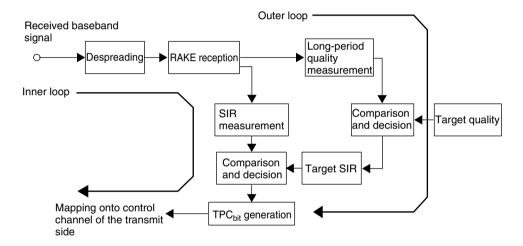


Figure 3.36 Conceptual diagram of TPC

loop-back channel (DPCCH) so that the required quality of the receiving communication channel would be satisfied.

Figure 3.36 illustrates the configuration of the reception of TPC applied to the BS and UE in the W-CDMA system. As shown in the figure, closed-loop TPC consists of two-step loops: (1) inner-loop control and (2) outer-loop control.

(1) Inner-Loop TPC

Under inner-loop TPC in the uplink (downlink) communication channel, BS (UE) measures the received Signal-to-Interference Ratio (SIR), compares it with the target SIR and, as TPC bits, sends an "UP" command if it is below the target SIR or a "DOWN" command if it is above the target SIR. UE (BS) receives the TPC bits and changes the transmission power by 1 dB according to the decoding results. Such closed-loop control is performed at a slot cycle of 0.667 ms.

(2) Outer-Loop TPC

In contrast with inner-loop TPC, which controls the SIR of the received communication channel so that it would reach the target value, outer-loop TPC controls the target SIR so that the Bit Error Rate (BER) and BLock Error Rate (BLER) would meet the target values. This involves the measurement of the communication quality over a relatively long period (between a few hundred milliseconds and a couple of seconds), and setting the target SIR at an adequate level to achieve the target quality.

3.3.1.10 Site Selection Diversity Transmit Power Control (SSDT)

SSDT is an optional power control method applied at the time of soft handover. UE regards one cell among the cells engaged in Diversity HandOver (DHO) as the "primary cell" and all other cells as "nonprimary cells". The main objective of SSDT is to prevent the amount of downlink interference from increasing by executing downlink transmission only from the primary cell. Another objective is to perform high-speed cell selection without increasing the load on the network.

For the purpose of selecting the primary cell, a temporary ID is assigned to each cell, and the UE periodically reports the primary cell to the cells engaged in DHO. The nonprimary cells switch off the transmission of DPDCH. The ID of the primary cell is transmitted using the uplink FBI field. Activation, termination, assignment and other tasks associated with SSDT are executed on the basis of signaling from the higher layer.

Definition of Temporary Cell ID

Each cell is given a temporary ID during the execution of SSDT. The ID is assigned as a binary bit sequence, and in terms of length, three types of IDs are applicable: "long", "medium" and "short". The length of the ID is specified by the network. When 1 bit is to be assigned in the uplink FBI, their respective lengths are 15 [slots], 7 to 8 [slots] and 3 [slots]; when 2 bits are to be assigned in the uplink FBI, their respective lengths are 7 to 8 [slots] and 3 to 4 [slots] and 3 [slots]. A shorter ID enables faster tracking of channel fluctuations for site diversity, whereas a longer ID helps prevent deterioration in performance caused by ID reception errors in each cell. Table 3.6 shows the update cycle of the primary cell ID.

Selection of Primary Cell

UE measures the Received Signal Code Power (RSCP) of CPICH in each cell engaged in DHO and determines the primary cell. The cell with the largest RSCP of CPICH is chosen as the primary cell.

Code length	The number of FBI bits per 1	slot assigned for SSDT 2
"Long" "Medium" "Short"	1 update per frame 2 updates per frame 3 updates per frame	2 updates per frame4 updates per frame5 updates per frame

Table 3.6 Primary cell update cycle

Transmission and Recognition of Primary Cell ID

UE periodically transmits the primary cell ID using the *S* field in FBI. BS recognizes its own cell as a nonprimary cell when the following conditions are met:

- 1. The received ID of the primary cell is different from the ID of its cell;
- 2. The quality of the uplink reception signal exceeds the threshold set by the network;
- 3. The quality of uplink reception signal does not have an excessive level in compressed mode and
- 4. The puncture level of the permissible ID is $N_{\rm ID}/3$ symbol.

If these conditions are not satisfied, the cell is regarded a primary cell.

TPC Operation in BS and MS

If BS determines that its own cell is a primary cell, it performs the normal TPC operations. If it decides that its own cell is a nonprimary cell, it turns off the transmission of DPDCH (normal operation of DPCCH is performed).

MS decides the TPC bits to be transmitted uplink on the basis of the received SIR of DPCCH from the primary cell. Uplink TPC is performed in the same manner as at the time of normal DHO.

3.3.1.11 Random Access Control

Overview of Random Access Control

Preamble power ramping is applied to random access in W-CDMA. Preamble is a short signal that is sent before the transmission of the RACH message, and is spread by a prescribed spreading code. The preamble can be easily detected by using a simple Matched Filter (MF) in BS. BS can know the reception timing of the following message part and the used scrambling code beforehand by receiving the preamble in advance, which helps reduce the load on the message-part reception process and improve the reception performance of BS.

Furthermore, the adverse impact of interference to other users caused by control errors in open-loop TPC can be reduced through power ramping using the preamble. Specifically, UE repeatedly transmits the preamble until it receives the AI on AICH, which indicates the detection of the preamble by BS, and gradually increases the transmission power every time the preamble is sent. UE stops the transmission of the preamble once it receives the AI, and sends the message part at the level of power equal to the preamble transmission power at that point.

Random Access Transmission

Random access transmission consists of one or more preamble(s), and a message that is either 10 ms or 20 ms. Figure 3.37 shows this arrangement.



Figure 3.37 Structure of random access transmission

The preamble is 4096-chips long, and consists of a signal sequence based on the iteration of a 16-chips-long signature 256 times. The signature consists of Hadamard codes of length 16, and there are 16 different types of signatures.

Random Access Subchannel

The timing at which the UE can send the preamble is divided by random access subchannels. A random access subchannel is a subset comprising the combination of all uplink access slots. There are 12 random access subchannels in total. Random access subchannel #I (I = 0, 1, ..., 11) consists of the access slots referred to in Table 3.7.

Random Access Control Procedures

Random access control is initiated after receiving the following information from RRC:

- Scrambling code for preamble;
- Message length (10 ms or 20 ms);
- AICH transmission timing $(0 \text{ or } 1)^1$;
- Available signature set and available random access subchannel for each access service class (ASC);
- Step width of power ramping: Power_Ramp_Step;
- Maximum number of preamble retransmission attempts: Preamble_Retrans_Max;
- Initial transmission power of preamble: Preamble_Initial_Power;
- Power offset, which is the ratio of power of the message part (control channel only) to the preamble part and
- Set of transport format parameters.

SFN modulo 8 of					Sı	ıbchan	nel nu	mber				
corresponding P-CCPCH frame	0	1	2	3	4	5	6	7	8	9	10	11
0	0	1	2	3	4	5	6	7				
1	1	1	1						8	9	1	1
	2	3	4								0	1
2				0	1	2	3	4	5	6	7	
3	9	1	1	1	1	1						8
		0	1	2	3	4						
4	6	7					0	1	2	3	4	5
5			8	9	1	1	1	1	1			
					0	1	2	3	4			
6	3	4	5	6	7					0	1	2
7						8	9	1	1	1	1	1
								0	1	2	3	4

 Table 3.7
 Access slots that can be used in each random access subchannel

¹ A different AICH transmission timing is used according to cell size. Normally, AICH transmission timing is 0. If the cell size is large, AICH transmission timing is 1.

On the other hand, the following information is received from MAC:

- Transport format used for the message part of PRACH;
- Access service class of PRACH transmission and
- Transmission data (Transport Block (TB) set).

Random access control is performed according to the procedures below:

- 1. In the random access subchannel that can be used for the ASC concerned, one access slot is chosen randomly from access slots that can be used in the next full access slot sets². If there are no access slots available, one access slot is chosen randomly from access slots that can be used in the next full access slot sets.
- 2. One signature is randomly chosen from the set of available signatures within the given ASC.
- 3. The preamble retransmission counter is set at Preamble_Retrans_Max, which is the maximum number of preamble retransmission attempts.
- 4. The preamble transmission power is set at Preamble_Initial_Power, which is the initial transmission power of the preamble.
- 5. The preamble is transmitted on the basis of the chosen uplink access slot, signature and set transmission power.
- 6. If no "ACK" or "NACK" corresponding to the selected signature is detected in the downlink access slot corresponding to the selected uplink access slot.
 - The next available access slot is selected from the random access subchannel within the given ASC.
 - A new signature is randomly selected from the available signatures within the given ASC.
 - The preamble transmission power is increased by Power_Ramp_Step, which is the step width of the power ramping.
 - The preamble retransmission counter is reduced by 1.
 - The procedures from step 5 are repeated for the duration in which the preamble retransmission counter exceeds 0. When the retransmission counter reads 0, the higher layer (MAC) is informed of the fact that "ACK" has not been received on AICH, and the random access control procedures in the physical layer are finished.
- 7. If "NACK" corresponding to the selected signature is detected in the downlink access slot concerned, the higher layer (MAC) is informed of the fact that "NACK" has been received on AICH, and the random access control procedures in the physical layer is finished.
- 8. The random access message is transmitted 3 or 4 uplink access slots³ after the uplink access slot of the last transmitted preamble depending on the AICH transmission timing parameter. The transmission power of the control channel of the random access message is set at a level higher than the transmission power of the last preamble transmitted by power offset.
- 9. The higher layer is informed of the transmission of the random access message, and the random access control procedures in the physical layer are finished.

 $^{^{2}}$ PRACH is based on a combination of two access slots. Access slot set 1 consists of PRACH access slots 0–7, and access slot set 2 consists of PRACH access slots 8–14.

 $^{^{3}}$ If the AICH transmission timing is 0 and 1, it is sent 3 and 4 access slots after the last preamble access slot transmitted, respectively.

Random Access Transmission Timing

Downlink AICH is divided by the spacing of downlink access slots. Each access slot is 5120 chips, with the same timing as P-CCPCH. Similarly, uplink PRACH is divided according to the spacing of uplink access slots. Uplink access slot number *n* is sent from UE at T_{p-a} before the reception of downlink access slot number *n* (n = 0, 1, ..., 14). Downlink AICH is transmitted only at the time of commencing the downlink access slots. Likewise, the random access preamble part and message part are transmitted only at the time of commencing uplink access slots. Figure 3.38 illustrates the relationship between PRACH and AICH transmission timings.

Preamble spacing T_{p-p} is no smaller than the minimum spacing of preamble $T_{p-p,min}$. The spacing between the preamble and transmission acknowledgement T_{p-a} and the spacing between the preamble and the message-part T_{p-m} are defined as follows.

If the AICH transmission timing is 0,

 $T_{p-p,min} = 15,360$ chips (3 access slots) $T_{p-a} = 7680$ chips $T_{p-m} = 15,360$ chips (3 access slots)

If the AICH transmission timing is 1,

$$T_{p-p,min} = 20,480$$
 chips (4 access slots)
 $T_{p-a} = 12,800$ chips
 $T_{p-m} = 20,480$ chips (4 access slots)

AICH transmission timing is set at 1 in cases in which the cell radius is large and the propagation delay is substantial.

3.3.1.12 Transmission Diversity

Transmission diversity is a diversity technology for executing transmission with the use of two ANTs. Normally, transmission is carried out using two BS ANTs for uplink reception diversity.

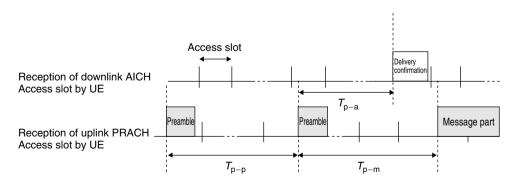


Figure 3.38 Relationship between PRACH and AICH transmission timings

Transmission diversity is designed to generate a higher gain in the UE (receiver) by manipulating the amplitude, phase and symbol pattern of the two ANTs.

Transmission diversity can be broadly divided into two groups: open-loop mode, which operates on the basis of a predetermined pattern; and closed-loop mode, in which UE (receiver) specifies the transmission pattern using the opposite link (uplink).

The W-CDMA system applies transmission diversity technology on the BS side. Table 3.8 shows the types and characteristics of transmission diversity technologies applied in W-CDMA.

Time-Switched Transmit Diversity (TSTD)

TSTD is the most elementary transmission diversity technology, which generates diversity effects by switching the transmission ANTs every slot. TSTD is applied only to SCH in the W-CDMA system.

Space-Time Block Coding Based Transmit Antenna Diversity (STTD)

STTD is a diversity technology that enables Maximal Ratio Combining (MRC) of signals from two ANTs by manipulating the symbol pattern of ANT 2. FEC, rate-matching and interleaving tasks are performed in the same manner as in the case of no STTD.

Figure 3.39 illustrates the encoding and decoding methods of STTD. In the figure, α_1 and α_2 refer to the fading vectors of the propagation path from ANTs 1 and 2, respectively.

Category	Type of transmission diversity	Overview	Characteristics
Open loop mode	Time-Switched Transmit Diversity (TSTD)	Transmission antennas are switched for each slot	-
	Space-Time Block Based Transmit Antenna Diversity (STTD)	The symbol pattern of antenna 2 is manipulated so as to achieve Maximal Ratio Combining (MRC) diversity on signals from two antennas.	Large effect on common CH
Closed loop mode	Closed loop mode1	The phase (4 patterns) is manipulated so as to achieve the maximum gain.	Applicable only to DPCH
	Closed loop mode2	The amplitude (2 patterns) and the phase (8 patterns) are manipulated so as to achieve the maximum gain.	Applicable only to DPCH

 Table 3.8
 Types of transmission diversity

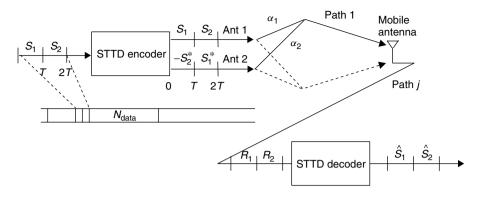


Figure 3.39 STTD encoding and decoding methods

As illustrated in the figure, the STTD encoder time inverts 2 symbols as a pair to the output of ANT 2, and reverses the polarity of the odd-number symbol as a conjugate complex. At the receiving side, the value received is as follows (ignoring the impact of noise and interference).

$$R_1 = \alpha_1 S_1 - \alpha_2 S_2^*$$
$$R_2 = \alpha_1 S_2 - \alpha_2 S_1^*$$

The output is determined as follows using the STTD decoder.

Output
$$1 = \alpha_1^* R_1 + \alpha_2 R_2^* = (|\alpha_1|^2 + |\alpha_2|^2) S_1$$

Output $2 = \alpha_2^* R_2 + \alpha_2 R_2^* = (|\alpha_1|^2 + |\alpha_2|^2) S_2$

The equations above make it possible to perform MRC on fading vectors α_1 and α_2 for each symbol.

Closed-Loop Mode

Closed-loop mode is a diversity scheme in which the UE selects the phase of the transmission signal symbol from both ANTs (the phase and the amplitude in the case of mode 2), and then controls it using FBI mapped in uplink DPCCH in a closed loop so as to increase the reception power on the UE side. The number of patterns of the phase and amplitude that can be chosen by the UE differs between modes 1 and 2. (Mode 1 = 4patterns; mode 2 = 16 patterns.) As more patterns can be chosen in mode 2, the gain due to controlling the phase and the amplitude is larger. However, it is more difficult to generate the diversity effect because control tends to lag behind in a high-speed moving environment, as more time is required to specify one phase and amplitude.

Figure 3.40 shows the transmitter block configuration of closed-loop transmission diversity applied to DPCH. Channel coding, interleaving and spreading are carried out as in no transmission diversity mode. The complex signal after spreading is multiplied by complex weight factor w_1 and w_2 to control the phase or amplitude of both ANTs. Weight factors w_1 and w_2 are chosen by the UE, and are signaled to the BS using D-bits in the FBI field of uplink DPCCH.

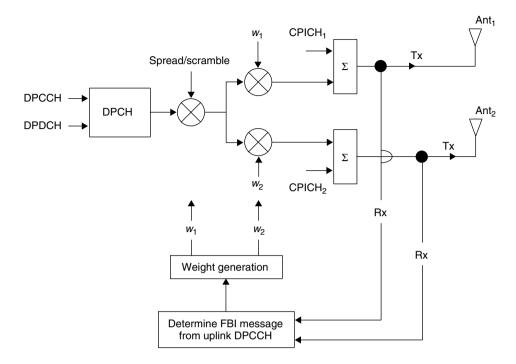


Figure 3.40 Configuration of transmitter with transmission diversity

 Table 3.9
 Basic specifications of closed loop transmission diversity

Mode	$N_{ m FBD}^{ m a}$	$N_{ m W}^{ m b}$	Update rate	Feedback bit rate	$N_{\rm po}^{\rm c}$	$N_{\rm ph}^{\rm d}$
1	1	1	1500 Hz	1500 bps	0	1
2	1	4	1500 Hz	1500 bps	1	3

^a N_{FBD} : Number of FBI bits in slot.

 ${}^{b}N_{W}$: FB signal message length.

^cN_{po}: Number of phase bits in 1 FB signal message.

^dN_{ph}: Number of amplitude bits in 1 FB signal message.

Table 3.9 shows the basic specifications of closed-loop transmission diversity modes 1 and 2.

(1) FBI Determination Method

UE estimates the propagation path from the two transmission ANTs using CPICH, selects the combination of weight vector $w = (w_1, w_2)$ so that the reception power would be maximized, and determines the corresponding Feedback Signaling Message (FSM).

After the determination of FSM, UE maps it in the *D* field in FBI of uplink DPCCH and sends it to the BS. The length of the FSM is $N_{\rm w} = N_{\rm po} + N_{\rm ph}$. Figure 3.41 shows the format of FSM. In order, transmission starts from MSB (Most Significant Bit). The

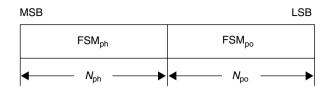


Figure 3.41 Format of Feedback Signaling Message (FSM)

sub-fields of FSM_{po} and FSM_{ph} are used for information on transmission power and transmission phase, respectively.

BS updates the weight vector w according to the received FSM. Updating starts from the downlink DPCCH pilot field. Updating is delayed by 1 or 2 slots in association with the cell radius, which is notified by the higher layer to the UE. In mode 1, the notified delay information is used upon ANT verification by UE, as described later [refer to Section 3.3.1.12 (2)].

(2) Closed-Loop Mode 1

The weight vector in closed-loop mode 1 is represented by the equation below, and may follow one of the 4 patterns.

$$(w_1, w_2) = (1, \exp(j\phi))$$
 $\phi \in \{1/4\pi, 3/4\pi, 5/4\pi, 7/4\pi\}$

Upon the selection of the vector, UE selects $\phi = 0$ or π in even-numbered slots and $\phi = 1/2\pi$ or $3/2\pi$ in odd-numbered slots, and sends them as feedback commands. BS averages out the phase information acquired from the immediately preceding 2 bits upon reflecting the feedback commands received, and determines one vector from the four patterns referred to above.

Antenna Verification in Mode 1

In mode 1, if UE can estimate the weight vector generated at the transmitting side, it can apply CPICH to channel estimation upon RAKE combining, as in the case of the nonimplementation of closed-loop transmission diversity. To achieve this, UE estimates the weight vector generated at the BS, using (1) FBI transmitted from the UE itself; (2) the estimated error rate of FBI at BS and (3) the received dedicated pilot signals. If the estimation accuracy is poor, RAKE combining would be executed on the basis of a wrong channel estimation value, which may severely undermine the reception characteristics.

(3) Closed-Loop Mode 2

The weight vector in closed-loop mode 2 is represented by the following equation. Unlike closed-loop mode 1, not only the phase but also the amplitude is subject to control.

$$(w_1, w_2) = \left[\sqrt{p_1}, \sqrt{p_2} \exp(j\phi)\right]$$

The chosen combination of transmission power of each ANT p_1 and p_2 is mapped to the aforementioned FSM_{po}, and the selected phase difference ϕ is mapped to FSM_{ph}.

0.8

0.2

Table 3.10	Combination	of	FSM _{po}	and
transmission	power p_1 and	p_2		
FSM	p_1			p_2

0.2

0.8

Phase difference ϕ (degrees)
180
-135
-90
-45
0
45
90
135

Table 3.11 FSM_{ph} and phase difference ϕ

0

1

Table 3.10 shows the combination of p_1 and p_2 relative to FSM_{po}, and Table 3.11 depicts ϕ relative to FSM_{ph}.

As shown above, in mode 2, FSM is 4 bits, and 4 slots form 1 FSM. In order to improve the trackability of rapid fluctuations in the propagation path, mode 2 updates FSM stepby-step. Figure 3.42 illustrates the framework. In the figure, $b_i (0 \le i \le 3)$ refers to each bit in FSM, and *m* indicates the FSM number in one frame. (*m* assumes a value between 0 and 3 because 1 frame consists of 15 slots.) Upon the transmission of the first bit of FSM, UE selects the optimal FSM from the 16 patterns and transmits it from the MSB side. Upon the transmission of FSM over 4 slots, UE gradually updates the options. Specifically, the following updating method is applied.

Definition

The 4 bits of FSM to be transmitted from slot #k to slot #k + 3 are defined as $\{b_3(k)b_2(k) b_1(k)b_0(k+3)\}$ (in which k = 0, 4, 8 or 12). The power calculated upon the selection of FSM is defined as $p(\{x_3, x_2, x_1, x_0\})$ (in which $\{x_3, x_2, x_1, x_0\}$ refers to the 16 FSM bit patterns).

Algorithm

$$b_3(4m) = X_3$$

Provided that X_3 refers to x_3 based on the combination of $\{x_3, x_2, x_1, x_0\}$ that maximizes $p(\{x_3, x_2, x_1, x_0\})$, chosen from 16 patterns.

$$b_2(4m+1) = X_2$$

Slot 4m	Slot 4 <i>m</i> + 1	Slot 4 <i>m</i> + 2	Slot 4 <i>m</i> + 3	
Send <i>b</i> ₃ (4 <i>m</i>)	Send $b_2 (4m + 1)$	Send $b_1 (4m + 2)$	Send $b_0 (4m + 3)$	
$\{x_3 \ x_2 \ x_1 \ x_0\}$	${x_3 x_2 x_1 x_0}$	${x_3 x_2 x_1 x_0}$	$\{x_3 \ x_2 \ x_1 \ x_0\}$	
0000	$b_3(4m) 0 0 1$	$b_3 (4m) b_2 (4m + 1) 0$ $b_3 (4m) b_2 (4m + 1) 0$ $b_3 (4m) b_2 (4m + 1) 1$ $b_3 (4m) b_2 (4m + 1) 1$ $b_3 (4m) b_2 (4m + 1) 1$	$\begin{pmatrix} 1 \\ 0 \\ b_3 \end{pmatrix} \begin{pmatrix} 4m \\ b_2 \end{pmatrix} \begin{pmatrix} 4m \\ b_2 \end{pmatrix} \begin{pmatrix} 4m \\ b_2 \end{pmatrix} \begin{pmatrix} 4m \\ b_3 \end{pmatrix} $	+ 1) $b_1 (4m + 2) 0$ + 1) $b_1 (4m + 2) 1$
		4 values	2 values	
1111	b ₃ (4 <i>m</i>) 1 1 1			
16 values	8 values			

Figure 3.42 Step-by-step weight updating scheme executed by UE

Provided that X_2 refers to x_2 based on the combination of $\{b_3(4m), x_2, x_1, x_0\}$ that maximizes $p(\{b_3(4m), x_2, x_1, x_0\})$, chosen from 8 patterns.

 $b_1(4m+2) = X_1$

Provided that X_1 refers to x_1 based on the combination of $\{b_3(4m), x_2, (4m + 1), x_1, x_0\}$ that maximizes $p(\{b_3(4m), x_2, (4m + 1), x_1, x_0\})$, chosen from 4 patterns.

$$b_0(4m+3) = X_0$$

Provided that X_0 refers to x_0 based on the combination of $\{b_3(4m), b_2, (4m + 1), b_1(4m + 2), x_0\}$ that maximizes $p(\{b_3(4m), b_2(4m + 1), b_1(4m + 2), x_0\})$, chosen from 2 patterns.

In each slot, the weight vector reflecting the latest FSM bit received by BS is determined and adopted.

3.3.1.13 Compressed Mode

Compressed mode is a function that enables the measurement of cells with different frequencies for the purpose of carrying out handover between different frequencies. The support of downlink compressed mode is essential for a single-receiver UE.

The decision to migrate to compressed mode is made by UTRAN, which informs UE of the parameters required for compressed mode.

In compressed mode, no data transmission takes place in the slot referred to as the *transmission gap*. Figure 3.43 shows an example of compressed mode. In a frame of

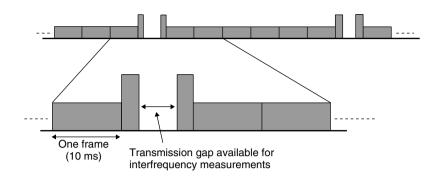


Figure 3.43 Example of transmission in compressed mode

Table 3.12	Types	of	compressed	mode
-------------------	-------	----	------------	------

Method	Overview		
Compressed mode by puncturing	A way to reduce the number of transmitted bits by using rate-matching (puncturing) function. The same SF is used in compressed mode as in the normal mode. (Refer to Section 3.3.1.6 for rate matching.)		
Compressed mode by reducing the spreading factor by 2 (SF/2)	A way to temporarily increase the transmission speed by halving SF so that the same number of bits can be transmitted as in the normal mode in slots other than the transmission gap.		
Compressed mode by higher layer scheduling	A way to limit the Transport Format Set (TFS) by the higher layer according to the number of bits that can be transmitted in slots other than the transmission gap. The same SF is used in compressed mode as in the normal mode. It is basically applicable only to non-real-time services, such as packet transmissions.		

compressed mode, the transmission power is raised temporarily to prevent the deterioration in quality (BER and BLER) due to the lower gain caused by the suspension of transmission.

As illustrated in the example referred to in Figure 3.43, the transmission gap can be iterated in compressed mode. The type of compressed mode (referring to the number of slots of the transmission gap, the interval between transmission gaps, the number of repetitions of transmission gaps etc.) can be changed through measurement request and so on.

There are three types of compressed modes, as explained in brief in Table 3.12.

3.3.2 Media Access Control (MAC) Sublayer

The Media Access Control (MAC) [4] protocol transfers data between the UE and each MAC-Entities of RNC at a one-to-one ratio. The data separation/combining function is provided by the higher-layer protocol because MAC does not have this capability. MAC

can reallocate radio resources and change MAC parameters in response to instructions from RRC. MAC also has the function to measure the traffic volume and quality, and inform RRC of the measurement results.

3.3.2.1 Logical Channel

MAC provides data forwarding functions different from RLC as a logical channel. MAC handles the mapping between the logical channels and the transport channels provided by Layer 1.

The logical channels can be divided into CCH, which are used for forwarding information in the control plane, and Traffic CHannels (TCH), which are used for forwarding user plane information. The available channels are as described in Table 3.13.

3.3.2.2 Mapping of Logical Channel and Transport Channel Support

Table 3.14 shows the mapping between logical channels and transport channels. Mapping is allowed only between the circled combinations of logical and transport channels.

3.3.2.3 Overview of MAC Functions

The functions of MAC are as follows.

(1) Mapping of Logical Channels and Transport Channels

MAC handles the mapping of logical channels and transport channels as described in Section 3.3.2.2.

Category	Logical channel	Application		
Control Channel (CCH)	Broadcast Control Channel (BCCH)	A downlink channel used for broadcasting system control information.		
	Paging Control Channel (PCCH)	A downlink channel used for broadcasting paging information.		
	Common Control Channel (CCCH)	A bidirectional channel used for transmitting control information between UE and the network. Used when there is no RRC connection or when accessing a new cell.		
	Dedicated Control Channel (DCCH)	A bidirectional point-to-point channel used for transmitting dedicated control information between UE and the network. Established by RRC Connection Setup.		
Traffic Channel (TCH)	Dedicated Traffic Channel (DTCH)	A bidirectional channel used for transferring user data, dedicated to one UE. Has both uplink and downlink.		
	Common Traffic Channel (CTCH)	A uni-directional point-to-multipoint channel for broadcasting user data to all UEs or certain UEs.		

 Table 3.13
 Logical channel

Logical/Transport	BCH	PCH	CPCH	RACH	FACH	DSCH	DCH
BCCH	0	_	_	_	0	_	_
PCCH	-	0	_	_	_	_	_
CCCH	-	_	_	0	\bigcirc	_	_
DCCH	-	_	0	0	0	0	0
DTCH	-	_	0	0	0	0	Ó
CTCH	-	_	_	_	0	_	_

 Table 3.14
 Mapping between logical channels and transport channel

(2) Selection of Transport Format Combination

MAC selects the adequate TFC for each transport channel from the Transport Format Combination Set (TFCS) specified by RRC. For RACH, MAC selects only the TF because there is no Layer 1 multiplexing of TFs.

(3) Priority Control

Upon the selection of the TF, MAC selects the TF that can transmit high-priority data on the basis of the ordering of priority according to the radio bearer attribute and RLC buffer status.

Also, MAC schedules data transmission between UEs on the common channel.

(4) Identification of UEs on Common Channel

MAC identifies UE with reference to UE-Id in the MAC header.

(5) Multiplexing and Demultiplexing to/from Transport Channels in RLC-PDU

MAC multiplexes and demultiplexes logical channels carried on the same transport channel using the MAC header.

(6) Observation of Traffic Volume

MAC measures the traffic volume according to the observation method specified by RRC for each transport channel and reports it to RRC.

(7) Switching Between Common and Dedicated Transport Channels

RRC switches between common and dedicated transport channels through MAC based on the measured traffic.

(8) Cyphering Process

MAC executes cyphering when Transparent Mode (TM) is used in RLC.

(9) Selection of Access Service Class (ASC) for RACH Transmission

MAC on the UE side controls and selects the Layer 1 PRACH resources (signature and access slots) and RACH parameters on the basis of ASC specified by RRC.

3.3.2.4 Data Format

MAC Protocol Data Unit (PDU)

Figure 3.44 shows the configuration of MAC PDU. The MAC header and MAC Service Data Unit (SDU) are of variable lengths. The configuration of the MAC header depends on the logical channel. MAC SDU serves as RLC-PDU.

Table 3.15 shows the components of MAC header.

MAC Header and Logical Channel Support

The configuration of the MAC header depends on the logical channel. As illustrated in Figure 3.45, there are 6 types of MAC header configurations. Table 3.16 shows the logical channels used in each configuration.

3.3.2.5 Selection of Transport Format Combination

MAC handles the mapping of data onto Layer 1 via L1 and L2 interfaces formed by the transport channels. The definition of terms relating to data mapping by MAC is as described below.

Transport Block

TB is a basic unit exchanged between MAC and Layer 1, for Layer 1 processing. RLC-PDU corresponds to the TB, and is the unit added with Cyclic Redundancy Check (CRC) in Layer 1.

Transport Block Set

This is defined as a set of TBs. It is the unit exchanged between L1 and MAC at the same time instance using the same transport channel.

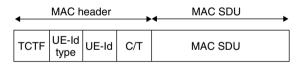


Figure 3.44 MAC data PDU

Table 3.15 Compo	nents of MAC header
------------------	---------------------

Components	Purpose
TCTF (Target Channel Type Field) UE-Id type UE-Id C/T	Used for identifying logical channels on FACH and RACH. Format of UE-Id used. Used for identifying UE. Used for identifying logical channels on dedicated transport channels, and for identifying logical channels upon the transmission of user data with RACH and FACH.

Logical channel	Transport channel	Existence of multiplexing	MAC header type (refer to Figure 3.45)
DTCH/DCCH	DCH	No multiplexing of dedicated channels	(1)
		Multiplexing of dedicated channels	(2)
	RACH/FACH	_	(3)
	DSCH	Multiplexing	(4)
		No multiplexing	(5)
BCCH	BCH	_	(1)
	FACH	_	(6)
PCCH		_	(1)
CCCH	RACH/FACH	_	(6)
CTCH	FACH	-	(6)

 Table 3.16
 MAC header structure and mapping with logical channel

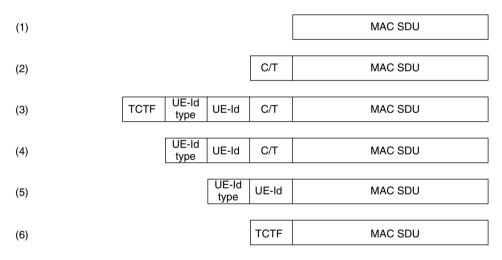


Figure 3.45 Structure of MAC header

Transport Block Size

This refers to the length of the TB, represented in bit units.

Transport Block Set Size

This refers to the length of the TB set, represented in bit units.

Transmission Time Interval (TTI)

This is defined as the interval of the time at which TB sets arrive between layers, and is equal to the time interval at which TB sets are forwarded by Layer 1 on the radio interface. TTI is the integral multiple of the minimum interleave period (10 msec); in practice, 10, 20, 40 and 80 msec. MAC supplies data to Layer 1 at every TTI.

Transport Format (TF)

This refers to the format in which the TB is supplied at every TTI on a transport channel. It consists of a Dynamic part and a Semistatic part.

- Dynamic part: Transport block size and transport block set size.
- Semi-static part: TTI, error-correction method and CRC size.

Transport Format Set (TFS)

This is defined as a set of TFs used in the transport channel. Within the TFS, the Semistatic parts of the same TF assume the same value. The Dynamic parts may assume different values for every TTI in order to assure variable rates.

Transport Format Combination (TFC)

As Layer 1 can multiplex multiple transport channels, there is a combination of transport channels that can be forwarded simultaneously on Layer 1. This combination is referred to as the TFC. The Coded Composite Transport CHannel (CCTrCH) of a particular UE is defined as the unit of transport channels that are combined as TFC.

Transport Format Combination Set (TFCS)

A set of TFCs carried on a CCTrCH is referred to as the TFCS (TFCS).

Transport Format Indicator (TFI)

This is the identifier of a TF assigned to every TB set forwarded to Layer 1 from MAC, and indicates which TF is being used in TFS.

Transport Format Combination Indicator (TFCI)

This corresponds one-to-one to TFC. It is generated on the basis of TFI by Layer 1, and transmitted over the radio interface. It is used in the Layer 1 of the receiving side for decoding the received data and demultiplexing the TB.

When mapping data to Layer 1, MAC selects the adequate TFC from TFCS specified by RRC, assigns TFI to TFS and forwards it to Layer 1. As the Semistatic part is common to TFCs, the selection task is actually done only for the Dynamic part.

3.3.3 Radio Link Control (RLC) Sublayer

3.3.3.1 Overview of RLC Functions

The RLC sublayer [5] establishes the RLC connection between UE and UTRAN, and offers the following three data transfer modes to the higher layer.

(1) Transparent Mode

This executes only the segmentation and reassembly of RLC-SDU, and no additional information is provided, such as header information or PADding (PAD).

(2) Unacknowledge Mode

Under the unacknowledge mode, segmentation, concatenation and reassembly of RLC-SDU as well as error detection are performed. When the same RLC-PDU is received erroneously, it will be discarded.

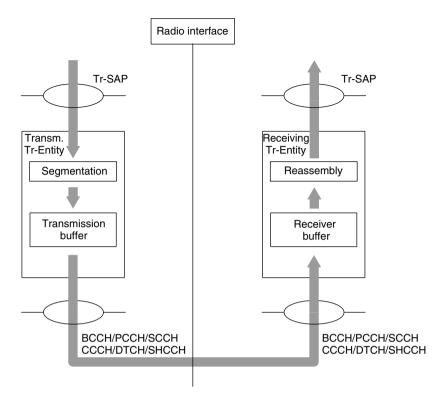


Figure 3.46 Architecture of transparent mode

(3) Acknowledge Mode

Under the Acknowledge mode, the segmentation, concatenation and reassembly of C-SDU is performed. It also retransmits RLC-PDU. In the event the same RLC-PDU is received erroneously, it will be discarded. The retransmission mechanism can be coordinated according to the QoS class by the RRC. Options are available for reversing the data order and for guaranteeing the data order in the event of retransmission.

RLC offers the following functions to realize the data transfer modes described in the preceding text.

- RLC-SDU segmentation/reassembly function
- RLC-SDU concatenation/reassembly function
- Padding function
- RLC-PDU transfer
- Error-correction function through retransmission mechanism
- Order control function
- Function to detect and discard the same RLC-PDU
- Flow control function
- Protocol error detection and recovery function
- Ciphering function
- Suspend and resume function

3.3.3.2 Transparent Mode

The functions of the transmission entity in TM are limited to generating RLC-PDU by segmenting RLC-SDU, and to transmitting RLC-PDU to the receiving entity. The receiving entity has the function to reassemble the received RLC-PDU and generate RLC-SDU.

As the ciphering function in TM is realized by the MAC sublayer, the TM RLC Entity does not have a ciphering function.

3.3.3.3 Unacknowledge Mode

The transmitting entity of the unacknowledge mode has the function to segment RLC-SDU into data that would be short enough to be stored in PU (Payload Unit) of RLC-PDU. The size of PU is specified by RRC. It also has the function to concatenate the next RLC-SDU in cases in which the data is to be stored into PU of RLC-PDU but cannot fill PU completely (referred to as *concatenation*). It can also insert PADs into the latter half of PU when PU cannot be completely filled with data.

The transmitting entity has the function to assign a sequence number to the RLC-PDU header. The sequence number is checked by the receiving entity, so that the same RLC-PDU would not be transmitted by mistake.

The transmitting entity attaches length information of RLC-SDU to enable the receiving entity to reassemble RLC-SDU in cases in which concatenated multiple RLC-SDUs or parts of them are stored in the same RLC-PDU or in cases in which PAD is required.

Both the transmitting and receiving entities have ciphering functions.

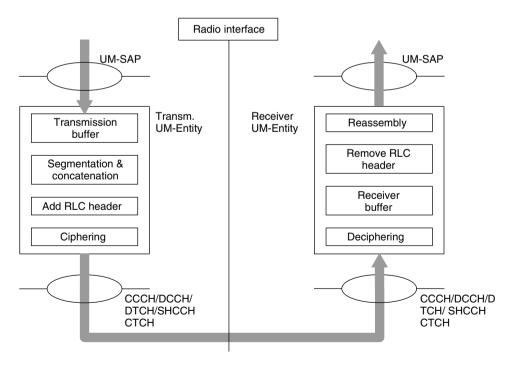


Figure 3.47 Architecture of unacknowledged mode

3.3.3.4 Acknowledge Mode

The transmitting unit of the acknowledge mode entity has the function to segment RLC-SDU into data that would be short enough to be stored in PU of RLC-PDU. The receiving unit has the function to reassemble the segmented data to generate RLC-SDU. The size of PU is specified by RRC. Under Release 99, only 1 PU is stored in 1 RLC-PDU. (In other words, RLC-PDU = PU.)

If data is to be stored in PU of RLC-PDU but cannot fill PU with data, the transmission unit can concatenate the next RLC-SDU (a function called *concatenation*) or insert PAD information in the latter half of PU. The transmission unit also has the function to insert information necessary for control such as retransmission (piggybacked information) in place of PAD. The transmission unit attaches length information on RLC-SDU to enable the receiving unit to reassemble RLC-SDU when multiple concatenated RLC-SDUs or a part of such RLC-SDU are stored in the same RLC-PDU or when PAD is required.

On the other hand, the receiving unit has the function to segment the RLC-SDU concatenated in the PU of RLC-PDU, the function to remove PAD and the function to extract piggybacked information.

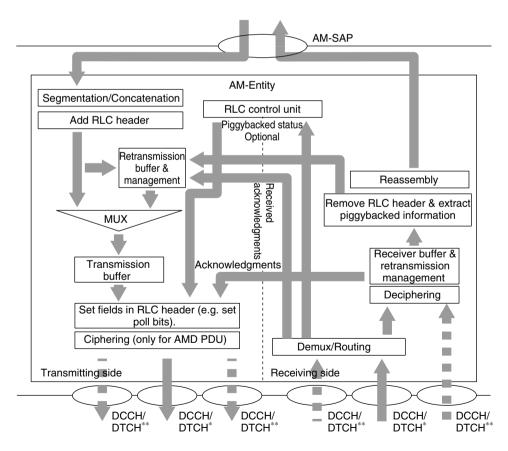
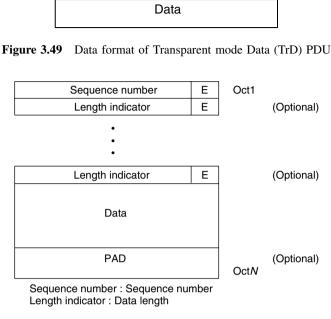


Figure 3.48 Architecture of acknowledge mode



E (extension bit) : dentifier bit of data/length indicator PAD : PADding

Figure 3.50 Data format of UMD PDU

The acknowledge mode entity has a retransmission function as well as the function to assign a sequence number to the RLC-PDU header. The receiving unit has the function to transmit Acknowledge as control information to the transmitting side when RLC-PDU was received normally and to execute control so that the RLC-SDU transfer order would not be reversed when sent to the higher-layer entity. It is also capable of discarding the same data when the same RLC-PDU has been transmitted.

The retransmission mechanism is equipped with a window flow control function. Both the transmitting and receiving sides of the retransmission mechanism have reset and resume functions.

The transmitting entity and the receiving entity both have a ciphering function.

3.3.3.5 Data Format

TrD PDU

TrD PDU is used in the transmission of RLC-SDU in TM. No information is added to it. TrD PDU is bit-aligned.

UMD PDU

UMD (Unacknowledged Mode Data) PDU is used in the transmission of RLC-SDU in Unacknowledged Mode. A sequence number is assigned to it. A Length Indicator is also assigned to enable the insertion of PAD and concatenation of RLC-SDU. UMD PDU is octet-aligned.

AMD PDU

AMD (Acknowledged Mode Data) PDU is used in the transmission of RLC-SDU in AM. A sequence number is assigned to it. A Length Indicator is also assigned to enable the insertion of PAD and concatenation of RLC-SDU. The empty space can be filled with Piggybacked STATUS PDU. AMD PDU is octet-aligned.

STATUS PDU

STATUS PDU is used in the transmission of RLC-SDU in AM. It is used for transmitting control information used in the retransmission mechanism. STATUS PDU is octet-aligned.

Piggybacked STATUS PDU

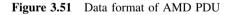
This is a control-oriented PDU that can be inserted in place of the PAD of AMD PDU. It has the same structure as STATUS PDU, apart from the D/C bit, which is the reserved

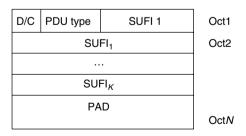
D/C	Sequence r	Oct1			
Se	equence number	Р	Н	E	Oct2
Length indicator					Oct3 (Optional) (1)

Length indicator	E
Data	
PAD or a piggybacked STATUS	PDU

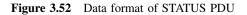
D/C : Identifier bit of AMD PDU and control PDU
Sequence number : Sequence number
P (polling bit) : Request bit of control information
HE (Header Extension) : dentifier bit of Length indicator length
Length indicator : Data length
E (extension bit) : dentifier bit of data/length indicator
PAD : PADding

Oct_N





D/C : Identifier bit of AMD PDU and control PDU PDU-type : Type of control PDU



bit (R bit) in the case of Piggybacked STATUS PDU. Piggybacked STATUS PDU is octet-aligned.

RESET/RESET ACK PDU

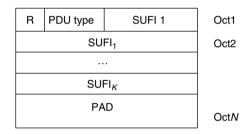
RESET PDU and RESET ACK PDU are used in the transmission of Reset instructions and acknowledgement information in AM. RESET PDU and RESET ACK PDU are octet-aligned.

3.3.3.6 Basic Procedures

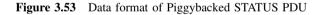
Transparent Mode (TM) Data Transfer Procedures

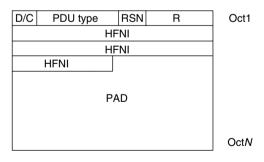
Figure 3.55 illustrates the data transfer procedures in Transparent Mode (TM).

The procedures are activated on the basis of a TM data transfer request from the higher layer. If the transmitting side is in a Data Transfer Ready state, data received from the higher layer is transmitted as TrD PDU. If necessary, RLC-SDU is segmented and TrD PDU is generated.



R (Reserved) : Reserved bit PDU-type : Type of control PDU SUFI : Parameters for retransmission mechanism PAD : PADding





D/C : Identifier bit of AMD PDU and control PDU PDU-type : Type of control PDU R (Reserved) : Reserved bit HFNI : HFN (Hyper Frame Number) PAD : PADding

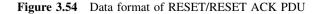




Figure 3.55 TM data transfer procedures

On the receiving side, RLC-SDU is configured from TrD PDU (if they have not been segmented, TrD SDU automatically becomes RLC-SDU), and transferred to the higher layer via Tr-SAP.

In cases in which RLC-SDU has been segmented, if the RLC-SDU reassembled on the receiving side is not the permissible size, the RLC-SDU is discarded.

Unacknowledged Mode (UM) Data Transfer Procedures

Figure 3.56 illustrates the data transfer procedures in Unacknowledge Mode (UM).

The procedures are activated on the basis of a UM data transfer request from the higher layer. If the transmitting side is in a Data Transfer Ready state, data received from the higher layer is transmitted as UMD PDU. If necessary, RLC-SDU is segmented and UMD PDU is generated, or conversely, multiple RLC-SDUs are filled into the same UMD PDU to generate UMD PDU.

At the transmitting side, a Length Indicator is assigned so that RLC-SDU can be generated at the receiving side. The Length Indicator is inserted at the head of the payload part. In the RLC header part, a sequence number is assigned. The initial value is 0, and a value of 1 is added for each RLC-PDU.

On the receiving side, the data part is extracted from UMD PDU using the Length Indicator to configure an RLC-SDU, which is transferred to the higher layer via UM-SAP.

In cases in which there are any inconsistencies in the sequence number, the RLC-SDU that contains the RLC-PDU is discarded.

Acknowledge Mode (AM) Data Transfer Procedures

Figure 3.57 illustrates the data transfer procedures in AM.

The procedures are activated on the basis of an AM data transfer request from the higher layer or in response to the activation of data retransmission. If the transmitting side is in a Data Transfer Ready state, data received from the higher layer or the retransmission data is transmitted as AMD PDU. In the case of retransmission, the retransmitted data is given higher priority over the first data transfer attempt. If necessary, RLC-SDU is segmented and AMD PDU is generated, or conversely, multiple RLC-SDUs are filled into the same



Figure 3.56 UM data transfer procedures



Figure 3.57 AM data transfer procedures

AMD PDU to generate AMD PDU. In the case of retransmission, only the final part of the retransmitted data can be filled into the same AMD PDU.

In the case of the control plane, PDU is transmitted on DCCH. In the case of the user plane, it is transmitted on DTCH.

On the transmitting side, a Length Indicator is assigned so that RLC-SDU can be generated at the receiving side. The Length Indicator is inserted in the head of the payload part. In the RLC header part, a sequence number is assigned. The initial value is 0 and a value of 1 is added for each RLC-PDU.

On the receiving side, the data part is extracted from AMD PDU using the Length to configure an RLC-SDU, which is transferred to the higher layer via AM-SAP. If the Polling bit is flagged in the RLC header or if the data with a certain sequence number has not arrived as expected, the transmission of STATUS PDU is activated.

The triggers to set the polling bit can be freely selected from any of the following instances, apart from Option 1 and either Option 2 or 7 that must always be chosen:

- 1. At the time of transmitting the last PU in the transmission buffer;
- 2. At the time of transmitting the last PU in the retransmission buffer;
- 3. Upon the expiration of the Timer_Poll;
- 4. When $VT(PU) = Poll_PU$ (Polling for each PU);
- 5. When $VT(SDU) = Poll_SDU$ (Polling for each SDU);
- 6. When the Window usage rate >= Poll_Window (Window-based Polling);
- 7. Upon the expiration of Time_Poll_Periodic (Timer-based Polling); and
- 8. Upon the expiration of Timer_Poll_Prohibit and the flagging of the Poll trigger. (Upon the cancellation of Poll when Timer_Poll_Prohibit is activated.)

RLC Reset Procedures

Figure 3.58 illustrates the RLC reset procedures.

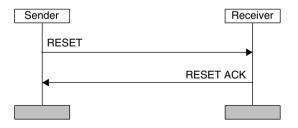


Figure 3.58 RLC reset procedures

The RLC reset procedures are activated when there are protocol errors between RLC entities in AM. The RESET PDU transmitting side sends RESET PDU in a Data Transfer Ready state, goes into RESET Pending mode and activates the reset timer. Upon the expiration of the reset timer, RESET PDU is retransmitted.

The RESET PDU receiving side sends back RESET ACK PDU and initializes the RLC entity. In response to the reception of RESET ACK PDU, the RESET PDU transmitting side stops the timer and returns to the Data Transfer Ready state.

In the RLC reset procedures, the Hyper Frame Number (HFN) is notified by RESET PDU and RESET ACK PDU in order to synchronize HFN between UE and UTRAN.

RESET PDU and RESET ACK PDU are PDUs that are given higher priority than data PDU. In the case of the control plane, PDU is transmitted on DCCH. In the case of the user plane, it is transmitted on DTCH.

STATUS PDU Transfer Procedures

Figure 3.59 illustrates the STATUS PDU transfer procedures.

STATUS PDU is used for transferring status information between RLC entities in AM. The receiving side in Figure 3.59 refers to the receiving side of AMD PDU.

The procedures are activated when the following conditions are met:

- Polling bit is set in AMD PDU;
- Missing in RLC-PDU is detected; or
- The timer expires in cases in which timer-based Status PDU transfer is set.

STATUS PDU is a PDU that is given higher priority than data PDU. In the case of the control plane, PDU is transmitted on DCCH. In the case of the user plane, it is transmitted on DTCH. The status information can also be sent by piggybacked PDU instead of STATUS PDU.

It is also possible to set a separate transport channel from AMD PDU for STATUS PDU.

If the transmission side receives Negative Acknowledge by STATUS PDU or Piggybacked PDU, the data retransmission function is activated.

Suspend/Resume Procedures

The suspend/resume function is a function to suspend the transmission of data when changing the ciphering key (CK: Refer to the section on Parameters Relating to Ciphering Algorithm in Section 3.3.3.8) during AM.

RRC informs RLC of CRLC-SUSPEND-Req. In response, RLC informs RRC of CRLC-SUSPEND-Conf and goes into Local Suspend mode. By CRLC-SUSPEND-Req,



Figure 3.59 STATUS PDU transfer procedures

the transmission of data with a sequence number notified by RRC and data with subsequent numbers is suspended. Meanwhile, RRC changes the CK and then switches from Local SUSPEND mode to Data Transfer Ready mode using CRLC-RESUME-Req.

3.3.3.7 Discard Function

Each mode of RLC has a discard function. Table 3.17 shows the modes of the discard function and applicable RLC modes.

3.3.3.8 Ciphering

Allocation of Ciphering Function

The ciphering function is executed by SRNC (Serving Radio Network Controller) and UE. Their respective RLC and MAC have ciphering functions, which are used differently according to the following rules:

- The logical channels that require ciphering and are transferred on common transport channels do not use TM in RLC, but use UM.
- Logical channels that use AM mode or UM mode in RLC perform ciphering with RLC.
- Logical channels that use TM in RLC perform ciphering with MAC.

Parameters Relating to Ciphering Algorithm

When ciphering is done by RLC, the payload part is subject to ciphering. In other words, the sequence number part in the first 1 byte of RLC-PDU is not ciphered in UM mode;

Discard mode	Applicable RLC mode	Overview
Timer-based discarding (with signaling)	АМ	The transmission side activates the timer for each SDU received from the higher layer and discards SDU when the timer exceeds a predefined value. In the event of discarding, the transmission side sends a Move Receiving Window command by RLC STATUS PDU or Piggybacked PDU to update the Window.
Timer-based discarding (with no signaling)	TM/UM	The transmission side activates the timer for each SDU received from the higher layer and discards SDU when the timer exceeds a predefined value. The transmission side does not inform the reception side of the activation of the discard function.
Discarding by maximum retransmission attempts	АМ	The transmission side discards the SDU when the maximum number of retransmission attempts of SDU received from the higher layer exceeds a certain level. In the event of discarding, the transmission side sends a Move Receiving Window command by RLC STATUS PDU or Piggybacked PDU to update the Window.

 Table 3.17
 Discard function

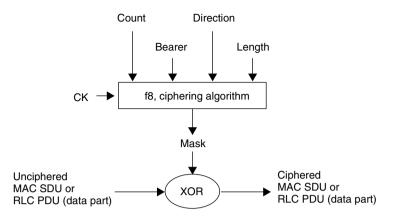


Figure 3.60 Ciphering algorithm and parameters

the same part in the first 2 bytes of RLC-PDU is not ciphered in AM mode. If ciphering is done by MAC, the ciphering process is carried out on the entire MAC SDU (RLC-PDU). Figure 3.60 illustrates the ciphering process.

Ciphering is done by the exclusive OR (XOR) between the Mask-which is the ciphering algorithm output-and the data subject to ciphering. The parameters are as shown in Table 3.18.

3.3.4 Packet Data Convergence Protocol (PDCP) Sublayer

3.3.4.1 Overview of Functions

The functions of PDCP are as follows:

- 1. IP data header compression function (e.g. Transmission Control Protocol/Internet Protocol (TCP/IP), Real Time Protocol (RTP)/User Datagram Protocol (UDP)/IP).
- 2. User data transfer function.
- 3. Transfer of PDCP-SDU [1] from Non-Access Stratum (NAS) to RLC and conversion of RLC-SDU from RLC into PDCP-SDU and transferring it to NAS.
- 4. PDCP sequence number management for loss-free SRNS (Serving Radio Network Subsystem) relocation support.
- 5. Multiplexing of different radio bearers onto the same RLC entity (Release 2000 and later).

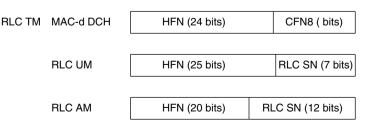


Figure 3.61 Structure of ciphering sequence number

Parameter name	Overview
COUNT	This is a sequence number of 32 bits (ciphering sequence number), and consists of a combination of a long sequence number (referred to as Hyper Frame Number (HFN)) and a short sequence number. In AM and UM, a ciphering sequence is set for each logical channel (the COUNT is set independently), whereas in TM, a ciphering sequence common to all logical channels is set. The configuration of the ciphering sequence number depends on the RLC mode, which is as illustrated in Figure 3.61. HFN is initialized by UE before ciphering and notified to SRNC. Also, it is added at every cycle of each short sequence number. For the short sequence number, the Connection Frame Number (CFN) of UEFN is set when RLC is in TM, whereas the sequence number of RLC is used when RLC is in AM or UM.
CK (Ciphering Key)	This is set between UE and SRNC in the authentication phase. In the case of two-key solution, the most recent CK agreed upon between User, 3G and MSC is used assuming a CS-Domain bearer and that between User, 3G and SGSN is used assuming a PS-Domain bearer. The most recent CK is used for the signaling link in both CS and PS Domains. Either CS-Domain or PS-Domain is forwarded on each logical channel so as to make the ciphering function of RLC and MAC work properly. RRC informs RLC and MAC of the mapping between radio bearer and CS/PS, and the mapping between radio bearer ID and CK.
BEARER	This is an identifier of a logical channel, which is uniquely assigned in the RRC connection. It is used to set the ciphering algorithm so that the same MASK would not be applied to multiple logical channels with the same CK and COUNT. In such cases, each logical channel is ciphered independently.
DIRECTION	This distinguishes between uplink and downlink.
LENGTH	This shows the length of Keystream Block (MASK) generated by algorithm. However, it is not an entry parameter for the Keystream generation function.

 Table 3.18
 Parameters relating to ciphering algorithm

Note: MSC: Mobile Switching Center; SGSN: Serving GPRS Support Node; UEFN: User Equipment Frame Number.

3.3.4.2 Data Format

There are two types of data formats defined for PDCP. The data format specified by RRC is used.

The PDCP-No-Header PDU format provides data transfer with no overheads, as illustrated in Figure 3.62. The PDCP-Data-PDU format performs the transfer of PDCP-SDU after compressing the header, as shown in Figures 3.63 and 3.64.

The PDU-Type of the PDCP-Data-PDU format identifies PDCP-Data-PDU and PDCP-SeqNum-PDU in Release 99. Release 99 stipulates only the method of using PID for

Bit	8	7	6	5	4	3	2	1
Oct 1								
•••			D	Data se	egmer	nt		
N								

Figure 3.62 PDCP-No-Header PDU format

Bit	8	7	6	5	4	3	2	1
Oct 1	PDU	PDU type PID						
•••		Doto cogmont						
Ν	Data segment							

Figure 3.63 PDCP-Data-PDU format

Bit	8	7	6	5	4	3	2	1
Oct 1	PDU type PID							
Oct 2	Sequence number							
Oct 3								
•••	Data segment							
N								

Figure 3.64 PDCP-SeqNum-PDU format

identifying the header compression scheme. The sequence number refers to the PDCP-PDU sequence number used for loss-free SRNS relocation.

3.3.4.3 Assigning PDCP Sequence Numbers

PDCP has the function to manage the transmitted/received PDCP-SDU based on sequence numbers, in order to prevent data loss in the event of SRNS Relocation (refer to Section 3.3.4.4).

The following applies to each radio bearer:

- 1. The uplink transmit PDCP sequence number increases by 1 when transmit PDCP-PDU is handed to RLC in UE.
- 2. The downlink transmit PDCP sequence number increases by 1 when transmit PDCP-PDU is handed to RLC on UTRAN.
- The uplink received PDCP sequence number increases by 1 when received PDCP-PDU is handed over from RLC in UE or when the discarding of RLC-SDU is notified by RLC-SDU Discard function.
- 4. The downlink received PDCP sequence number increases by 1 when received PDCP-PDU is handed over from RLC in UTRAN, or when the discarding of RLC-SDU is notified by RLC-SDU Discard function.
- 5. After the execution of RLC reset, after the execution of RB reconfiguration, and upon the unexpected reception of an invalid PDCP sequence number, the PDCP sequence number is exchanged between the entities of the same layer by PDCP-SeqNum-PDU to synchronize the sequence number.

3.3.4.4 SRNS Relocation

In the event of SRNS relocation, the following controls are activated between UE, Old-SRNC and Target SRNC:

- 1. SRNC relocation is activated from RRC at UE, Old-SRNC and Target SRNC.
- 2. The corresponding entity of PDCP of Old-SRNC transfers all unconfirmed PDCP-PDUs and transmission sequence numbers to Target SRNC.
- 3. The corresponding entity of PDCP of Old-SRNC transmits the reception sequence number corresponding to the PDCP-PDU to be received next to Target SRNC.
- 4. Target SRNC and UE transmit the reception sequence number corresponding to the PDCP-PDU to be received next to their respective destination entities and compare the transmitted sequence number with the notified reception sequence number to assure consistency between them.
- 5. During the execution of SRNS relocation, all compression entities are reset.

3.3.5 Radio Resource Control (RRC)

3.3.5.1 Relationship Between RRC and Other Layers

RRC [6] is a Layer 3 protocol in the radio interface and offers the following services to the higher layer:

- 1. Notification of broadcast information to all UEs in the area;
- 2. Paging of a specific UE; and
- 3. Setting, changing and releasing connections, as well as signaling.

RRC also has the function to control lower layer protocols in order to provide the services stated above. Figure 3.65 illustrates its relationship with lower-layer protocols.

3.3.5.2 Overview of RRC Functions

Functions of RRC are as follows:

(1) Broadcasting of Non-Access Stratum (NAS)-Related Information from CN

RRC broadcasts system information (including information on higher layer) to all Ues from the network. RRC makes scheduling, division and iteration of system information possible. System information includes cell-specific information and information common to all cells.

(2) Broadcasting Access Stratum (AS)-Related Information

RRC broadcasts system information to all UEs from the network. RRC makes scheduling, division and iteration of system information possible.

(3) Establishment, Reestablishment, Maintenance and Release of RRC Connection Between UE and UTRAN

The establishment of RRC connection is activated from the higher layer on the UE side to establish the signaling connection used by UE. RRC connection control includes

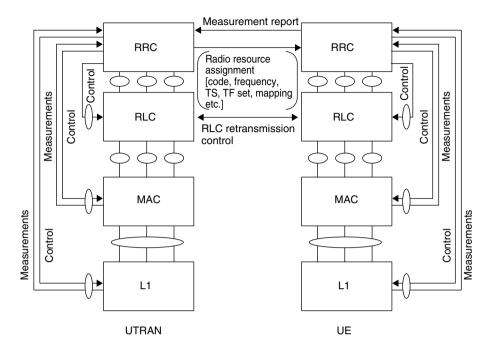


Figure 3.65 Interaction between RRC and lower-layer protocol

the reselection of additional cells, admission control and the establishment of Layer 2 signaling links.

RRC connection is released when the higher layer requests the release of the last signaling link or when the RRC itself decides to release it because of faults in the RRC connection. When RRC connection is lost, UE requests its reestablishment.

(4) Establishment, Alteration and Release of Radio Bearer

On the basis of the higher layer's request, RRC sets, alters or releases the radio bearer in the user plane. RRC can set more than one radio bearer simultaneously with one UE. When setting and changing the radio bearer, RRC selects parameters for admission control on the basis of the information from the higher layer and for setting the radio bearer by the lower layer.

(5) Allocation, Alteration and Release of Radio Resources Required for RRC Connection

RRC controls the allocation of radio resources required for RRC connection (such as codes). RRC can allocate radio resources asymmetrically between uplink and downlink. It can also change the radio resources in the established RRC connection – this function includes the reallocation of multiple radio bearers related to the same RRC connection. RRC can also transmit signals that allow radio resources to be allocated from UTRAN to UE, for the purpose of performing handover to Global System for Mobile Communications (GSM) or other radio systems.

(6) Mobility Management of RRC Connection

RRC assesses, decides and executes movements of the established RRC connection. RRC functions include the preparation of handover based on measurement results executed by UE, preparation of handover to GSM and other radio systems, reselection of cells and procedures for updating the paging area.

(7) Paging and Notification Functions

RRC broadcasts paging information to a specific UE in response to a request from the higher layer on the network side. It also pages UE with which RRC connection is established.

(8) Transfer of Higher Layer PDU

This is a function to route the higher-layer PDU on the UE side to the corresponding higher-layer entity on the UE side and the same to the Radio Access Network Application Part (RANAP) entity on the UTRAN side.

(9) QoS Control

RRC secures a radio bearer that meets the required QoS. This includes the allocation of radio resources.

(10) Control of Measurement Notice and Measurement Report to UE

RRC controls the items to be measured in the radio interface (including other systems), the measurement period and timing and the reporting method by UE. RRC also has the function to forward the UE report results to UTRAN.

(11) Outer-Loop Power Control

RRC has the function to control the settings of the target value of closed-loop power control.

(12) Ciphering Control

RRC offers the procedures to set ciphering between UE and UTRAN.

(13) Initial Cell Selection and Reselection

RRC has the function to select the optimal cell based on the measurement results and selection criteria in idle mode.

3.3.5.3 Basic Procedures

Table 3.19 shows the basic procedures of RRC.

3.3.5.4 Message

Table 3.20 shows the messages used by RRC.

3.3.5.5 Protocol Status

There are two RRC protocol statuses: Idle mode and UTRAN-connected mode. The UTRAN-connected mode consists of CELL_PCH, CELL_FACH, CELL_DCH and URA_PCH.

Procedure name	Overview				
RRC connection management proce	edures				
Broadcast of information	This is a procedure for broadcasting system information from UTRAN to UE in idle mode and connected mode inside the cell.				
Paging	This is a procedure for forwarding paging information to UE in idle mode (CELL_PCH/URA_PCH state) using PCCH.				
RRC connection release	This is a procedure for releasing signal links and all radio bearers between UE and UTRAN.				
Transmission of UE capability information	This is a procedure for UE to inform UTRAN of the UE's capabilities.				
UE capability enquiry	This is a procedure for UTRAN to make UE inform its capabilities.				
Initial direct transfer	This is a procedure for establishing signaling between UE and CN, and is used for forwarding the first higher layer (NAS) message in uplink via the radio interface.				
Downlink direct transfer	This is a procedure for forwarding the higher layer (NAS) message in downlink via the radio interface.				
Uplink direct transfer	This is a procedure for forwarding the higher layer (NAS) message in uplink via the radio interface.				
UE dedicated paging	This is a procedure for forwarding dedicated paging information to a specific UE in connected mode in CELL_DCH or CELL_FACH states.				
Security mode control	This is a procedure for requesting the commencement of ciphering or alteration to a ciphering key, and is used for radio bearers in general including signal links. It also instigates the commencement of integrity protection simultaneously.				
Signalling connection release procedure	This is a procedure for UTRAN to inform UE of the release of one signaling connection to CN Domain. RRC connection release is not activated based on this procedure.				
Radio bearer control procedures					
Radio bearer establishment	This is a procedure for establishing a new radio bearer. It also establishes the transport channel for forwarding signals transparently. Hard handover can be performed by the execution of this procedure.				
Radio bearer reconfiguration	This is a procedure for reconfiguring the parameters of the radio bearer or the signal link associated with changes in QoS. Hard handover can be performed by the execution of this procedure.				

Table 3.19Basic procedures

(continued overleaf)

Table 3.19(continued)

Procedure name	Overview
Radio bearer release	This is a procedure for releasing a set radio bearer. Hard handover can be performed by the execution of this procedure.
Transport channel reconfiguration	This is a procedure for changing the parameters of the transport channel. Hard handover can be performed by the execution of this procedure.
Transport format combination control	This is a procedure used for controlling the transport format combination in the uplink transport format combination set.
Physical channel reconfiguration	This is a procedure used for establishing, reconfiguring and releasing physical channels. Hard handover can be performed by the execution of this procedure.
RRC connection mobility procedures	5
Cell update	This is a procedure for UE in CELL_FACH or CELL_PCH states to inform UTRAN of the current cell after cell reselection. It may also be used to diagnose the status of RRC connection through periodical URA Update. It can be used for reconfiguring the AM RLC Entity for signal link, and UE can notify the occurrence of irrecoverable errors in the AM RLC Entity. It includes the reconnection of dedicated channels.
UTRAN Registration Area (URA) update	This is a procedure for UE in CELL_PCH state to inform UTRAN of the current URA after URA reselection. It may also be used to diagnose the status of RRC connection through periodical URA Update. It can be used for reconfiguring the AM RLC Entity for signal link, and UE can notify the occurrence of irrecoverable errors in the AM RLC Entity.
UTRAN mobility information	This is a procedure for newly assigning C-RNTI or U-RNTI to UE in connected mode.
Active set update in soft handover	This is a procedure for updating the active set of the connection between UE and UTRAN, and is used under CELL_DCH state.
Hard handover	This is a procedure for switching between TDD and FDD modes when the frequency of the connection between UE and UTRAN is to be changed, or when UE has moved from cell to cell in a network without micro-diversity. It is used under CELL_DCH state.
Measurement procedures	
Measurement control	This is a procedure for setting, changing and releasing the measurement functions of UE.
Measurement report	This is a procedure for UE to forward measurement results to UTRAN.

Note: C-RNTI: C-Radio Network Temporary Identity.

Procedure name		Message name		
	$(UE \rightarrow UTRAN)$	\rightarrow (UTRAN \rightarrow UE)	\rightarrow (UE \rightarrow UTRAN)	
	RRC connection	n management procedures		
Broadcast of system information	-	System information (TM:BCCH)	-	
Paging	_	Paging Type1 (TM:PCCH)	-	
RRC connection establishment	RRC connection request (TM:CCCH)	RRC connection reject (UM:CCCH)	_	
		RRC connection setup (UM:CCCH)	RRC connection setup complete (AM:DCCH)	
RRC connection release	_	RRC connection release (UM:DCCH)	When dedicated CH is used: RRC connection release complete (UM:DCCH) When common CH is used: C connection release complete	
		RRC connection release	(AM:DCCH)	
T · · · ·		(UM:CCCH)		
Transmission of UE capability information	UE capability information (AM:DCCH)	UE capability information confirm (AM/UM:DCCH)	-	
UE capability enquiry	(AM.DCCH) -	UE capability enquiry (AM/UM:DCCH)	-	
Initial direct transfer	Initial direct transfer (AM:DCCH)	- -	_	
Downlink direct transfer	_	Downlink direct transfer (AM:DCCH)	-	
Uplink direct transfer	Uplink direct transfer (AM:DCCH)	-	-	
UE dedicated paging	_	Paging Type2 (AM:DCCH)	-	
Security mode control	_	Security mode command (AM:DCCH)	Security mode complete (AM:DCCH) Security mode failure (AM:DCCH)	

Table 3.20 List of messages

Procedure name		Message name	
	$(UE \rightarrow UTRAN)$	\rightarrow (UTRAN \rightarrow UE)	\rightarrow (UE \rightarrow UTRAN)
Counter check	_	Counter check (AM:DCCH)	Counter check response (AM:DCCH)
Signalling connection release request procedure	Signalling connection release request (AM:DCCH)	_	-
Signalling connection release procedure	_	Signalling connection release (AM:DCCH)	-
	Radio beare	r control procedures	
Radio bearer establishment	-	Radio bearer setup (AM/UM:DCCH)	Radio bearer setup complete (AM:DCCH) Radio bearer setup failure (AM:DCCH)
Radio bearer reconfiguration	_	Radio bearer reconfiguration (AM/UM:DCCH)	Radio bearer reconfiguration complete (AM:DCCH) Radio bearer reconfiguration failure (AM:DCCH)
Radio bearer release	-	Radio bearer release (AM/UM:DCCH)	Radio bearer release complete (AM:DCCH) Radio bearer release failure (AM:DCCH)
Transport channel reconfiguration	-	Transport channel reconfiguration (AM/UM:DCCH)	Transport channel reconfiguration complete (AM:DCCH) Transport channel reconfiguration failure (AM:DCCH)

Table 3.20 (continued)

(continued overleaf)

Procedure name		Message name	
	(UE→UTRAN)	\rightarrow (UTRAN \rightarrow UE)	\rightarrow (UE \rightarrow UTRAN)
Transport format combination control	_	Transport format combination control (TM/UM/AM:DCCH)	_
			Transport format combination control failure (AM:DCCH)
Physical channel reconfiguration	_	Physical channel reconfiguration (AM/UM:DCCH)	Physical channel reconfiguration complete (AM:DCCH) Physical channel reconfiguration failure (AM:DCCH)
	RRC connection	on mobility procedures	
Cell update	Cell update (TM:CCCH)	Cell update confirm (UM:CCCH/DCCH)	- UTRAN mobility information confirm (AM:DCCH) Physical channel reconfiguration complete (AM:DCCH) Transport channel reconfiguration complete (AM:DCCH) Radio bearer release complete (AM:DCCH) Radio bearer reconfiguration complete
		RRC connection release	(AM:DCCH)
URA update	URA update (TM:CCCH)	(UM:CCCH) URA update confirm (UM:CCCH/ DCCH)	_

Table 3.20 (continued)

Procedure name	Message name		
	(UE→UTRAN)	\rightarrow (UTRAN \rightarrow UE)	\rightarrow (UE \rightarrow UTRAN)
			UTRAN mobility information confirm (AM:DCCH)
		RRC connection release	_
UTRAN mobility information	-	(UM:CCCH) UTRAN mobility information (AM/UM:DCCH)	UTRAN mobility information confirm (AM:DCCH)
			UTRAN mobility information failure (AM:DCCH)
Active set update in soft handover	-	Active set update (AM:DCCH)	Active set update complete (AM:DCCH) Active set update
Hard handover (example: no changes in RB or TrCH)	-	Physical channel reconfiguration (AM/UM:DCCH)	failure (AM:DCCH) Physical channel reconfiguration complete (AM:DCCH) Physical channel reconfiguration failure (AM:DCCH)
	Measurem	nent procedures	
Measurement control	_	Measurement control (AM: DCCH)	_
		Measurement control (AM:DCCH)	Measurement control failure (AM:DCCH)
Measurement report	Measurement report (AM/UM:DCCH)	_	_

Table 3.20(continued)

UE is in the idle mode when the power is turned on up until the establishment of RRC connection. The connection of UE is limited to the AS. UE in the idle mode is identified by identifiers (International Mobile Subscriber Identity (IMSI), Temporary Mobile Subscriber Identity (TMSI) and packet TMSI (P-TMSI) in the NAS. UTRAN has no information on UE. The establishment of RRC connection is activated in response to a request from the higher layer of UE or a paging request from the network. When UE receives a

confirmation message about the establishment of RRC connection from UTRAN, UE goes into the UTRAN-connected mode (CELL_FACH or CELL_DCH state). If it fails to establish RRC connection, UE stays in the idle mode.

UE in the UTRAN-connected mode is assigned a Radio Network Temporary Identity (RNTI), which is used to identify UE on the common transport channel. The status of RRC in the UTRAN-connected mode depends on the level of the transport channel that can be used by UE, which may be CELL_PCH, CELL_FACH, CELL_DCH or URA_PCH. If UE in the UTRAN-connected mode releases RRC connection, it goes into the idle mode.

Figure 3.66 shows the RRC protocol status. The figure also illustrates the idle mode and the connected mode of other radio systems (GSM and General Packet Radio Service (GPRS)).

The following is the description of the 4 statuses in the UTRAN-connected mode.

(1) CELL_DCH

A DPCH is assigned to UE. The UE has identified the cell level by the current Active Set. The dedicated transport channel, the Downlink-Shared Transport Channel and the combination of them have also been identified in this state.

(2) CELL_FACH

No DPCH is assigned to UE. In this state, UE receives FACH in downlink, and in uplink it can use a common channel that can execute transmission from time to time according to the access procedures of each transport channel. UTRAN is aware of the location of UE at cell level (the cell updated by UE most recently).

(3) CELL_PCH

No dedicated channel is assigned to UE. In downlink, UE receives PCH via PICH by Discontinuous Reception (DRX) (refer to the section on PICH in Section 3.3.1.2). In

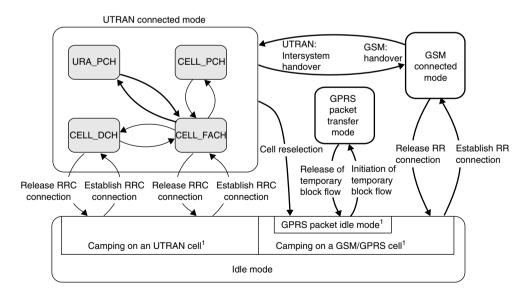


Figure 3.66 Protocol status

Node-B	 Connection parameters (i.e. parameters required for generating a different transport channel) UE ID Uplink scrambling code CPICH timing difference (calculated based on UE measurement results)
UE	Channelization code Timing difference T_{offset} (refer to Figure 3.67)

 Table 3.21
 Information notified when cell is added

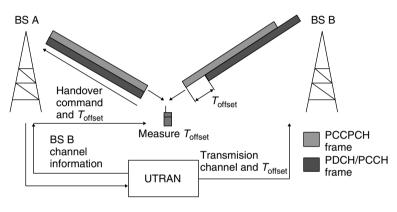


Figure 3.67 Timing difference

uplink, UE is not engaged in anything. UTRAN is aware of the location of UE at cell level (the cell updated most recently by UE in CELL_FACH state).

(4) URA_PCH

No dedicated channel is assigned to UE. In downlink, UE receives PCH via PICH by DRX (intermittent reception). In uplink, UE is not engaged in anything. UTRAN is aware of the location of UE at UTRAN Registration level (the URA assigned most recently to UE during URA Update in CELL_FACH state).

Transition of the protocol states in the UTRAN-connected mode takes place according to the rules below.

(1) $CELL_DCH \rightarrow CELL_FACH$

Transition takes place when all dedicated channels are released using signals.

(2) $CELL_FACH \rightarrow CELL_DCH$

Transition takes place when a dedicated channel is established using signals.

(3) CELL_FACH \rightarrow CELL_PCH

Transition takes place when there is a signal that orders UE to shift to CELL_PCH from UTRAN.

(4) $CELL_PCH \rightarrow CELL_FACH$

Transition takes place when there is a paging from UTRAN or when UE makes an uplink access.

(5) $CELL_FACH \rightarrow URA_PCH$

Transition takes place when there is a signal that orders UE to shift to URA_PCH from UTRAN.

(6) $URA_PCH \rightarrow CELL_FACH$

Shifts to CELL_FACH in the event of accessing the network while in URA_PCH status.

3.3.5.6 Control Scheme

Handover Control

(1) Handover Categories Handover is categorized as follows:

1. Inter-3G Handover

- Inter-FDD Handover
 - FDD soft/softer handover (Handover between the same frequencies)
 - FDD hard handover, between different frequencies
- Handover from FDD to TDD
- Handover from TDD to FDD
- Inter-TDD handover
- 2. Handover from 3G to 2G
- 3. Handover from 2G to 3G

The following description relates to 3G inter-FDD handover.

(2) Hard Handover (HHO)

HHO is activated by reports on measurement results from UE or on the basis of the network's initiative.

(3) Soft Handover

Soft handover is a handover scheme that uses macrodiversity when a radio link is to be established with a new Node B in the same frequency. Here, Active Set refers to all radio link groups associated with DHO.

Soft handover is based on the following process:

- Measurement
- Filtering of measurements
- Reporting of measurement results
- Soft handover algorithm
- Handover decision.

The measurement results from UE serves as the input into the soft-handover algorithm. A decision is made as to whether a new Node B should be added (Radio Link Setup/Addition), deleted (Radio Link Removal) or replaced (Radio Link Setup/Addition/ Radio Link Removal) and Active Set Update is executed. The soft handover algorithm is carried out by RNC.

When adding a new cell, the following information is notified by RNC to Node B and UE.

Radio Bearer Control

(1) Radio Bearer Setup

The establishment of a radio bearer requires the setup of RRC connection. After RRC connection setup, a Radio Bearer Setup message is sent from UTRAN to UE, and the common transport channel or the dedicated transport channel is set according to the required service.

(2) Physical Channel Reconfiguration

Physical Channel Reconfiguration is activated when reallocating spreading codes and at the time of HHO, depending on the transmitted data volume in uplink and downlink.

Control in the Event of Increase in Data

1. Transition from RACH/FACH to DCH/DCH (When there is an increase in UL data): UE in RACH/FACH state can transmit small volumes of user data on the common transport channel. If the data volume in the uplink buffer exceeds the threshold, it can shift to DCH/DCH state. UTRAN determines the transition of transport channel according to the load incurred by the system overall.

On the basis of System Information from UTRAN or Measurement Control, UE measures the data volume in the RLC buffer. If the data stored exceeds the threshold, UE notifies it to UTRAN by a Measurement Report. On the basis of the measurement results, UTRAN calculates the transmission capacity required by UE and decides the transition to a dedicated channel. If an adequate TF and TFC that can be used after channel switching has been set in advance, the Physical Channel Reconfiguration procedures would be applied.

2. If there are no changes to the transport channel [When there is an increase in DL (Down Link) data]: If there is an increase in downlink transmitted data, that is, if the data volume in the RLC buffer in the network exceeds the threshold, UTRAN changes the configuration of the physical channel. If UE is using dedicated channels, it reduces SF on the basis of the Physical Channel Reconfiguration procedures in order to increase the transmission rate of the physical channel. If no TF or TFC to be used after such change has been defined in advance, Transport Channel Reconfiguration procedures would be required.

Control in the Event of Decrease in Data

- If there are no changes to the transport channel (When there is a decrease in DL data): As in the case of increase in downlink data, if the data volume in the RLC buffer in the network side stays below the threshold over a certain period, UTRAN increases SF on the basis of the Physical Channel Reconfiguration procedures in order to decrease the transmission rate of the physical channel. The TF and TFC that are no longer used after changing SF are kept in order to deal with potential increases in data volume.
- 2. Transition from DCH/DCH to RACH/FACH: The network monitors and assesses the traffic from UE. If uplink traffic volume decreases and downlink traffic is limited, it

switches from a dedicated channel to a common channel. Monitoring of traffic from UE by the network may be on the basis of measurement of data received by the network or on the Measurement Report from UE.

If the configuration of an adequate RACH/FACH to be used after changing the channel has not been defined in advance, Transport Channel Reconfiguration would be applied. If it has been defined in advance, Physical Channel Reconfiguration procedures would be applied.

For example, if uplink and downlink traffic volume is limited, switching from a dedicated channel to a common channel takes place. As RACH/FACH configuration is already defined on the basis of the broadcast information, Physical Channel Reconfiguration procedures are applicable.

(3) Transport Channel Reconfiguration

If there are no Changes to the Transport Channel

If the volume of data stored in the RLC buffer exceeds the threshold, UE transmits a Measurement Report to UTRAN. UTRAN can increase the uplink speed of that UE after determining the load status in the network. To increase the uplink speed, a higher-ranking (small SF) code (SF = 4 in the case of uplink) can be used as the code tree used by UE is unique to that UE (there is no need to take interference between UEs into account because the scrambling code is different). Therefore, there is no need to assign a channelization code as in the case of downlink. The uplink speed can be changed by modifying and restricting the TF and TFC.

Transport Channel Reconfiguration or TFC Control message is used for altering the uplink speed. TFC Control message is used to impose and remove restrictions on the TFC. Transport Channel Reconfiguration message is used when it is necessary to change the TFCS.

Transition from DCH/DCH to RACH/FACH

The network monitors and assesses the traffic to UE. If downlink traffic volume to UE decreases and uplink traffic is also limited, it switches from a dedicated channel to a common channel. If the configuration of RACH/FACH to be used after changing the channel has not been defined in advance, Transport Channel Reconfiguration procedures would be applied. If it has been defined in advance, Physical Channel Reconfiguration would be applied.

For example, switching to a common channel takes place when UE moves from cell to cell during communication on a dedicated channel and uplink and downlink traffic volume is limited. As RACH/FACH configuration is not defined in the destination cell, Transport Channel Reconfiguration is applied and the common channel to be used is specified by Physical Channel parameters.

Radio Bearer Reconfiguration

In MAC, logical channels mapped over onto different radio bearers can be multiplexed on a transport channel. Radio Bearer Reconfiguration is used to change the MUltipleXer (MUX) configuration.

The Radio Bearer Reconfiguration message specifies the parameters relating to the MUX configuration and the reconfiguration (including deletion) of the transport channel in MAC. Other parameters relating to radio bearers are not changed.

3.3.6 Control Sequence

The 3GPP specifications stipulate the basic procedures by protocol such as RRC, NBAP (Node B Application Protocol), RNSAP (Radio Network Subsystem Application Protocol) and RANAP, but do not provide for the overall procedures. The sequence described below is an example of a sequence specified by the combination of basic procedures.

3.3.6.1 Description of Nodes

Nodes in the sequence are as follows:

- User Equipment (UE)
- Base Transceiver Station (Node B)
- Radio Network Controller (RNC)
- Circuit-Switched Core Network (CS-CN)
- Packet-Switched Core Network (PS-CN).

The RNC is further composed of the following functional entities:

• RACFa: RACFa is a functional entity that controls the terminals.

The ratio of generating RACFa to UE is 1:1.

RACFa generation is triggered by RRC connection setup and its release is triggered upon RRC connection release. In the event of SRNC relocation, RACFa of the relocation source is released. (RACFa corresponding to UE is sustained at the relocation destination.)

- *RACFc*: RACFc is a functional entity that receives CCCH (e.g. RRC CONNECTION REQUEST)
- RACFd: RACFd is a functional entity that performs control to BS.
- *RACFpg*: RACFpg is a functional entity that executes paging.

The ratio of generating RACFpg to RNC is 1:1.

If multiple Node Bs exist in the connection between UE and UTRAN, they are distinguished from each other e.g. Node B1 and Node B2.

If SRNC and DRNC (Drift Radio Network Controller) exist in the connection between UE and UTRAN, they are described separately from each other.

3.3.6.2 Description of Internode Messages

This sequence describes the messages of the following protocols:

- RRC (between UE and UTRAN);
- NBAP (between Node B and RNC);
- RNSAP (inter-RNC);
- RANAP (between RNC and CS-CN and between RNC and PS-CN);
- Q.aal2 (between NodeB and RNC, inter-RNC and between RNC and CS-CN) and
- Broadband ISDN User Part (B-ISUP) (between RNC and PS-CN).

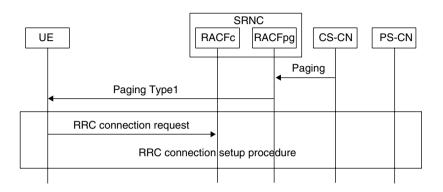
In the sequence, the message names (Q.aal2 and B-ISUP) are described for messages of Q.aal2 and B-ISUP in order to distinguish them from other messages. Messages between functional blocks of RNCs are shown by broken lines.

3.3.6.3 Sequence

Paging

(1) Paging to an Idle Mode UE (CS)

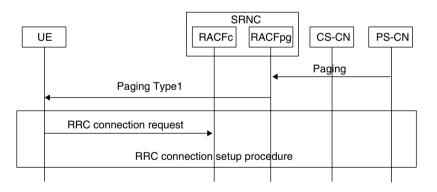
This refers to paging for an incoming call from a CS network to UE in idle mode.



- Incoming speech call to UE in idle mode.

(2) Paging to an Idle Mode UE (PS)

This refers to paging for an incoming call from a PS network to UE in idle mode.



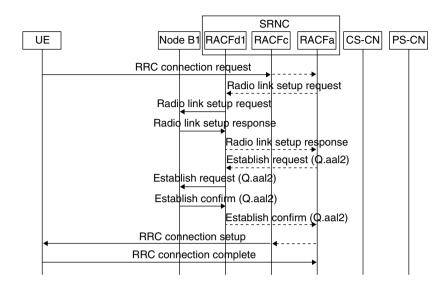
- Incoming packet call to UE in idle mode.

RRC Connection Setup

(1) RRC Connection Setup (CELL_DCH)

This refers to the settings of RRC connection using dedicated channels. The connection for signaling is established by this sequence.

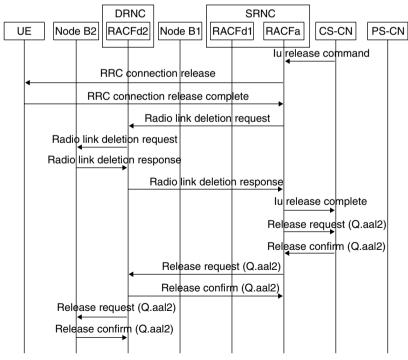
- When making an outgoing call from UE
- When an idle mode US receives paging.



RRC Connection Release

(1) RRC Connection Release (CELL_DCH) (CS)

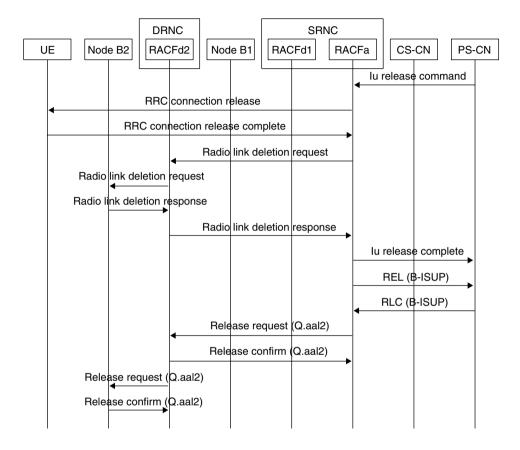
This refers to the release of RRC connection of dedicated channels activated by a CS network.



- When all calls to UE cease to exist following the release of speech calls

(2) RRC Connection Release (CELL_DCH) (PS)

This refers to the release of RRC connection of dedicated channels activated by a PS network.

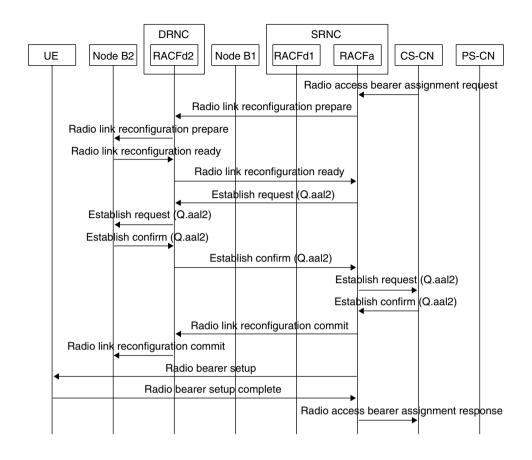


 When all calls to UE cease to exist following the release of packet calls using dedicated channels.

Radio Bearer Setup

(1) Radio Bearer Setup (CELL_DCH-CELL_DCH) (CS)

This refers to the settings of the radio access bearer from a CS network in CELL_DCH state.

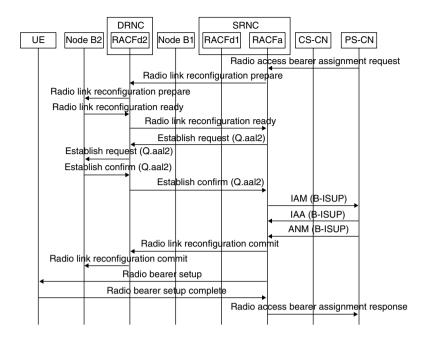


- Upon speech call connection setup, a CS network requests radio access bearer setup for user data after completing RRC connection setup for signaling.

(2) Radio Bearer Setup (CELL_DCH-CELL_DCH) (PS)

This refers to the settings of the radio access bearer from a PS network in CELL_DCH state.

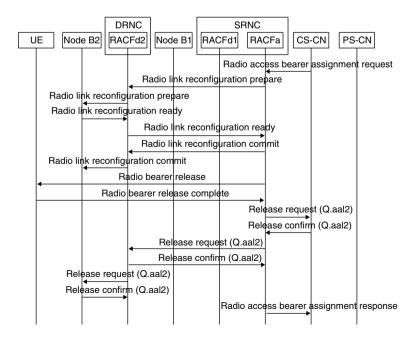
- Upon packet call connection setup, a PS network requests radio access bearer setup for user data after completing RRC connection setup for signaling.
- A PS network requests radio access bearer setup in cases in which a packet call is to be added during speech communications.



Radio Bearer Release

(1) Radio Bearer Release (CELL_DCH-CELL_DCH) (CS)

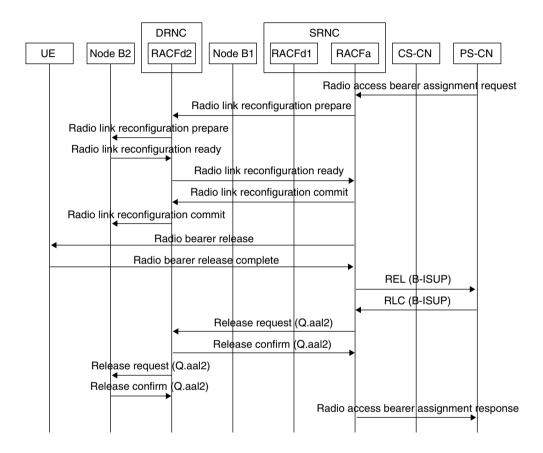
This refers to the release of the radio access bearer from a CS network (RRC status: $CELL_DCH \rightarrow CELL_DCH$).



 In a sequence of releasing one speech call in a multi-call state, dedicated channels are used even after releasing the speech call.

(2) Radio Bearer Release (CELL_DCH-CELL_DCH) (PS)

This refers to the release of the radio access bearer from a PS network (RRC status: CELL_DCH \rightarrow CELL_DCH).

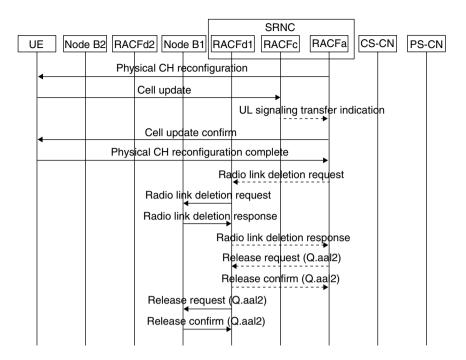


 In a sequence of releasing one packet call in a multi-call state, dedicated channels are used even after releasing the packet call.

Physical CH Reconfiguration

(1) Physical CH Reconfiguration (CELL_DCH-CELL_FACH)

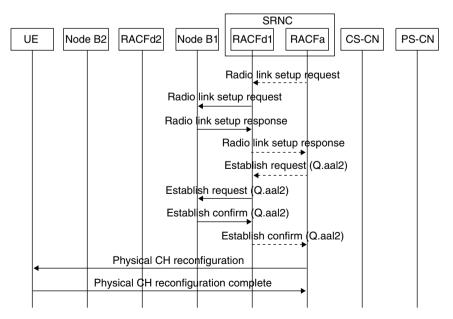
This refers to changing the settings of the physical channel. (RRC status: CELL_DCH \rightarrow CELL_FACH).



 If the packet call traffic falls below the threshold, switching from dedicated channel to common channel takes place.

(2) Physical CH Reconfiguration (CELL_FACH-CELL_DCH)

This refers to changing the settings of the physical channel. (RRC status: CELL_FACH \rightarrow CELL_DCH).

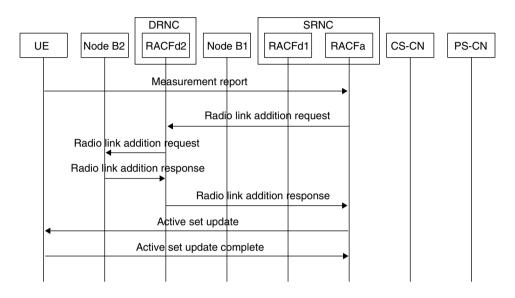


 If the packet call traffic exceeds the threshold, switching from common channel to dedicated channel takes place.

Radio Link Addition and Deletion

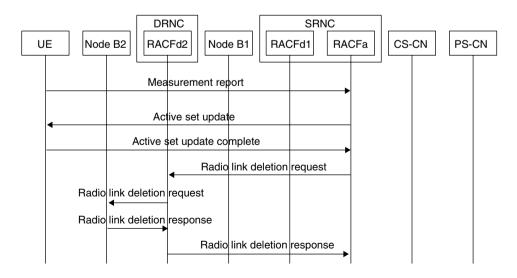
(1) Radio Link Addition (in Node B)

This refers to adding a radio link in Node B with a preexisting radio link at the time of executing softer handover.



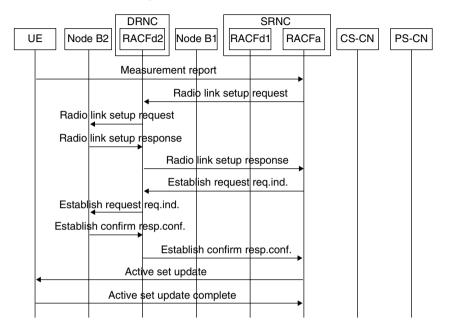
(2) Radio Link Deletion (in Node B)

This refers to releasing one of the radio links when a number of radio links exist in the same Node B in a soft-handover state.



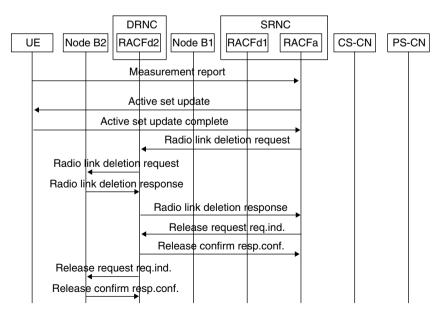
(3) Radio Link Addition (Internode B)

This refers to adding a radio link between Node Bs with no preexisting radio link with the UE at the time of executing softer handover.



(4) Radio Link Deletion (Internode B)

This refers to releasing a radio link with a specific UE that runs across Node B upon soft handover. (After release, no radio link will exist in Node B.)



3.4 Radio System Design

3.4.1 W-CDMA Radio System Design

As with conventional systems, the W-CDMA system is based on a cellular configuration in which the service area is covered by many radio zones ("cells"). W-CDMA is different from TDMA in that the same frequency carrier can be used in all cells, which eliminates the need for any frequency allocation design that had been required in FDMA and TDMA. Figure 3.68 is a conceptual diagram of frequency reuse. While any communications performed over the same frequency carrier in adjacent cells will normally cause interference, it is suppressed to an insignificant level in the despreading process at the time of demodulation in the W-CDMA system. Put differently, W-CDMA is a system that operates in an environment in which interference always exists (but at a level that does not affect quality) and enables cells to flexibly share a common frequency band without any special complex design or control.

As mutual interference between calls is inherent in the operation of the W-CDMA system, the amount of interference must be taken into account in the radio link design. The amount of interference is believed to depend heavily on the propagation status and traffic, thus it is necessary for the W-CDMA radio system design to consider radio propagation as well as traffic.

It must also be noted that the radio link design method differs between uplink and downlink. In uplink, BS receives signals transmitted from many MSs simultaneously, whereas in downlink, many MSs receive signals transmitted from BS *en bloc* at their respective locations. W-CDMA applies SIR-based TPC to both uplink and downlink. The mechanism of interference differs between the two: in uplink, the strength of signals from other users in the same BS are more or less the same, whereas in downlink, MSs closer to BS suffer from the interference caused by strong signals transmitted to MSs far away from BS. Other differences to be noted upon system design include the following: downlink requires orthogonal encoding, the presence of the Common Control Channel (CCCH) for transmitting broadcast information and paging information, the implementation of CPICH.

This section reviews the W-CDMA radio system design with reference to the relationship between traffic (capacity) and the amount of interference, in addition to differences between uplink and downlink. It also provides a brief description of the characteristics and noteworthy aspects of the cell/sector configuration in W-CDMA.

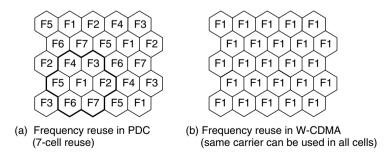


Figure 3.68 Conceptual diagram of frequency reuse

3.4.2 Concept of W-CDMA Capacity

3.4.2.1 Uplink

For the sake of simplicity, the concept of W-CDMA capacity will be described assuming that there are multiple users using the same service in uplink without any interference from other cells (i.e. isolated cell). Assuming that the number of users is N, the required average E_b/I_0 to achieve the required quality at the receiving side is $(E_b/I_0)_{req}$, the transmission speed of user information is R [bps], the ratio of chip rate B [cps] to Ris pg and the thermal noise power density is N_0 [W/Hz] (the thermal noise takes into account the noise factor NF of the receiver), the following equation must be upheld to satisfy the required quality.

$$\frac{E_{\rm b} \cdot R \cdot pg}{E_{\rm b} \cdot R \cdot (N-1) + N_0 \cdot B} = (E_{\rm b}/I_0)_{\rm req} \tag{1}$$

 $(E_{\rm b}/I_0)_{\rm req}$ can use the so-called link-level simulation results, which takes into account FEC, error-correction decoding, spreading and despreading. Upon the calculation of $E_{\rm b}$, it must be noted that pilot symbols, power control bits and the energy of parts other than the user information need to be included in the calculation. Marginal capacity C_0 can be calculated as follows, by solving the equation above with respect to N and attaining the extreme with respect to $E_{\rm b}$.

$$C_0 = \lim_{E_b \to \infty} N = \lim_{E_b \to \infty} \left(\frac{pg}{(E_b/I_0)_{\text{req}}} + 1 - \frac{N_0 \cdot pg}{E_b} \right) = \frac{pg}{(E_b/I_0)_{\text{req}}} + 1$$
(2)

As Equation (1) shows the required received power (E_b) rapidly increases and eventually diffuses in line with the increase in the number of users N; the marginal capacity referred to in Equation (2) is sometimes called *pole capacity*. In practice, however, it is impossible to infinitely increase received power. Because of the limits of transmission power of MS, the required quality cannot be achieved by MS near the edge of the cell (i.e. large propagation loss) as the amount of interference increases.

In order to study the capacity of W-CDMA in cases in which the amount of interference is limited, interference margin η is introduced, which indicates how large the total amount of interference (including thermal noise power) is in comparison to the thermal noise. The definition of η is as follows.

$$\eta = \frac{E_{\rm b} \cdot R \cdot N + N_0 \cdot B}{N_0 \cdot B} \tag{3}$$

Assuming that the amount of interference is limited, the capacity can be worked out on the basis of the equation below, after solving Equation (3) with respect to E_b and using Equation (1).

$$C = N|_{\eta} = \left(\frac{pg}{(E_{\rm b}/I_0)_{\rm req}} + 1\right) \cdot (1 - \eta^{-1}) = C_0 \cdot (1 - \eta^{-1}) \tag{4}$$

The relationship in Equation (4) shows that a larger η leads to a larger capacity C. However, this also implies that the reception power at BS becomes larger. In turn, the MS requires huge transmission power, as the traveling distance (cell radius) decreases if

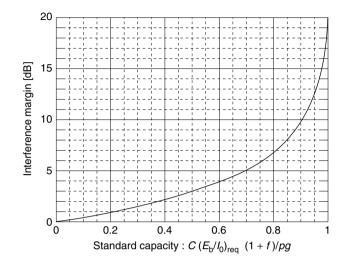


Figure 3.69 Relationship between interference margin and normalized capacity

the same transmission power is used. Put differently, there is a trade-off between capacity and cell radius (or transmission power) [7]. As such, interference margin η is an important parameter that determines both radio link design and capacity. Also, relationship $C/C_0 =$ $1 - \eta^{-1}$ can be proven from Equation (4). Accordingly, $1 - \eta^{-1}$ – which is calculated on the basis of the interference margin – can be interpreted as the equivalent traffic normalized by marginal capacity [8]. Figure 3.69 shows the relationship between normalized C and η . Table 3.22 shows the relationship between typical values of the interference margin and the normalized capacity. The interference margin is also referred to as the load margin.

These results are applied to a multiple-cell model. f is introduced as the ratio of interference power from other sectors to interference power in the own sector and the interference power from other sectors is indicated by $E_b \cdot R \cdot N \cdot f$. By calculating in the same way as working out Equation (4) from Equation (3), the following equation can be determined.

$$C = \left(\frac{pg}{(E_{\rm b}/I_0)_{\rm req}} + 1\right) \cdot \frac{1}{1+f} \cdot (1-\eta^{-1}) = C_0 \cdot \frac{1}{1+f} \cdot (1-\eta^{-1})$$
(5)

Parameter f may take various values, depending on distance factor of propagation loss, standard deviation of shadowing fluctuations and so on [9] Assuming the standard conditions in urban districts, it is around 0.7. Although the capacity determined in theory cannot be assured in practice, owing to TPC and other error factors, in addition to irregularities in the traffic distribution and radio wave propagation in the plane, Equation (5) is extremely useful as a basic equation that indicates the fundamental relationship between interference margin and capacity in W-CDMA. For the sake of simplicity, the system capacity was estimated using the average amount of interference. It should be noted that the amount of interference is constantly changing because of the number of users in connection, the positioning of each user, Voice Operated Transmission (VOX) and DTX. The accuracy

Interference margin	Capacity
1 dB	20% of marginal capacity
3 dB	50% of marginal capacity
6 dB	75% of marginal capacity
10 dB	90% of marginal capacity

 Table 3.22
 Relationship between typical interference margins and normalized capacity

of calculating the system capacity can be further improved by taking into account the characteristics of the changes in amount of interference [10, 11].

As explained in the preceding text, in W-CDMA uplink, the amount of uplink interference in BS has a great impact on the capacity and cell radius (radio link design). Accordingly, the interference margin is an important design parameter.

3.4.2.2 Downlink

On the other hand, in downlink, the total transmission power of BS itself is a resource that is shared by multiple users and the CCCH. As there is a limit to the total transmission power of BS, the power assigned to each channel and the number of channels that can be accommodated are limited. Therefore, important design parameters are the total transmission power of BS and the power allocation, especially the allocation of power to the CCCH [12].

In uplink, the capacity is determined by the power shared at the receiving side, where a certain level of quality is maintained through TPC by MS. In downlink, the capacity is determined by the transmission power, which depends on the location of MS and the amount of interference received. The calculation of downlink capacity is therefore relatively more complicated than uplink. The basic concept of downlink system capacity is described in Refs. [9] and [13]. In Refs. [9] and [13], the allocation of power to each CCCH and each user is calculated assuming that all BSs are executing transmission at the total transmission power, and the probability of the allocated power exceeding the original total transmission power is calculated. Then, the maximum number of users required to keep the probability below a certain level is calculated as the system capacity. The allocation of power to each user depending on the location of each MS is based on a model of cell/sector configuration and propagation loss and the Monte Carlo simulation is applied. Recent studies include attempts to analytically represent downlink system capacity [14, 15]. In any case, a scheme that can support versatile, dynamic designs should be implemented in actual system operations.

This section shows the basic concept of downlink system capacity. A certain MS receives a channel transmitted in unit power (1 W) from BS (total number of stations: J), assuming that r_1, r_2, \ldots, r_J represents the reception power in descending order. Assuming that the total transmission power of BS is P_{total} , and the proportion of power allocated to the subject channel in total transmission power is ξ , the received E_b/I_0 of the channel is

calculated by the equation below.

$$E_{\rm b}/I_0 = r_1 \xi P_{\rm total} \left[\frac{1}{pg} \left(r_1 P_{\rm total} \gamma + \sum_{j=2}^J r_j P_{\rm total} \right) + N_0 R \right]^{-1}$$
(6)

The above equation assumes that γ is an orthogonality factor ($0 \le \gamma \le 1$), and the use of the orthogonal codes reduces interference in the own sector. The value of γ , which depends on multipath conditions and so on, is believed to be around 0.5. By bringing $(E_b/I_0)_{req}$ to the left-hand side of Equation (6), and solving it with respect to ξ , the following equation will be attained.

$$\xi = \frac{(E_{\rm b}/I_0)_{\rm req}}{pg} \left(\gamma + \sum_{j=2}^{J} r_j / r_1 + \frac{N_0 B}{r_1 P_{\rm total}} \right)$$
(7)

Assuming that $\gamma = 1$, Equation (7) is the same as the equation for calculating the power allocation referred to in Refs. [9] and [13]. (Because of the difference in definition, it should be noted that $\phi\beta$ in Refs. [9] and [13] corresponds to ξ in this volume.) Assuming that $1 - \beta$ in total transmission power P_{total} is assigned to CPICH and CCCH for broadcast information, paging information and so on, and that the remaining β is shared by each user, system capacity *C* can be estimated by the following equation.

$$C = \beta / E[\xi] \tag{8}$$

As in Refs. [9] and [13], the capacity can be estimated on the basis of Equation (8) regardless of cell radius and other conditions as such, ignoring the thermal noise term. It is believed that thermal noise can normally be ignored because the amount of interference is larger in downlink than in uplink. In uplink, the signal power of many users is more or less at the same level, as BS receives signals from users *en bloc*. In downlink, the location of reception depends on each MS, operating under larger interference power. It is worth mentioning that the average value of ξ was used in Equation (8); by determining the distribution of ξ , it is possible to come up with a design that keeps the probability of total transmission power shortage (outage) below a certain level. The accuracy of calculating capacity can be further improved by taking DHO into account in Equation (7) for calculating ξ .

Recent link-level simulations for downlink widely use $\hat{I}_{or}/(I_{oc} + N_0)$ as the parameter for differentiating location-dependent interference, which is calculated using the power density when MS receives the total power transmitted from the own sector \hat{I}_{or} [W/Hz] the power density when MS receives the total power transmitted from all other sectors I_{oc} [W/Hz], and the thermal noise power density of the receiver N_0 [W/Hz] (assumings that the noise factor NF of the receiver is taken into account). The value of $\hat{I}_{or}/(I_{oc} + N_0)$ is large when MS is located near BS, because \hat{I}_{or} is large and I_{oc} is small. Conversely, the value is small when MS is located near the edge of the cell, far away from BS. As the value changes according to the location of MS, this parameter is sometimes referred to as *geometry* [16]. Equation (7) can be expressed in terms of geometry g in the following manner.

$$\xi = \frac{(E_{\rm b}/I_0)_{\rm req}}{pg} \left(\gamma + \sum_{j=2}^{J} r_j / r_1 + \frac{N_0 B}{r_1 P_{\rm total}} \right) = \frac{(E_{\rm b}/I_0)_{\rm req}}{pg} \left(\gamma + \frac{1}{g} \right)$$
(9)

Capacity C can be calculated by executing link simulation so that the required power allocation $\xi(g)$ for a certain g can be directly determined without evaluating received $(E_b/I_0)_{req}$, according to the equation below.

$$C = \frac{\beta}{\int_{-\infty}^{\infty} \xi(g) \cdot p(g) \, dg} \tag{10}$$

p(g) is the probability density distribution of the geometry, which depends on the cell/sector configuration and the propagation properties. p(g) can be determined through simulations based on models of cell/sector configuration and propagation properties. The accuracy of calculating the capacity can be further improved by regarding the denominator as a distribution, rather than an average value.

3.4.2.3 Admission Control

In W-CDMA systems, many users share the same frequency band, and each signal is statistically multiplexed without any special timing control and so on. Hence, the amount of interference might temporarily increase and as a result, the quality of communication for each user might fall below the required level, as far as probability is concerned. The admission of many users increases the probability of falling below the required quality. However, strict restriction on user admission reduces capacity, even though it reduces the amount of interference and sustains quality. In this manner, there is a trade-off between capacity and quality [10, 17]. Thus, in W-CDMA, admission control (which decides whether a new channel can be established or not) is regarded important for controlling capacity as well as the quality of communication for each user. In system design, it is vital to evaluate the properties not only in consideration of the so-called Erlang capacity, which is calculated by taking temporal traffic fluctuations into account, but also admission control [10, 17–19].

According to the capacity characteristics in W-CDMA described in the preceding text, it is believed that it would be adequate to determine uplink capacity on the basis of the amount of uplink interference (the total power received in the spreading band) of the BS, and downlink capacity on the basis of the total transmission power (in the applicable carrier) of the BS. In uplink, if the setup of a new channel is restricted by the amount of interference corresponding to the interference margin described in the radio link design section, cell breathing (described later) can be suppressed and the shrinking of the cell radius can be prevented even at times of heavy traffic. Of course, the system capacity can be increased by setting up the system so that it would actively admit users in excess of the interference margin of the link design, or by controlling the system so that it would admit only MSs close to BS that can withstand greater interference, although the cell radius would shrink and the QoS would deteriorate for surrounding MSs. On the other hand, in downlink, if the setup of a new channel is restricted so that the total transmission power of BTS would not exceed the maximum level, cell breathing can be suppressed. As in the case of uplink, system capacity can be increased by setting up the system so that it would actively admit users regardless of the total transmission power of BTS, or by controlling the system so that it would admit only MSs close to BS that require minimal

transmission power at the BS, even though the cell radius would be reduced in size and surrounding MSs would suffer from quality degradation.

Accordingly, admission control is capable of controlling the quality of communications, system capacity and even area coverage. While flexible operation is possible on the basis of parameter and algorithm settings, these properties are heavily affected by such settings and proper operation is constantly required. The evaluation of the so-called Erlang capacity, which simply takes into account variations in the number of users, is insufficient in assessing these properties—it is important to perform dynamic, system-level simulations, which simulate admission control or conduct evaluations considering the amount of interference reduced by admission control [8, 10].

3.4.3 Radio Link Design

3.4.3.1 Uplink Design

In uplink, BS receives signals from many users. There is a trade-off between system capacity and cell radius (interference margin): larger capacity requires a proportionate increase in the interference margin. The cell radius decreases in size unless there is an increase in the maximum transmission power of MS. The interference margin indicates the extent to which an increase in the amount of interference in dB is permissible relative to zero interference (i.e. when there is only thermal noise in the BS receiver). Radio link design for uplink is primarily concerned with setting the interference margin according to the required cell radius and traffic. The introduction of the interference margin enables the same method to be applied to uplink design as conventional radio link designs [20, 21]. The details are discussed in Section 3.4.3.3.

A larger interference margin assures larger capacity; however, the required transmission power of MS increases, as BS needs higher reception power. Conversely, a smaller interference margin reduces the transmission power of MS, but leads to smaller capacity. Put differently, cells with smaller radii increase the interference margin to expand capacity, whereas cells with larger radii decrease the interference margin at the sacrifice of capacity. These settings are made in consideration of the density of traffic demand and the required number of BSs and so on.

3.4.3.2 Downlink Design

In downlink, many users and common CCH share the total transmission power of BS. The link design for downlink is purely concerned with deciding the total transmission power of BS and the allocation of power to each channel. As the total transmission power of BS can only assume a specific set of values because of the configuration of equipment, in practice, the design is mainly concerned with setting the allocation of power according to the required cell radius and capacity relative to the predetermined total transmission power. Basically, if the cell radius is small, the allocation of power to common CCH is increased at the sacrifice of capacity. Hence, the relationship between cell radius and capacity is more or less the same as in the case of uplink design. Actual design resorts to not only simple link design schemes but also computer-based design tools for verifying the allocation of power and the amount of interference. The following

is a description of a scheme that incorporates the verification of total transmission power of BS and the allocation of power to common CCH [22], extending the basic concept of downlink capacity [9, 13]. The basic concept of a detailed design scheme for allocating power to common CCH will also be discussed, introducing parameters in consideration of coordination with link-level simulations.

On the basis of Equation (7) used in the explanation of capacity, the optimal design for total transmission power P_{total} and ξ is sought simultaneously. The following definitions are applied to Equation (7).

 $x = \gamma + \sum_{j=2}^{J} r_j / r_1$ (Ratio of total amount of interference from other sectors to interference of own sector)

$$1/y = \frac{(E_{\rm b}/N_0)}{(E_{\rm b}/N_0)_{\rm cell_edge}}$$
(Normalized received power) (11)
$$\eta = \frac{(E_{\rm b}/N_0)_{\rm cell_edge}}{(E_{\rm b}/N_0)_{\rm req}}$$
(Margin)

By rearrangement, the equation below is derived.

$$x = \xi \cdot \frac{pg}{(E_{\rm b}/I_0)_{\rm req}} \cdot \left(1 - \frac{y}{\eta}\right) = \xi \cdot (C_0 - 1) \cdot \left(1 - \frac{y}{\eta}\right) \tag{12}$$

Accordingly, assuming that P_{out} is the ratio of locations that cannot meet the required reception quality in the area with a specific power allocation ξ (outage probability by location), P_{out} can be determined as follows.

$$P_{\text{out}} = \operatorname{Prob}\left[x \ge \xi \cdot (C_0 - 1) \cdot (1 - y/\eta)\right]$$
(13)

As x and y depend on the cell/sector configuration and propagation characteristics, it would be adequate to evaluate the rate on the basis of simulation using model values for x and y. Figure 3.70 illustrates the relationship between interference margin η and power allocation determined by computer simulation. In the simulation, MS was placed at a specific location, x and y were calculated and Equation (12) was assessed. This process was repeated many times to determine P_{out} . The horizontal axis represents the power allocation normalized by pg and $(E_{\text{b}}/I_0)_{\text{req}}$. In practice, power is allocated such that ξ is halved when pg is doubled, and ξ decreases by 3 dB when $(E_{\text{b}}/I_0)_{\text{req}}$ is reduced by 3 dB. The vertical axis η is a parameter that relates to actual transmission power. Once the reception power at the edge of the cell, ξ and η are determined, the total transmission power required with the specified cell radius and the absolute value of the transmission power of the channel concerned are decided.

Similar to the *geometry* introduced in the capacity calculations, geometry is also applied to the allocation of power to common channels. This enables the direct application of link simulation results. With respect to the required outage probability by location P_{out} , G is determined according to the following equation.

$$P_{\rm out} = \int_{-\infty}^{G} p(g) \, dg \tag{14}$$

Link design is completed when the power allocation determined by link-level simulation $\xi(G)$ is set with respect to this. As changes in the total transmission power P_{total} are

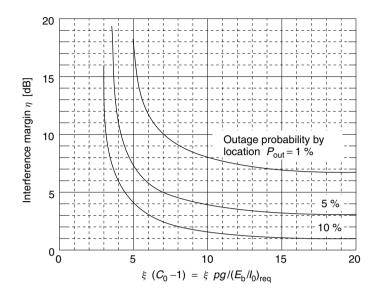


Figure 3.70 Relationship between interference margin η and normalized power allocation

reflected in geometry distribution p(g), the total transmission power must be altered to determine the design through trial and error. In many cases, the total transmission power of BS is determined at a relatively early stage, that is, in the design of the equipment itself. Hence, in the very early stages of scheme design, evaluations should be carried out using various different values for the total transmission power of the BS. However, in a routine system design, it is considered adequate to carry out system designs closely associated with link-level simulation using geometry distribution.

In radio link design for downlink, special attention needs to be paid to the power allocation of common pilots. Common pilots are used as a phase reference for demodulating each downlink channel, as well as for sector selection by MS. As insufficient power of the common pilot increases the channel estimation error upon demodulation and results in poorer quality of the demodulated channel, consideration in design must be given so that a certain amount of power would always be allocated. On the other hand, for MS to go into idle mode in a particular sector, the received power and quality of the common pilot must be above a predetermined level [23]. The level must be set adequately in consideration of the balance of both factors.

3.4.3.3 Example of Link Budget

Link budget is applied for making a rough estimation on the cell radius, as in the case of conventional systems. While the same method as those based on conventional link budgets can be applied to uplink with the introduction of the interference margin, the link budget cannot be used for downlink. This is because, in downlink, the interference power from the desired wave and other stations fluctuates dramatically depending on the cell radius and various losses, which makes it impossible to calculate the accumulated gain and losses. The following paragraph explains a simple way to confirm the allocation of power to downlink common CCH developed independently. An example of uplink link budget will be discussed first, followed by a simple way to confirm the allocation of power to downlink common CCH.

Example of Link Budget in Uplink

Table 3.23 shows an example of link budget in uplink in a W-CDMA system. The example assumes that the required E_b/I_0 is the Vehicular A channel specified by ITU-R [24, 25] (maximum Doppler frequency 222 Hz equivalent to 120 km/h), citing Ref. [26]. DHO gain is an item that takes into account the macrodiversity gain from soft handover in the link budget. Tx Power Increase is an item for considering the fluctuations in transmission power through fast TPC as the margin.

It should be noted that the propagation loss differs by about 9 dB between the conventional 900-MHz-band TDMA system and the 2-GHz-band W-CDMA system. Nonetheless, W-CDMA can have a cell configuration with coverage equivalent to the conventional system, because of the shorter wavelength, which enables the use of a higher-gain ANT of the same size, higher required sensitivity assured by high-performance error-correction technology and improved quality through the application of RAKE receiving, DHO and various other technologies.

Service Quality of service	12.2 k-speech BER 1e-3	64 kbps BER 1e-6	384 kbps BER 1e-6
(a) Max TX Power per TCH (dBm)	21.00	24.00	24.00
(b) Tx cable loss (dB)	0.00	0.00	0.00
(c) Tx ANT gain (dBi)	0.00	0.00	0.00
(d) Tx EIRP per TCH = $(a) - (b) + (c) (dBm)$	21.00	24.00	24.00
(e) Rx ANT gain (dBi)	17.00	17.00	17.00
(f) Rx cable loss (dB)	1.00	1.00	1.00
(g) Rx Nf (dB)	5.00	5.00	5.00
(h) Thermal noise density (dBm/Hz)	-174.00	-174.00	-174.00
(i) Interference margin (dB)	6.00	6.00	6.00
(j) Total noise+ Interference	-163.00	-163.00	-163.00
density = $(g) + (h) + (i) (dBm/Hz)$			
(k1) Information rate (kbps)	12.20	64.00	384.00
(k2) Information rate (dBHz)	40.86	48.06	55.84
(1) Required $E_b/(N0 + I0)$ (dB)	6.10	3.80	2.70
(m) Rx sensitivity = $(j) + k2 + (1)$ (dB)	-116.04	-111.14	-104.46
(n) DHO gain (dB)	3.00	3.00	3.00
(o) Log-normal fade margin (dB)	5.30	5.30	5.30
(p) Tx power increase	2.00	2.00	2.00
(q) Building penetration loss (dB)	6.00	6.00	6.00
(r) Antenna tilting compensation (dB)	0.00	0.00	0.00
(s) Max. Path Loss = $(d) + (e) - (f) - (m) + (n) + (n) - (m) + (n) + ($	142.74	140.84	134.16
(o) - (p) - (q) - (s) (dB)			
(t) Max. Range (km)	1.88	1.66	1.06

 Table 3.23
 Example of link budget in uplink

Example of Link Budget in Downlink

As mentioned earlier, conventional link budget cannot be applied to downlink. Although link budget can be applied in cases in which there is no interference assuming the limits to the cell radius [26], link budget in consideration of the interference from other cells and the interference from other users in the same cell is somewhat complex compared to uplink. Moreover, special consideration must be given to common pilots.

The following paragraph reviews an example of link budget of common pilot and downlink common CCH. It is assumed that the link budget of the common pilot is designed so that the common pilot could be received at the edge of the cell at a predetermined reception power. For the downlink common CCH, the procedures of strictly designing the power allocation by link-level simulation and system-level simulation were explained in Section 3.4.3.2. The technique explained in this section may be regarded as a simpler way to confirm its validity. It should be noted, however, that certain assumptions are made to simplify the calculations, as explained below.

Table 3.24 shows an example of a link budget of a downlink CPICH. By design, the power of the transmission source is determined so that the reception power at the edge of the cell reaches a predetermined level (in this example, -115 dBm) [23]. Hence, the concept is exactly the same as conventional link budgets. As a matter of course, the power must be checked simultaneously as to whether it is insufficient or not as a phase reference for demodulating each channel and if it is insufficient, the power must be further increased. As explained before, this must be carefully studied to a certain degree by link-level simulation and so on.

Table 3.25 shows an example of link budget of other downlink common CCH. The orthogonality factor is a parameter reflecting that the interference in the own sector is reduced by downlink orthogonality. When interference in the own sector is zero at perfect orthogonality, the orthogonality factor is also set at 0; otherwise, the orthogonality factor is set at 1. Other-sector interference coefficient is a parameter indicating the size of the interference from other sectors at the edge of the cell relative to the power from the own

		Definition	CPICH
(a) Tx	power of CPICH [dBm]		33.00
(b) Tx	feeder loss [dB]		3.00
(c) Tx	Ant gain [dB]		17.00
(d) CP	PICH Tx EIRP [dBm]	(a) - (b) + (c)	47.00
(e) Rx	Ant gain [dB]		0.00
(f) Rx	cable loss [dB]		0.00
(g) Re	quired Rx Lev [dBm]		-115.00
(h) DH	HO gain [dB]		0.00
(i) Lo	g-normal fade margin [dB]		5.30
	power increase [dB]		0.00
(k) Bu	ilding penetration loss [dB]		6.00
(l) Ma	ax.allowable path loss [dB]	(d) - (g) + (e) - (f) + (h) - (i) - (j) - (k)	150.70
	ax.range [km]	According to Ref. [27], etc.	2.96

 Table 3.24
 Example of link budget of downlink common pilot channel

		Definition	ССРСН
(a)	Total Tx power of BTS [dBm]		42.00
(b)	Tx power of channel [dBm]		36.00
(b1)	The ratio of channel [%]		25.12
(c)	Tx feeder loss [dB]		3.00
(d)	Tx Ant gain [dB]		17.00
(e)	Total Tx EIRP [dBm]	(a) - (c) + (d)	56.00
(f)	Channel Tx EIRP [dBm]	(b) - (c) + (d)	50.00
(g)	Rx Ant gain [dB]		0.00
(h)	Rx cable loss [dB]		0.00
(i)	Thermal noise density [dBm/Hz]		-174.00
(j)	Receiver noise figure [dB]		5.00
(k)	Symbol rate [ksps]		15.00
(1)	Symbol rate [dBHz]	10*LOG((<i>k</i>)*1000)	41.76
(m)	Noise power [dBm]	(i) + (j) + (l)	-127.24
(n)	Chip rate [Mcps]		3.84
(0)	Other-sector interference coeff. [dB]		8.00
· I /	Required Es/Io [dB]		7.00
· •	Orthogonality factor		0.50
	Required Rx Lev [dBm]	•	-116.95
	DHO gain [dB]		0.00
(t)	Log-normal fade margin [dB]		5.30
(u)	Tx power increase [dB]		0.00
(v)	Building penetration loss [dB]		6.00
	Antenna tilting compensation [dB]		0.00
	Max. allowable path loss [dB]	(f) - (r) + (g) - (h) + (s) - (t) - (u) - (v)	155.65
(y)	Max. range [km]	According to Ref. [27], etc.	4.12

 Table 3.25
 Example of link budget of downlink common control channel

sector. Here, other-sector interference coefficient is the reciprocal of G corresponding to the prescribed P_{out} using the geometry referred to in Section 3.4.3.2.

3.4.4 Cell/Sector Configuration

This section explains matters requiring attention in W-CDMA cell configuration.

3.4.4.1 Impact of BS Location

As there are no limits to frequency reuse in W-CDMA, it is strong against shifts from the regular cell layout. The overall efficiency of the system can be improved by allocating BSs according to the traffic distribution rather than deploying them regularly.

3.4.4.2 Sector Configuration

Sectoring technologies configure cells based on multiple directional ANTs at one BS. They have been adopted as technologies for efficient frequency usage, effectively shrinking the size of cells without changing the number of BSs in FDMA and TDMA systems. In systems that assume frequency reuse, the reduction in the number of cells in which the same frequency is reused because of sectoring leads to an increase in spectrum utilization, but the effects are not necessarily proportionate to the number of sectors. W-CDMA is distinctive in that the capacity expansion effects are more or less proportionate to the number of sectors, because the same frequency an be used in adjacent cells. Strictly speaking, the effects are not exactly proportionate because of changes in the so-called other-cell (sector) interference, depending on the number of sectors and the properties of directional ANTs, but it is fair to say that sectoring is more effective. Furthermore, more sectors normally mean the application of ANTs with narrower beams, which has the merit of increasing the ANT gain.

In a sector configuration, the ANT beam width is optimized at a point that is narrower than 360° divided by the number of sectors [27]. However, excessively narrow beams must be avoided in light of problems in coverage and difficulties in ANT implementations.

3.4.4.3 BS Antenna Tilting Angle

Normally, BS ANTs are tilted vertically to reduce interference. If the ANT goes out of the line of sight of the edge of the cell because of tilting, it has to be taken into account in link budget generally in the form of tilt compensation. As transmission power is related to capacity in W-CDMA, more tilting aimed at lowering interference does not necessarily help increase capacity, as the amount of power compensation increases. The optimal tilting angle is believed to be more or less toward the edge of the cell.

3.4.4.4 Cell Breathing

As the cell radius relates to capacity, a phenomenon called *cell breathing* occurs, which refers to virtual changes in the cell radius depending on traffic. Although the cell radius might automatically be adjusted by traffic, it is regarded problematic for assuring stable area quality. This phenomenon can be controlled by schemes to determine whether UE is out of the coverage or by call admission control. For example, when UE out-of-coverage determination is done by SIR, cell breathing is observed; however, cell breathing can be avoided by determining the out-of-coverage status using an absolute reception level assuming a predetermined capacity. Also, a horizontal area coverage can be further assured by executing admission control so that no calls would be admitted after the capacity is exceeded.

3.4.4.5 Layered Cell Configuration

Cells of different sizes (e.g. macrocells, microcells, picocells) and indoor cells can be implemented in layers, overlapping each other. To overlay cells, a different frequency must be applied to each layer. In cases in which capacity is an issue, it is basically more effective to actively introduce smaller cells and microcells, rather than overlaying macro and microcells. In cases in which large cell coverage is required in light of economy, overlay operations may be necessary. Picocells, which are indoor cells, are effective especially in high-rise buildings where substantial interference is exposed to other cells because the interference signal for the outside environment reaches in the far distance. However, when they are implemented at different frequencies, they would be effective only if their effects supersede the reduction in the number of carriers operable in macro cells, as the operation involves frequency division. Therefore, their effects must be assessed in consideration of the number of carriers operable in the system.

3.5 Radio Access Network Equipment

3.5.1 Overview of System Configuration of Radio Access Equipment

Section 3.2.3 discussed the architecture under the standard. This section describes the concrete examples of system configuration and equipment configuration based on the logical architecture. Figure 3.71 shows an example of W-CDMA system configuration. Radio access equipment consists of the UE, BTS, the Radio Network Control equipment (RNC) and the Multimedia signal-Processing Equipment (MPE). Although the BTS is called Node B as a logical node in terms of architecture, it is referred to as BTS as a physical device in this figure. Signal processing functions of MPE may be equipped in RNC, as it constitutes part of RNC in terms of architecture – however, the signal-processing device is depicted as a separate piece of equipment in this network configuration. As some of the signal-processing functions of CN are also collectively equipped in MPE, the MPE is connected with the local switch as well, as illustrated in Figure 3.71. CN is based on an example of the integrated CS and PS physical configuration using the Asynchronous Transfer Mode (ATM) technology referred to in Section 4.2.

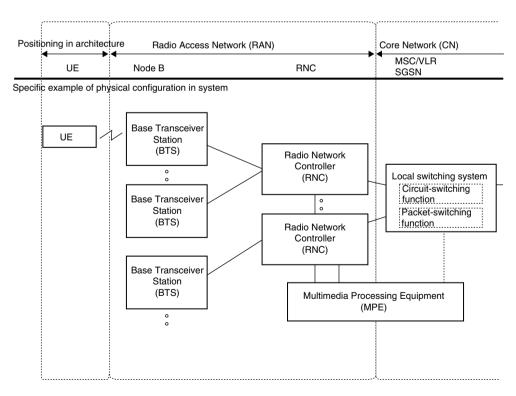


Figure 3.71 W-CDMA radio system configuration (example)

3.5.2 BTS

3.5.2.1 Functional Configuration

Figure 3.72 shows the functional configuration of BTS. BTS consists of the Open Air Receive Amplifier (OA-RA), the Open Air Receive Amplifier Supervisory Controller (OA-RA-SC), the AMP and the Modulation and Demodulation Equipment (MDE). MDE consists of functional modules including the Transmitter/Receiver (TRX), the controller, the highway interface and the Base Band signal-processing unit (BB). AMP, OA-RA and TRX are configured as independent units for each sector, whereas other functional modules of the MDE are shared by multiple sectors.

3.5.2.2 Basic Specifications of BTS

Table 3.26 shows the basic radio specifications of BTS. The BTS radio characteristics specifications are compliant with TS25.104 "UTRA (BS) FDD; Radio Transmission and Reception" and TS25.141 "Base station conformance testing (FDD)" prepared by the 3GPP Terminal Specification Group (TSG) RAN Working Group (WG) 4. Further efforts are being made to increase the number of carriers and the channel capacity through miniaturization, less power consumption and further circuit integration.

3.5.2.3 Key Technologies in Each Functional Block

The key functions of BTS are outlined below.

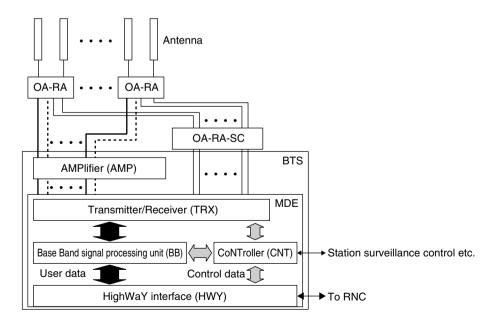


Figure 3.72 Functional configuration of Base Transceiver Station (BTS) (example)

ltem number	Item	Spe	ecifications
1	Radio access system	DS-CDMA FDD	
2	TX/RX Frequency band	IMT-2000 Band	
3	TX/RX Frequency spacing	190 MHz	
4	Carrier spacing	200-kHz Carrier raster	
5	Chip rate	3.84 Mcps	
6	Symbol rate	7.5 ksps \sim 960 ksps	
7	Modulation scheme		, spreading modulation: QPSI
8	Demodulation scheme	Pilot-symbol aided coher	rent detection
9	Data transmission speed	Max. 384 kbit/s (2 Mbi	t/s)
10	Max. transmission power	20 W \pm 2 dB/carrier/sector (10 W \pm 2 dB/carrier/sector/antenna when subject to transmission diversity)	
11	Frequency stability	± 0.05 ppm or less	
12	Occupied bandwidth	5 MHz or less (99% bar	ndwidth)
13	Adjacent Channel Leakage Power Ratio (ACLR)	5 MHz off-carrier	45 dB or higher/3.84 MHz bandwidth
		10 MHz off-carrier	50 dB or higher/3.84 MH bandwidth
14	Spurious emission	$9\sim 150~{ m kHz}$	-13 dBm (1 kHzBW)
		150 kHz \sim 30 MHz	-13 dBm (10 kHzBW)
		30 MHz 1 GHz	-13 dBm (100 kHzBW)
		$1 \sim 12.75 \text{ GHz}$	-13 dBm (1 MHzBW)
		However,	
		$1893.5 \sim 1919.6 \text{ MHz}$	-41 dBm (300 kHz BW)
		$1920 \sim 1980 \text{ MHz}$	-96 dBm (100 kHz BW)
		frequency bands, the s	overlapping in the above strictest figure is applicable ter frequency is less than
15	Transmit intermodulation	<i>,</i>	CLR and spurious emission
		must be satisfied when entering a signal ± 5 MHz, ± 10 MHz or ± 15 MHz offset from desired wave	
16	Roll-off factor	and lower by 30 dB (W-CDMA modulated wave). Square root raised cosine Nyquist filter $\alpha = 0.22$	
10	Modulation accuracy	17.5% RMS or less	e Nyquist filter $\alpha = 0.22$
18	Reference sensitivity	Data rate	12.2 kbit/s
10	Reference sensitivity	Input signal level	-121 dBm
		Bit error rate	10^{-3} or less
19	Spurious emissions in a receiver	9 kHz – 1 GHz	-57 dBm (100 kHzBW)
		1–12.75 GHz	-47 dBm (1 MHzBW)
		However,	
		1920–1980 MHz	-78 dBm (3.84 MHzBW)

 Table 3.26
 Basic specifications of base transceiver station (BTS)

Note: RMS: residual vector error.

AMP

AMP collectively amplifies the power of the output signals of the MDE (multiple codemultiplexed signals, multiple carriers) up to the required ANT input level. The gain is about 40 to 50 dB. Since the 3GPP specification requires to satisfy an ACLR of 45 dBc at 5 MHz off-carrier during multicode and multicarrier transmission; hence, a common AMP with superior linearity is necessary. Distortion compensation technologies [28] for common AMPs include feed forward [29] and predistortion. Feed forward (Figure 3.73) is the dominant technology due to its high distortion compensation performance. The reduction in the size of AMP has been achieved by creating a common AMP meeting these requirements. Predistortion is also expected to help achieve higher efficiency.

OA-RA and OA-RA-SC

The receive AMP used in OA-RA normally consists of Low Noise Amplifiers (LNA) in parallel configuration aimed at improving reliability. The gain is around 40 dB, and power is fed on the basis of the phantom power supply method, which involves the use of high-frequency coaxial cables in light of economy. This requires a power separating filter, coaxial arrestors and other surge-protection measures are to be taken as it is prone to power surge caused by lightning. Figure 3.74 shows an example of an OA-RA connection configuration at a station where transmission diversity is applied. Since low-loss, small duplexer and reception filter are used, a two-branch support OA-RA equipment with *NF* of 3 dB or less is realized in a size of approximately 15 liters. Provisions for spurious emissions fully take into account the Personal Handy phone System (PHS) frequency band and so on.

TRX

TRX converts transmission signals spread in base band from digital to analog, converts them into Radio Frequency (RF) signals though quadrature modulation, quasi-coherently detects the received signals from OA-RA, converts them from analog to digital and sends them to BB. TRX is configured with an independent unit for each sector. It has a redundancy configuration with a spare TRX, and the spare TRX automatically takes over in the event of any problems. One TRX cabinet can handle up to 6 sectors.

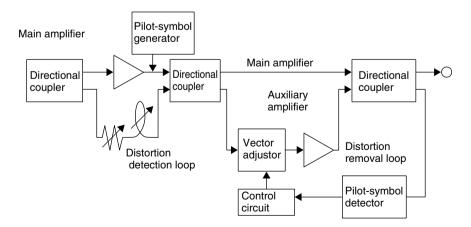


Figure 3.73 Basic configuration of self adjustment feed forward amplifier (example)

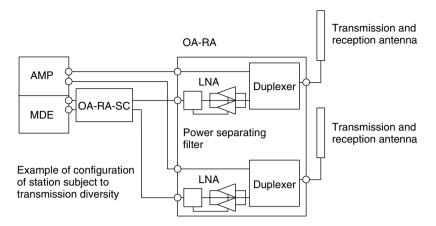


Figure 3.74 Basic configuration of Open Air Receive Amplifier (OA-RA) (example)

BB

BB is a functional unit responsible for the FEC, framing, data modulation, spreading modulation of transmitted signals and despreading, chip synchronization, error-correction decoding, data multiplexing/demultiplexing, MRC upon diversity handover from sector to sector, and other signal processing of the received signals.

In contrast with TRX, which has a separate hardware for each sector, the hardware resources of the base band signal-processing card may be assigned to any sector or carrier. By sharing the card in BTS, it is possible to flexibly assign channels to cope with uneven services and traffic. Even the BTS introduced first was able to offer a base band processing capability of more than 720 speech channels per cabinet, which makes it possible to operate the system at full radio channel capacity in a two-carrier, three-sector configuration. Further efforts are being made to achieve higher density and lower power consumption, aimed to further expand capacity.

Highway Interface

BTS and RNC are interconnected via a 1.544 Mbps, 6.312 Mbps or ATM mega-link transmission line. Highly efficient transmission of control data and user data is achieved by ATM transmission technology. The highway interface has an ATM processing function, the signal processing function of ATM Adaptation Layer (AAL)-Type-2 and Type-5 and the Service Specific Connection Oriented Protocol (SSCOP) function. The Highway (HWY) interface also supplies various reference clocks and reference timing phasing functions required for the BTS operation, as well as a time stamp function for internode synchronization between BTS and RNC.

Control Function

This unit is responsible for transmitting/receiving call-control signals to/from RNC, managing radio channels and setting/releasing radio channels. The control software (CC, maintenance monitor and control) and various system parameters can be stored in a nonvolatile memory via a PCMCIA (Personal Computer Memory Card International Association) Card, so that software upgrading can be managed at a central location. Also, the layer function and the application software are stratified to enable the development of applications independently of each other, which makes functional enhancement or modification easier. The maintenance interface function is a standard one like Common Object Request Broker Architecture (CORBA). It also has an Initial Program Loader (IPL) function and a remote file transfer function and is capable of monitoring and controlling the status of the card mounted in BTS, as well as monitoring and controlling station information outside BTS and peripheral equipment through an external monitor/control interface.

3.5.3 RNC

RNC has a control signal processing function, Operation and Maintenance (O&M) function, common channel multiplexing/demultiplexing function, ATM switching function, diversity-handover function and so on. RNC, which is connected with the local switch, MPE and BTS, performs radio link connection control and handover control.

Figure 3.75 illustrates the configuration of RNC and each function of RNC as one block. Table 3.27 describes the processing tasks of each functional block in brief. It should be noted that Figure 3.75 is a functional block diagram; in practice, the hardware configuration may be such that multiple functions are implemented in one hardware or software.

RNC needs to flexibly accommodate a wide range of areas, from traffic-intensive cities to suburbs. Accordingly, it must be able to process at least tens of thousands of Busy Hour Call Attempts (BHCA), have a switching capability of at least several Gbit/s, be able to accommodate dozens of BTSs, and be flexible enough to adapt to area designs. Also, call connection processes should not be the only focus-sufficient consideration must be given to O&M as well. RNC therefore has a standard O&M interface function like CORBA.

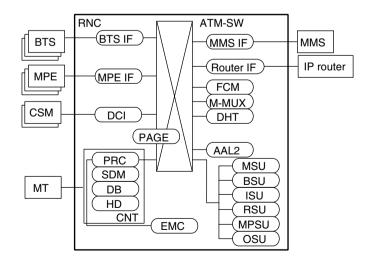


Figure 3.75 Functional configuration of RNC

Name	Overview of functions	
ATM-SW	ATM Switch	
BTSIF	BTS InterFace	
Switch IF	Switch InterFace	
MPE IF	MPE InterFace	
PAGE	Paging signal processing unit	
FCM	Frame Clock synchronization Module	
DHT	Diversity Handover Trunk	
AAL2	AAL2 cell multiplexer and demultiplexer	
CNT	CoNTroller	
PRC	ProCessor	
SDM	System Data Module	
DB	DeBugger	
HD	Hard Disk	
MSU	Mobile Signaling termination Unit	
BSU	BTS Signaling termination Unit	
ISU	Iu interface Signaling termination Unit	
RSU	RNC Signaling termination Unit	
MPSU	MPE Signaling termination Unit	
OSU	Operation system Signaling termination Unit	
Router IF	IP Router InterFace for surveillance	
M-MUX	MAC MultipleXer	
DCI	Digital clock supply Clock Interface	
EMC	Emergency Controller	

 Table 3.27
 Overview of RNC functional blocks

3.5.4 MPE

MPE has packet signal processing functions including the protocol conversion function for PS data, as well as speech signal processing functions to convert speech data from Adaptive MultiRate (AMR) to μ -law Pulse Code Modulation (PCM) and vice versa. As illustrated in Figure 3.71 of Section 3.5.1, the packet signal processing function constitutes part of RAN and the RNC function is implemented in MPE, which is a physically separate device.

Hence, packet signal processing takes place via connection with multiple RNCs. Hardware resources for signal processing are centralized in MPE and can be shared by multiple RNCs. Signal processing functions for circuit switching services, such as speech signal processing, are regarded as CN functions in the standard specifications, and are therefore performed in connection with the local switch. As MPE is a piece of hardware that integrates the signal processing functions of RAN and CN, it is able to perform both functions with a single device.

Figure 3.76 illustrates the functional configuration of MPE. It shows the functional configuration rather than the hardware configuration. Table 3.28 describes the processes carried out by each functional block in brief.

MPE must be able to process at least hundreds of thousands of BHCA, have a switching capability of at least several Gbit/s and be able to accommodate dozens of RNCs, in order to flexibly adapt to a wide range of areas.

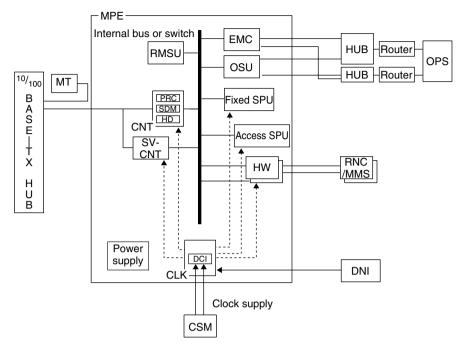


Figure 3.76 Functional configuration of MPE

Table 3.28	Overview of functions of each functional block of MPE
Name	Description
CNT	CoNTroller
RMSU	RNC/MMS Signaling termination Unit
SV-CNT	SuperVisory CoNTroller
SPU	Signal Processing Unit
HW	HighWay (RNC/MMS transmission line termination)
Internal bus o switch	r Signal transmission unit inside equipment
CLK	CLocK (generates the reference bit clock)
Power supply	Power supply
OSU	Operation system Signaling termination Unit

Emergency Controller

Table 2 28 f functions of each functional black of MDE

MMS: Multimedia Messaging Service.

3.5.5 BS Antenna

EMC

3.5.5.1 BS Antenna for IMT-2000

In the Personal Digital Cellular (PDC) system based on 0.9 GHz and 1.5 GHz bands, the radius of the radio zone is less than 1 km in some urban areas, with a few places to install more BS ANTs. As IMT-2000 uses the 2.0 GHz frequency band, it is necessary to install new BS ANTs. BS ANT design therefore requires miniaturization (smaller ANT diameter) to minimize the mechanical stress incurred when the ANT is mounted, as well as frequency sharing aimed at reducing the number of ANTs that need to be installed. IMT-2000 BS ANT design is also distinctive in that the horizontal directive pattern (sectoring) is heavily dependent on subscriber capacity. Figure 3.77 compares the relationship between the number of sectors and subscriber capacity in W-CDMA with the ideal fan-shaped ANT pattern, using the directivity represented by $f(\theta)$, which shows that more sectors leads to larger subscriber capacity. In practice, however, subscriber capacity is about 20% smaller than in the case of ideal fan-shaped ANT pattern as a result of overlap with adjacent sectors in the horizontal direction. The beam widths assumed in the 3-sector radio zone configurations are 120° and 60°, respectively.

Figure 3.78 shows the relationship between a basic ANT arrangement and beam width. In case A, one dipole ANT is placed in the middle of a flat reflector (spacing with ANT = 0.25 wavelength) and the beam width is 120° with a reflector of width 0.7 wavelength. The beam does not become narrower any further even if the width of the reflector is increased. In case B, two dipole ANTs are placed on a flat reflector with a spacing of 0.5

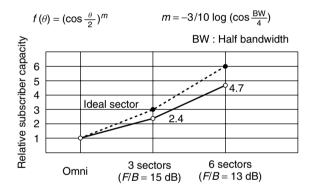


Figure 3.77 Number of sectors and subscriber capacity

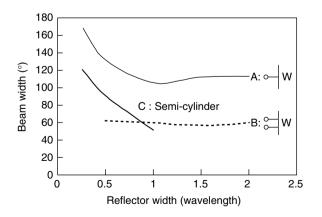


Figure 3.78 Reflector width and beam width

wavelength in between and the waves are combined at the same phase shift and amplitude (spacing with ANT = 0.25 wavelength). The beam width is about 60° , almost independent of the width of the reflector. As the beam width does not depend on the width of the ANT reflector, a reflector with a width of 0.5 wavelength or so might be sufficient to decrease the radius, but the width must be 0.7 to 1.0 wavelength because the front-to-back ratio of horizontal directivity deteriorates to about 13 dB. In case C, the reflector is shaped in a semi-cylinder, with one dipole ANT placed in the middle of the cylinder. The width of the reflector refers to the width of the aperture (diameter) in this case. As illustrated in the figure, a larger diameter results in a narrower beam. For example, a beam that is 120° wide can be generated by a reflector with a diameter of 0.25 wavelength, which means that the equivalent radius can be reduced more than in case A. Moreover, a beam that is 60° wide can be generated by a reflector with a diameter of 0.8 wavelength, which is more or less the same dimensions as in case B. In general, the ANT arrangement in cases A and B is covered by a dielectric radar dome (radome) to improve its mechanical robustness, enhance its durability against weather, reduce the load from wind pressure and so on. Accordingly, the ANT dimensions are normally assessed on the basis of equivalent radius. For a BS ANT designed to generate beams with a width of 120°, the figure shows that the ANT arrangement in case C is the smallest.

The beam width can also be controlled by placing the reflector at an angle, to make a corner-reflector ANT. Corner-reflector ANTs can reduce the beam width by narrowing the angle, although optimization (smaller ANT diameter) is required for each beam width.

3.5.5.2 Frequency Sharing

IMT-2000 dedicated ANTs need have smaller radii in order to reduce the load incurred by wind pressure. As mentioned earlier, one way to do this is to effectively reduce the number of ANTs that need to be installed by resonating multiple frequencies with one ANT.

As PDC has already achieved frequency sharing for 0.8 GHz and 1.5 GHz bands, this section will concentrate on describing the structure and properties of ANTs for sharing three frequency bands, namely, 0.8 GHz, 1.5 GHz and 2.0 GHz.

First, three-frequency-sharing ANT with a horizontal beam width of 120° will be discussed. Efforts were made to determine how a beam width of 120° could be achieved in the 2.0 GHz band on the basis of the ANT structure shown in Figure 3.79, designed for PDC. Figure 3.80 illustrates the beam width against frequency, which shows that a beam width of 120° can be achieved in the 2.0 GHz band as well. While corner-reflector ANTs normally tend to generate narrower beams at higher frequencies, this ANT structure is distinctive in that the beam width properties are relatively stable against the frequency. The ANT structure assures that the equivalent cylindrical diameter is no different from conventional PDC systems by attaching parasitic elements for the 2.0 GHz band in the front panel of the two-frequency-sharing ANT. The merit of this arrangement is that the load incurred by wind pressure remains the same even if the ANT is replaced. In addition, a reflector and a parasitic element short-circuited on one side are attached to both ends of each radiating element in order to reduce the return loss in the 0.8 GHz band.

Figure 3.81 illustrates the return loss against the frequency, showing that resonance occurs in each frequency band.

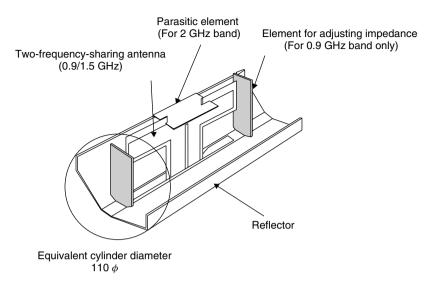


Figure 3.79 Structure of three-frequency-sharing 120° beam antenna

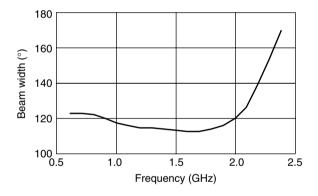


Figure 3.80 Frequency characteristics of three-frequency-sharing 120° beam antenna

The BS ANT practically used for the conventional PDC system has 26 elements that are spaced 200 mm from one another. In contrast, in W-CDMA systems, it has 36 elements that are spaced 150 mm between each other, in order to reduce changes in the gain due to beam tilting. The ANT aperture is 5.4 m, and the effective gain is 20 dBi (2 GHz band) excluding the loss due to power lines and so on.

Figure 3.82 illustrates the structure of a three-frequency-sharing ANT with beam width of 60° and 120° . A PDC ANT has a beam width of 120° (3 sectors), whereas an IMT-2000 ANT has a beam width of 60° (6 sectors). One reflector consists of two ANT systems. The reflector is shared, its edges are bent at right angle to the height of 20 mm or so, the distance from the reflector is set at 70 mm for 0.9 GHz and 1.5 GHz and 37.5 mm for 2.0 GHz, and the width of the reflector is set at 140 mm so as to assure a beam width of 120° at 0.9 GHz and 1.5 GHz and 60° at 2.0 GHz.

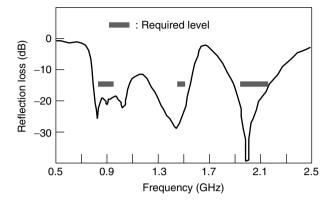


Figure 3.81 Return loss characteristics of three-frequency-sharing 120° beam antenna

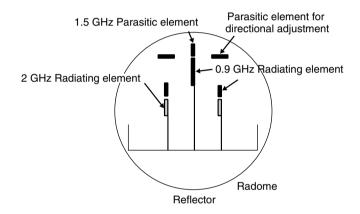


Figure 3.82 Structure of three-frequency-sharing 60°/120° beam antenna

In practice, as multiple elements are aligned for the two ANT systems in the same reflector, their respective radiating elements have a negative impact on each other in terms of directivity and so on. In particular, the directivity in the 2.0 GHz band substantially deteriorates in the direction of $\pm 90^{\circ}$ to the direction of the main radiation, because of electric current induced in the radiating elements of the 0.9 GHz and 1.5 GHz bands. On the other hand, radiating elements in the 2.0 GHz bands. Parasitic elements are installed between the radiating elements in the 0.9 GHz and 1.5 GHz bands in order to improve directivity in the 2.0 GHz band. This ANT configuration helps improve directivity. Figure 3.83 illustrates the return loss characteristics against the frequency. The 0.9 GHz/1.5 GHz ANT achieves dual resonance properties shown in Figure 3.83a by mounting parasitic elements on the front side of the radiating elements for the 2.0 GHz band to support a wider carrier bandwidth, by which impressive return loss properties can be achieved as shown in Figure 3.83b.

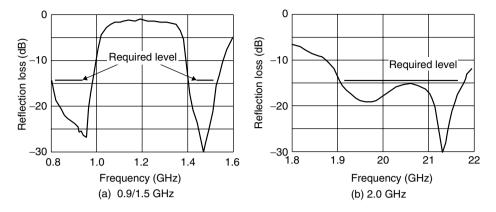


Figure 3.83 Return loss characteristics of three-frequency-sharing $60^{\circ}/120^{\circ}$ beam antenna

3.6 Mobile Terminals

3.6.1 Implementation of Mobile Terminals

3G mobile terminals are wide in variety, ranging from so-called portable phone types, PCcard types dedicated to data communications, video-phone types that can display video, to Personal Digital Assistant (PDA) types that are combined with PDA. Whilst these variations can be categorized by the combination of their abilities, notable characteristics of 3G mobile terminals include the hardware multimedia capabilities (e.g. display), the transmission capability in the radio interface, and the ability to carry various types of multimedia applications.

For example, Figure 3.84 maps out the transmission capability of the radio interface on the horizontal axis and multimedia capabilities of the terminal hardware (e.g. display) on the vertical axis. As the figure shows, even the portable phone types, which have traditionally been speech-centric, are increasingly required to accelerate their transmission speeds because of the installation of browser and mail applications such as *i-mode*, even though cameras and video CODEC for video services may not yet be essential components.

What makes 3G terminals distinctive more than anything else is its high-speed transmission capability. However, there is a trade-off between speed (which stresses the hardware such as memory and processing power) and terminal size, as well as portability. With W-CDMA, many mobile phone types are expected to achieve speed up to 384 kbit/s, as it is easy to achieve 384 kbit/s packet transmission because of the fact that increases in downlink speed have a relatively small impact on hardware, thanks to wideband transmission. In environments in which faster uplink transmission is required, variations such as Small Office Home Office (SOHO) terminals and vehicle terminals shown in Figure 3.85 are expected to appear: for example, terminals will be connected with a server and multiple computers and car navigation systems will be connected with each other via Bluetooth and Local Area Network (LAN). Such terminals are expected to have fewer restrictions in terms of size and costs, and by then, it should be possible to increase the uplink transmission speed depending on demand.

Figure 3.85 maps the installed applications on the horizontal axis and the multimedia capability on the vertical axis. Even if the terminal's hardware multimedia capability (e.g.

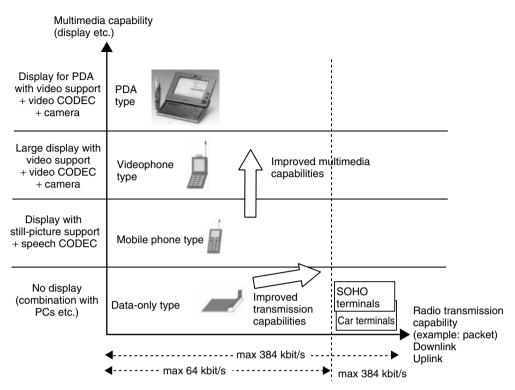


Figure 3.84 Relationship between multimedia capabilities and radio transmission capabilities (example)

display) is high, its full potential cannot be appreciated without application services that provide a wide range of contents. As illustrated in Figure 3.85, terminals with superior multimedia capabilities would have a wider range of applications. In particular, videophone and PDA types require not only a high-definition display with video support, but also substantial memory and processing power, as they receive advanced services such as video distribution and have multiple applications.

Figure 3.86 shows an example of terminals that were actually released in the spring of 2001.

3.6.2 Radio Access Specifications and Hardware Configuration Technologies

3.6.2.1 UE Transmission and Reception Characteristics and Example of Hardware Configurations

As explained before, W-CDMA terminals are wide in variety. However, the same radio access specifications compliant with 3GPP specifications are applied [30, 31]. Tables 3.29 and 3.30 show the key radio transmission and reception characteristics of UEs, respectively. Figure 3.87 illustrates an example of hardware configuration that implements such radio characteristics.

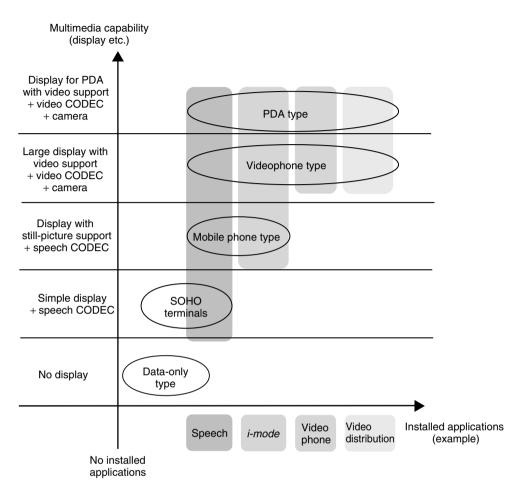


Figure 3.85 Relationship between multimedia capabilities and supported applications (example)

UE Maximum Output Power: 3GPP specifications define: class 1 = +33 dBm; class 2 = +27 dBm; class 3 = +24 dBm; and class 4 = +21 dBm. In Japan, UE maximum output power specified under the equipment regulations of the Radio Law is +24 dBm.

Spurious Emission: This specification primarily refers to ITU-R recommendation SM.329. In addition to this, the spurious emission level for PHS band, etc. is specified by 3GPP specifications.

Reference Sensitivity Level: This is specified in terms of the required reception power that achieves a BER = 10^{-3} upon the reception of a reference measurement channel, which is a 12.2 kbps DTCH and a 2.5 kbit/s DCCH multiplexed on one 30 ksps DL DPCH. According to simulations, the required E_b/N_0 (including channel coding gain) in this channel is 5.3 dB. The value -117 dBm is derived in consideration of the equivalent NF = 9 dB including the margin at the time of mass production, and a margin of 1.8 dB for the Base Band part.



Figure 3.86 Example of W-CDMA terminals

Adjacent Channel Selectivity (ACS): Conventional ACS was often defined in terms of the power ratio of desired waves to that of interference waves (dBc). In 3GPP systems, the spreading gain to interference waves needs to be considered and the power ratio of desired waves to interference waves is 51 dBc according to the specifications. This is equivalent to having a filter with ACS (5 MHz offset) of 33 dB.

In addition to the transmission and reception characteristics mentioned in the preceding text, there are specifications on reception performance in fading environments, radio characteristics when the radio interface (Layer 1) function is used and the radio properties of UE with respect to functions required for controlling the radio interface, as described in Table 3.31.

In relation to UE radio properties with respect to functions required for controlling the radio interface, items that need to be reported to the network and the accuracy of the reported values among the radio parameters measured in UE are specified as measurement items in Table 3.32.

The following distinctive technical issues on the hardware front must be solved to assure the radio performance of W-CDMA UE:

- Low-distortion, high-efficiency power AMP;
- Transmitter supporting fast, precise, and wide dynamic range power control;
- Synchronization with BS transmission signals through fast cell search (reception dynamic range and MF);
- High-sensitivity reception in multipath fading environments (RAKE reception) and
- High frequency stability (AFC function).

Item	Specification	
Transmit frequency	1920 MHz \sim 1980 MHz	
Symbol rate	$15 \sim 9$	960 ksps
Modulation	Channelization	BPSK (Downlink: QPSK)
	Scrambling	HPSK (Downlink: QPSK)
UE maximum output power	Class 3	+24 dBm
	Class 4	+21 dBm
Frequency error	Within ± 0.1 ppm to BS carr	ier upon reception AFC control
Minimum transmit output power	-50 dBn	n/3.84 MHz
Transmit off power	-56 dBn	n/3.84 MHz
Occupied bandwidth	5	MHz
Adjacent Channel Leakage Power	5 MHz offset	-33 dBc
Ratio (ACLR)	10 MHz offset	-43 dBc
Transmit intermodulation	5 MHz offset	-31 dBc
	10 MHz offset	-41 dBc
Spurious emission	$9 \text{ kHz} \leq f < 150 \text{ kHz}$	-36 dBm/1 kHz
	$150 \text{ kHz} \le f < 30 \text{ MHz}$	-36 dBm/10 kHz
	$30 \text{ MHz} \le f < 1 \text{ GHz}$	-36 dBm/100 kHz
	$1 \text{ GH z} \le f < 12.75 \text{ GHz}$	-30 dBm/1 MHz
	1893.5 MHz	-41 dBm/300 kHz
	$\leq f < 1919.6 \text{ MHz}$	
Error vector magnitude	17.5% (When UE transmissio	on power is -20 dBm or more)
Peak code domain error	-15 dB (When s	preading factor is 4)

Table 3.29 UE transmit characteristics

3.6.2.2 Power-Saving Technologies

UEs are restricted in the sense that they must be light, small and battery operated. For years, efforts have been made to reduce the size of devices and decrease power consumption and technologies for reducing power consumption have been applied in the radio system as well, such as intermittent reception control.

W-CDMA requires more sophisticated signal processing functions than conventional systems, in order to provide a wider range of services and applications at high speed. However, the size of the battery that can be installed is more or less the same as in conventional terminals, because of miniaturization requirements. Despite recent efforts to make batteries thinner and lighter and to increase their capacity, as demonstrated by lithium ion batteries and lithium polymer batteries, it is important to take an integrated power-saving approach in each device that constitutes UE–including the RF unit, base band unit and display–so as to make them no inferior to 2G mobile terminals in terms of usage time and size.

Power Saving in Devices

Transmission Devices

W-CDMA executes careful TPC to reduce interference and expand capacity. In order to harness this feature to achieve lower power consumption, it is not enough to reduce distortion and improve efficiency when the power is maximum in the power AMP; the

Item	Specification 2110 MHz ~ 2170 MHz	
Receive frequency		
Reference sensitivity level	DPCH Ec	-117 dBm
Maximum input level	Total input power	-25 dBm
Adjacent Channel Selectivity	DPCH_Ec of desired wave	-103 dBm
(ACS)	Interference waves from adjacent channel	-52 dBm
Blocking characteristics	DPCH_Ec of desired wave	-114 dBm
-	10 MHz off-carrier	-56 dBm
	15 MHz off-carrier	-44 dBm
	RX band \pm 15 MHz \sim 60 MHz off-carrier	-44 dBm
	RX band \pm 60 MHz \sim 80 MHz off-carrier	-30 dBm
	RX band ±80 MHz off-carrier or more	-15 dBm
Spurious response	DPCH_Ec of desired wave	-114 dBm
	Unmodulated interference wave power	-44 dBm
RX intermodulation	DPCH_Ec of desired wave	-114 dBm
	10 MHz off-carrier interference wave power (Unmodulated)	-46 dBm
	20 MHz off-carrier interference wave power (Modulated)	-46 dBm
RX spurious emission	9 kHz $< f < 1$ GHz	-57 dBm/100 kHz
*	$1 GHz \leq f < 12.75 GHz$	-47 dBm/1 MHz
	$1920 \text{ MHz} \le f < 1980 \text{ MHz}$	-60 dBm/3.84 MHz
	2110 $MHz \le f < 2170 \text{ MHz}$	-60 dBm/3.84 MHz

Table 3.30	UE receive	characteristics
-------------------	------------	-----------------

key in extending communication time lies in improving the efficiency overall, from low output to maximum output. In light of maximum transmission power, linearity and power efficiency, GaAs hetero-junction elements and SiGe bipolar elements are used.

Reception Devices

As wideband spreading modulation signals need to be synchronized and demodulated at high sensitivity, it is important to achieve a low NF in the analog front-end. As W-CDMA is based on full duplex operations, small-loss RF switches and so on. cannot be used as devices that are shared in transmission and reception, unlike TDMA terminals. Therefore, such hybrid transmission-reception devices are implemented on the basis of a combination of a dielectric filter or a wideband duplexer using SAW filter technology, with a transmit/receive filter. For the purpose of extending standby time, the requirement of the RF unit is to suppress increases in the current in the receive LNA so as to acquire enough gain to compensate for the loss. Moreover, studies are being conducted on direct conversion transmission and reception, software radio and other technologies to integrate RF components aimed at further miniaturization and lower power consumption.

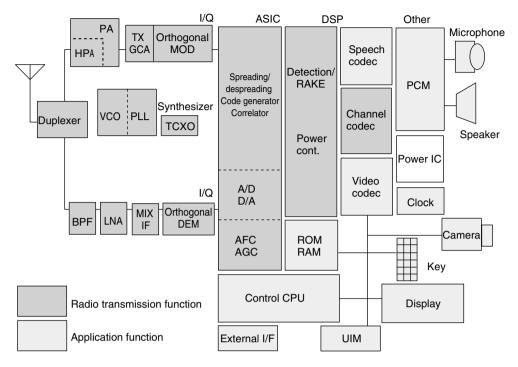


Figure 3.87 Example of UE configuration

W-CDMA also requires fast demodulation operations through MFs, cell search and despreading process, even in intermittent reception mode as described in the following section. As these contribute to increase in reception current, the key for power saving lies in careful control by Digital Signal Processor (DSP), Central Processing Unit (CPU) and other base band Large Scale Integration (LSI).

First, it is important to reduce the base current (dark current) when reception is off. This requires the reduction of leaking current and a power-saving design for the interface between base band devices such as CPU, DSP and memory.

The second issue is to manage the power of circuits in each component. Examples of such power management include the adequate supply of power to DSP and CPU according to the demodulation operation during intermittent reception and sleep control during reception-off mode (e.g. reduction or suspension of clock frequency, shutdown of power supply to RF unit, shortening of warm-up time). Creative efforts are being made in the form of power-controlling ICs, which control the power supply to each unit by CPU and clock supply control circuits.

In addition to such efforts to save power in the radio device, power saving in the display device is also an issue. The challenge is to reduce the power consumed by the display driver circuit to accommodate the increase in current consumed because of larger screen size, color display and video playback. In order to reduce power consumption in idle mode, the display is switched to sub-display during standby and the display mode is changed.

Table 3.31 UE performance provisions

Reception characteristics in fading	Radio characteristics of UE with
environment and radio characteristics	respect to functions required
using radio interface (Layer 1) function	for radio interface control
 DCH demodulation performance under various propagation conditions, including various multipath fading model and dynamic path fluctuation model. DCH demodulation performance when various modes of transmission diversity function are used. Performance of DCH demodulation and TPC command combining capabilities when signals are received from multiple BTSs upon soft handover. Inner- and outer-loop power control performance. DCH demodulation performance when compressed mode is activated. Performance of blind transport format detection without using TFCI bits. 	 Performance of cell reselection Performance of soft handover and hard handover Performance of random access Performance of transmission timing and reception timing Measurement

Table 3.32Measurement items

- CPICH RSCP
- CPICH Ec/Io
- UTRA carrier RSSI
- Transport channel BLER
- UE transmitted power
- SFN-CFN observed time difference
- SFN-SFN observed time difference
- UE-RX-TX time difference

Intermittent Reception Control

Intermittent reception is a technology that activates UE in idle mode only when it is necessary to receive signals from BS, for battery saving purposes.

In W-CDMA, intermittent reception is performed using the PICH, in order to improve the battery saving performance of UE in idle mode. BS sends PIs in a short period to inform UE of any incoming call. Normally, UE in idle mode receives only the timing of PI and when it is informed of a call termination by PI, it shifts to the operation of receiving PCH mapped onto S-CCPCH associated with PICH.

As illustrated in Figure 3.88, the intermittent reception cycle of frames on which PIs are mapped is specified in the broadcast information from BS, as DRX cycles length [2^k frame (k = integer between 6 and 9)]. PIs are divided into multiple groups ($N_p = 18, 36, 72, 144$). UE in idle modes only has to receive short PIs and it can keep the intermittent

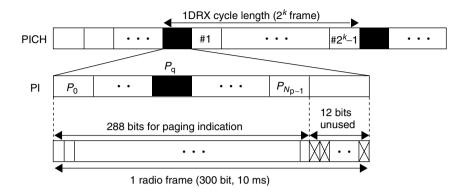


Figure 3.88 PICH structure (FDD)

reception rate low by reducing the incidence of call termination per group. The location of the bit sequence of the received PI in the frame (P_q referred to in Figure 3.88) moves at a cycle of every 512 frames. The aim of this arrangement is to reduce the impact of cyclic level fluctuations and to improve the accuracy of cell search thereby, assuming that the cell search process for shifting to an adjacent cell would be performed simultaneously upon the reception of PI.

In regard to the intermittent reception cycle, the system operation conditions must be designed so that there is a good balance between the effects of extending the idle time by reducing reception power and the control delay time up to the point of connection in the event of call termination. In PI reception, power is consumed in the event of call termination to other users in the same group and erroneous reception of PI, which gives rise to invalid S-CCPCH reception. Therefore, it is necessary to set an adequate number of groups and work out an adequate downlink power allocation to assure that the PICH reception performance is as required in the area. MS is informed of the transmission power ratio of PICH to P-CCPCH from the network, on the basis of PICH_Power_offset in the broadcast information. This value is used to set the adequate threshold for the PI error-detection rate and the detection override rate.

In idle mode, level detection is performed on adjacent cells in parallel so as to detect the cell to which MS moves. As it is important to save power in the cell search process, the process is simplified by broadcasting the scrambling code of the adjacent cell specified in the broadcast information as well the timing information.

3.6.3 UIM

In cellular systems, subscriber information must be written in the mobile phone so that it can be identified to control incoming calls and charge communication fees. The ITU refers to the media memorizing the subscriber information as the User Identity Module (UIM) in its recommendations for IMT-2000 systems. UIM is an IC card with a built-in CPU. Two types are available: one in the size of an ordinary credit card and the other in a plug-in size, which consists only of a terminal strip for miniaturization purposes. UIM has already been introduced in GSM systems, which are widely in use in Europe, where it is called Subscriber Identity Module (SIM) [32]. In conventional cellular systems commercialized in Japan, subscriber information is stored in the nonvolatile memory of the mobile phone itself, excluding some car phones. In such systems, a special device is required to write and erase subscriber information.

There are two merits of introducing UIM: mobile phones can be easily swapped and security can be improved. If subscriber information is written in the phone itself, the user cannot substitute his/her phone with an alternative phone in the event of malfunctioning, until his/her subscriber information is rewritten on an alternative phone with a special writing device. In contrast, UIM allows users to swap mobile phones whenever necessary, simply by taking the card out of a terminal and inserting it into a different one. This merit is appreciated not only in the event of malfunction, but also by subscribers who have more than one mobile terminal (owners of a small mobile phone, a PDA with a built-in phone capability etc.), as it enables them to swap their terminals flexibly. I-C cards are structurally robust against electric and mechanical attacks aimed at undermining security (i.e. they are strong against tampering), and require a Personal Identity Number (PIN), which is a type of password, for accessing and writing any memorized data. Security is maintained at an extremely high level by these two features.

In addition to subscriber information, UIM stores the user's phone number, phonebook, Short Message Service (SMS), accumulated call charges and so on. Another important function of UIM is the authentication function. Authentication is a function to prove that the user is an authorized subscriber in response to the request from the network. In conventional systems, this function was built inside the mobile phone itself. UIM not only offers the same function, but can also check that the authentication request is coming from an authorized network, which enables the terminal and the network to confirm each other's authenticity (mutual authentication function).

UIM specifications are strictly standardized so that independently developed cards operate with mobile phones. Contact-type IC cards are specified by ISO standard (ISO7816) and UIM card functions in IMT-2000 systems are prescribed under 3GPP specifications (TS31 Series, etc.).

Exchange between UIM and UE is carried out by half duplex serial communications. While the basic transmission speed is 9600 bit/s, it can be increased to up to 111,500 bit/s on the basis of transmission-speed negotiation. There are two types of transmission protocols: asynchronous character transmission (T = 0 protocol) and asynchronous block transmission (T = 1 protocol) both of which need to be supported by the mobile phone.

Figure 3.89 shows the structure of a normal IC card. EEPROM, which is a nonvolatile memory that can be electrically rewritten, stores subscriber information, phonebook and other data. ROM, which cannot be rewritten, stores CPU programs that control that card and encryption algorithms for authentication tasks. Figure 3.90 illustrates the file structure in UIM. No detailed explanation will be provided here, as the content and meaning of each file is defined by 3GPP specifications, while the method of accessing each file is prescribed by ISO specifications.

IC cards are likely to be adopted in the management of credit card information and so on., as they can assure an extremely high level of security. In the future, UIM may have credit card functions and store electronic money information to enable mobile phones to electronically settle accounts while assuring a high level of security.

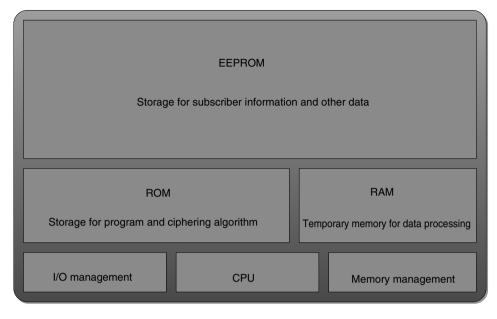


Figure 3.89 IC card structure

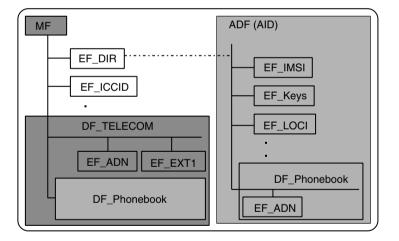


Figure 3.90 UIM file structure

3.6.4 Terminal Display Technologies

3.6.4.1 Semitransmissive Liquid Crystal Display (LCD)

Recent models of mobile phones are equipped with a small LCD (but relatively large for mobile phones) to support mobile Internet services such as *i-mode*. As mobile phones are driven by batteries, their components must meet strict requirements in terms of power consumption. For this reason, Super Twisted Nematic (STN) LCDs, which consume the

least power, were used in mobile phones. Generally, STN LCDs are inferior to activematrix systems like LCDs based on Thin-Film Transistor (TFT) in terms of response speed and color contrast. However, STN-type LCDs were sufficient because mobile Internet services provided under PDC and other 2G mobile communications systems were mostly used for the distribution of text-based information. Table 3.33 compares the characteristics of LCDs [33].

On the other hand, W-CDMA supports not only text-based information distribution services but also entertainment services, including video distribution and high-quality games, for which displays should have a response speed of no more than 60 msec. Normal STN-type LCDs have a response speed of approximately 400 msec, not to mention the insufficient number of colors. A strong candidate for W-CDMA systems is active-matrix displays, which have a response speed as fast as 60 msec and sharp color contrast. The most commonly used active-matrix display is the TFT-type LCD, which has fast respond speed and is able to display many colors. However, the problem is that it consumes 40 times more power than STN. The solution to this was the development of TFD-type LCD. It should be noted that TFT-type LCD achieves sharp contrast and high response speed by mounting a TFT in each pixel where electrodes covering the board intersect with each other and controlling each pixel by switching the transistor. TFD-type LCDs are made by replacing the TFTs in TFT-type LCDs with TFDs, which makes its highspeed response and sharp contrast on par with TFT. TFD-type LCDs consist of a digital drive circuit aimed at achieving lower power consumption. Currently, they consume only 4 mW when displaying still pictures, which is comparable to STN-type LCDs.

3.6.4.2 Organic Electro Luminescence (EL)Display

The problem with color semi-transmissive LCDs is that it requires a backlight, which increases power consumption and makes it difficult to make the film lighter and thinner.

System	STN-type LCD semitransparent	TFD-type LCD semitransparent	Amorphous Si-TFT	Low- temperature polycrystal Si-TFT-driven organic EL
Screen size	Type 2	Type 2.6	Type 2	Type 2.4
Number of pixels or dots	120×160 pixels	160×240 pixels	$560 \times 220 \text{ dots}$	852×222 dots
Number of colors	256 colors	4096 colors	260,000 colors	260,000 colors
Contrast	10:1	15:1	5:1	-
Brightness	F	Reflection rate:30%		_
Viewing angle	70°	80°		_
Response time	400 msec	60 msec	50 msec	A few micro seconds
Power consumption	2 mW	4 mW	80 mW	440 mW

 Table 3.33
 Specifications of key displays for mobile phones

Note: STN: Super Twisted Nematic; TFD: Thin Film Diode; Si-TFT: Si Thin Film Transistor.

A strong candidate to solve this issue is a display that uses organic EL as a light emitter. As the name suggests, organic EL emits light, and materials that emit red, blue and green light have been developed, making it possible for organic EL to demonstrate full-color display [34]. The basic structure of organic EL display is similar to Light Emitting Diode (LED) and it can respond faster than liquid crystal, which relies on changes in molecular arrangement. W-CDMA mobile terminals have started to adopt organic EL, whose response speed is already on the order of μ sec. It is expected to be a strong solution for decreasing the thickness and weight of large displays that would be required in large-capacity communications in the years to come.

3.6.4.3 Future Issues and Prospects of Display Technologies

In W-CDMA mobile phones, active-matrix-type displays are strong candidates. However, further reduction in power consumption is a must. In fact, efforts are still being made to accelerate the response speed of STN-type LCD displays in which lower power consumption can be easily achieved, and according to some reports, STN-type LCD displays with properties similar to their active-matrix counterparts have been developed. Organic EL displays, which have been developed to make displays thinner, have the potential to completely take over LCDs provided that it reduces power consumption, achieves higher brightness and extends the life of organic materials.

3.6.4.4 Compact Hyper Text Markup Language (CHTML)-Based Micro-Browser

i-mode browser is written with a description language called Compact-Hyper Text Markup Language (HTML). CHTML is a subset of HTML, which is a standard language used in PC-based Internet browsers. CHTML is designed to operate efficiently in a mobile environment, taking into account the capabilities of mobile terminals and the radio interface characteristics. For example, the memory capacity is limited in mobile phones because of implementation and cost constraints: PDC phones have a memory of about 1 or 2 Mbytes, and even W-CDMA phones with video support are expected to have a memory of 5 Mbytes or so. The CPU clock speed also has to be suppressed compared to PCs in light of power consumption, and thus is projected to be around 30 to 50 MHz even in 2002. In addition to such restrictions on processing power, mobile phones only have number keys and a few control keys. Taking this into account, CHTML deletes some functions supported in HTML, and limits the navigation of the browser to UP/DOWN keys only. This simplification has actually made the *i-mode* browser user-friendly. The basic design concept has been passed onto W-CDMA phones.

3.6.5 External Interface

3.6.5.1 Role of Mobile Phones and External Interface

Mobile phones are one of the few terminals that users always carry around. The role of mobile phones is becoming increasingly important and diversified, as illustrated in

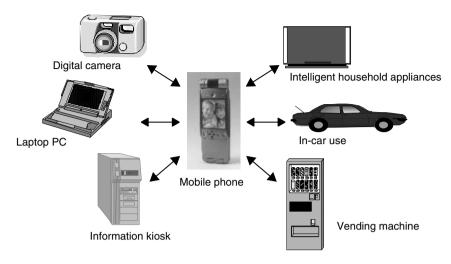


Figure 3.91 Connection with various external devices

Role of phone	Connected device	Content type	Example of application
Radio module	PDA, PC and intelligent household appliances	AT command, user data	PC communications, Web browsing, speech and video transmission
PIM/Content storing device	PDA, PC and mobile phones	Object files	Business card exchange, phonebook exchange and exchange of various files
Electronic wal- let/authentication terminal	POS terminal and vending machine	Electronic value, authentication data	Settlement of accounts as an electronic wallet and/or credit card

 Table 3.34
 Functions and interface of mobile phone

Figure 3.91. Table 3.34 shows an example of the roles of mobile phones, in addition to the handled content.

(1) Application as a Radio Module

When connected with PCs or PDAs, mobile phones can be used to make Internet/Intranet accesses to send and receive emails or to browse Web sites. Real-time speech and video transmission requires an analog interface or a synchronous digital interface.

(2) Application as a Personal Information Manager (PIM) and Content Storage

Certain types of PDC digital mobile phones are able to exchange business cards and phonebook information with PC, PDA and other mobile phones via an infrared interface. In addition to such functions, W-CDMA mobile phones are able to exchange information such as messages (mail) and URLs.

(3) Application as Electronic Wallet and Authentication Terminal

For a mobile phone to be used in place of an electronic wallet or a credit card, it is necessary to provide a secure interface that protects electronic value and authentication data.

In addition to the above, an external interface may be applied to monitor the status of the mobile phone – which may not be a function made available to the user.

3.6.5.2 Requirements of External Interface of W-CDMA Mobile Phones

The requirements of the external interface of W-CDMA mobile phones are as follows.

Speed: W-CDMA radio interface offers a maximum transmission speed of 384 kbit/s (2 Mbit/s in indoor environments in the future). The external interface must be fast enough to fully harness the transmission speed of the radio interface.

Power Consumption: As mobile phones are carried around and thus battery-driven, the power consumed by mobile phones must be minimal even if the external interface is made faster.

Size: The connector and the external interface module must be small so that the portability and the design of mobile phones would not be undermined.

Versatility: The external interface must be a generic one so as to connect with PCs, PDA and many other types of external equipment.

3.6.5.3 Types, Characteristics and Applications of External Interface

Table 3.35 shows the characteristics of various interfaces.

Infra-red Data Association (IrDA): This is an infrared interface widely used in laptop PCs and PDAs. IrMC (IrDA Mobile Communication) is defined as the communication standard for mobile phones [35]. For mobile phones, it would be practical to apply an interface (device and lens) that can communicate over a distance of 30 cm or so for the purpose of reducing power consumption. As IrDA requires line-of-sight transmission, it is suitable for securely transmitting short data without setting the destination of communications, such as business card exchange.

Bluetooth: This is a new radio interface [36]. Its use of the Industrial Scientific Medical (ISM) band may result in the deterioration of throughput due to the impact of interference, but deterioration in throughput due to narrowband interference is limited as it performs frequency hopping over extremely short slots. As for the connection arrangement, the interface allows multiple Bluetooth-supporting terminals to be interconnected via a network; hence, it is suitable for electronic conferences and home networks that interconnect multiple intelligent household appliances.

IEEE802.11b: This is an interface frequently used for wireless LAN [37]. However, it would not be appropriate to apply the specifications immediately as a mobile phone interface, as a result of the large power consumption.

	Transmission speed (maximum)	Communication range (maximum)	Main application	Demerits
IrDA	1 Mbps(MIR) 4 Mbps(FIR)	0.3 m (Mobile application)	Object exchange (business cards, phonebook, etc.)	 Short communication range Requires line-of-sight positioning
Bluetooth	0.7 Mbps	10 m (CLASS 1)	Speech/data transmission, LAN (e.g. electronic conference, home network)	 Poorer quality due to interference Difficult to specify destination of communication Authentication required
IEEE802.11b	11 Mbps	50 m	Radio LAN (data transmission)	 Poorer quality due to interference Large power consumption Authentication required
USB	12 Mbps	5 m	Music/data transmission	 Cable connection required Impossible to interconnect USB devices
RS-232C	115.2 kbps	Approx. 15 m (depending on voltage applied)	Data transmission	Cable connection requiredSlow transmission speed

 Table 3.35
 Characteristics of various interfaces

IrDA: Infra-red Data Association;

USB: Universal Serial Bus.

Universal Serial Bus (USB): This is an interface for connecting PCs with peripheral devices [38]. ARIB connector standards [39] specify the use of USB at default due to its reasonable speed and ability to transmit digital synchronous data. As for the connection arrangement, USB is based on host-device connection, which has certain shortcomings: for example, digital cameras and such devices. cannot be directly connected if the mobile phone is configured as USB device.

RS-232C: This is an extremely versatile, cheap interface. It is a legacy interface for PCs and is used also in measuring instruments and consumer terminals. However, when connected with IMT-2000 mobile phones, its application is limited because of its speed: its maximum transmission speed is only 115.2 kbps.

3.6.5.4 Future Prospects of External Interfaces

No particular specifications exist for external interfaces of IMT-2000 mobile phones under 3GPP, as they are considered to be out of the scope [40].

3GPP takes this stance so that the interface can be freely selected, rather than inflexibly determining the interface technologies, which are diversifying and undergoing rapid progress. Nevertheless, it may be more convenient to have the external interface for mobile phones standardized, considering the fact that the role of mobile phones is becoming increasingly important as personal terminals: devices subject to connection with mobile phones were mainly PCs and PDAs in the past, but in the future, the demand for data exchange with intelligent household appliances and intelligent kiosks via mobile phones is expected to increase. As described in Table 3.35, each interface has its own merits and demerits, so it is vital to apply each interface in the area where it excels.

Accordingly, mobile phones must have 2 or 3 types of interfaces and be equipped with a function to switch from one interface to another by simple user operation or by automatic recognition.

3.6.6 Future Prospects of Mobile Terminals

As a result of efforts to make mobile phones smaller and consume less power for improved portability, the gadgets have evolved from "cellular phone-to-talk into cellular phone-to-use." They should now be called *mobile terminals* rather than mobile phones, because they are not only equipped with speech functions; they also incorporate a wide range of functions as information terminals. For instance, they now have so-called mobile Electronic Commerce (EC) functions (a system to purchase products via mobile terminals) in addition to the *i-mode* functions and have even started to adopt functions for controlling air conditioners at home [41]. Mobile terminals are expected to make life more convenient for users in such a manner (personalization). An alternative trend is that mobile terminals are now being used like PCs; for example, they have turned into platforms of entertainment applications, migrating from game arcades to the home domain (multimedia). Needless to say, mobile Internet is included in this category. Figure 3.92 shows the directions in which terminal technologies may progress depending on the provision of services.

As illustrated in the figure, mobile terminals must constantly fulfill miniaturization and low-power-consumption requirements in order to sustain their convenient properties based on portability. The power consumption issue is expected to be solved, albeit gradually, through developments in LSI processes (which have been progressing rapidly in recent years) and the development of high-energy, high-density batteries, in addition to the radio link technologies mentioned in Section 3.5. In the future, the important issue would be: how can large-capacity data rendered in the W-CDMA system be operated and controlled by small mobile phones? In other words, improvements in the human interface of mobile terminals or the interface with other devices are believed to become increasingly important.

If the technologies referred to in Figure 3.92 enjoy progress in the future, LCDs that are currently applied in small sizes because of various constraints might evolve into large, easy-to-see displays on par with books and glossy magazines. The existing dialing buttons might evolve into easier-to-use menu formats or touch-panels. By having mobile terminals

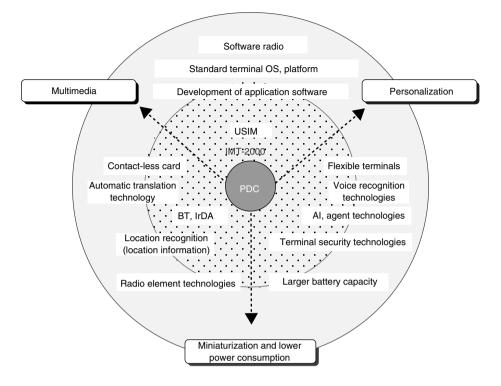


Figure 3.92 Direction of future mobile terminals

as such, we might be able to be completely liberated from restrictions imposed because of locality. The simple possession of mobile terminals might give people a sense of security and peace of mind, if IT technologies and security systems undergo further progress in the existing traffic systems, hospitals and other social infrastructures. Mobile terminals are expected to play an important part as the front-end for people living in the contemporary social infrastructure.

References

- [1] 3GPP TS 25.321 V3.6.0, December 2000.
- [2] 3GPP TS 25.211 V3.5.0, December 2000.
- [3] 3GPP TS 25.212 V3.5.0, December 2000.
- [4] 3GPP TS 25.213 V3.4.0, December 2000.
- [5] 3GPP TS 25.214 V3.5.0, December 2000.
- [6] 3GPP TS 25.322 V3.5.0, December 2000.
- [7] Ohno, K. and Adachi, F., 'Uplink Capacity and Transmission Power in DS-CDMA', *The IEICE Transac*tion on Communications, B-II, J79-B-II(1), 1996, 17–25.
- [8] Ishikawa, Y. and Iwamura, M., 'Method of Estimating Interference Power Distribution and Call Loss Rate in W-CDMA Uplink', Proceeding of the 2000 IEICE General Conference, B-5-31, March 2000.
- [9] Viterbi, A.J., 'CDMA-Principles of Spread Spectrum Communication', Addison-Wesley, Reading, MA, 1995.
- [10] Ishikawa, Y. and Umeda, N., 'Capacity Design and Performance of Call Admission Control in Cellular CDMA Systems', *IEEE Journal on Selected Areas in Communication*, 15(8), 1997, 1627–1635.

- [11] Viterbi, A.M. and Viterbi, A.J., 'Erlang Capacity of a Power Controlled CDMA System', *IEEE Journal Selected Areas in Communication*, **11**(6), 1993, 892–900.
- [12] Hayashi, T., Usuda, M., Ishikawa, Y., Nakamura, T. and Onoe, S., 'Studies on Transmission Power Allocation to W-CDMA Downlink Common Channel', Proceeding of the 2000 IEICE General Conference, B-5-81, March 2000.
- [13] Gilhousen, K.S., Jacobs, I.M., Padovani, R., Viterbi, A.J., Weaver, L.A. and Wheatley, III, C.E., 'On the Capacity of a Cellular CDMA Systems', *IEEE Transaction on Vehicular Technology*, 40(2), 1991, 303–312.
- [14] 'Analysis of Downlink Capacity in Wideband DS-CDMA', 1999 IEICE Society Conference, B-5-24, September 1999.
- [15] Furukawa, H., 'Theoretical Analysis of CDMA Cellular Downlink Capacity Subject to Transmit Power Control', Technical Report of IEICE RCS 99-93, August 1999.
- [16] TIA TR45. 5, 'The cdma2000 ITU-R RTT Candidate Submission', IMT-2000 Radio Transmission Technology (RTT) Proposals (8-2), June 1998.
- [17] Ishikawa, Y. and Umeda, N., 'Method of Designing CDMA Capacity Considering Rate of Deterioration in Communication Quality', Technical Report of IEICE RCS95-49, July 1995.
- [18] Ishikawa, Y. and Umeda, N., 'Method of Designing CDMA Capacity considering Geographic Traffic Distribution', Technical Report of IEICE RCS-95-132, January 1996.
- [19] Ishikawa, Y., 'CDMA Capacity and Call Admission Control in SIR-based Transmit Power Control', 1996 IEICE Society Conference, B-379, September 1996.
- [20] Hata, M., Kinoshita, K. and Hirade, K., 'Radio Link Design of Cellular Land Mobile Communication Systems', *IEEE Transaction on Vehicular Technology*, VT-31(1), 1982, 25–31.
- [21] Hata, M., 'Radio Circuit Design Scheme for Terrestrial Mobile Communications', *Electrical Communication Laboratories Technical Journal*, 31(10), 1982, 1861–1872.
- [22] Ishikawa, Y., Nakano, E. and Uebayashi, S., 'Downlink Radio Design Scheme for DS-CDMA Mobile Communications', 1997 IEICE Society Conference, B-5-8, September 1997.
- [23] 3GPP Specification, 'RRC Protocol Specification', 3GPP TS 25.331 V3.3.0, June 2000.
- [24] ITU-R Recommendations, Recommendation ITU-R M. 1225, 'Guidelines for Evaluation of Radio Transmission Technologies for IMT-2000', 1997.
- [25] Hata, M., 'Empirical Formula for Propagation Loss in Land Mobile Radio Services', *IEEE Transaction on Vehicular Technology*, VT-29(3), 1980, 317–325.
- [26] ARIB, 'Self Evaluation Report on Japan's Proposal for Candidate Radio Transmission Technology on IMT-2000: W-CDMA-Part II Revised RTT Proposal', Self Evaluation Report Submitted to ITU-R, September 1998.
- [27] Iwamura, M., Ishikawa, Y., Ohno, K. and Onoe, S., 'Optimization of Beam Width of W-CDMA Sector Antenna', Proceeding of the 1999 IEICE General Conference, B-5-157, March 1999.
- [28] Murata, and Yamamoto, 'Non-linear Properties of High-frequency Amplifier and Compensation Method', *The Journal of the Institute of Electronics, Information and Communication Engineers*, 61(9), 1978, 999–1007.
- [29] Nojima, T. and Narahashi, S., 'Ultra-low-distortion, Multi-frequency Common Amplifier for Mobile Communications – Auto-adjusting Feed Forward', Technical Report of IEICE RCS90-4, May 1990.
- [30] 'UE Radio Transmission and Reception (FDD)', 3GPP TSG-RAN WG4, TS 25.101 v3.5.0, December 2000.
- [31] 'Requirements for Support of Radio Resource Management (FDD)', 3GPP TSG-RAN WG4 TS 25.133 v3.4.0, December 2000.
- [32] Mouly, M. and Pautet, M.B., 'The GSM System for Mobile Communications', C & Sys, France, 1993.
- [33] 'Frontline of Mobile Internet: From i-mode to the Next-generation IMT-2000', Nikkei BP, 1999.
- [34] Forrest, S., Burrows, P. and Thompson, M., 'The Dawn of Organic Electronics', *IEEE Spectrum*, August, 2000, pp. 29–34.
- [35] IrDA, 'Specifications for Ir Mobile Communications Version 1.1', March 1999.
- [36] Bluetooth-SIG, 'Bluetooth Specification Version 1.0B', December 1999.
- [37] IEEE, 'IEEE Std 802.11b', 1999.
- [38] USB-IF, 'Universal Serial Bus Specification Revision 2.0', April 2000.
- [39] ARIB, ARIB TR-T12-27.A01 v3.0.0, March 2000.
- [40] 3GPP TR 27.901 v3.0.0, January 2000.

- [41] NTT DoCoMo 2010 Vision Editing Group, '2010: NTT DoCoMo's Vision for the Future', NTT Publishing, 1999.
- [42] 3GPP TS 25.331 V3.5.0, December 2000.
- [43] Adachi, F., et al., 'Coherent DS-CDMA: Promising Multiple Access for Wireless Multimedia Mobile Communications', Proceedings of the IEEE ISSSTA '96, September, 1996, pp. 351–358.
- [44] 3GPP TS 25.323 V3.3.0, December 2000.
- [45] Onoe, S., Ohno, K., Yamagata, K. and Nakamura, T., 'Wideband-CDMA Radio Control Techniques for Third-Generation Mobile Communication Systems', Proceedings of the IEEE VTC '97, May 1997, pp. 835–839.

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

4

Network Technologies

Makoto Furukawa, Hiroshi Kawakami, Mutsumaru Miki, Daisuke Igarashi, Yukichi Saito, Toyota Nishi, Mayuko Shimokawa, Katsumi Kobayashi, Yasuhiko Kokubun and Masayuki Nakanishi

4.1 Overview

Mobile communication networks were commercially launched as Circuit-Switched (CS) systems centering on speech communication services. The First-Generation (1G) analog system evolved into the Second-Generation (2G) digital system, followed by the introduction of Packet-Switched (PS) communication system. These conventional mobile communication systems were realized with different technologies by country and region, and there was no internationally unified standard.

The standardization and system development of the Third-Generation (3G) International Mobile Telecommunications-2000 (IMT-2000) was instigated in response to meet the increasing need to accomplish high-speed data communications adequate for mobile multimedia services and to develop a global system that would allow mobile terminals to be used worldwide.

IMT-2000 is standardized on the basis of two regional standard development groups, namely, 3GPP and 3GPP2 (3rd Generation Partnership Projects). This chapter reviews the network technologies with reference to 3GPP, which adopts Wideband Code Division Multiple Access (W-CDMA) for the Radio Access Network (RAN) and an evolved Global System for Mobile Communications (GSM) Core Network (CNs) for CN systems.

Figure 4.1 illustrates the reference model for the CN architecture specified by 3GPP [11]. The functional entities inside CN more or less correspond to the functions in the Personal Digital Cellular (PDC) model referred to in Chapter 1.

The signaling method of CN under 3GPP is based on GSM and General Packet Radio Service (GPRS), which are used for 2G mobile communication systems worldwide, with some newly added functionality and capabilities to meet IMT-2000 requirements. As network components, the CS domain and the PS domain are defined separately from each other. These represent a group of logical function units; in the actual implementation, these functional domains can be arbitrarily mapped with physical equipment and nodes.

For example, by implementing the CS functionality [Mobile Switching Center (MSC)/ Gateway MSC (GMSC)] and PS functionality [Serving GPRS Support Node (SGSN)/Gate

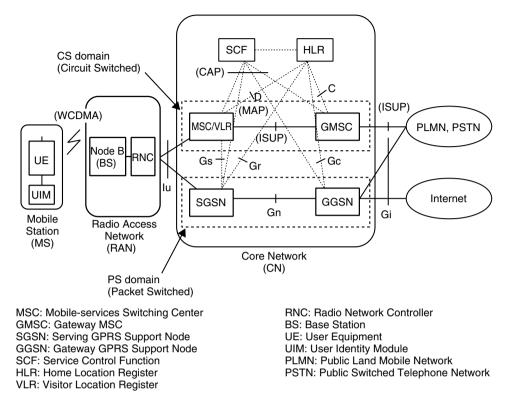


Figure 4.1 CN Architecture model under 3GPP

GPRS Support Node (GGSN)] in a single node, it is possible to build an integrated system capable of switching and transmitting various types of media, ranging from speech traffic to large-capacity data traffic. This is where Asynchronous Transfer Mode (ATM) communication technology is effective, which enables adequate traffic control and quality control with respect to traffic that requires different types of Quality of Service (QoS). Figure 4.2 shows an example of the physical node configuration for an integrated CS and PS network within CN.

In response to the demand to *use mobile phones worldwide*, the CNs used for IMT-2000 are virtually converged into two systems as explained in the preceding text, and thus are expected to radically facilitate globalization. Three functions are required for the achievement of global services: *terminal mobility* (the ability to receive services with the same terminal regardless of location); *personal mobility* (the ability to receive services at the roaming destination in the same service environment as in the home network). Network technologies aimed at meeting these requirements include the Virtual Home Environment (VHE) using an advanced Intelligent Network (IN), which is currently under study.

In response to demands for *high-speed data communications*, IMT-2000 will achieve data transmission speed of up to 2 Mbit/s in mobile networks. As represented by NTT DoCoMo's *i-mode* provided over the 2G mobile packet communication systems, mobile

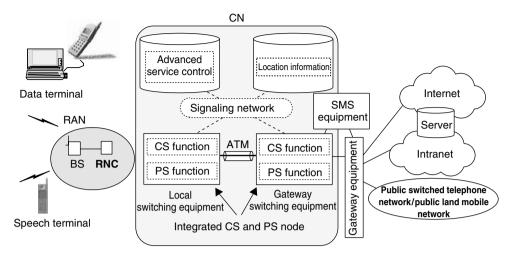


Figure 4.2 Example of physical configuration of integrated CS and PS network

phones have widely penetrated the market as devices for Internet access owing to their easy-to-operate features. In IMT-2000, their data storage/notification functions [such as Short Message Service (SMS)] are expected to be enhanced, and advanced multimedia services are likely to emerge through a connection with the Internet and the corporate Local Area Network (LAN). For example, content that is likely to be available over the Internet from various service providers in the future would include video phone service based on video information communications, music and video distribution, mail with video attachments, chat lines, Virtual Private Network (VPN) in mobile networks, advanced e-commerce capitalizing on the authentication capability of mobile terminals and applications for the Intelligent Transport System (ITS). The scope of mobile multimedia services is thereby expected to increase dramatically.

In this chapter, Section 4.2 discusses ATM, which is an effective data transmission technology in IMT-2000, and describes the QoS assurance mechanism. Sections 4.3 and 4.4 describe the CN-related signal schemes and the basic control procedures (CS and PS) with respect to the IMT-2000 system standardized under 3GPP. Section 4.5 reviews the trends in and the outline of IN technologies that are indispensable for accomplishing Supplementary Services (SSs) and VHE. Sections 4.6 and 4.7 introduce technologies for connecting the mobile network and the Internet-highlighting SMS, various gateway equipment and multimedia platform technologies – and forecast the future of advanced services.

4.2 ATM Technology

4.2.1 Switching Scheme for Multimedia Communications

IMT-2000 offers CS services including speech, video and unrestricted digital information services and PS services primarily aimed at Internet access. CS is a scheme in which the switching equipment executes communications by setting up a connection that secures network resources in the event of call origination. The switching equipment used for CS services in 2G mobile communications carries out switching based on 64-kbit/s circuits,

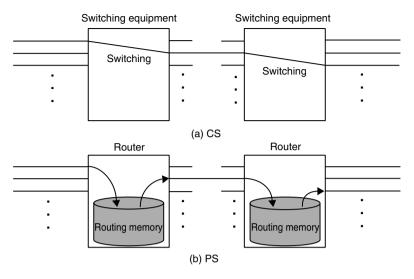


Figure 4.3 Circuit Switching (CS) and Packet Switching (PS)

and is therefore suitable for the transfer of application services that operate at a fixed speed of 64 kbit/s, namely, Pulse Code Modulation (PCM) coded speech.

PS is a scheme that divides user data into blocks of a certain length referred to as *packets*, and switches the data in packet units according to the destination information attached to each packet. The technology is applied to Internet Protocol (IP) communications. Figure 4.3 shows the structure of CS and PS. Packet communications are discussed in further detail in Section 4.4.

3G mobile communications required a switching scheme that could efficiently transmit compressed speech and data information for Internet access (for which traffic has been steadily increasing in recent years). ATM is a technology that divides information to be transmitted into 53-byte frames called *cells* for transmission and switching. The use of ATM in the RAN is specified under 3GPP Release 1999. There are substantial merits in applying ATM in CN, including the ability to perform traffic management in coordination with RAN, implement CS and PS functions in the same architecture and carry out quality control and operations in an integral manner. In the future, data transmission including today's CS service is expected to be carried out in a comprehensive manner as IP communications, which will enable a flexible service provision based on the convergence of Internet services and mobile communication services. The use of "All-IP" networks is considered to enable an economical IP data transfer, yet there are some technical challenges to be solved including the assurance of communication quality and the reliability of the network. ATM has extensive traffic management and quality-control functions for handling traffic characteristics, and is an effective technology for forwarding not only CS services but also PS services.

4.2.2 Basic Configuration of ATM

A cell, which is the data transfer unit in ATM, consists of a 5-byte header (which includes routing information etc.) and a 48-byte payload (storing user data). The ATM switching

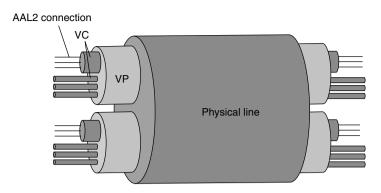


Figure 4.4 ATM connections

equipment achieves fast switching based on hardware switching with reference to the routing information in the header, without detecting data errors in each cell. The routing information in the header consists of the Virtual Path (VP) and the Virtual Channel (VC). The stratified connection control consisting of VC (which corresponds to the user channel) and VP (which is a bundle of VCs) enable highly flexible, extensible operation and administration. ATM Adaptation Layer type 2 (AAL2) can set multiple-user connections in VC. Figure 4.4 illustrates the structure of ATM connections. Normally, VP is setup on the basis of system data at the time of building the network. VC connections can be divided into Permanent Virtual Channel (PVC), which is inflexibly set up at the time of network construction, and Switched Virtual Channel (SVC), which is established and released on the basis of signaling upon call origination and termination. The establishment and release of the user connection through the operation of SVC helps efficiently use ATM connection and bandwidth resources.

4.2.3 ATM Adaptation Layer (AAL)

AAL is a protocol for coordinating the higher layer, which has various traffic properties including speech, video streaming and IP packets, and the ATM layer, which is specified regardless of the higher-layer application. Four types of AAL are specified, namely, AAL1, AAL2, AAL3/4 and AAL5 [1–4].

AAL1 is used for forwarding continuous, fixed-rate data, such as PCM-coded speech. AAL2 was originally standardized for the purpose of efficiently forwarding short frames in ATM; such as compressed speech data used in mobile communications, and is applied as the standard for transferring user data in IMT-2000 RAN. AAL3/4 was developed for data communication purposes, and is distinctive in that it can transfer up to 1024 types of higher-layer data on one VC connection with a Multiple Identifier (MID). AAL5 is a simpler protocol compared to AAL3/4, and is widely used for forwarding data packets and control signals. The following is a brief description of AAL2 and AAL5, which are applied in IMT-2000 specifications.

4.2.3.1 AAL2

Figure 4.5 shows the frame structure of AAL2. AAL2 has the function to multiplex up to 256 user connections on one VC connection, and is able to transmit short frames in

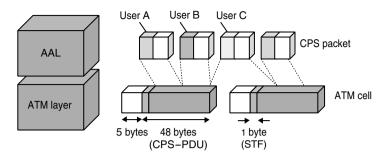


Figure 4.5 AAL2 structure

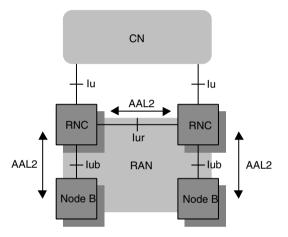


Figure 4.6 RAN interfaces

a highly efficient manner with limited delay. The Common Part Sublayer (CPS) packet consists of a 3-byte header and a payload between 1 and 45 bytes. The header includes a Channel IDentifier (CID) for user identification. Multiple user connections can be transmitted by multiplexing them on one VC connection. CPS packet is carried by CPS-Protocol Data Unit (PDU) to which a one-byte STart Field (STF) is assigned, and converted into cells. Figure 4.6 illustrates the IMT-2000 interfaces and the AAL type applied to the user-plane transfer. AAL2 is an important protocol applicable from Node B to CN. Although there are no specifications under 3GPP in regard to transmissions inside CN, the application of AAL2 in CN enables the traffic on the interface between Radio Network Controllers (RNCs) (Iur Interface) to be physically relayed by CN and communications between IMT-2000 terminals to be transmitted by AAL2 between Node Bs, which allows efficient operations.

4.2.3.2 AAL5

AAL5 is suitable for forwarding signaling data and IP packet data. Figure 4.7 illustrates the frame structure of AAL5. The higher-layer user data are attached with a trailer (which includes length information and error detection codes) and PADding (PAD) for length adjustment, to construct a Common Part Convergence Sublayer (CPCS)-PDU. The

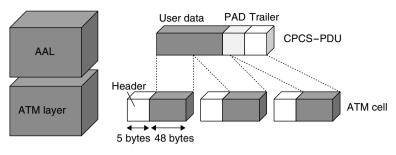


Figure 4.7 AAL5 structure

maximum length of CPCS-PDU payload is 65535 bytes. Under IMT-2000, control signals in RAN and PS data on the ATM-based Iu interface and CN are transmitted in AAL5.

4.2.4 Quality of Service (QoS) and ATM Traffic Management

4.2.4.1 QoS Classes Under 3GPP

The subscriber count of wireless Internet access services provided on 2G mobile communication systems has been increasing remarkably. Under IMT-2000, which offers substantially faster transmission speeds, Internet access and other data communications traffic is expected to increase further. Meanwhile, IMT-2000 will also be used for voice communications. Hence, each type of traffic would need to assure a certain level of QoS according to the service application. QoS classes specified by 3GPP are as follows [5].

Conversational Class: Interactive communication that requires low delay (e.g. speech).

Streaming Class: Unidirectional communication, requiring streaming service with low delay (e.g. real-time video distribution).

Interactive Class: Requires response within a certain period and low error rate (e.g. Web browsing, server access).

Background Class: Requires best-effort services performed in the background (e.g. e-mail, file download).

These QoS need to be assured end-to-end. The traffic capabilities of ATM can be used to achieve this in RAN and CN, as illustrated in Figure 4.8. In particular, ATM can disperse the processing load associated with QoS control and execute network quality control with certain standards by guaranteeing the QoS in the lower layer of the network, without resorting solely to traffic control of end-to-end protocols. The traffic management functions of ATM include

- efficient use of network resources by statistical multiplexing,
- provision of various service classes,
- assurance of communication quality by securing bandwidth at the time of connection setup and
- function to monitor contract traffic violations.

The ATM traffic management functions are described in the following sections.

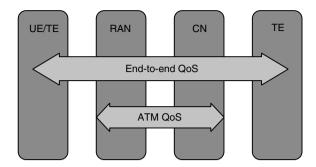


Figure 4.8 End-to-end QoS and ATMQoS

Service category	CBR	rt-VBR	nrt-VBR	UBR	ABR
Characteristics	Guarantees PCR	Guarantees SCR with low delays	Guarantees SCR for non-real-time com- munications	Best effort	Guarantees MCR with flow control
Bandwidth parameters	PCR	PCR SCR MBS	PCR SCR MBS	PCR ^a	PCR ^a MCR

^aNetwork is not required to guarantee.

4.2.4.2 ATM Service Class

Table 4.1 shows the service classes within the scope of ATM service categories.

Constant Bit Rate (CBR) is suitable for quality assurance applications with a fixed speed, such as PCM coded speech and unrestricted bearer services, as it secures bandwidth based on the Peak Cell Rate (PCR). Real-time Variable Bit Rate (rt-VBR) and non-realtime Variable Bit Rate (nrt-VBR) secure bandwidth with the use of PCR, Sustainable Cell Rate (SCR) and Maximum Burst Size (MBS). MBS specifies the permissible level of burstiness in the traffic exceeding SCR, and assures the speed of SCR in communications. rt-VBR is suitable for variable bit rate, compressed speech data, as it guarantees cellforwarding delays, whereas nrt-VBR is suitable for bursty packet communications with an assured data loss rate, as it does not provide for quality in terms of delay. Because of the fact that nrt-VBR has a low delay requirement and allows more queuing delay, its statistical multiplexing effect is greater than rt-VBR. Unspecified Bit Rate (UBR) is a service class of a best-effort type, for which there are no bandwidth or quality provisions. As UBR does not normally secure bandwidth, the inflow of data in excess of transmission capacity results in data loss. Available Bit Rate (ABR) secures bandwidth at the Minimum Cell Rate (MCR) and enables communications up to PCR using flow control. Figure 4.9 illustrates the transmission image of each service class.

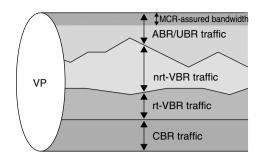


Figure 4.9 ATM service classes and bandwidth usage

According to the characteristics mentioned in the preceding text, the QoS classes under 3GPP may be mapped to the ATM service classed as follows:

- Conversational class \rightarrow CBR
- Streaming class \rightarrow rt-VBR
- Interactive class \rightarrow nrt-VBR
- Background class \rightarrow UBR

4.2.4.3 Connection Admission Control (CAC)

In order to assure communication quality, there must be a mechanism to reject the admission of calls if the quality cannot be assured because of insufficient network resources in consideration of the bandwidth and the required QoS. Software control that makes the decision as to whether the call should be admitted or not upon connection setup by SVC is referred to as CAC. When the communication quality assurance is controlled by CAC, insufficient network resources result in higher blocking probability. Therefore, network operation not only requires communication quality assurance by CAC but also call-connecting quality assurance based on adequate traffic engineering.

4.2.4.4 Usage Parameter Control (UPC)

If more traffic enters the network than declared upon the admission of a connection by CAC, not only the contract-breaching connection but also other connections engaged in communications might be affected in terms of quality. The function to monitor whether the admitted connection is adhering to the contract made upon admission is referred to as UPC. Contract-breaching traffic is either disposed of, given lower priority in transmission by attaching a tag, or transmitted by adjusting the speed to comply with the contract traffic. Figure 4.10 illustrates the application of UPC to an IMT-2000 network. Mobile communication services largely consist of speech communications, video communications and other application services that are rendered at a predetermined speed, and the speed limit is determined by the radio channel speed, meaning that UPC is not an essential function. However, in Internet access services, the traffic flowing into the IMT-2000 network from the Internet may not always be accurately predicted; thus, UPC control at the gate node in an ATM-based CN is effective in providing quality-assurance-type

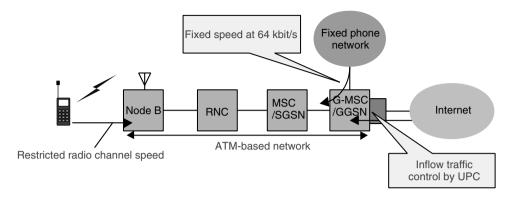


Figure 4.10 Example of UPC applied to mobile communications network

Internet access services, or in effectively using the network bandwidth by keeping the inflow of traffic to a level that can be processed by radio resources.

4.3 Network Control and Signaling Scheme

This chapter describes the signaling system and procedures specified on the basis of the IMT-2000 network architecture.

4.3.1 CN Signaling Systems in IMT-2000

The signaling systems for IMT-2000 CN are an evolution for the signaling systems for a GSM CN. The technical specifications that specify the signaling system have been produced and maintained by 3GPP and approved as the Japanese standards by the Telecommunication Technology Committee (TTC). The signaling system for an IMT-2000 CN has been evolved in order to achieve providing global mobile multimedia, pursuing economy, offering flexible network services and assuring a communication quality equivalent to that of fixed networks.

Figure 4.11 shows an example of a signaling system in IMT-2000 CNs. The following is an explanation of the functions of the signaling system in each interface and the characteristics of the applicable protocols.

4.3.1.1 Interface between User Equipment (UE) and MSC/SGSN

Two protocols are specified for providing CS services: the Call Control (CC) protocol, which controls CS connection between the UE and the MSC; and Mobility Management (MM) protocol, which is a protocol for supporting location management, security management and mobile equipment management. These protocols are specified by extending the GSM, and CC has additional features including multi-CC procedures, which provides multiple active calls simultaneously on the same terminal, speech calls through the use of a new speech coding scheme known as Adaptive Multi/Rate (AMR), and multimedia call (videophone) control functions based on 3G-H.324/M, which is an extension of H.324/M.

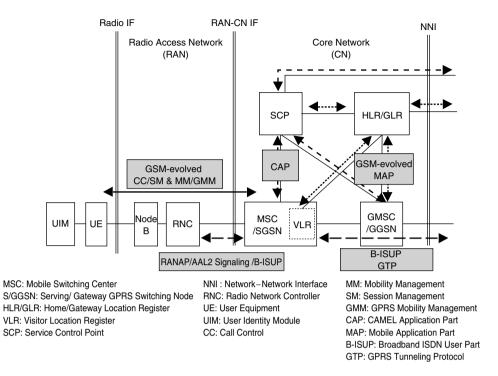


Figure 4.11 IMT-2000 CN signaling scheme

Functional additions have also been made to MM, such as the authentication procedures for executing new security steps including network authentication.

On the other hand, two control protocols are prescribed for providing PS services: Session Management (SM), which is used for session activation, session modification and session deactivation between the UE and the SGSN; and GPRS Mobility Management (GMM), which is an MM protocol for PS. These protocols were developed by extending GPRS. As for SM, functional enhancements have been made incorporating new attributes to specify the QoS and procedures to inform the UE of the name of the Internet Service Provider (ISP) in the event of receiving PDU from ISP.

In addition, two protocols are defined for providing various SSs and SMS: SS, which adds new service procedures for IMT-2000 (e.g. multicall) to the GSM specifications; and SMS pursuant to the GSM specifications.

4.3.1.2 Interface between RNC and MSC/SGSN

For this interface, the Radio Access Network Application Part (RANAP) was newly developed, which has the function to transparently forward CC/MM, SM/GMM and other signals exchanged between UE and MSC/SGSN, control RAN including paging and ciphering executed from CN to RAN, instruct the establishment and release of radio access bearers, and carry out maintenance. For PS domain, the GPRS Tunneling Protocol (GTP) is further applied in order to create, modify and delete tunnels for forwarding user packets.

Between RNC and MSC, AAL2 transmission protocol is applied, which efficiently transmits speech and other low-speed traffic with low delays. AAL2 Signaling Protocol CS.1 is applied in order to establish and release AAL2 connections. Meanwhile, PS services are provided by AAL5 transmission based on PVC or SVC. In the case of SVC, Broadband ISDN User Part (B-ISUP) is applied.

These signaling systems operate on the broadband SS7 stacks RNC and MSC/SGSN. MTP3b and Signaling Connection Control Part (SCCP) (Connection less (CL)/Connection-Oriented (CO)) are used as lower-layer protocols. (Note: SCCP is applicable only in the case of RANAP.)

4.3.1.3 Interface between MSC/SGSN and GMSC/GGSN

For PS domain, GTP is applied in order to create, modify and delete tunnels.

In this interface, ATM can be applied in order to transfer IP packets. When applied, B-ISUP is used in addition to the aforementioned GTP. B-ISUP is used in order to establish and release ATM connection on demand. B-ISUP has ATM connection control and CC functions. TTC specified additional functions for B-ISUP, including functions for SSs and functions to transfer billing information used for interconnection between domestic telecom carriers and so on, so as to perform CC equivalent to that of ISDN User Part (ISUP) protocol.

4.3.1.4 Interface between Home Location Register (HLR)/Gateway Location Register (GLR) and MSC/Visitor Location Register (VLR) and SGSN/GGSN

The Global System for Mobile Communications (GMS)-evolved Mobile Application Part (MAP) is used in this interface for location management, authentication of the subscriber, transferring the subscriber data to the visited network, paging and other MM control tasks. With ease, this enables global roaming with GSM/GPRS networks and IMT-2000 networks based on evolved GSM/GPRS. Key functional enhancements of MAP in IMT-2000 include the GLR-based procedures, which help reduce the number of MAP signals between networks upon global roaming, and the prepaging procedure, which executes paging before setting up CS connection between the visited MSC receives and the GMSC.

Interface between Service Control Point (SCP) and MSC

In this interface, service control is executed using IN technologies, and the CAMEL Application Part (CAP) is used. CAMEL, which stands for Customized Applications for Mobile network Enhanced Logic, is a technology developed by the European Telecommunications Standards Institute (ETSI) with the aim to apply IN to GSM/GPRS mobile communications. In CAMEL, the information indicating the trigger, the MSC to request service control to the SCP, is included in the subscriber information stored in HLR, and forwarded to VLR upon location updating. On the basis of this trigger, MSC requests SCP to carry out service control, and various SSs are offered.

Service control procedures by CAMEL are discussed in Section 4.5.

Table 4.2 summarizes the network capabilities, protocols and specifications relating to IMT-2000 CNs.

No.	Network function and protocol	Specification
1	Mobile Radio Interface Layer 3 specification (Basic call	TS 24.007, TS 24.008,
	control: CC, MM, SM, GMM, supplementary service	TS 24.010, TS 24.011,
	control: SS, short message control: SMS etc.)	TS 24.080, and others
2	UTRAN Iu Interface RANAP signaling	TS 25.413 and others
3	Mobile Application Part (MAP) specification	TS 29.002 and others
4	General Packet Radio Service (GPRS); GPRS Tunneling Protocol (GTP) across the Gn and Gp interface	TS 29.060 and others
5	Customized Applications for Mobile network Enhanced Logic (CAMEL) Phase 3; CAMEL Application Part (CAP) specification	TS 29.078 and others
6	AAL2 signaling protocol CS1	JT-Q. 2630 and others
7	Broadband ISDN User Part (B-ISUP)	JT-Q. 2763 and others
8	ISDN User Part (ISUP)	JT-Q. 763 and others
9	Bearer Independent Call Control (BICC)	ITU-T Q. 1902.X ITU-T Q. 765.5
10	Multicall supplementary service	TS 24.135 and others
11	Multimedia	TS 29.007 and others
12	Gateway Location Register	TS 23.119, TS 29.119, TS 29.120, and others
13	Prepaging	TS 23.018 and others
14	Out of band transcoder control	TS 23.153 and others
15	Security features and security mechanisms	TS 33.102 and others

 Table 4.2
 Main protocols, network functions and specifications relating to IMT-2000

Note: UTRAN: UMTS Terrestrial Radio Access Network.

4.3.2 Control Scheme

This section elaborates on the basic procedure of IMT-2000 CN, multicall, multimedia control and so on, which were added as new network capabilities.

4.3.2.1 Basic Procedure

IMT-2000 networks adopt the VLR scheme for MM control. The following is an overview of the basic procedure in CS services with reference to MM, call origination and termination and handover control. The signaling system and procedure for PS services is discussed in Section 4.4.

Mobility Management

For MM, location updating procedure and attach/detach procedure are specified pursuant to the GSM system. Location updating is performed in the event of moving across the location area, and in the process, the subscriber information is downloaded by the visited VLR from HLR. Attach/detach procedure is defined for the network to know the reachability of paging to the UE: in attach state, the UE can respond to paging, whereas in detach state, the UE cannot respond to paging. This procedure helps prevent the unnecessary paging signals by not performing paging in detach state. The attachment/detachment status managed by the network changes under the following circumstances: when the terminal explicitly informs the network of the attach/detach status as soon as the power is turned on/off; when the terminal fails to periodically perform location registration, which is regarded an implicit detachment; and when the network initiated detach procedures are performed at an arbitrary timing. There are also procedures specified for requesting location updating and attach/detach procedures simultaneously to CS and PS networks, which help reduce the number of signals compared to executing such procedures independently in CS and PS networks. Figures 4.12 and 4.13 show the location updating procedures and the attach/detach procedures, respectively.

In the location updating and attach procedures, a Temporary Mobile Subscriber Identity (TMSI) can be assigned to the UE that is identified by an International Mobile Subscriber Identity (IMSI), which is allocated permanently to each UE.

When a user requests service after location registration, the use of TMSI for user identification on the radio bearer results in improved security compared to IMSI because it conceals IMSI and reduces the signaling volume carried over the Paging Channel (PCH), as the signaling volume of TMSI is only about half of IMSI.

Figure 4.14 illustrates the location registration and attachment procedures.

Mobile Originating Call Establishment

As illustrated in Figure 4.14, in location updating, the VLR number visited by the user is kept in the memory of HLR, and the subscriber information is downloaded from HLR into VLR visited by the user. As the information required for call origination is kept in VLR visited by the user at the time of location registration, there is no need to access HLR upon call origination after completing location registration. The call-origination procedures can be divided into four steps.

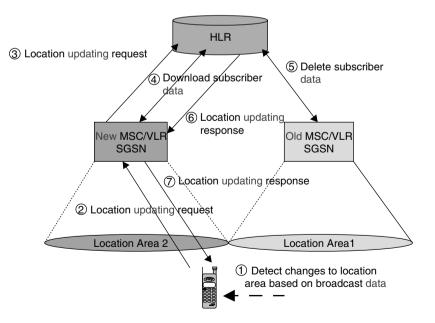


Figure 4.12 Overview of location registration procedures

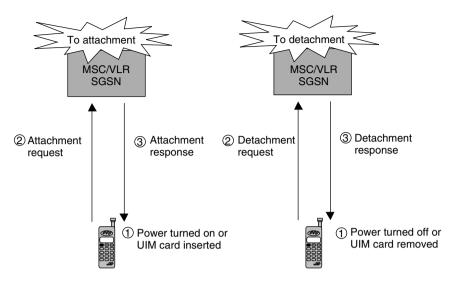


Figure 4.13 Overview of attachment and detachment procedures

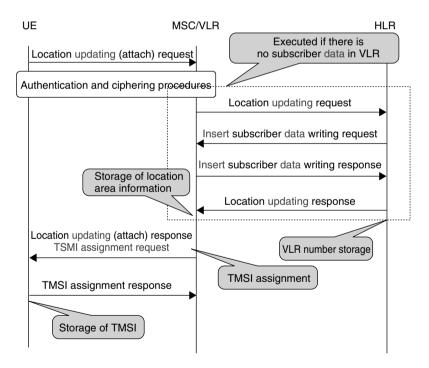


Figure 4.14 Location registration and attachment procedures in IMT-2000

(1) Establishment of Signaling Connection Between UE and MSC/VLR

For the UE to start communicating with the network, signaling connection between UE and RAN, and SCCP connection between RAN and MSC/VLR is established as an initial message is transported from the UE to the MSC/VLR. This enables the transmission and reception of message between UE and MSC/VLR. Subsequently, authentication and ciphering is performed between UE and MSC/VLR to establish a secure control connection.

(2) Initiation of Call Setup

After the completion of step (1), the UE transmits the called number, bearer type and information relating to SSs to MSC/VLR to start call setup. MSC/VLR checks as to whether the information related to the requested call is consistent with the user's contract information.

(3) Radio Access Bearer and Bearer Setup

On the basis of the bearer type and other information notified from the UE in step (2), the radio access bearer and the Transcoder/InterWorking Function is set up. Also, Initial Address Message (IAM) is transmitted to the terminating network based on the called number and so on and bearer establishment is performed.

(4) Completion of Call Establishment

After the completion of step (3), the called party is alerted. Communication begins when the called party responds to alerting of the call.

Figure 4.15 shows the call-origination procedures in IMT-2000.

Mobile Terminating Call Establishment

(1) Routing upon Call Termination

When the GMSC receives an incoming call, it interrogates HLR about the current location of the UE and notifies the visited MSC of an incoming call by HLR. On the basis of the response, the number required for routing the IAM to the visited MSC is acquired. The number for routing is defined as the Mobile Station Roaming Number (MSRN), which is assigned to each call in visited VLR. The MSRN for Public Switched Telephone Network (PSTN)/ISDN routing shall have the same structure as international ISDN numbers in the area in which the roaming number is allocated. MSRN can be identical to the MSISDN (The MS international ISDN numbers are allocated from the ITU-T Recommendation E.164 numbering plan) in certain circumstances in mobile communication networks.

MSRN assigned by the visited VLR is released after receiving IAM, the link connection request to efficiently use resources for MSRN. The visited MSC/VLR subject to MSRN request can execute paging before receiving the link connection request signals based on the prepaging procedure described later.

Figure 4.16 shows the procedures for routing to the visited MSC/VLR.

(2) Paging

In mobile communications, a UE must be informed when there is an incoming call to the UE. The network manages the location area of the UE, and a Location Area Identifier (LAI) is assigned to each location registration area. The network broadcasts that there

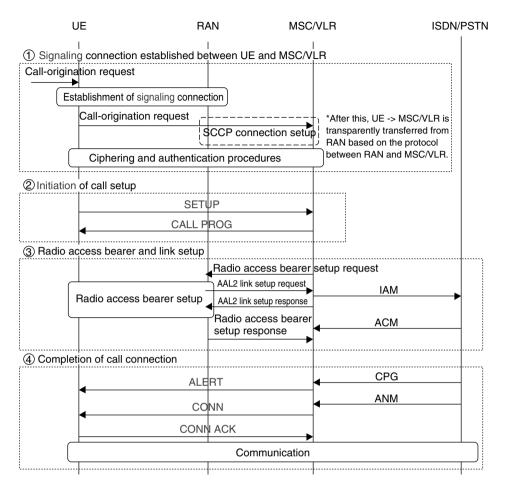


Figure 4.15 Call-origination procedures in IMT-2000

has been an incoming call to all UE in the location area in which the UE has its location registered. This procedure is referred to as *paging*.

Paging is performed by transmitting a paging request signal from MSC/VLR to all RNCs belonging to LAI registered in the visited VLR. The use of TMSI as the user identifier at this stage is preferred in terms of security and signal volume than IMSI, as mentioned before. If the UE is in the idle mode, it is constantly monitoring the PCH so that it can recognize a paging addressed to itself. If LAI and TMSI (IMSI) in the paging request turns out to be the same as LAI and TMSI (IMSI) memorized by the terminal, the terminal sends a response to the network. The network can assure security by executing authentication and ciphering after receiving the response.

Figure 4.17 shows the paging procedures in IMT-2000.

Handover

In IMT-2000, the functions of RAN and CN are clearly separated, and functions dependent on the radio system are concealed inside RAN. Therefore, for example, handover may take

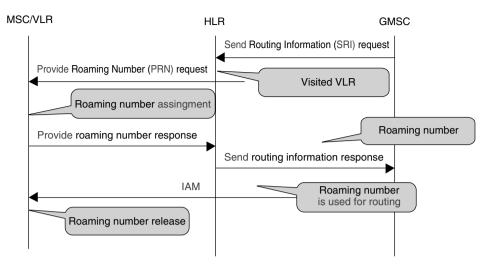


Figure 4.16 Routing procedures in IMT-2000

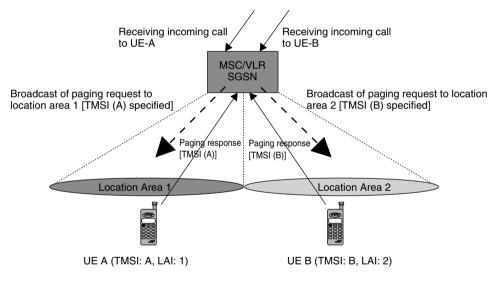


Figure 4.17 Paging procedures in IMT-2000

place between radio zones under two MSCs as illustrated in Figure 4.18; the handover procedures of the CN in IMT-2000 are limited to the relay functions in cases in which the radio bearer (both control channel and user information channel) between handover RNC and UE runs via MSCs except of RNC relocation.

4.3.2.2 New Network Technologies

This section reviews the new network technologies introduced in IMT-2000, succeeding to GSM.

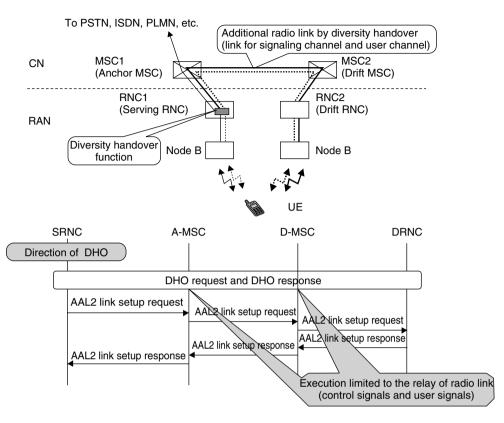


Figure 4.18 Handover control in IMT-2000 CN

GLR Mechanism

IMT-2000 is a global system, which is assumed to be applied in Europe, Japan, the USA, South Korea, China and many other countries worldwide. Roaming across national borders is expected to increase as a result, and the signal volume over transnational SS7 is likely to swell proportionately.

GLR is a node that is positioned between VLR/SGSN and HLR. The introduction of this node makes it possible to reduce the signal volume when location updating or SS procedures are carried out between different Public Land Mobile Network (PLMNs). GLR acts like a visited VLR/SGSN for the HLR and functions like an HLR for the visited VLR/SGSN. Figure 4.19 is a conceptual diagram showing how the signal volume is reduced by the introduction of GLR.

The network-control procedures of the GLR Mechanism are as follows.

(1) Initial Location Registration from Roaming Network

Figure 4.20 shows the procedures for performing location updating for the first time from the roaming network. HLR memorizes the GLR node number as the VLR number, and VLR memorizes the GLR node number as the HLR number.

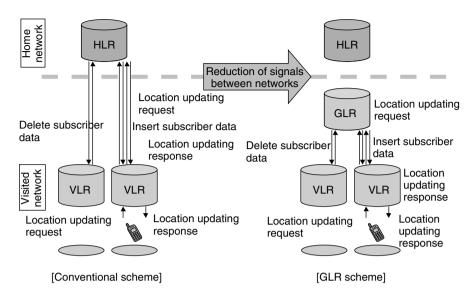


Figure 4.19 Difference between conventional scheme and GLR scheme

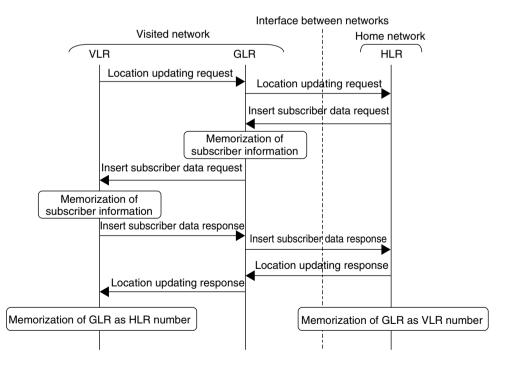


Figure 4.20 Initial location registration from roaming network

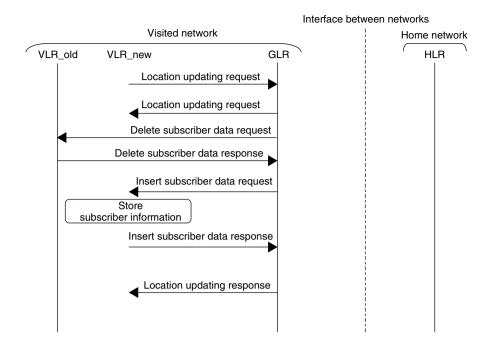


Figure 4.21 Location registration in roaming network

(2) Location Registration in Roaming Network

Figure 4.21 illustrates the procedures for performing location registration in the roaming network. As HLR access is not necessary, no signals are transmitted between HLR and GLR.

The introduction of GLR as explained in the preceding text makes it possible to cut the signal volume between networks.

Prepaging Scheme

Prepaging is a function that executes paging when HLR interrogates MSC/VLR about the MSRN in the event of mobile terminating call. Conventionally, paging has been performed after MSC receives IAM. Figure 4.22 shows the difference between the normal paging procedure and the prepaging procedure.

The introduction of the prepaging procedure enables an earlier detection of "Notreachable" because the UE is out of coverage and so on compared to the normal paging procedure. The following effects are appreciated as a result.

(1) Improved Efficiency of Resource-Utilization between GMSC and MSC

When the probability of "link setup not completed" is compared to that of Notreachable in the network, the probability to have incomplete paging procedures is higher. Therefore, it is easy to discern that it is more likely for the paging procedures to be incomplete after the completion of link setup than the bearer setup to be incomplete after the completion of paging procedures. The implementation of the prepaging procedure makes it possible

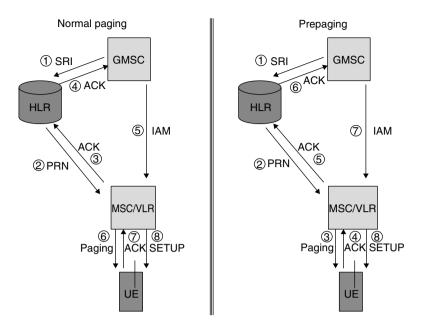


Figure 4.22 Difference between normal paging scheme and prepaging scheme

to confirm the response from the UE in advance and complete the paging procedures, and therefore prevent ineffective bearer setup.

(2) Quick Activation of Services Triggered by Not Reachable

IMT-2000 can provide ones or announcements to the calling party or activate SSs for the called party using Not Reachable as a trigger. In the event of Not Reachable, services can be activated faster than in conventional systems. Figure 4.23 shows the procedures.

CODEC/Transcoder Control (Out-of-Band Transcoder Control)

In mobile communications, low-speed CODEC is generally used in the radio channel so as to use radio resources in an efficient manner. As illustrated in Figure 4.24, in conventional mobile communication systems (such as PDC and GSM), transcoding has been performed in the switching equipment and G.711-coded speech was exchanged between MSCs. This type of connection–referred to as *tandem connection*–requires the network to carry out transcoding twice, which gives rise to poorer speech quality due to coding errors and delays. In order to prevent the execution of transcoding like this, the CODEC types must be negotiated between the MSCs and transcoder control is required so that conversion into G.711 does not take place when the CODEC types match. GSM adopts Tandem-Free Operation (TFO), which executes low-speed CODEC control by stealing part of the G.711 codes. PDC adopts CODEC-bypass control, which involves the execution of speech quality is prevented on the basis of a connection without transcoder as illustrated in Figure 4.25.

Although GSM and PDC have been able to prevent speech quality from deteriorating by TFO and CODEC-bypass control, respectively, they have not been able to efficiently

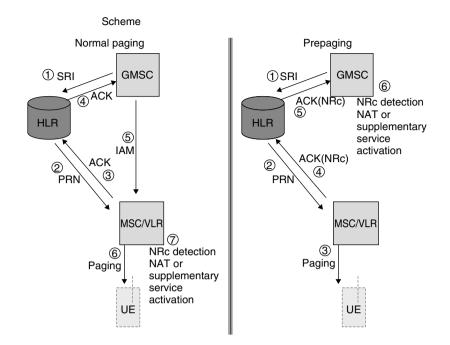


Figure 4.23 Service activation in the event of not reachable under prepaging scheme

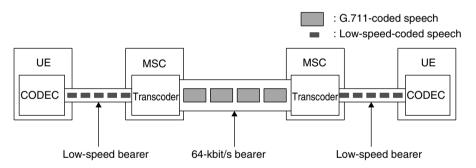


Figure 4.24 Tandem connection

use the transmission bandwidth between MSCs as shown in Figure 4.25, as they are not capable of setting up bearers with a speed lower than 64-kbit/s between MSCs and are therefore forced to use 64-kbit/s bearers even for the transmission of low-speed coded speech.

In order to utilize the transmission bandwidth between MSCs more efficiently, IMT-2000 can use the Bearer Independent Call Control Protocol (BICC) and Q.AAL2 as the protocols between MSCs. BICC is different from conventional ISUP in that the former only has CC capabilities, whereas the latter has both CC and bearer control capabilities. It is used in combination with bearer control protocols such as Q.AAL2. BICC has the ability to carry out low-speed CODEC control, such as end-to-end CODEC negotiation

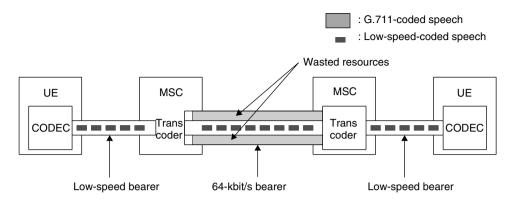


Figure 4.25 Bypass connection of transcoding in PDC/GSM

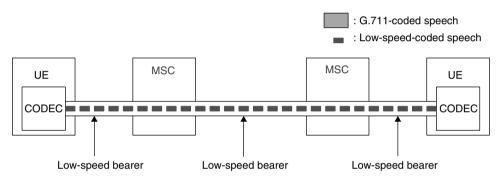


Figure 4.26 Bypass connection of transcoding in IMT-2000

and modification of CODEC during communications. It can set up a bearer for lowspeed CODEC between switching stations in combination with Q.AAL2 and other bearer control protocols. On the other hand, the air interface between UE and the MSC of IMT-2000 is different from GSM in that it adds CODEC negotiation and the ability to change CODEC during communication according to the capabilities of BICC. According to these capabilities, IMT-2000 negotiates the CODEC to be used end-to-end before setting up the bearer, selects the CODEC to be used, and the bearer suitable for the selected CODEC is set up. This Out-of-Band Transcoder Control enables the MSC to bypass unnecessary transcoding as illustrated in Figure 4.26, and to establish a connection that effectively uses the transmission bandwidth between MSCs.

Figure 4.27 illustrates the flow of Out-of-Band Transcoder Control in concrete terms.

In order to apply Out-of-Band Transcoder Control, it is necessary to adopt an extended CN architecture model in which C-Plane control and U-Plane control are separated from the CN architecture model referred to in Figure 4.1.

Security Mechanism

Some of the typical security functions of IMT-2000 are described in the following text.

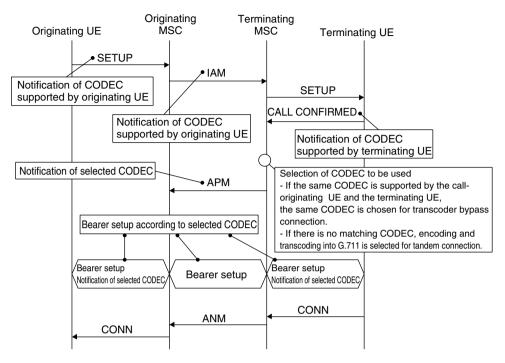


Figure 4.27 Out-of-Band transcoder control

(1) User Authentication Function

This is a function to confirm that the user accessing the network is legitimate. As the key and algorithms relating to the user authentication process are kept by the USIM and the network in advance, the process is carried out by the UE and the network based on random numbers generated by the network at every instance of user authentication process, and the processing results sent from the UE are checked by the network with reference to the processing results of the network. In IMT-2000, the parameter length is extended for the user authentication processing results so as to enable highly secured authentication processing.

(2) Network Authentication Function (New Function Added to GSM)

This is a function to check the legitimacy of the network accessing the user. As the key and algorithms relating to the network-authentication process are kept by the USIM and the network in advance, the process is carried out by the UE and the network based on random numbers generated by the network at every instance of user authentication process, and the processing results sent from the network are checked by the UE with reference to the processing results of the UE.

(3) Ciphering Function

This is a function for preventing overhearing of control signals and user information in the radio bearer. Ciphering is performed by encrypting the control signals and user information using a Ciphering Key (CK) unique to each user and a ciphering algorithm.

(4) Integrity Protection Function (New Function added to GSM)

This is a function for the receiving entity to be able to verify that signaling data has not been modified in an unauthorized way since it was sent by the sending entity in the radio bearer. A Message Authentication Code for Integrity (MAC-I) is generated using an integrity key unique to each user and an integrity algorithm, and is transmitted together with signaling information. Integrity is verified at the receiving side by comparing MAC-I to the MAC-I computed by the receiving side.

Among the functions referred to in the preceding text, functions (1) and (2) are mainly implemented between UE and CN, whereas functions (3) and (4) are primarily applied between UE and RAN. Figure 4.28 shows the procedures of functions (1) and (2), and Figure 4.29 illustrates the procedures of functions (3) and (4).

Multicall

IMT-2000 can provide multicall services, in which multiple CS calls can be communicated on the basis of an independent radio bearer assigned to each call. Standard specifications (R99) do not allow multicalls of two or more speech calls but do permit other combinations (e.g. speech call and unrestricted digital information call, speech call and multimedia call), making it possible to offer a wide range of multimedia services simultaneously to users. New functions of IMT-2000 for providing multicall services are described in (1) and (2) in the following text.

(1) Stream Identifier (SI)

GSM uses a Transaction Identifier (TI) on CC protocol to identify different calls. In multicall, it is necessary to identify not only the calls but also the radio bearers used by the calls, as multiple radio bearers are used for each call. Stream Identifier (SI) is used to associate a particular call with a radio bearer and to identify whether a new Traffic CHannel (TCH) is requested for the call. When making an outgoing call or receiving an incoming call, a user can specify SI corresponding to the call. The user can differentiate

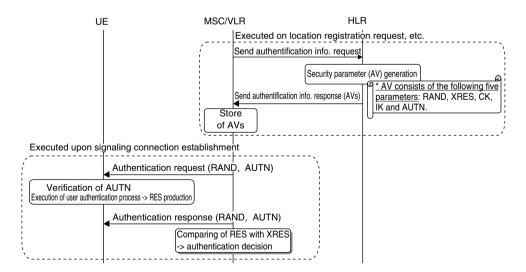


Figure 4.28 Security procedures between UE and CN

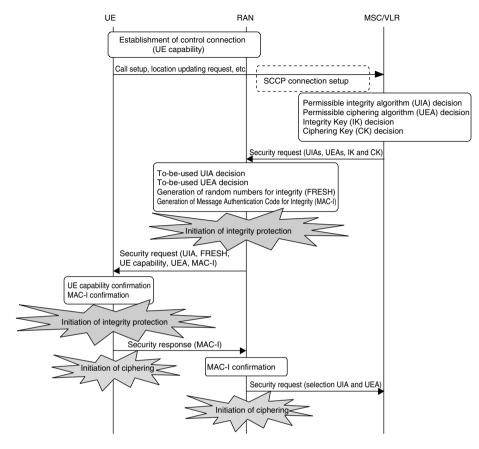


Figure 4.29 Security procedures between UE and RAN

the use of Call Waiting (CW) and Call Hold (CH) services from multicalls, as a radio bearer already setup can be shared by multiple calls if the UE specifies an SI value already used by a call engaged in communication upon the origination and termination of additional calls. Also, in the event of call termination, the user is not able to know about the termination of the call until user alerting is initiated, and the SI value cannot be determined as a result. Therefore, the SI value may be determined when the user replies to the alerting, instead of setting the SI value when the user initially replies to the call-termination request from the network.

Figure 4.30 shows the concept of this function.

(2) Functions to Change and Confirm Permissible Number of Multicalls

The maximum number of multicalls offered to users can be determined by the network. It is also possible for each user to set the number of calls allowed to him/her on an individual basis, provided that it does not exceed the number of permissible calls. Figure 4.31 shows the procedures for changing and confirming the number of permissible calls based on user operation.

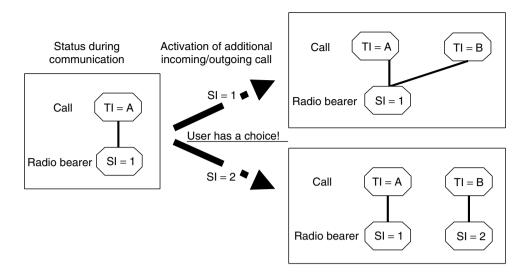


Figure 4.30 Concept of SI functions

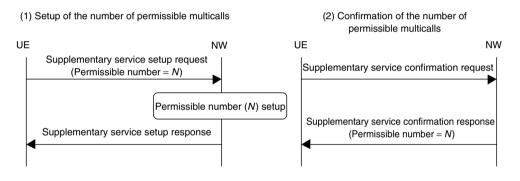


Figure 4.31 Supplementary service procedures associated with multicall

Figures 4.32 and 4.33 illustrate the call-origination and call-termination procedures, respectively, relating to multicall connection enabled by the functions mentioned in the preceding text.

Multimedia Call Connection

IMT-2000 has the ability to offer multimedia communications using 3G-H.324/M, which is an extension of H.324/M. Two types of multimedia communications are supported by IMT-2000: 3.1 kHz audio Multimedia, which is used to communicate with H.324; and Unrestricted Digital Information (UDI) Multimedia, which is used to communicate with H.324/I. In order to differentiate the use of these two types of multimedia communications, and to enable the set up of suitable radio bearers for each

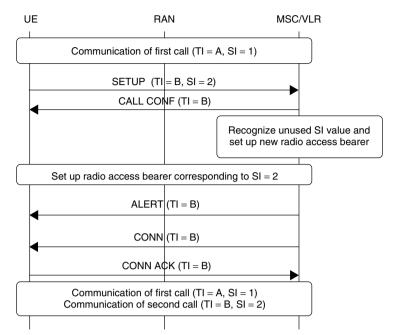


Figure 4.32 Connection procedures in the event of multicall origination

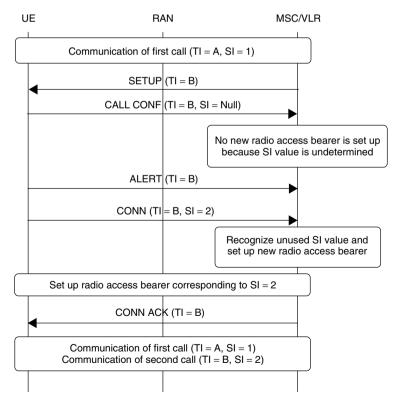
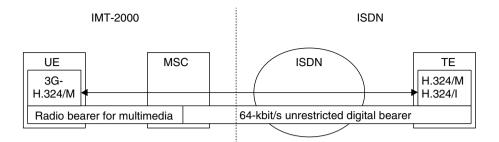
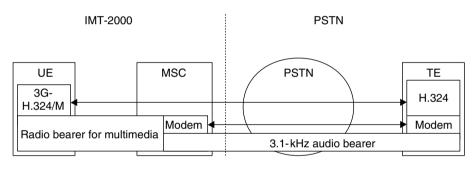


Figure 4.33 Connection procedures in the event of multicall termination



BC IE setup valueInformation transfer capability = UDIRate adaption= H.223 & H.245Fixed network user rate= 64 kbit/s

Figure 4.34 UDI multimedia



BC IE setup valueInformation transfer capability = UDIRate adaption= H.223 & H.245Modem type= V.34Foxed network user rate= 28.8 kbit/s or 33.6 kbit/s

Figure 4.35 3.1-kHz audio multimedia

multimedia type, Bearer Capability Information Element (BCIE) of CC protocol in the interface between UE and MSC is extended. UDI Multimedia uses UDI bearer (Unrestricted Digital bearer) as the bearer between MSCs as depicted in Figure 4.34. On the other hand, a 3.1-kHz audio Multimedia uses a modem in the switching equipment and uses a 3.1-kHz audio bearer to execute communications with other terminals, as illustrated in Figure 4.35. A 3.1-kHz audio Multimedia has the ability to automatically fall back on speech communications if the terminal at the other end turns out to be a speech terminal. As shown in Figure 4.36, the MSC automatically switches to speech communications when the modem in the MSC cannot detect modem signals from the other side after connecting with the destination terminal.

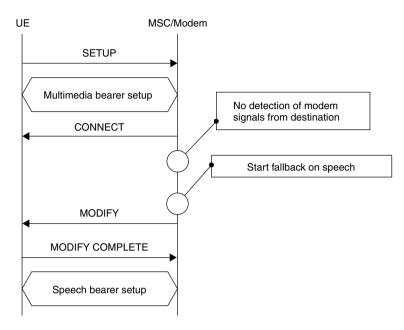


Figure 4.36 Fallback to speech in 3.1-kHz audio multimedia

4.4 Packet Communication Scheme

4.4.1 Overview of Mobile Packet Communications

In recent years, Internet access and other forms of data communications have carried out transmission and reception of data by dividing them into small parcels called *packets*. Serverclient applications, such as the World Wide Web (WWW), are characterized by asymmetric communications in which the data volume sent from the server to the client is greater than that sent from the client to the server. In addition, the duration in which the user actually transmits or receives data often constitutes a small proportion of the total time of the connection.

If normal CS connections are used to enable data communications characterized by the above-mentioned communication properties (refer to Table 4.3), network resources allocated for CS cannot be utilized to the full extent.

For this reason, mobile packet communication schemes, which forward the data transmitted and received by data terminals in the form of packets inside the mobile communications network, have been commercially implemented in 2G systems in Japan (PDC-P), the United States [Cellular Digital Packet Data (CDPD)] and Europe (GPRS) [7].

The Japanese *i-mode* is an application based on PDC-P, and is aimed at converging the Internet and mobile communications. *i-mode*, which was commercially launched in February 1999, represents a case in which the data communication needs of users have been met. The fact that *i-mode* acquired 17 million subscribers in just over 18 months since its service launch (as of December 2000) shows how high the demand is for packet communications. It shows that packet communications have much potential to become the mainstream of mobile data communications.

Traffic type	Feature	Application (example)
Transaction type	Relatively small data generated on an irregular basis	– E-mail etc.
Block type	Relatively large data generated infrequently	WWWFTP etc.
Stream type	Average-sized data generated periodically	Internet phoneSpeechVideoconference (video) etc.

Table 4.3 PS traffic

Note: FTP: File Transfer Protocol.

Packet mode in IMT-2000 is attracting a great deal of attention as a communication scheme that carries a wide range of multimedia services, including but not limited to *i-mode* mentioned in the preceding text.

The following paragraphs discuss the service target, mobile packet communication technologies and connection schemes required in IMT-2000.

4.4.2 Service Target

Figure 4.37 shows an example of mobile multimedia services presumed under IMT-2000. IMT-2000 aims to achieve a maximum transmission speed of 2 Mbit/s (indoors) and 384 kbit/s (outdoors), and is able to carry a wide range of media, including speech, still pictures and video. It has the potential to offer services with different uplink/downlink speeds according to the application, such as uplink-downlink symmetric communications, asymmetric communications and broadcast communications. These services may be applied not only to CS communications but also to packet communications; CS would be more suitable if the forwarded information is continuous, whereas PS would be preferred if the forwarded information is dispersed. Table 4.4 shows the service targets of IMT-2000.

Real-time information (e.g. speech, video), which has primarily been offered through CS, can also be provided on the basis of the introduction of QoS control in PS so as to meet service quality requirements in terms of speed, delay and so on.

4.4.3 Network Architecture

Figure 4.38 shows the configuration of the logical nodes and key functions for providing mobile PS communications standardized under 3GPP.

SGSN, which is a subscriber node (or visited node), manages subscriber information of the visiting mobile user in the area (e.g. available QoS service class, information on connectible connection destination, authentication information). The subscriber information is downloaded from HLR when the power of the mobile station (i.e. user equipment: UE) is turned on. SGSN determines whether to connect on the basis of the aforementioned subscriber information when there is an incoming call or outgoing call to and from the UE and executes connection control with the connection destination.

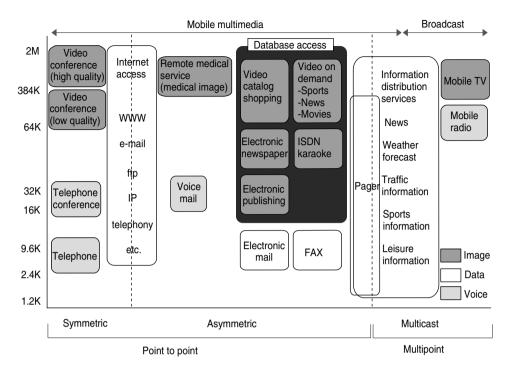


Figure 4.37 Mobile multimedia

Gateway GPRS Support Node (GGSN), which is a gateway node, performs access control (e.g. connection notice and call-termination notice etc.) with an ISP connected with an IMT-2000 network. GGSN serves as an interface with a wide range of services that are likely to be provided by ISPs, as described in Section 4.7. GGSN also manages the address for data communications assigned to UE required for packet communications, and controls inquiries to HLR about the current location of UE in the event of call termination.

Domain Name System (DNS) is a same function used over the Internet, and converts the Access Point Name (APN), which is the name specified by the user, into the IP address of the destination GGSN.

Also, both SGSN and GGSN gather billing information so as to provide packet volumebased billing.

4.4.4 Mobile Packet Communications Technologies

4.4.4.1 Transfer Speed and QoS Control

The transfer speed of 2G mobile packet communication systems is 28.8 kbit/s at the maximum in PDC-P and approximately 100 kbit/s in GPRS. Whereas 2G systems improve the transmission speed by bundling multiple Time Division Multiple Access (TDMA) time slots assigned to each user in the radio interface, 3G systems based on W-CDMA offer

	Target	Description
Anytime	 No time restrictions (Available any time) 	Superior user-friendliness and throughputStorage services
Anywhere	 Relaxation of geographical constraints (Uniform services available at any locality) 	 Users need not worry about location and distance (Global roaming)
With anyone	 Communication with a wide range of media (Communication with PC, PDA, MS etc.) WWW and mail access 	 Users need not worry about the communication media of the other party or differences in application software Connection to the Internet, Intranet and mobile terminals
Desired information	 Communication based on complex expression media (Speech, still pictures, video, data etc.) 	 Various traffic support (Large capacity, real time)
Affordable and reasonable pricing	 Reduced communication costs Stratified communication quality and communication fee 	 Cost-cutting technologies Various service classes secured (QoS classes)

Table 4.4 Service targets in IMT-2000

Note: PDA: Personal Digital Assistant.

much faster speeds than 2G systems because they are not constrained by the time slot concept.

As for the QoS control scheme (Table 4.5), the 2G PDC-P system does not have any parameters to specify QoS, and all calls are therefore delivered on a best-effort basis. GPRS is able to specify QoS parameters, but they are merely specified as target values to be sought when the user engages in communication, rather than for the strict control of QoS. For example, the average speed can only be specified by setting the parameters so that the hourly target data volume can be transmitted on average.

In contrast with the abstract QoS designation method in 2G systems, 3G systems are able to specify QoS in much further detail to execute QoS control in more concrete terms. For example, maximum Service Data Unit (SDU) size transmitted from UE and the assurance of the packet arrival order in the network can be specified. Also, for asymmetric communication services, the data transfer speed can be specified separately in uplink and downlink.

The fact that QoS parameters can be specified in concrete terms in 3G systems, as explained in the preceding text, shows that the technologies for flexible bandwidth control using W-CDMA as the radio transmission technology and for more reliable QoS control in the backbone of CN are now mature enough for commercial use.

Although the 3GPP standard does not specify any particular method to implement QoS, one potential solution is to use ATM technology to achieve that. For example, as

SGSN: Serving GPRS Support Node GGSN: Gateway GPRS Support Node HLR: Home Location Register ISP: Internet Service Provider RAN: Radio Access Network UE: User Equipment DNS: Domain Name System

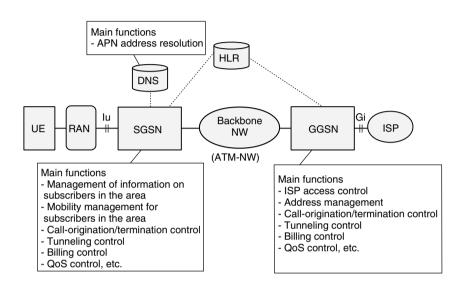


Figure 4.38 Mobile packet logical nodes and main functions

discussed in Section 4.2.4, reliable QoS services can be offered to users by mapping the ATM service classes provided by ATM (Constant Bit Rate (CRB), UBR, VBR) onto the four traffic classes specified by 3GPP (Conversational Class, Streaming Class, Interactive Class and Background Class).

4.4.4.2 Tunneling Control

In the world of mobile communications, the data communications address used in fixed data communication networks (IP address) is not sufficient by itself for forwarding packets to UE, which is the destination of data transfer, because UE physically moves from one location to another in the mobile communications network. Therefore, tunneling technology is applied to mobile communications, which uses a separate address from the destination address (Figure 4.39).

Tunneling technology is aimed at forwarding packets to mobile terminals, which do not stay in one place. The process involves setting up a logical connection for forwarding purposes (tunnel) in advance between GGSN and SGSN that accommodates UE and controls the origination and termination of calls relating to that UE, and forwards packets received by GGSN from other data communication networks using the logical connection from GGSN to SGSN.

When UE moves from area to area and changes the visited SGSN, a new logical connection is reset between the new SGSN and GGSN.

Scheme		2G		3G	
		PDC-P	GPRS	IMT-2000	
Transfer speed		Maximum 28.8 kbit/s <i>i-mode</i> : 9.6 kbit/s	Approx. 100 kbit/s	384 kbit/s (Outdoors) 2 Mbit/s (Indoors)	
QoS Control	Characteristic	No QoS Control	Can designate same uplink and downlink speed only	Different uplink and downlink speed can be designated	
	Number of QoS parameters that can be specified		5 parameters (Delay, maximum speed, average speed, priority and reliability)	12 parameters (Traffic class, delay, maximum uplink and downlink speeds, minimum uplink and downlink speeds, priority, maximum SDU size, sequence assurance etc.)	
	Speed specifying method		Maximum speed: Oct/s Average speed: Oct/h	Maximum uplink speed: bit/s Maximum downlink speed: bit/s Minimum uplink speed: bit/s Minimum downlink speed: bit/s	

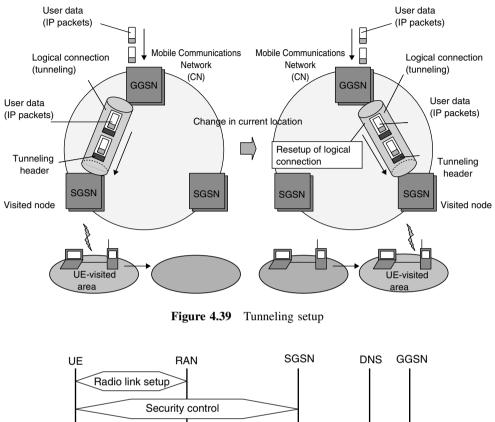
Table 4.5Transfer speed and QoS control [8]

Tunneling control under 3GPP1 is executed with the use of the GTP. The setup procedures are described in detail in the following section (Figure 4.40).

4.4.5 Connection Scheme

The IMT-2000 packet communication scheme is based on a functionally enhanced version of GPRS, which is highly compatible with the 2G CS GMS, in order to support global roaming. In CN, the aforementioned GTP is used as the protocol for tunneling setup. The QoS control and logical connection setup methods of 3G GTP are enhanced compared to 2G. 2G GTP and 3G GTP both carry version information, and are therefore interoperable between 2G networks and 3G networks through version negotiation.

This section discusses call-origination, call-termination and -relocation procedures including the tunneling setup method, using GTP.



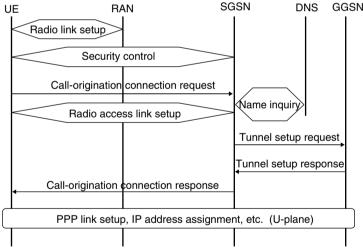


Figure 4.40 Call origination procedure

4.4.5.1 Call-Origination Procedures

As in the case of CS, packet call-origination procedures are based on dial-up connection. The difference with CS dial-up is that it uses APN as the number indicating the ISP in IMT-2000. APN is a scheme to search the IP address from the name using the Domain Name Server (DNS), which is also used for the Internet. Similar to specifying the name

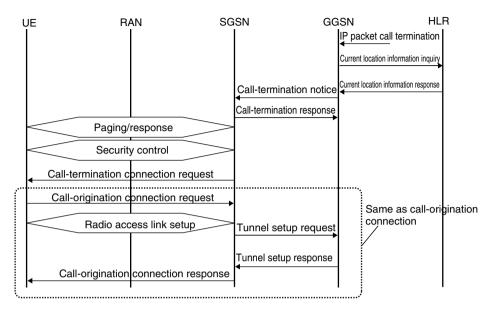


Figure 4.41 Call-termination procedures

when designating a URL with WWW, connection with the desired destination (ISP or Intranet) can be established by specifying APN from UE. The connection procedures are described in the following text (Figure 4.41).

UE firstly sets up the control link with RAN if there is no control link set up with the network. The procedure is the same as that in CS. This is followed by user authentication with SGSN. Once the legitimacy of the user has been proved, UE specifies the APN of the connection destination and the parameters for the required QoS, and sends a call-origination connection request to SGSN. SGSN searches the subscriber information of the user who made the call-origination request, and decides the destination APN and the specified QoS. Simultaneously, it inquires DNS to acquire the IP address of GGSN to be connected from the destination APN. On the basis of the decision result, SGSN sends a radio link setup request to RAN. In the process, information on QoS specified by the user (e.g. speed) is notified to RAN, and the corresponding radio channel is set up. On the other hand, CN transmits a tunnel connection request to GGSN using the acquired IP address of GGSN. In the process, the corresponding QoS information is forwarded. GGSN identifies the external network that matches with the destination information. GGSN sends back a tunnel connection response to SGSN. SGSN returns a call-origination connection response to UE. On the basis of the procedures up to this stage, the path between UE and GGSN is set up. Subsequently, the Point to Point Protocol (PPP) link with GGSN is set up from the UE. PPP link setup involves the same procedures as those associated with normal dial-up arrangements. Steps performed include user authentication and the assignment of IP address.

4.4.5.2 Call-Termination Procedures

Packet call termination is activated when an IP packet is forwarded to GGSN. As the IP address and the mobile phone number (IMSI) assigned to UE is registered in GGSN in

advance, GGSN can determine the IMSI with reference to the IP address in the header of the IP packet. GGSN inquires HLR about the address of SGSN visited by UE using IMSI as the key information. The address of SGSN is registered in HLR when UE registers communication with the network, or when it moves from SGSN to SGSN, which requires location registration procedures. GGSN notifies the visited SGSN of the call termination according to the SGSN address based on the inquiry to HLR. In response, SGSN sends back a response to GGSN. SGSN pages all UEs to identify the cell in which the UE is currently located, and executes security control (authentication and ciphering) on the UE. Once the UE has been proven to be legitimate, a call-termination connection request is notified to the UE. Then, the UE sends a connection request to SGSN. The subsequent steps are the same as in the call-origination connection procedures (Figure 4.42).

4.4.5.3 Relocation Procedures

In the case of CS, when UE moves from a governing subscriber node to another subscriber node during communication (i.e. internode handover), the call-controlling subscriber node is anchored to extend the subscriber line, which is referred to as the *anchor scheme*. PS also adopts the anchor scheme as in the case of CS if UE is subject to internode handover during communication. However, when hardly any data is transmitted or received, UE resorts to the *relocation scheme* for internode handover in order to change the current location. In the relocation scheme, the aforementioned logical connection for tunneling is reset in conjunction with the switching of the CC node.

The procedures are as follows. The old RNC decides the execution of relocation, and then gives a relocation order to the higher-node old SGSN. In response to the relocation

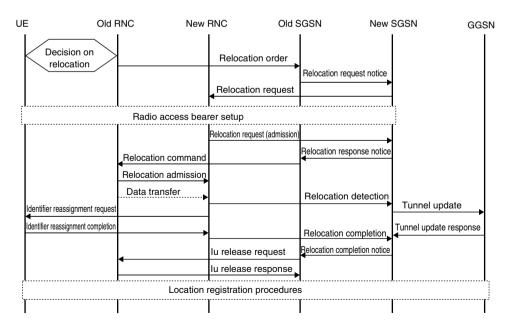


Figure 4.42 Relocation procedures

order, the old SGSN makes a relocation request to the new RNC via the new SGSN. After the radio access bearer between UE and the new RNC is set up, the new RNC informs the old RNC of the admission of relocation via the new SGSN and the old SGSN. At this stage, the preparation tasks for relocation are completed. The old RNC sends information on the radio access bearer, information on sequence numbers of packet data and other information held by the old RNC before the relocation to the new RNC, and notifies the completion of relocation admission. If there is any data to be forwarded, it is transferred to the new RNC. Subsequently, as part of the relocation procedures, the new RNC sends the relocation detection to the higher-node new SGSN and simultaneously sends a request to reassign the identifier to UE. The new SGSN sends a tunnel update request to GGSN in order to reset the tunneling. At this point, the downlink data packet is forwarded to the new SGSN. Then, the Iu link between the old SGSN and the old RNC is released, marking the end of the relocation procedures. This is followed by normal location registration procedures, and HLR is informed of the switching of SGSN and subscriber information is inserted into the new SGSN (Figure 4.40).

4.5 Intelligent Network (IN) Scheme

SSs in W-CDMA mobile communication systems will be based on GSM-standard services (e.g. call forwarding and CW) and IN-based services. This section outlines the IN scheme, compares IN with conventional networks, discusses the merits of IN and reviews the standardization trends.

4.5.1 Overview of IN Scheme

The IN scheme was proposed and implemented in wired networks during the 1980s. Before the advent of IN, switching equipment was in charge of basic CC as well as the control of SSs. The problem of this arrangement was that the software of all switching equipment had to be changed when adding and extending services, which was extremely time and labor consuming.

The IN scheme was advocated to cut the time consumed in such tasks, reduce the workload, advance the services and meet diversification requirements. Under the IN scheme, a node separate from the switching equipment is in charge of service control and the management and storage of service-related data, so that services can be added and extended without changing the software in switching equipment.

The IN scheme is described in brief in the following text (Figure 4.43).

The IN scheme takes the following measures to separate the service control node and the service execution node:

- 1. standardize the functions required for the execution of service,
- 2. specify the Basic Call-State Model (BCSM) and
- 3. specify the Detection Point (DP) in the state transition model.

The IN scheme divides the function for providing SSs into multiple functions, so that the services can be offered on the basis of a combination of functions. The service control node instructs the activation of functions to the service execution node, and the service execution node activates the functions according to the instructions so as to render the services. On the basis of this arrangement, the service control node and the service

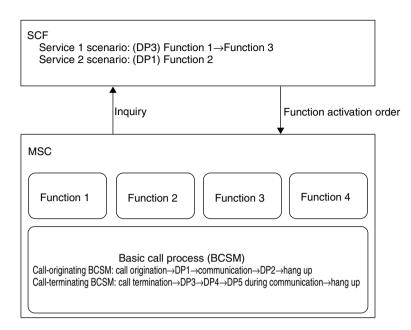


Figure 4.43 Overview of IN scheme

execution node can be clearly separated. Moreover, the service execution node is solely responsible for basic call-processing control, as the separation of service control and basic call-processing control helps reduce processing delays. The service execution node must recognize the conditions in which the activation of SSs might take place based on the state transition model of basic call processing, and inquire from the service control node as to whether it is necessary to activate SSs. Under the IN scheme, the activation of SSs in the course of basic call processing is made possible by prescribing BCSM.

4.5.2 Comparison with Conventional Systems

In conventional mobile communication systems, the service control node was separated from the service execution node by giving the function of the service control node to HLR, which manages user location information (Figure 4.44). However, the software of both the service control node and the service execution node had to be updated for the introduction of new SSs, because the functions were not standardized and the provisions for BCSM and so on were not specified.

In W-CDMA mobile communication systems, the Service Control Function (SCF) is placed as the service control node separated from MSC, which is the service execution node. The IN scheme is accomplished by assigning BCSM and various service functions to MSC (Figure 4.45). This arrangement makes it possible to add services without updating the software of the service execution node, which had been a challenge for a long time. In addition, services can be provided between different networks by SCF control (VHE), as the interface between SCF and MSC, BCSM and various service functions have been standardized.

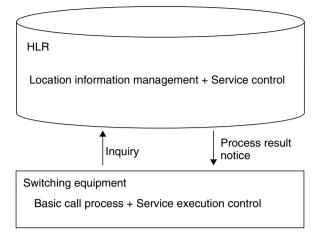


Figure 4.44 Conventional system

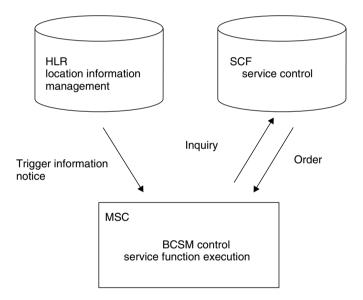


Figure 4.45 IN scheme in W-CDMA mobile communications system

VHE enables service control nodes to give instructions to nodes in other networks in the same manner as nodes in the home network by standardizing the interface and the functions of the execution nodes. It is a technology that enables all networks to be virtually controlled as the home network.

As the separation of HLR and SCF enables the complete separation of basic call processing and SS control, SSs can be operated on a trial basis or offered in certain regions only, making it possible to implement services in various ways.

The difference between the conventional scheme and the IN scheme is described in the following text with reference to the case of incoming call transfer as an example.

Figures 4.46 and 4.47 show the control procedures under the conventional scheme and the IN scheme, respectively. In the conventional scheme, (1) the switching equipment inquired HLR about the service contract status in the event of call process and (2) HLR sent back the contract status in consideration of the conditions in which the inquiry was made and the switching equipment decided the status of the call process and the activation of services.

In the IN scheme, when there is a call process, (1) MSC shifts the states up to the DP of BCSM and when the state reaches DP, it searches the trigger information that indicates whether to send an inquiry to SCF, and if any trigger information is found, (2) MSC sends an inquiry to SCF. SCF identifies the type of the trigger for the inquiry, decides the service to be activated on the basis of the user's contract status and the conditions in which the call is made and (3) instructs the activation of functions required for MSC and the timing at which an inquiry is to be made next time.

Accordingly, in contrast with the conventional scheme in which the switching equipment had to make decisions according to the contract service, MSC under the IN scheme merely has to decide the basic call state and whether to conduct an inquiry at each DP, and the subsequent control tasks are executed as instructed by SCF.

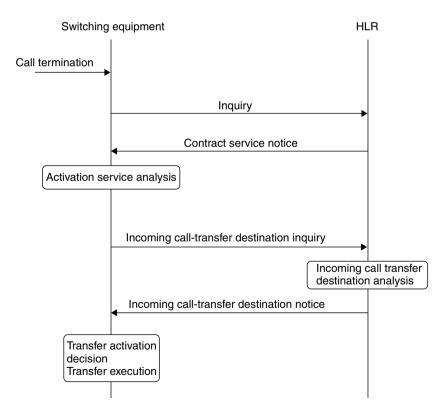


Figure 4.46 Incoming call-transfer procedures under conventional scheme

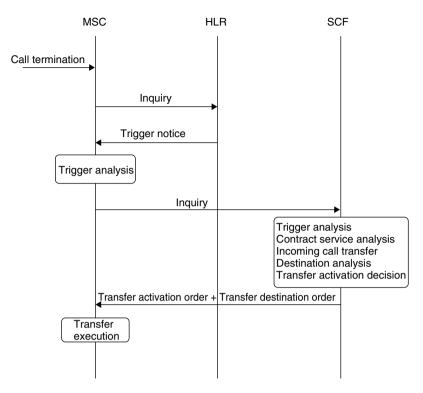


Figure 4.47 Incoming call-transfer procedures under IN scheme

4.5.3 Merits of the IN Scheme

There are three significant merits in the IN scheme as described in the following text:

- 1. quick provision of a wide range of services,
- 2. lower costs and shorter time required for the development of services and
- 3. ability to provide the same service over different networks without standardizing the service itself.

As mentioned before, under the IN scheme, all services are accomplished by the combination of standard functions and manipulating the activation timing, and these functions are implemented in the service execution node. As long as the interface between SCF and MSC is standardized, additional services can be provided simply by adding functions to SCF and the above-mentioned merits can be enjoyed thereby.

4.5.4 Standardization Trends

The GSM standard, currently used widely in Europe and other parts of the world, adopted CAMEL to apply IN for mobile communications. Since W-CDMA mobile communication systems use evolved-GSM as the Core Network, 3GPP is working on the standardization of the IN scheme for 3G in the direction to extend GSM CAMEL.

4.5.5 Future Prospects

In the world of mobile communications, IN-based services are currently limited to prepaid services and the IN has been applied only to Circuit Switch connections. Studies are being conducted with the aim to apply IN to a wider range of services and Packet Switch and to realize VHE.

4.6 Short Message Scheme (Figure 4.48)

4.6.1 Overview of Scheme

SMS is a service that allows users to send text data up to a certain size to UE. On the basis of this service, users can send and receive text data up to 140 octets (70 double-byte characters) via a server called the Short Message Service Center (SMSC) [10].

4.6.1.1 Message Origination and Delivery

To send and receive a short message between UEs, the message-originating UE sends a short message containing the address of the call-terminating UE to SMSC. When SMSC receives the message, it sends back a report indicating if the short message was successfully received or not by SMSC to the originating UE, and then delivers the short message to the terminating UE. After the reception of the short message, the terminating UE sends back a report to SMSC indicating if it was successfully received.

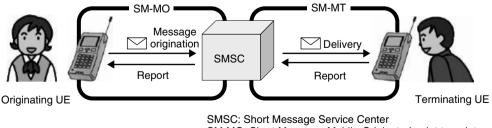
The procedures for delivering the short message from the originating UE to SMSC is called Short Message-Mobile Originated Point-to-Point (SM-MO), whereas the procedures for delivering it from SMSC to the terminating UE is referred to as Short Message-Mobile Terminated Point-to-Point (SM-MT).

Figure 4.48 illustrates the transmission of short messages.

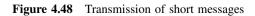
4.6.1.2 Repeated Delivery Attempt (RDA)

SMSC has the function to store a short message and try to deliver it repeatedly if it cannot be delivered temporarily because the power of the terminating UE is turned off or it is out of the service area and so on.

There are two forms of RDA, as described in the following text.



SM-MO: Short Message–Mobile-Originated point-to-point SM-MT: Short Message–Mobile-Terminated point-to-point



(1) RDA Activated by SMSC

This refers to autonomous RDA by SMSC after a certain period or the expiration of the storage period.

(2) RDA Activated by HLR

This refers to RDA performed through the following procedures. Information that there is a message in the originating SMSC waiting to be delivered to the terminating UE (hereinafter referred to as "Message-Waiting Indication") is managed by HLR as a part of the subscriber data of its own. When the UE becomes ready to receive the short message, the HLR sends an RDA request to the originating SMSC.

An RDA request from HLR to SMSC is triggered when:

- 1. terminating UE has resumed operations (e.g. location registration, power ON, call termination or origination etc.) and when
- 2. terminating UE has memory newly available for short message reception.

Figure 4.49 illustrates the HLR activated RDA.

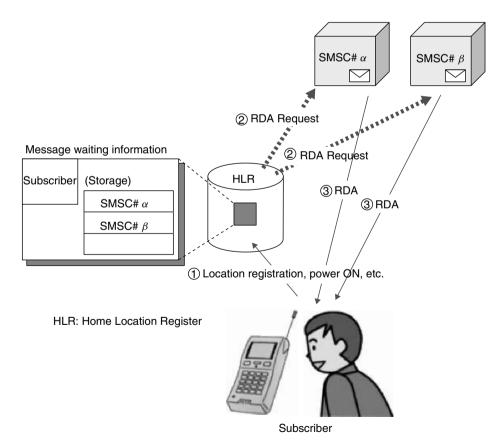


Figure 4.49 HLR-activated RDA

4.6.1.3 Status Report Capabilities

The SMS offers the capabilities of notifying the sender of the result of a previously sent short message. In cases in which such delivery result notice is designated, SMSC generates a *status report*, which records the delivery results, upon the reception of a delivery report from the terminating UE and delivers it to the originating UE. The status report is delivered on the basis of SM-MT procedures.

Figure 4.50 illustrates the status report.

4.6.2 Network Configuration

Figure 4.51 illustrates the network configuration for SMS. Table 4.6 shows the functions of each node.

4.6.3 Routing Scheme

Figure 4.52 illustrates the SM-MO.

Figure 4.53 depicts the SM-MT.

Figure 4.54 shows the RDA, with reference to an example in which retransmission is activated by HLR based on memory availability notice.

4.6.4 Main Extended Functions of SMS

SMS supports the following extended functions.

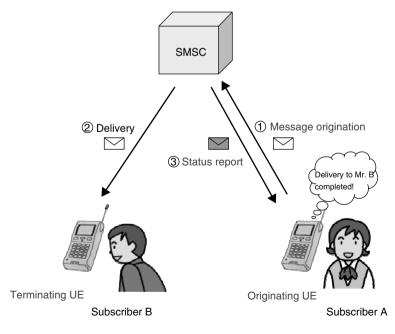


Figure 4.50 Status report

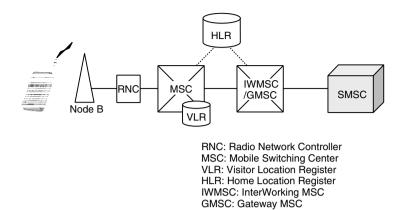


Figure 4.51 Network configuration

SMSC: Short Message Service Center

Table 4.6 Functi	ions
------------------	------

Function name	Function	
SMSC	 Relay/forward short messages from the originating UE etc., to the terminating UE. Temporarily store the short message if the transmission of the short message fails. 	
SMS-IWMSC	 Receives short messages forwarded from MSC and transfers them to SMSC. 	
SMS-GMSC	 Receives short messages delivered from SMSC and forwards them to the MSC visited by the terminating UE. Accesses HLR upon the reception of short messages from SMSC, inquires about information on the current location of UE and decides the routing. Registers message waiting indication in HLR in the event of failure to deliver the short message. 	
MSC	 SM-MO Receives short messages forwarded from Node B and forwards them to IWMSC connected with the designated SMSC. Inquires VLR about the service information of the originating UE upon the reception of short messages from Node B, and decides whether to transmit the messages. SM-MT Receives short messages forwarded from GMSC and forwards them to Node B. 	
VLR HLR	 Manages subscriber data relating to subscribers located under the MSC. Manages SMS information, current location information, and other subscriber data of each user. Manages information on undelivered short messages (address information of SMSC storing the short messages) for each user. Activates RDA request for short messages. 	

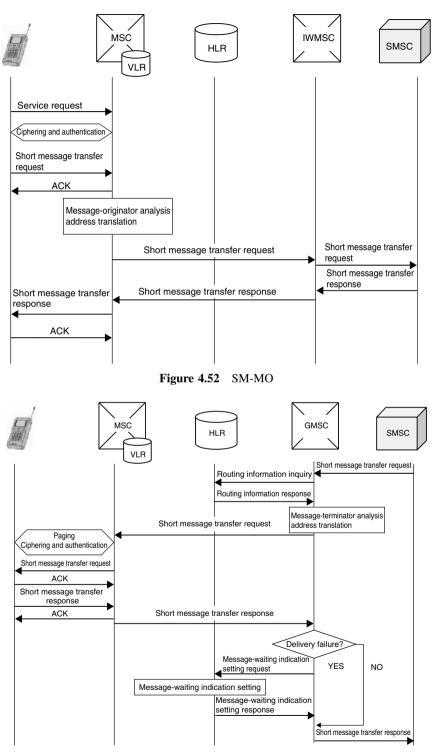


Figure 4.53 SM-MT

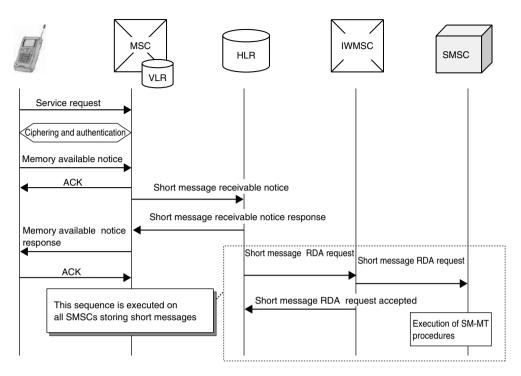


Figure 4.54 HLR-activated RDA (e.g. activation by memory available notice)

4.6.4.1 Text Concatenation

For transmitting data exceeding 140 octets, the text concatenation function enables the data to transmit by dividing it into multiple short messages. The data to be transmitted is divided by the originating UE and sent as multiple short messages. And the terminating UE recovers the original data by concatenating the short messages. Up to 255 short messages can be joined together.

4.6.4.2 Reply Path

Reply path secures a path for short messages for reply purposes as specified by the originating UE. For delivering a reply from the replying UE, the SMSC used for transmitting the short message by the originating UE is used for replying as well.

4.6.4.3 SMS and Internet E-mail Interworking

E-mail can be exchanged between UE and the Internet via short messages. In order to interwork with e-mail, the message must be written in the prescribed format.

4.6.5 Example of SMS Applications

As short messages can also be sent and received by devices other than UE, SMS may be applied to the following services.

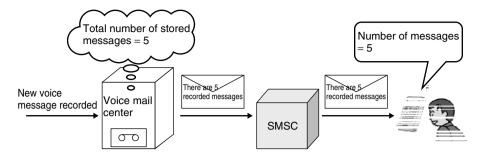


Figure 4.55 An example used for notice of the number of stored messages in voice mail service

[Example] Notification of the Number of Messages stored in Voice Mail Center

In voice mail services widely used in mobile communications, SMS can be used to inform subscribers of the number of voice messages stored in the voice mail center.

When the number of voice messages stored in the center changes as a result of additional recording or deletion of a voice message, the center generates a short message indicating the number of voice messages at that time and sends it to SMSC. SMSC delivers the short message to the terminating UE, based on the same procedures as SM-MT.

Figure 4.55 illustrates notice of the number of messages stored in Voice Mail Center.

4.7 Gateway Scheme

An effective way to add functions on the packet bearer and provide a wide range of services to users in a flexible and quick manner is to install a gateway between CN and an ISPs and so on, which provide the applications. A wide range of services can be rolled out though various gateways (Figure 4.56). Main functions of the gateway include the conversion of the IP into a protocol that is suitable for a mobile environment, and the conversion of addresses. The required gateway functions depend on how the terminals are used

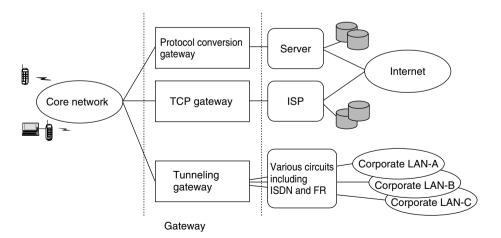


Figure 4.56 Basic configuration of gateway system

(UE standalone, UE used in combination with PC etc.) and the destination of connection (ISP, corporation etc.). This section reviews the functions of three key gateways: protocol conversion gateway, TCP gateway and tunneling gateway. The concept of the multimedia service platform, which is an evolved version of gateways, will also be discussed in brief.

4.7.1 Protocol Conversion Gateway

The protocol conversion gateway is a gateway between the GGSN of CN and the server. One of the functions is to convert protocols so that UE can directly access the Internet from the UE itself. This enables the provision of *i-mode*-like services (e.g. Internet access, information distribution, e-mail) on IMT-2000.

4.7.1.1 System Configuration

Figure 4.57 illustrates the system configuration and the protocol stack. CN sets up the communication channel for the high-speed packet bearer (best-effort type service, with maximum downlink speed of 384 kbit/s and maximum uplink speed of 64 kbit/s), over which IP communication is executed with UE. The transport protocol used is Wireless TCP (W-TCP), which is an improved version of the normal TCP designed to assure efficient transmission in a mobile communications environment. (The nature and effects of W-TCP will be discussed in Section 4.7.2.) Communication between the protocol conversion gateway and the server is based on the standard IP. Functions of the protocol conversion gateway include the conversion of the transport protocol (W-TCP into TCP and vice versa) and the Hyper Text Transfer Protocol (HTTP) termination process. It also supports protocols for CC and Operation And Maintenance (OAM). The protocol conversion gateway does not alter or process information flowing from end to end (data part).

4.7.1.2 Call Control and Functions

Figure 4.58 illustrates the CC sequence for UE to access the server's content. Figure 4.59 shows the sequence to notify the mail reception, UE has an IP address dynamically assigned to it by GGSN upon connection, rather than a static IP address.

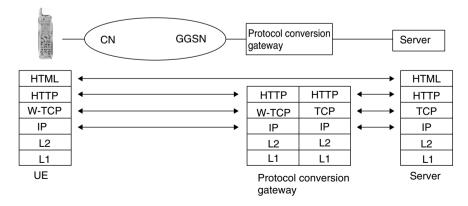


Figure 4.57 System configuration and protocol stack (when protocol conversion gateway is in use)

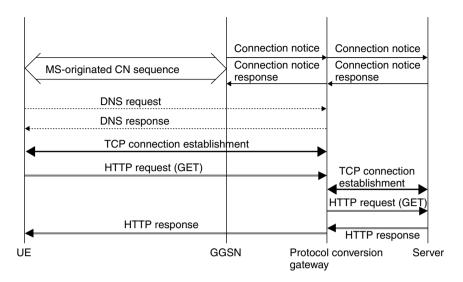


Figure 4.58 Call-origination sequence from UE

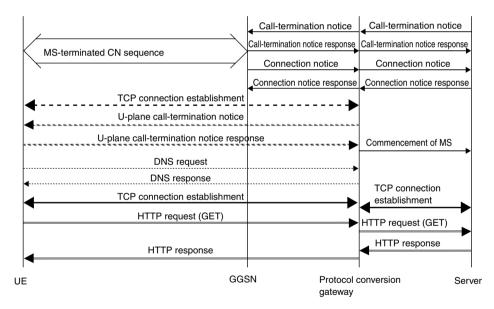


Figure 4.59 Call-termination sequence to UE

Table 4.7 shows the main functions of the protocol conversion gateway.

4.7.2 TCP Gateway

TCP gateway is a gateway between GGSN in CN and ISP and so on. It has the function to convert conventional TCP into W-TCP and vice versa. The existing TCP normally

Function	Description
Conversion of TCP protocol	Converts conventional TCP into W-TCP.
HTTP proxy	Relays HTTP requests and responses exchanged between UE and the server.
Mail notification	Notifies that the new messages arrive at the server.
DNS server	Notifies the IP address in response to inquiries about the address of the protocol conversion gateway from UE.
SSL tunneling	Executes transparent forwarding during SSL-encrypted HTTP communications.

 Table 4.7
 Main functions of protocol conversion gateway

Note: SSL: Secure Sockets Layer.

applied to PCs and so forth might not assure sufficient throughput for high-speed packet switching services of IMT-2000 (downlink speed: 384 kbit/s, uplink speed: 64 kbit/s). The objective of the TCP gateway is to improve the throughput of Internet applications in a mobile communications environment, through the installation of a gateway device equipped with W-TCP.

4.7.2.1 System Configuration

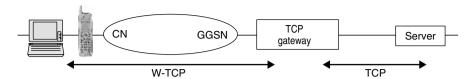
Figure 4.60 illustrates the system configuration and the protocol stack when the TCP gateway is in use. The TCP gateway functions in a connection arrangement in which information services provided by ISP and so on are to be used by UE and client PC. It terminates the TCP connection used for data transfer between the client PC and the WWW server, and relays W-TCP between the client PC and the TCP gateway and the conventional TCP between the TCP gateway and the WWW server. Packets subject to processing by the TCP gateway are limited to those for WWW services. The protocol of TCP applications other than those for WWW services and non-TCP protocols [User Datagram Protocol (UDP), Internet Control Message Protocol (ICMP)] are bypassed without being relayed.

4.7.2.2 Functions

The TCP gateway functions as a completely transparent gateway device: the destination of the connection looks like a WWW server from the client PC's point of view, and from the WWW server's point of view, the connected destination looks like a PC client. The TCP gateway has no CC function. Table 4.8 shows the main functions.

4.7.2.3 W-TCP

W-TCP [6] aims to improve throughput in a mobile communications environment, and it can be connected with the existing TCP since it adopts only the technologies publicly endorsed by the Internet Engineering Task Force (IETF), which is a standardization body



(1) Stack when transparent proxy is in operation (TCP/HTTP)

·			,			
	HTML					HTML
	HTTP]←───►	HTTP	HTTP	↓ →	HTTP
	W-TCP]←───►	W-TCP	TCP	← →	TCP
	IP]∢	IP	IP	↓ →	IP
	L2	L2 L2	L2	L2		L2
	L1] [<u>L1 L1</u>]	L1	L1		L1
	DTE	MS	TCP gate	eway		Server

(2) Stack when traffic bypass is in operation (TCP other than HTTP and UDP)

APL	•			 APL
TCP/UDP	•			 TCP/UDP
IP	<u>+</u>			 IP
L2	L2 L2	L2	L2	L2
L1	L1 L1	L1	L1	L1
DTE	MS	TCP G	ateway	Server

Figure 4.60 System configuration and protocol stack (when TCP gateway is in use)

Table 4.8 Main functions of TCP gateway

Function	Description
Transparent proxy function	 Interconnects W-TCP and TCP. Assures transparency of IP packets. IP address does not change even after passing TCP gateway. Data Terminal Equipment (DTE) and the server look as though they are communicating end-to-end.
Traffic bypass function	- Bypasses packets other than TCP/HTTP.

of Internet technologies. The technologies applied are as referred to in Table 4.9: the expansion of the TCP window size, Selective Acknowledgement (SACK), the expansion of the initial window size, and the expansion of the Maximum Transmission Unit (MTU). Figure 4.61 shows how the technologies are applied. Figure 4.62 illustrates the effects of W-TCP, showing that the throughput is three or four times greater than normal TCP.

4.7.3 Tunneling Gateway

The tunneling gateway is a gateway set between CN and corporate LAN, enabling packet terminals to securely access from mobile environment to corporate LANs via a mobile

Measure	Description	Reference
Expansion of TCP	Expands the transmission and reception buffer size,	RFC793
window size	and increases the volume of data that can be transmitted without waiting for the arrival of Acknowledgement (ACK) signals.	RFC1323
SACK	Requests the retransmission of lost packets only (Selective Repeat) in the event of any packet loss at the receiving side, rather than requesting the retransmission of lost packets as well as the subsequent packets (Go-back-N).	RFC2018
Expansion of	Increases the number of packets that can be	RFC2414
initial window size	transmitted upon the commencement of connection (i.e. reduced impact of slow start).	RFC2581
Expansion of Maximum Transmission Unit (MTU)	Increase the size of data that can be transmitted by 1 packet (i.e. reduced impact of overheads due to TCP header and slow start).	RFC793

Table 4.9 Technologies applied to W-TCP

PS network (IMT-2000 packet, PDC-P). The tunneling gateway and the corporate LAN can be connected in various forms, including frame relay, ISDN and ISP.

4.7.3.1 System Configuration

Figure 4.63 illustrates the system configuration and the protocol stack when the tunneling gateway is in use. The user packet (PPP frame) sent from the client PC on the UE side is relayed through the tunneling gateway into the corporate LAN side based on the Mobile Packet Tunneling protocol (MPT). The PPP frames including user data and data for user identification, is transmitted transparently to the access terminal installed on the corporate LAN side. The transport protocol used is W-TCP rather than the existing TCP. W-TCP described in Section 4.7.2 is also used in this system. Wireless PPP(W-PPP) is used in order to decrease PPP negotiation. W-PPP is a uniquely enhanced version based on PPP Vendor Extensions (RFC2153), with a much shorter PPP negotiation time. The access terminal is a piece of equipment for accessing the corporate LAN server via NTT DoCoMo's network and communication links, and has the function to terminate MPT and W-PPP.

4.7.3.2 Functions

The tunneling gateway has the function to relay and forward user packets sent from a mobile client (IMT-2000 terminal or a PDC-P terminal) to a corporate LAN server. It also has the function to establish a tunnel to relay and forward user packets transmitted by the server to the mobile client. Table 4.10 shows the main functions of the tunneling gateway.

4.7.4 Multimedia Service Platform

The multimedia service platform is based on various services rolled out through the aforementioned gateways, and serves as a common platform that can provide new services

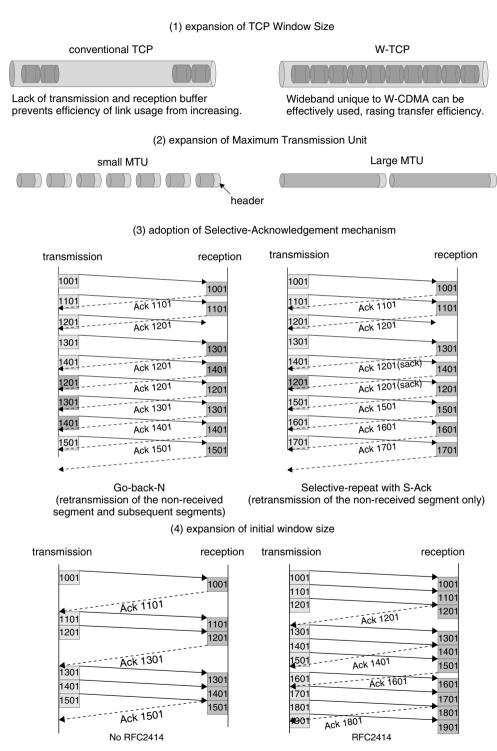


Figure 4.61 Characteristics of W-TCP

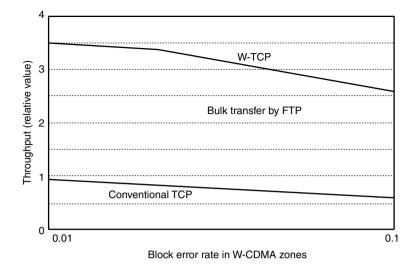


Figure 4.62 Effects of W-TCP (Improved throughput)

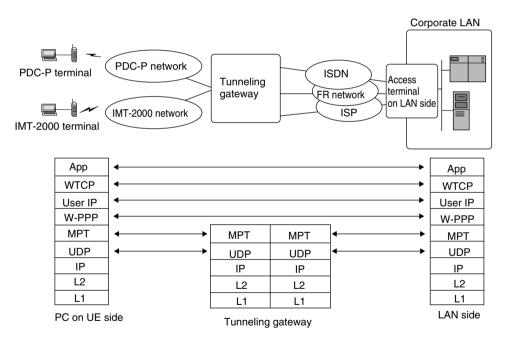


Figure 4.63 System configuration and protocol stack (when tunneling gateway is in use)

in a flexible and efficient manner through further integration and sophistication. Services customized for each ISP and terminal user can be provided by the integration of various gateways, the storage and management of profiles of users (terminal, ISP etc.) and the execution of services according to service scenarios.

Function	Description
Packet relay function	Sets up the link (FR, ISDN etc.) to registered corporate LANs in response to a connection request from a mobile terminal. Relays packets from mobile terminals to corporate LAN
Flow control function	Executes flow control between the mobile terminal and the tunneling gateway, and between the tunneling gateway and the corporate LAN
Call-termination function	Activates the call-termination sequence to GGSN in response to a call-origination request from the corporate LAN side

 Table 4.10
 Main functions of tunneling gateway

4.7.4.1 Architecture

Figure 4.64 shows the basic architecture of the multimedia service platform. Architectural designs discussed at various standardization venues for the next-generation network architecture, including Multiservice Switching Forum (MSF) and the International Softswitch Consortium (ISC), are incorporated in the platform. Specifically, the functions to implement multimedia services consist of three layers: Multimedia Service Management (MSM), which has SCFs; Multimedia Service Agent (MSA), which has service processing functions; and Multi-Featured Gateway (MFG), which is in charge of media conversion, protocol conversion and so forth.

(1) MSM

MSM is a logical node that stores and manages information required for providing services. Specifically, it is in charge of the centralized management of service scenarios required for service processing, various user profiles, customer information acquired from CN, quality control information and so on. The information is distributed to MSA for service processing.

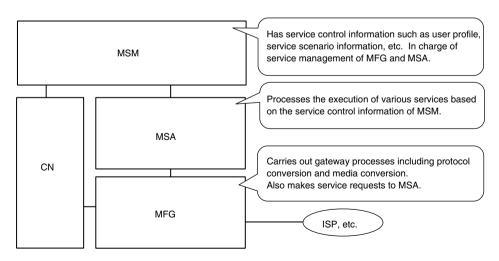


Figure 4.64 Architecture of multimedia service platform

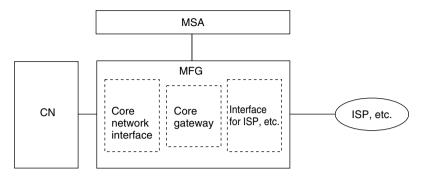


Figure 4.65 Architecture of MFG

(2) MSA

MSA is a logical node that executes service processing. MSA uses the service scenarios and the necessary customer information (various profile data) managed by MSM to execute the processing of services according to the service request from MFG.

(3) MFG

MFG is a logical node that is in charge of protocol conversion, media conversion, and so on between CN and ISP and so forth. It has an integrated architecture as shown in Figure 4.65, rather than a group of gateways referred to in Sections 4.7.1 through 4.7.3. Broadly speaking, MFG consists of a CN interface function, ISP interface function and a core gateway function that performs protocol conversion and so on. For calls that require service processing by MSA, a service request is made to MSA.

4.7.4.2 Example of Services

The following are examples of services provided on the basis of the multimedia service platform.

(1) Location Information Service

In services that provides location information of a UE and so on, to a user, the given location information is converted into longitude, latitude, address and so forth according to the user's request.

(2) Advanced Billing

On the basis of application level information, this service allows the billing address and billing units to be set and changed flexibly, according to the user's request.

References

- [1] ITU-T Recommendation I.363.1, 'B-ISDN ATM Adaptation: Type 1 AAL', August 1996.
- [2] ITU-T Recommendation I.363.2, 'B-ISDN ATM Adaptation Layer Specification: Type 2 AAL', September 1997.
- [3] ITU-T Recommendation I.363.3, 'B-ISDN ATM Adaptation Layer Specification: Type 3/4 AAL', August 1996.

- [4] ITU-T Recommendation I.363.5, 'B-ISDN ATM Adaptation Layer Specification: Type 5 AAL', August 1996.
- [5] 3GPP TS 23.107 V3.3.0, 'QoS Concept and Architecture', June, 2000.
- [6] IETF Internet Draft draft-inamura-docomo-00.txt, A TCP Profile for W-CDMA: 3G Wireless Packet Service.
- [7] 3GPP TS 23.060 V3.6.0, 'Digital Cellular Telecommunications System (phase 2+) General Packet Radio Service (GPRS); Service Description; Stage2', January, 2001.
- [8] 3GPP TS 23.107 V3.5.0, 'QoS Concept and Architecture', December, 2000.
- [9] 3GPP TS 24.008 V3.6.0, 'Mobile Radio Interface Layer3 Specification; Core Network Protocols-Stage3', December, 2000.
- [10] 3GPP TS 23.040 V3.5.0, 'Technical Realization of the Short Message Service', July, 2000.
- [11] 3GPP TS 23.002 V3.3.0, 'Network Architecture', March, 2000.

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

5

Operation System

Masafumi Onuki, Nobutaka Nakamura, Haruo Mizumoto, Takeshi Yamashita, Kazuhiko Hara and Kazuaki Terunuma

5.1 Overview

As an extremely large number of network equipment are installed by telecom carriers to build a mobile communications network, there is a significant number of NEs that affect communication services because of faults. It is important to monitor network equipments 24 hours a day to ensure the stable provision of high-quality services. Operation And Maintenance (OAM) by the Operation System (OpS) is therefore essential in communication networks nowadays.

Hitherto, mobile carriers had to resort to labor-intensive means to maintain the service quality whenever the network was expanded. In the future, it will be necessary to build an operation framework that can take swift action based on strategic information management and assure maximum quality with minimal personnel. This chapter reviews the functions and the mechanism of OpS, which is expected to become increasingly important with the introduction of International Mobile Telecommunications-2000 (IMT-2000), with reference to actual system construction.

5.1.1 Positioning of OpS

The layers depicted in Figure 5.1 show the OAM functions based on the Telecommunication Management Network (TMN) model generally specified by the ITU-Telecommunication Standardization Sector (ITU-T) [1]. The bottom layer represents Network Element (NE), whereas the element management layer positioned above that executes NE management on an individual basis. The NW management layer above the element management layer realizes functions to manage the entire network, which consists of multiple NEs. The service management layer above the NW management layer is positioned to support functions for managing services provided over the network.

On the basis of such a TMN model, the 3rd Generation Partnership Project (3GPP), which is a standardization body for the next-generation mobile communications system, defines the standard interface between the Network Manager (NM) and the Element Manager (EM), and between EM and NE as shown in Figure 5.2. The definition of the

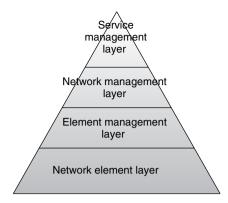


Figure 5.1 TMN layer model under ITU-T

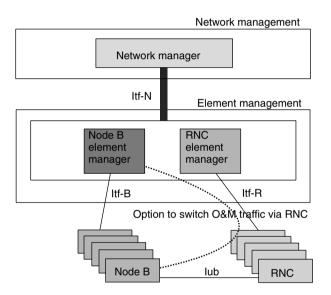


Figure 5.2 Management layer model under 3GPP

element and manager functions and the interfaces between them enable the interconnection of NEs from various vendors and EMs and coordination with NM.

The management functions required by these managers are defined in terms of Open Systems Interconnection (OSI) management [2]. The definition includes fault management, configuration management, accounting management, performance management and security management. 3GPP defines the management process groups using the Telecom Operation Map (TOM) of the Tele Management Forum (TM Forum) that specifies these functions in further detail [3]. By using the management processes referred to in Figure 5.3, interoperability is assured between network operators and service providers, including the exchange of fault information and billing information. In addition, the distribution of software that implements management process functions can cut the investment costs incurred at the time of system development.

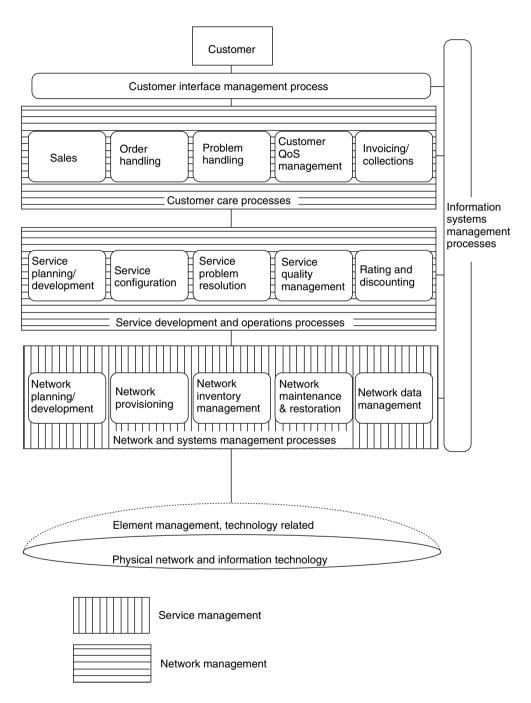


Figure 5.3 Telecom operation map of TM forum

For the provision of IMT-2000 services, the following enhancements were made to the OAM technologies accumulated thus far in addition to the configuration of process groups mentioned in the preceding text [4, 7, 8]:

- 1. the identification of network status and the enhancement of network control focusing on improved service quality;
- 2. the establishment of a flow-through data stream used in construction, maintenance, quality management and planning tasks and
- centralized OAM management of different Personal Digital Cellular (PDC) networks and IMT-2000 networks.

5.1.2 System Configuration

Figure 5.4 illustrates the architecture of IMT-2000 OpS according to TMN layers.

5.1.2.1 NE Management Layer

The NE management layer, which is in charge of OAM of each NE, consists of the following servers.

(i) NE Monitoring System

NE monitoring systems are installed to monitor radio equipment, switching equipment, transmission equipment and other network facilities. Each NE monitoring system manages the status of NE faults, performance, files and the status of system data updating and so on.

Also, the NE monitoring system informs the major alarm aggregate server of the most urgent alarms (major alarms).

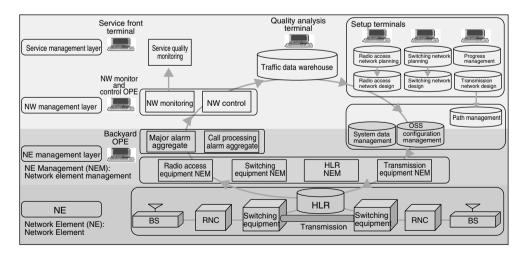


Figure 5.4 IMT-2000 OpS architecture

(ii) Major Alarm Aggregate Server

The major alarm aggregate server collects and manages major alarms of all NEs via the NE monitoring systems. Major alarms from the server are notified to the backyard OPeration Equipment (OPE) that constitutes the control desk. This arrangement enables the operator to identify the activation status of all NE major alarms at one backyard OPE.

(iii) Call-Processing Alarm Server

The call-processing alarm server gathers and stores call release factors because of seminormal processing from the radio equipment and switches. When semi-normal processing frequently occurs, the call-processing alarm server informs the backyard OPE, the NWmonitoring OPE and the service front terminal.

Moreover, in response to the customer's complaints about call abnormalities, the operator can check the status of the abnormalities with reference to the database of the call-processing alarm server using the subscriber number as the key.

(iv) NE File Management System

This system manages the version of systems files of each NE and system data, and executes the remote updating of files in coordination with the NE monitoring system. To let the data flow through using system data, system data is updated on the order of (design system \rightarrow NE file management system \rightarrow NE monitoring system \rightarrow NE), and the operation data of various OpS equipment (server/OPE) is created and updated on the order of (NE file management system \rightarrow OpS configuration management system \rightarrow NE monitoring system \rightarrow NE

(v) OpS Configuration Management System

This system appropriates NE files and system data from the system data management server, automatically creates the data required for the operation of various OpS equipment (server operation data and OPE screen display data), and downloads the data to the OpS equipment. This makes it unnecessary for the operator to make new entries associated with the creation of data for various OpS equipment, and enables the updating of operation data of OpS equipment in sync with the updating of the system data of NE itself.

(vi) Backyard OPE

This is a control desk for carrying out maintenance tasks associated with the NE management layer in general. One OPE monitors and controls various NEs based on common operation.

5.1.2.2 NetWork (NW) Management Layer

NW Monitoring System

In coordination with the NE monitoring system and the NW control system, the NW monitoring system gathers fault information and performance information (status of resource usage) of radio equipment and switching equipment required for network control, as well as traffic data such as circuit connection status and call loss state, and notifies the NW monitor and control OPE. This enables the operator to simultaneously identify the service quality status in the entire network – from the access network up to the switching network – and the status of NW components, that is, the network status.

NW Control System

The NW control system gathers fault information and performance information on switching equipment, detects congestion and executes network control by ordering restriction control to the switching equipment and Radio Network Controller (RNC) as required.

This enables prompt and accurate network control aimed at assuring service quality and maximizing the utilization of networks.

Traffic Data Warehouse

The traffic data warehouse gathers traffic information of the radio equipment and the switching equipment on a regular basis, via the NE monitoring system and the NW control system. The primary database, in which the data is stored, is designed for nonstandard forms, that is, the user can freely create the forms. Parts of the data stored in the primary database are automatically edited and processed at nighttime to suit standard forms required for periodic management, such as weekly reports, monthly reports and annual reports, and are stored in the secondary database.

The quality management operator applies general-purpose On-Line Analytical Processing (OLAP) tools to the network data stored in these databases to promptly analyze various data.

Path Management Server

The path management server receives the circuit order information of the transmission design system, automatically opens the circuit and manages the transmission path configuration information. It also detects faults in the path and informs the operator of such faults.

NW Monitor and Control OPE

This is a control desk for accessing the NW monitoring system and the NW control system, for the purpose of monitoring and controlling the network.

5.1.2.3 Service Management Layer

Service Monitoring System

The service monitoring system provides the service front with network information that is useful for identifying the service quality, including the call loss status and call-processing alarms identified by the NW monitoring system in the network management layer.

This enables the service front to identify the network quality status at real time and deal with customers accordingly.

Planning System

The planning system receives traffic information from the traffic data warehouse, assesses the traffic records, forecasts the traffic and develops facility plans.

Design System

On the basis of the planning information from the planning system, the design system is in charge of the logical NW design such as the allocation of NEs and physical designs including path design and capacity design. It also creates system data based on the design information.

As described in the preceding text, the IMT-2000 OpS is adapted to large-scale operations by integrating various servers and systems. The following sections explain network monitoring, network control, NE monitoring and NE management, which have been especially reinforced in the IMT-2000 OpS.

5.2 Network Monitoring

NTT DoCoMo's existing NW monitoring OpS monitors the switching network by calculating the call volume and the connection rate between switching equipment through common channel signal monitoring. In other words, it is a monitoring system specializing in switching networks.

NTT DoCoMo has developed a NW monitoring system with the following two objectives in mind in order to *identify the network status and execute network control focusing on the improvement of service quality*, which is one of NTT DoCoMo's basic OpS concepts [6].

- 1. Monitor the traffic status of the entire network, from the access network to the switching network and
- 2. Monitor the connection status between switches, and monitor call losses. Execute monitoring in a manner that complies with the customers' perception.

The NW monitoring OpS focuses especially on fault information and performance information (status of resource utilization) of the radio equipment and switching equipment required for network control and on traffic data including link connection status and call loss status.

5.2.1 Configuration of Network Monitoring Functions

Figure 5.5 illustrates an example of the configuration of NW monitoring functions.

The NW monitoring system monitors the traffic status and the equipment status of the network as a whole, from the access network to the switching network, on the basis of coordination between the NW-monitoring Core Network (CN) server, the NW-monitoring Radio-Access Network (RAN) server and the NW-monitoring fault server, as shown in Figure 5.5.

NW-Monitoring CN Server

NW-monitoring CN server gathers information on the status of resource utilization and the status of link connection from the switching NE in the switching equipment, in addition to various traffic data relating to call loss via the NW control system. The gathered information is subject to threshold decision and notified to the NW monitor and control OPE.

NW-Monitoring RAN Server

NW-monitoring RAN server is informed of the status of resource usage from the access NE of the Base Station (BS) and RNC, and various traffic data relating to call loss that

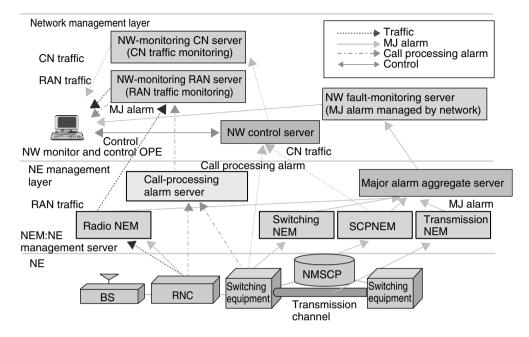


Figure 5.5 Configuration of network monitoring functions (example)

exceed the threshold via radio Network Element Management (NEM). Apart from the traffic data, information of abnormal call processing in excess of the threshold in the radio equipment and the switching equipment are notified via the call-processing alarm system. The information notified to these servers is passed on to the NW monitor and control OPE at real time.

NW-Monitoring Fault Server

NW-monitoring fault server is informed of significant faults in BS, switching equipment and transmission path via the major alarm aggregate server. The information notified to these servers is passed on to the NW monitor and control OPE at real time.

5.2.2 Characteristics of Network Monitoring

Figure 5.6 illustrates how NW monitoring works. The following is the description of the characteristics of NW monitoring.

(i) Total Monitoring: from Access Network to Switching Network

The operator monitors the service quality status of the nationwide network from RAN through CN. The operator also checks the level of impact on services in the event of access network failure by displaying the affected area in the map.

(ii) End-to-End Call Loss Monitoring

CN monitors the call loss status from the call-originating switching equipment to the call-terminating switching equipment, end-to-end. This makes it possible to monitor the connection quality and determine, for example, where it is difficult to establish calls.

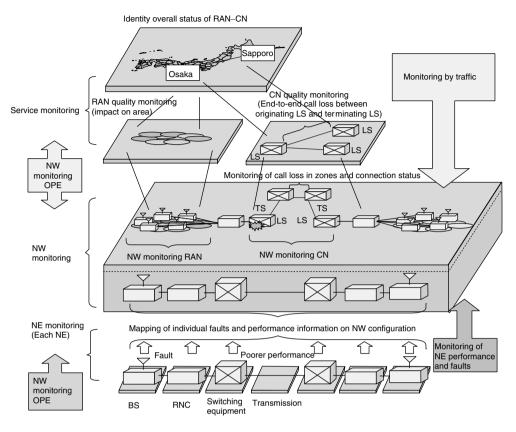


Figure 5.6 NW monitoring image

(iii) NE Performance Information and Fault Information are Displayed to Identify the Causes of NW Failure According to RAN Strata and CN Strata, Other than Traffic Information Relating to Call Loss and Connection Status

The operator executes NW control with reference to such detailed information. The operator can also identify the effects of control in a quantitative manner by checking the call loss status.

Table 5.1 shows the monitored items in the monitoring layer referred to in Figure 5.6.

Information to be monitored by operators engaged in network control mainly concerns the items monitored in the service-monitoring layer and the NW-monitoring layer. Information to be monitored by service staff at the service front is primarily based on the items monitored in the service-monitoring layer. Information to be monitored by operators involved in element maintenance is mainly based on items monitored in the NE-monitoring layer.

Using the information handled in these monitoring layers, NW monitoring and NE monitoring tasks are coordinated with each other so as to monitor, analyze and take measures with respect to NW and NE in a comprehensive manner.

Figure 5.7 shows the operation flow of NW monitoring and NE monitoring.

Monitoring layer	Objective and overview of monitoring	Monitoring item	Monitoring user
Service- monitoring layer	Monitoring focusing on identification of service quality status	 RAN connection quality items: traffic CH busy rate, PCH busy rate, call origination/termination completion rate etc. CN connection quality items: blocking probability, connection completion rate Call-processing alarm 	Service front network control
Network- monitoring layer	Identification of NW status focusing on network congestion and restriction control	 NE resource utilization status NE restriction status Major faults in NE Connection status, call loss and number of completed calls between NE and NE. 	Network control
NE monitoring layer	Identification of element status focusing on maintenance of individual NEs	 Major and minor faults in NE Operation status of NE (system, resources and file updating) 	Element maintenance

 Table 5.1
 Monitoring layers and monitoring items

Note: CH: Call Hold; PCH: Paging Channel

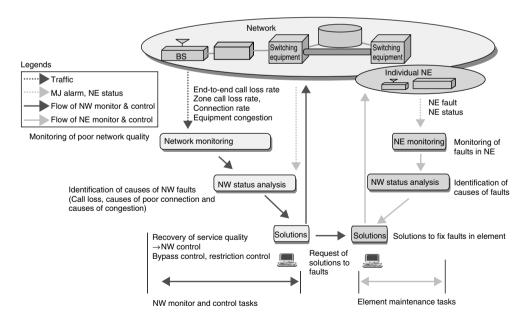


Figure 5.7 Operation flow of NW monitoring and NE monitoring

5.2.3 Building a Network Monitoring System

Monitoring the network built by NTT DoCoMo involves the identification of causes in the event of faults in NEs nationwide, NW congestion, restriction display and changes in the call connection rate and call loss rate. To monitor the nationwide network, the screen shown in Figure 5.8 is used.

In the event of faults and congestion, the CN strata screen shown in Figure 5.9 is displayed to clarify the source of the event. The important task here is to be able to constantly identify the status real time. With the displayed screen, it is easy to identify the status of NE and NW with reference to icon types and different colors. Also, the display of strata makes it easier to identify the impact of the event on other switching equipment and work out the necessary solutions.

In particular, for the identification of the connection status, the usage status of the set bandwidth in VP (Virtual Path) units of the Asynchronous Transfer Mode (ATM) link set up between switching equipment is displayed in terms of the connection rate, so that



Figure 5.8 Example of nationwide network monitoring

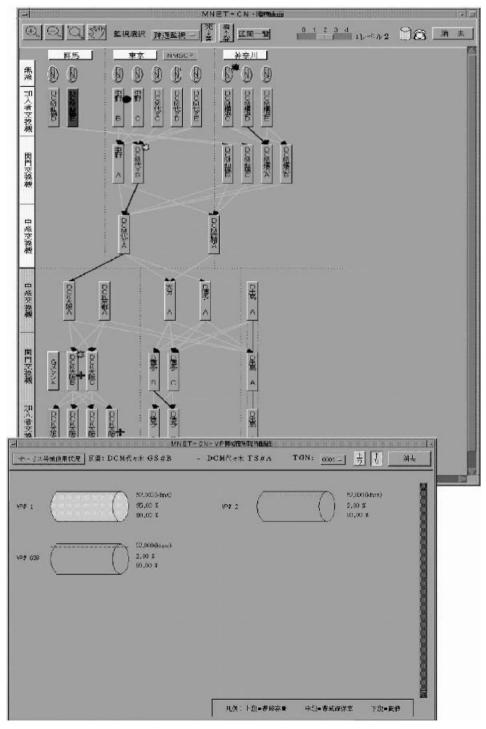


Figure 5.9 Example of CN strata display and VP bandwidth display

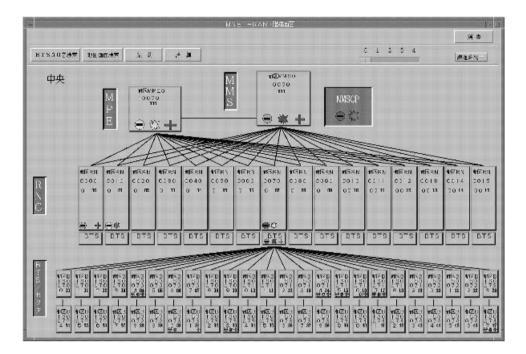


Figure 5.10 Example of RAN strata display

the traffic status of the switching equipment can always be identified. The impact of the event is therefore visually discernable and the status can be confirmed upon recovery by the display of the connection rate. Also, the connection rate in zones between specific switching equipment can be displayed, which offers a means to pin down the causes of the deteriorating connection rate.

Figure 5.10 shows the RAN strata screen, which displays the NEs and each sector of the BS. It is possible to switch from this screen to the RAN area display illustrated in Figure 5.11, and the service status can be visually represented in combination with the geographical location where the service is actually provided.

Services can always be provided in a stable manner as faults in NEs and the status of NW congestion and restrictions can always be identified using the NW monitoring screen.

5.3 Network Control

Normally, elements of the communication network are designed in accordance with traffic volume. If a large volume of traffic in excess of the element capacity flows in, the network would suffer from congestion and stop functioning. The congestion patterns and the automatic control algorithms in the IMT-2000 network is summarized below on the basis of past records relating to the congestion patterns and automatic Congestion-control algorithms in PDC and Personal Digital Cellular-Packet (PDC-P) networks of NTT DoCoMo.

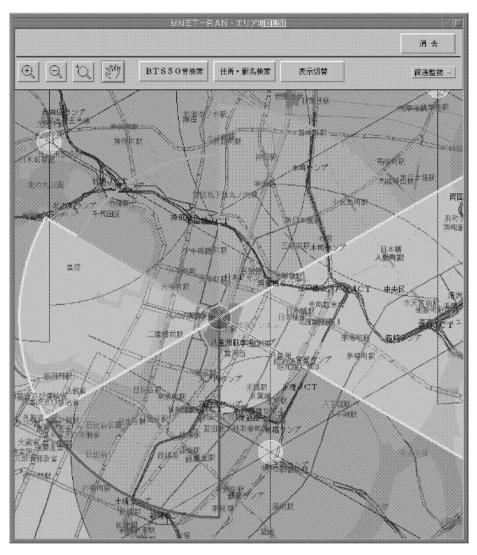


Figure 5.11 Example of RAN area display

Congestion Patterns

- Congestion caused by disasters and so on;
- Congestion due to many calls terminating at a particular phone number (fixed/mobile phone);
- Congestion due to traffic generated by events, for example, congratulation calls;
- Spot congestion at concerts and cherry-blossom viewing parties;
- Congestion caused by *i-mode* mail and large-volume Packet-Switched (PS) communications and
- Congestion caused by wide bandwidth communications, for example, audiovisual communications (assuming IMT services).

Automatic Congestion-Control Algorithms

There are two ways to carry out traffic control to deal with network congestion: autonomous control, which is done by the node itself, and control by the NW control system. The former is a self-defense mechanism against sudden increases in traffic, whereas the latter is a control method that takes the entire network into account with the aim to maximize the use of communication network resources.

Examples of control algorithms include the following:

- restriction of input from mobile terminals, Internet and so on;
- restriction of input into the Home Location Register (HLR);
- restriction of traffic in specific directions and
- flow control of packets during communication (speed adjustment).

Autonomous control by nodes is carried out in the following two ways.

(i) Traffic Control Functions of ATM Node

- Connection Admission Control (CAC) by which, the Quality of Service (QoS) is decided at the time of call setup and the bandwidth is assigned.
- Shaping control, which refers to control such that the ATM cells would not flow in excess of the bandwidth.

(ii) Traffic Control Function by Resource Decision

This function refers to the restriction of traffic from a mobile terminal and the automatic control of traffic toward a specific route when resources inside the node (rate of memory utilization etc.) exceed the threshold.

5.3.1 Positioning of Network Control System

The NW control system is a system that monitors congestion in the nodes constituting the communications network [5]. In the event of congestion, the node notifies the NW control system of the congestion in the form of an alarm or traffic information. In response to the notice, the NW control system controls the periphery nodes that are trying to send calls and signals to the congested node and suppresses calls and signals flowing into that congested node to alleviate the congestion. The effects of control is observed with reference to the statistics of the number of calls generated toward the congested node from nodes subject to control and the number of calls that were passed on to the congested node after restriction. The NW control system determines the effective control volume based on this information and makes control adjustments to each node being controlled.

By the repetition of this process, congestion in the network is alleviated while enabling the node to demonstrate its full capacity.

In network control, it is important to identify the signs of congestion before the traffic flows into the network and to take adequate measures to minimize the impact of the abnormal traffic on the network. Also, less network resources must be used on processing calls and signals annulled by control, and congestion detection and restriction control must be performed as close as possible to the source in order to use network resources in an efficient manner.

Figure 5.12 shows the concept of the NW control system. Figure 5.13 illustrates an example of the NW monitor and control OPE screen.

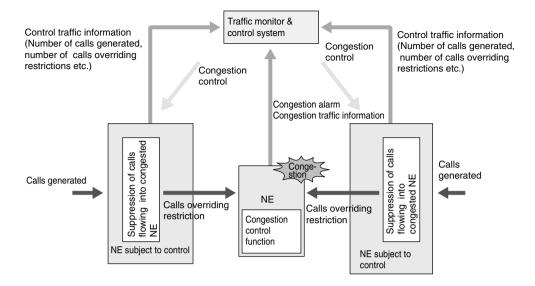


Figure 5.12 Concept of traffic monitor and control system

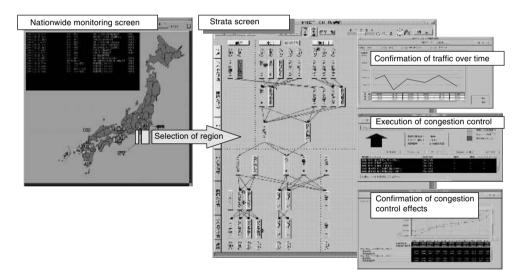


Figure 5.13 Screen of network monitor and control terminal (example)

5.3.2 Coordination between Systems in Different Types of Networks

NTT DoCoMo's IMT network shares HLR with its PDC network, and is connected to the PDC network via a switching equipment in the gateway strata. Because of such a system arrangement, network control is executed in coordination with the network monitor and control system of the PDC network. It is also coordinated to distribute information to other OpS.

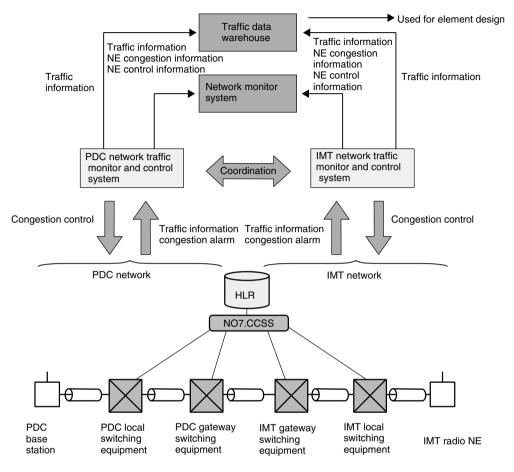


Figure 5.14 Traffic control based on coordination between PDC and IMT networks

Figure 5.14 illustrates network control based on coordination between PDC and IMT networks.

Coordination with Network Monitoring System

The NW monitoring system monitors and displays the connection status and fault status of the networks across Japan. The IMT NW control system provides traffic information gathered by nodes in the IMT network at short intervals as well as information on the congestion status and the control status of nodes in the IMT network.

Coordination with Traffic Data Warehouse

The traffic data warehouse executes processing on the basis of traffic data to be used in facility design. The IMT NW control system feeds traffic information gathered from each node at long intervals to the traffic data warehouse.

5.3.3 Network Control Functions

This section discusses restriction control associated with HLR congestion, which is a NW control function distinctive in the IMT-2000 system, as well as high-speed restriction control processing technologies.

HLR Congestion

One of the congestion patterns unique to mobile communications is congestion in HLR. HLR is connected with switching equipment via a Common Channel Signaling (CCS) network, and handles Mobile Application Part (MAP) signals. The key functions of HLR include the handling of mobile communication subscriber data, which involves the location registration of subscribers, number translation, subscriber authentication and so on. HLR congestion is unique to mobile communications because location registration is not required for fixed phones. HLR is shared by PDC, PDC-P and IMT-2000 systems. Thus, the HLR handles MAP signals compliant to PDC, P-MAP signals compliant to PDC-P and GSM-MAP (GSM: Global System for Mobile Communications) signals compliant to IMT-2000. The causes of congestion and the control methods associated with these signals are the same.

Congestion Factors

The concentration of various MAP signals from switching equipment may cause HLR congestion, in the following two cases.

- 1. *Increase in HLR access calls from mobile communication subscribers*: This refers to congestion resulting from more accesses to HLR from the switching equipment caused by more incoming or outgoing calls from/to mobile communication subscribers.
- 2. *Increase in HLR Access Calls from Other Networks*: This refers to congestion caused by more accesses to HLR from the gateway switching equipment as a result of more calls coming from NTT and other networks to mobile communication subscribers.

Congestion Detection

There are two ways to detect HLR congestion:

- 1. Constantly gather switching equipment traffic information in terms of the number of times HLR is accessed by the switching equipment and detect congestion when it exceeds a certain level or
- 2. Constantly monitor the Central Processing Unit (CPU) usage rate of HLR and detect congestion if an alarm is sent when the load exceeds a set level.

If congestion is detected by either of these methods, it is regarded as HLR congestion and restriction control is executed on periphery switching equipment.

Congestion-Control Method

Congestion is alleviated by restricting access to HLR from switching equipment nationwide based on the control level and control volume given to HLR. Switching equipment nationwide is subject to HLR access restriction because the current location of the subscriber accommodated by the congested HLR is unknown.

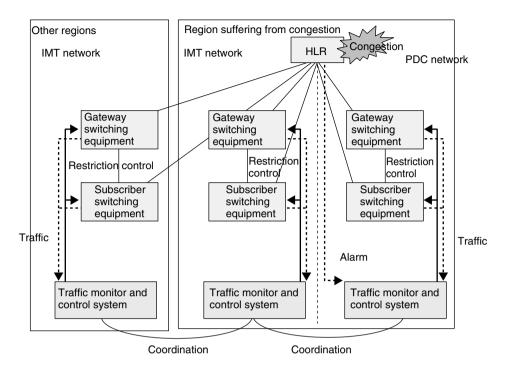


Figure 5.15 Restriction control in the event of HLR congestion

Figure 5.15 illustrates restriction control in the event of HLR congestion.

5.3.4 Congestion Control During Packet Communications

The earlier paragraphs described congestion in the call setup process, which is a primal cause of congestion in the existing mobile communications network. In the future, engaged calls involving *i-mode* mail, Internet access and wideband communications of video data are expected to give rise to congestion. In addition, congestion occurs in the PDC-P network due to a large volume of data communication by certain subscribers.

In order to tackle this, NTT DoCoMo plans to add functions to execute restriction control on specific URLs and dynamically control engaged packets.

5.3.5 Achieving High-Speed Restriction Process

In the event of node congestion, the node is given a restriction order by the control terminal of the NW control system. The problem here is that it takes time to actually execute the restriction order inside the node after the order made by the control terminal reaches the node. The more the time consumed in processing the restriction order, the more the traffic that flows into the node, which in the worst-case scenario would force the node to shut down. The clogging of internode traffic may result in network congestion and paralyze the network functions.

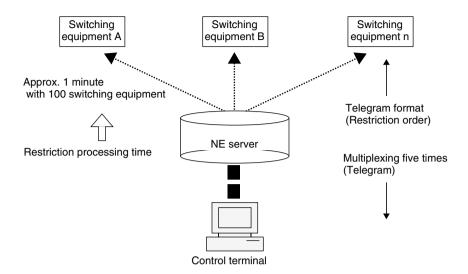


Figure 5.16 Restriction processing function

Therefore, it is important to figure out a way to swiftly execute restriction orders against node congestion.

NTT DoCoMo managed to dramatically shorten the time consumed in processing the restriction orders by converting them into a telegram format and by multiplexing the telegram five times.

Figure 5.16 shows the restriction processing function.

5.4 NE Monitoring

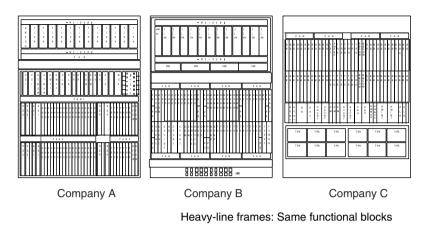
In order to efficiently perform OAM on a wide range of NEs based on a multivendor configuration, which are huge in numbers, an NE monitoring system is built according to the different systems used in the network [radio system, transmission system, switching system, Intelligent Network (IN) etc.], as shown in Figure 5.4.

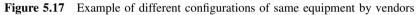
This section describes NTT DoCoMo's IMT-2000 NE monitoring system with reference to the radio-access system.

5.4.1 NEs in a Multivendor Environment

For example, NTT DoCoMo's RAN consists of the Base Transceiver Station (BTS), the RNC and the Multimedia Signal Processing Equipment (MPE), as discussed in Section 3.4.1 of Chapter 3. Each equipment is supplied by multiple vendors, and dozens of different types of NEs require operation. As illustrated in Figure 5.17, the design philosophy varies between vendors even if the equipment is the same, resulting in a totally different equipment configuration.

Thus, in general, an OpS unique to each NE has to be installed, as shown in Figure 5.18. IMT-2000 manages to do the same with only one OpS, as illustrated in Figure 5.19. The same concept has been applied to switching systems, transmission systems and other NEs.





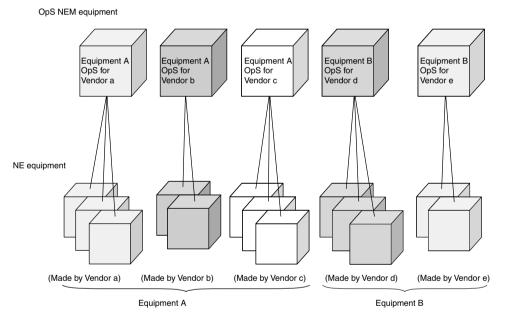


Figure 5.18 Equipment/vendor-specific operation

5.4.2 NE Monitoring Functions

In general, there are two functions required in NE operations.

- 1. Execute real-time monitoring and control in response to reports from NE, and identify the operation status of NE. At the same time, identify the impact on the network and services.
- 2. Collect various traffic information to plan additional installation of NEs.

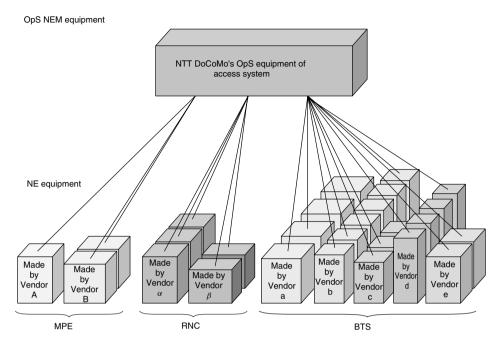


Figure 5.19 Configuration of NTT DoCoMo's OpS for radio-access system

The wide range of NEs supplied by multiple vendors must be equipped with these operation functions. However, if the monitoring conditions and the information gathered vary between each NE, it would be difficult to monitor the entire network, analyze the information and promptly deal with faults in a uniform manner. At the worst, it would increase the number of OpS units and maintenance tasks.

For the network element operation of a multivendor radio-access system, NTT DoCoMo considered the following factors, including NE, so that the maintenance staff would hardly recognize the differences in vendors and equipment.

(1) Uniform NE Monitoring Conditions

Maintenance staff should not have to worry about the vendors when monitoring NEs.

Vendor-unique monitoring should be limited to cards at the very bottom of the topology, and other parts should not require any vendor-specific operations.

(2) Standardized Format of Gathered Information and Itemization

The functions to gather information including traffic data should be standardized so that the maintenance staff would not have to worry about the differences in vendors. Also, data processing should be made easier.

(3) Standardized Signal Interface Between NE and OpS

The signal interface should be standardized for all NE types, so that all NEs can be accommodated in one NE monitoring OpS. The arrangement should help reduce the number of devices on the OpS side and cut the costs.

By fulfilling these requirements, the maintenance personnel are now able to normally engage in operations without having to worry about the differences in equipment due to vendors.

Also, the standardization of screen operations has helped prevent operational mistakes and reduce maintenance tasks, and the costs for developing OpS equipment have been reduced, as the OpS equipment is now common to all vendors.

5.4.3 Development of Element Operations

An NE monitoring system for the radio-access system was developed on the basis of the following methods in order to apply the aforementioned concepts into practice:

- 1. As the hardware configuration of NEs varies considerably between vendors, a vendorcommon AP specifying the APplication Interface (API) was installed on the NE side. Information can be exchanged with the operations staff through a common interface, enabling maintenance personnel to monitor and control equipment of various vendors under the same conditions. (Figure 5.20).
- 2. The monitor and control functions referred to as OAM functional requirements were standardized as vendor-common items by function in order to enable vendor-common operation. This arrangement helps standardize functional block levels required for determining the impact of faults in elements and so on in customer services, regardless of equipment and vendor (Figure 5.21).
- 3. Vendor-specific monitoring and control is limited to each card unit constituting each functional block (Figure 5.22).

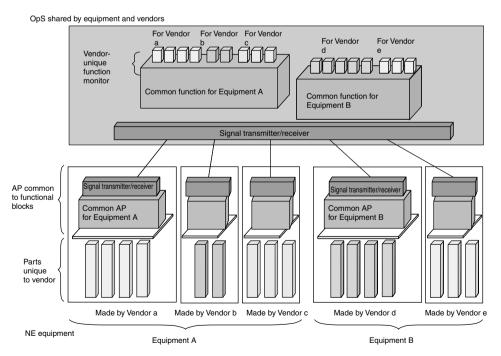
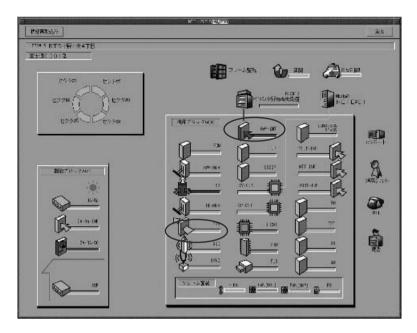
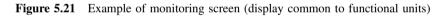


Figure 5.20 Example of element operation absorbing differences among vendors



The function and the screen do not depend on the equipment vendor as far as functional blocks are concerned. The status of equipment can be discerned and the impact on services can be identified commonly among vendors.



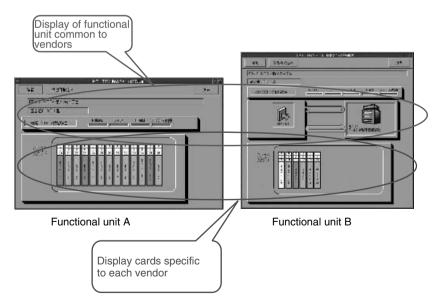


Figure 5.22 Example of detailed display of functional blocks

5.5 Network Element Management

5.5.1 Network Element Management

It is important to offer services with better quality by installing various NE in an accurate and speedy manner in response to customer's demand. Equipment as such includes many BSs, various switching equipment, Service Control Points (SCP) and CCS equipment that are installed nationwide, as well as transmission equipment that interconnect switching equipment, and entrance equipment that connect switching equipment with BSs.

The following three functions are required for element management in such large-scale networks:

- 1. Function to efficiently gather and analyze large volumes of network quality data (NW quality management function);
- 2. Function to carry out facility construction required for the installation of additional elements and the provision of new services in an efficient and speedy manner (remote file updating function); and
- 3. Function to efficiently create and edit graphic data for facility designing (equipments design function).

Element management is accomplished by coordinating these three functions (Figure 5.23).

NW quality management and remote file updating functions are described in further detail below.

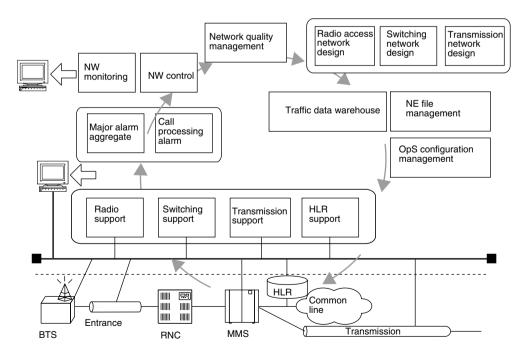


Figure 5.23 System coordination for element management

5.5.2 Network Quality Management

The traffic data warehouse stores various traffic data for NW quality management, and improves the efficiency of traffic data analysis. The main functions of the traffic data warehouse are

- 1. automatic traffic data gathering function
- 2. automatic traffic data editing function and
- 3. End User Computing (EUC) function based on a simple OLAP tool.

NTT DoCoMo is able to analyze various traffic data in a short period by implementing these functions.

Figure 5.24 shows an example of the configuration of a traffic data warehouse.

The traffic data warehouse offers the following functions.

(1) Data Gathering and Loading Function

Traffic data is gathered on a regular basis (30 minutes-1 hour) from the server of various NE monitoring systems that have an interface with NEs. The data is then stored in the

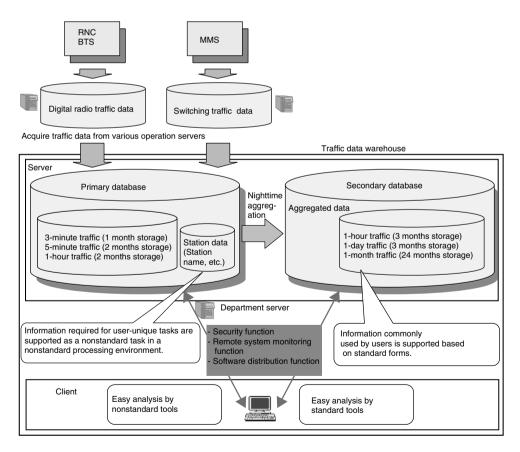


Figure 5.24 System configuration and data flow of traffic data warehouse

primary database. (The server gathers data from the NE at a cycle between 30 seconds and 5 minutes.)

(2) Data Aggregation Function

At nighttime, various traffic data stored in the primary database are aggregated and items for standard forms are created. Data aggregated in this process is stored in the aggregate database shown in Figure 5.24.

(3) Search and Form Output Function

Two types of forms are supported: standard and nonstandard. The standard form is used for information that is analyzed in a fixed pattern and can be shared by multiple users. Nonstandard forms are arranged by the user at his discretion for specific tasks, which can be analyzed in detail.

(4) Function to Display Traffic to BS in a Map

Traffic items in the nonstandard form can be displayed as traffic to a BS plotted on a map based on longitude and latitude. This helps identify the geographical characteristics of traffic. Figure 5.25 shows the screen image.

(5) Other Functions

For PDC, there are functions to simulate future traffic based on past traffic fluctuation trends. NTT DoCoMo plans to extend these functions to IMT-2000.

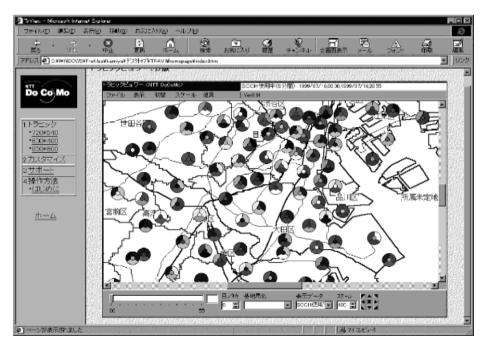


Figure 5.25 Example of traffic display on a map

5.5.3 Remote File Updating

The remote file updating system accelerates the process of updating NE files (system data, system files etc.) upon the expansion of NEs and improves the efficiency of updating operation data of OpS.

NTT DoCoMo provides the following functions under the NE file monitoring system to accelerate downloading tasks.

- 1. On-line exchange of NE files made by the teams responsible for facilities and development and
- 2. Remote downloading of NE files from OpS to NE.

Furthermore, the following functions were added to the OpS configuration management system to simplify the NE file injection tasks.

- 1. Automatic generation of operation data based on system data,
- 2. Automatic distribution of common system data.

Figure 5.26 illustrates the flow of NE files and operation data between systems.

The NE file management system offers the following functions.

(1) NE File Registration and Version Management Function

Multiple-versions management and file-status management by version and unit are performed on NE files and system data, which make it possible to distinguish the version applicable to each NE unit. This helps prevent the selection of the wrong file.

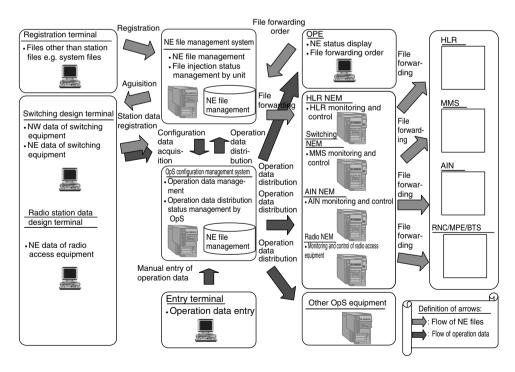


Figure 5.26 Flow of NE files and operation data

Also, the version management function sets the number of versions required in each NE file, which helps reduce the disk space required for memorization.

(2) NE File Transfer Function

Files are forwarded to NE on-line in coordination with the OpS for HLR, switching equipment and radio equipment. The status of file transfer and updating results are also managed. This function, which eliminates the need to go to the site where the NE is installed, enables centralized NE file updating by remote control.

On the other hand, the OpS configuration management system offers the following functions.

(3) Operation Data Management Function

The portion of OpS operation data that overlaps with NE system data is acquired from the NE file management system as configuration data from the system data design system (switch design system, radio design system etc.). Data that are missing from the NE system data are entered by the operation data entry tool, and the combinations of these data are managed as operation data.

The consistency of the operation data is checked within the system for data assurance purposes.

References

- [1] ITU-T Recommendation M.3010, 'Principles for a Telecommunication Management Network', 1992.
- [2] ITU-T Recommendation X.701, 'Information Technology-Open Systems Interconnection, System Management Overview', 1992.
- [3] TM Forum, Smart TMN Telecom Operations Map, Evaluation Version Release 1.0, TMF GB910, October 1998.
- [4] Onuki, Tanikawa, and Hara, 'Operation System Development Trends in DoCoMo', NTT DoCoMo Technical Journal, 2(1), 2000, 4–7.
- [5] Mizumoto, Tsumita, and Kuroki, 'Mobile Network Traffic Control and Management System', NTT DoCoMo Technical Journal, 2(1), 2000, 22–25.
- [6] Kamiya, Nakamura, and Terunuma, 'Traffic Monitoring System combining Radio Access Network with Switching Network', Technical Report of IEICE, SSE2000-173, 2000, pp. 59–64.
- [7] Yumiba, Yamamoto, and Nakamura, 'Overview of IMT-2000 Network Systems', NTT DoCoMo Technical Journal, 6(4), 2000, 8–13; (Japanese).
- [8] Sawada, and Arima, 'IMT-2000 Network Architecture', *The Journal of The Institute of Electronics, Information and Communication Engineers*, 82(2), 1999, 145–152; (Japanese).

6

Multimedia Processing Scheme

Minoru Etoh, Hiroyuki Yamaguchi, Tomoyuki Ohya, Toshiro Kawahara, Hiroshi Uehara, Teruhiro Kubota, Masayuki Tsuda, Seishi Tsukada, Wataru Takita, Kimihiko Sekino and Nobuyuki Miura

6.1 Overview

The Introduction of International Mobile Telecommunications-2000 (IMT-2000) has enabled high-speed data transmission, laying the groundwork for full-scale multimedia communications in mobile environments. Taking into account the characteristics and limitations of radio access, multimedia processing suitable for mobile communication is required.

In this chapter, signal processing, which is a basic technology for implementing multimedia communication, is first discussed. It contains descriptions on the technology, characteristics and trends of the Moving Picture Experts Group (MPEG-4) image coding method, the Adaptive MultiRate (AMR) speech coding, and 3G-324M. MPEG-4 is regarded as a key technology for IMT-2000, developed for use in mobile communication and standardized on the basis of various existing coding methods. AMR achieves excellent quality, designed for use under various conditions such as indoors or on the move. 3G-324M is adopted by 3rd Generation Partnership Project (3GPP) as a terminal system technology for implementing audiovisual services.

A functional overview of mobile Internet Service Provider (ISP) services using the IMT-2000 network is also provided together with some other important issues that must be taken into account when providing such services, including the information distribution method, copyright protection scheme and trends in the content markup language. The standardization direction in Wireless Application Protocol (WAP) Forum – a body responsible for the development of an open, globally standardized specification for accessing the Internet from wireless networks, and the technical and standardization trends of a common platform function required for expanding and rolling out applications in the future will also be discussed, with particular focus on such technologies as messaging, location information and electronic authentication.

6.2 Multimedia Signal Processing Scheme

6.2.1 Image Processing

The MPEG-4 image coding method is used in various IMT-2000 multimedia services such as videophone and video distribution. MPEG-4 is positioned as a compilation of existing image coding technologies. This section explains its element technologies and the characteristics of various image-coding methods developed before MPEG-4.

6.2.1.1 Image Coding Element Technology

Normally, image signals contain about 100 Mbit/s of information. To process images, various efficient image coding methods have been developed taking advantage of the characteristics of images. Element technologies common to these methods include interframe motion prediction, Discrete Cosine Transform (DCT), and variable length coding [1-3].

Interframe Motion-Compensated Prediction

Interframe motion-compensated prediction is a technique used to determine how much and in which direction the specific part of an image has moved by referencing the previous and subsequent images rather than by encoding each image (Figure 6.1). The direction and amount of movement (motion vector) vary depending on the block of each frame. Therefore, a frame is divided into a size of about 16 by 16 pixels (called macro block), to obtain the motion vector of each block. The difference between the macro blocks of the frame and the previous frame is called predicted error. DCT mentioned in the following section is applied to this error.

DCT

Each frame in a video is expressed by the sum of weights ranging from simple image components (low-frequency components) to complex image components (high-frequency components) (Figure 6.2). It is known that information generally is concentrated in the low-frequency components and plays a visually important role. DCT is aimed at extracting only the important frequency components at the end to perform information compression.

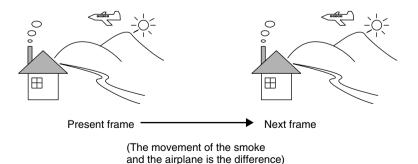


Figure 6.1 Basic idea of interframe motion-compensated prediction

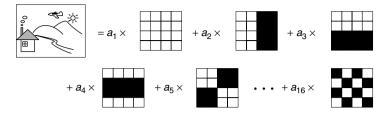


Figure 6.2 Concept of decomposing screen into frequency components

This method is widely adopted as the conversion into the space frequency domain can be carried out efficiently.

In practice, DCT is applied to each block of a frame that is divided into blocks with a size of about 8 by 8 pixels. In Figure 6.2, " a_i " denotes the DCT coefficient. This coefficient is further quantized and rounded to a quantization level, and then variable length coding is applied as mentioned in the following section.

Variable Length Coding

Variable length coding is used to compress information exploiting the uneven nature of input signal values. This method allocates short codes to signal values that occur frequently and long codes to less frequent signal values.

As mentioned in the previous section, many coefficients of high frequency components become zero in the process of rounding to the quantization representative value. As such, there are many cases in which "all subsequent values are zero (EOB: End of Block)" or "a value L follows after a certain number of zeros." Information can also be compressed by allocating short code to frequently occurring combinations of the number of zeros (zero run) and L value (Level). The methods explained in the preceding text are schemes that allocate one code to a combination of two values. This method is called two-dimensional variable length coding.

6.2.1.2 Positioning of Various Video-Coding Methods

Internationally standardized video-coding methods include H.261, MPEG-1, MPEG-2, H.263, and MPEG-4. Figure 6.3 shows the applicable areas of each scheme. The subsequent sections describe how each method uses the above-mentioned element technologies to improve compression efficiency and the functional differences of these methods.

H.261 Video Coding

This method is virtually the world's first international standard for video coding, designed for use in ISDN videophone and videoconference, standardized by International Telecommunication Union-Telecommunication (ITU-T) in 1990 [4]. H.261 uses all the element technologies mentioned in the preceding text. That is:

- 1. Predicts motion vector of a macro block containing 16 by 16 pixel in units of pixels to perform interframe motion-compensated prediction.
- 2. Applies DCT to the predicted error with the previous frame of size 8 by 8 pixels. For areas with rapid motion that exceeds a certain quantity of predicted error, interframe

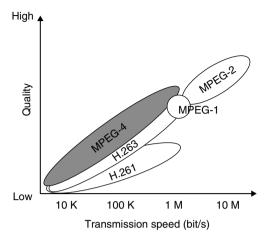


Figure 6.3 Relationship between MPEG-4 video coding and other standards

motion-compensated prediction is not performed. Instead, 8×8 pixel-DCT is applied within a frame to increase coding efficiency.

3. Performs variable length coding on the motion vector obtained with interframe motion compensation and the result of DCT processing, respectively. Two-dimensional variable length coding is used on the result of DCT processing.

H.261 assumes the use of conventional TV cameras and monitors. TV signal formats (number of frames and number of scanning lines), however, vary depending on the region. To cope with international communications, these formats have to be converted into a common intermediate format. This format is called Common Intermediate Format (CIF), defined as "352 (horizontal) by 288 (vertical) pixels, a maximum of 30 frames per second, and noninterlace." Quarter CIF (QCIF) that is a quarter of the size of CIF was defined at the same time and used also in subsequent video-coding applications.

MPEG-1/MPEG-2 Video Coding

MPEG-1 was standardized by International Organization for Standardization/International Electrotechnical Commission (ISO/IEC) in 1993 for use with storage media such as CD-ROM [5]. This coding method is designed to handle visual data in the vicinity of 1.5 Mbit/s. Since this is a coding scheme for storage media, requirements for real-time processing are relaxed compared with H.261, thereby increasing chances to adopt new technologies that require capabilities such as random search. While basically the same element technologies such as H.261 are used, the following new capabilities have been added:

- 1. All-intraframe image is periodically inserted to enable random access replay.
- 2. H.261 predicts motion vector from the past screen to perform interframe motioncompensated prediction (this is called *forward prediction*). In addition to this, MPEG-1 has enabled prediction from the future screen (called *backward prediction*), by taking advantage of the characteristics of the storage media. Moreover, MPEG-1 evaluates

forward prediction, backward prediction, and average of backward prediction and forward prediction and then selects the one having least prediction error among the three averages to improve the compression rate.

3. While H.261 predicts motion vector in units of 1 pixel, MPEG-1 introduced prediction in units of 0.5 pixel. To achieve this, an interporation image is created by taking the average of adjacent pixels. Interframe motion prediction is performed with the interporated image to enhance the compression rate.

With these capabilities added, MPEG-1 is widely used as a video encoder and player for personal computers.

MPEG-2 is a generic video-coding method developed by taking into account the requirements for telecommunications, broadcasting, and storage. MPEG-2 was standardized by ISO/IEC in 1966 and has a common text with ITU-T H.262 [6]. MPEG-2 is the coding scheme for video of 3 to 20 Mbit/s, widely used for digital TV broadcast, High Definition Television (HDTV), and Digital Versatile Disk (DVD). MPEG-2 inherits the element technologies of MPEG-1 and has the following new features:

- 1. The capability to efficiently encode interlace images used in conventional TV signals.
- 2. A function to adjust the screen size and quality (called spatial scalability and SNR scalability, respectively) as required by retrieving only part of the coded data.

Since capabilities are added for various uses, special attention must be paid to ensure the compatibility of coded data. To cope with this issue, MPEG-2 has introduced new concepts as "profile" and "level" that classify the difference of capabilities and complexity of processing. These concepts are used in MPEG-4 as well.

H.263 Video Coding

This is an ultra low bit rate video-coding method for videophones over analog networks, standardized by ITU-T in 1996. This method assumes the use of 28.8 kbit/s modem and adopts part of the new technologies developed for MPEG-1. Interframe motion-compensated prediction in units of 0.5 pixel is a mandatory basic function (baseline). Another baseline is three-dimensional coding including EOB that extends the conventional two-dimensional variable length coding (run and level). Furthermore, interframe motion-compensated prediction in units of 8 by 8 pixel blocks and processing to reduce block distortion in images are newly added as options.

With these functional additions, H.263 is now used in some equipment for ISDN videophones and videoconference.

6.2.1.3 MPEG-4 Video Coding

MPEG-4 video coding was developed by making various improvements on top of ITU-T H.263 video coding, including the error-resilience enhancement. This coding method is backward compatible with the H.263 baseline.

MPEG-2 was designed to mainly process image handling on computers, digital broadcasting and high-speed communications. In addition to these services, MPEG-4 was standardized with a special focus on its application to telecommunications, in particular, mobile communications. As a result, in 1999, MPEG-4 established a very generic video-coding method [7] as the ISO/IEC standard. Hence, MPEG-4 is recognized as a key

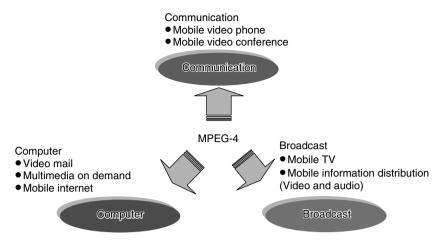


Figure 6.4 Scope of MPEG-4

technology for image-based multimedia services including video mail, video distribution as well as videophone in IMT-2000 (Figure 6.4).

Profile and Level

To ensure the interchangeability and interoperability of encoded data, the functions of MPEG-4 are classified by profile, while the computational complexity is classified by level as in the case of MPEG-2. Defined profiles include Simple, Core, Main, and Simple Scalable, among which the Simple profile defines the common functions. The interframe motion-compensated prediction with 8 by 8 pixels, which is defined as an option in H.263, is positioned as Simple profile.

With Simple profile, QCIF images are handled by levels 0 and 1, and CIF images by level 2.

The Core and Main profiles define an arbitrary area in a video as an "object", so as to improve the image quality, or to incorporate the object into other coded data. Other more sophisticated profiles such as those composed with CG (Computer Generated) images are also provided with MPEG-4.

IMT-2000 Standards

3GPP 3G-324M, the visual phone standard in IMT-200 detailed in Section 6.4 requires the H.263 baseline as a mandatory video-coding scheme and highly recommends the use of MPEG-4 Simple profile level 0. The Simple profile contains the following error-resilience tools:

- Resynchronization: Localizes transmission errors by inserting resynchronization code in variable length coded data and partitioning it at an appropriate position in a frame. Since header information follows the resynchronization code to specify coding parameters, a swift recovery from the state of decoding errors is enabled. Insertion interval of resynchronization code can be optimized taking into account the overhead of the header information, visual scene in input type and transmission characteristics.
- 2. *Data Partition*: Enables error concealment by inserting Synchronization Code (SC) at boundaries of different types of coded data. For example, by inserting SC between the

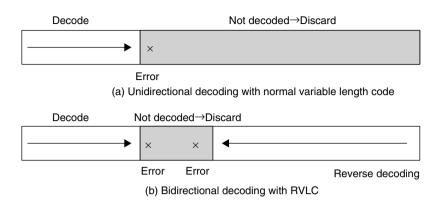


Figure 6.5 Example of decoding reversible variable length code (RVLC)

motion vector and the DCT coefficient, the motion vector can be transmitted correctly even if a bit error is mixed into the DCT coefficient, enabling more natural error concealment.

- 3. *Reversible Variable Length Code (RVLC)*: As shown in Figure 6.5, this code is a variable length code that can be decoded from the reverse direction. This is applied to the DCT coefficient. With this tool, all the macro blocks can be decoded except for those that contain bit errors.
- 4. *Adaptive Intrarefresh*: This tool prevents error propagation by performing intraframe coding on highly motive area.

As described in the preceding text, MPEG-4 Simple profile level 0 constitutes a very simple CODEC suitable for mobile communications.

6.2.2 Speech and Audio Processing

6.2.2.1 Code Excited Linear Prediction (CELP) Algorithm

There are typically three speech coding methods, namely, waveform coding, vocoder and hybrid coding. Like Pulse Coded Modulation (PCM) or Adaptive Differential PCM (ADPCM), waveform coding encodes the waveform of signals as accurately as possible without depending on the nature of the signals. Therefore if the bit rate is high enough, high-quality coding is possible. If the bit rate becomes low, however, the quality drops sharply. On the other hand, vocoder assumes a generation model of speech and analyzes and encodes its parameters. Although this method can keep the bit rate low, it is difficult to improve the quality even if the bit rate is increased because the voice quality largely depends on the assumed speech generation model. Hybrid coding is a combination of waveform coding and vocoder. This method assumes a voice generation model and analyzes and encodes its parameters and then performs waveform coding on the remaining information (residual signals) not expressed with parameters. One of the typical hybrid methods is CELP. This method is widely used for mobile communication speech coding as a generic algorithm for implementing highly efficient and high-quality speech coding.

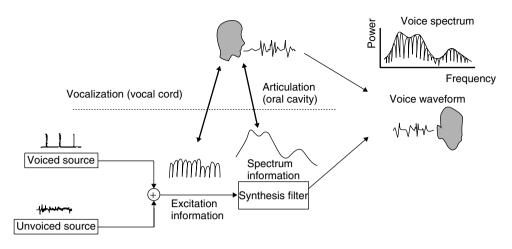


Figure 6.6 Voice generation model used in CELP coding

Figure 6.6 shows the speech generation model used in CELP coding. The CELP encoder has the same internal structure as the decoder. The CELP decoder consists of a linear prediction synthesis filter and two codebooks (adaptive codebook and stochastic codebook) that generate excitation signals for driving the filter. The linear prediction synthesis filter corresponds to the human vocal tract to represent spectrum envelope characteristics of speech signals and the excitation signals generated from the excitation codebook correspond to the air exhaled from the lung, which passes through the glottis. This means that CELP simulates the vocalization mechanism of human beings.

The subsequent sections explain the basic technologies used in CELP coding.

Linear Prediction Analysis

As shown in Figure 6.7, linear prediction analysis uses temporal correlation of speech signals and predicts the current signal from the past inputs. The difference between the predicted signal and the original signal is prediction residual.

The CELP encoder calculates the autocorrelation of speech signals and obtains linear prediction coefficients α_i using the Levinson-Dervin-Itakura method and so on. The order of the linear prediction coefficient in telephone band coding is normally ten. Since it is difficult to determine filter stability, linear prediction coefficients are converted to equivalent and stable coefficients such as reflection coefficients or Line Spectrum Pair (LSP) coefficients and then quantized for transmission. The decoder constitutes a synthesis filter with transmitted α_i and it drives the synthesis filter with the prediction residual to obtain the decoded speech. The frequency characteristics of the synthesis filter correspond to the speech spectrum envelope.

Perceptual Weighting Filter

The CELP encoder has the same internal structure as the decoder. It encodes signals by searching patterns and gains in each codebook so that the error between the synthesized speech signal and the input speech signal is minimized. Such techniques are called Analysis-by-Synthesis (A-b-S), one of the characteristics of CELP.

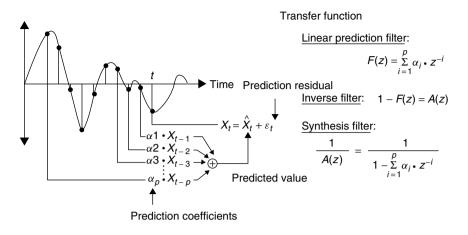


Figure 6.7 Linear prediction analysis

A-b-S calculates error using the weighted error based on the perceptual characteristics of human beings. The perceptual weighting filter is expressed as an ARMA (Auto Regressive Moving Average)-type filter that uses the coefficient obtained through linear prediction analysis. This filter minimizes the quantization error of spectrum valleys that are relatively easy to hear, by having frequency characteristics of vertically inverted speech spectrum envelope.

Although using nonquantized linear prediction coefficient improves the characteristics, the computational complexity increases. Because of this, there were some cases in the past in which the computational complexity was reduced by offsetting the quantized linear prediction coefficient against the synthesis filter at the cost of quality. Today, however, calculation is mainly performed using the impulse response of the synthesis filter and perceptual weighting synthesis filter.

Adaptive Codebook

The adaptive codebook stores past excitation signals in memory and changes dynamically. If the excitation signal is cyclic, like voiced sound, the excitation signal can be efficiently expressed using the adaptive codebook because the excitation signal repeats at the pitch cycle that corresponds to the pitch of the voice. The pitch cycle chosen is the one in which the difference between the source voice and the output of the adaptive codebook vector, from the synthesis filter, is the smallest in the perceptually weighted area. As an average voice pitch cycle, cycles of about 16 to 144 samples are searched for an 8 kHz sampling input. If the pitch cycle is relatively short, it is quantized to an accuracy of noninteger cycle by over sampling to increase the frequency resolution.

Since error calculation involves considerable computational complexity, normally the autocorrelation of speech is calculated in advance to obtain an approximate pitch cycle and then error calculation is performed including over sampling around that pitch cycle to significantly reduce the computational complexity. Exploring only around the previously obtained pitch cycle and quantizing the difference is also effective to reduce the amount of information and computational complexity.

Stochastic Codebook

The stochastic codebook expresses residual signals that cannot be expressed with the adaptive codebook and therefore has noncyclic patterns. Traditionally, the codebook contained Gaussian random noises and noise signals it had learned. But now the algebraic codebook, which can express residual signals with sparse pulses, is often used. With this, it is now possible to significantly reduce memory required for storing noise vectors, orthogonalization operation with the adaptive codebook and the amount of error calculation.

Post Filter

The post filter is used in the final stage of decoding in order to improve the subjective quality of the decoded voice by reshaping it. The Formant emphasis filter, a typical post filter, is of the ARMA type and has the inverse characteristics of the perceptual weighting filter, capable of suppressing spectrum valleys to make quantization errors less noticeable. Normally this filter is added with a filter for correcting the spectrum tilt of output signals.

6.2.2.2 Peripheral Technologies for Mobile Communications

In mobile communications, various peripheral technologies are used to cope with special conditions such as the utilization of radio links, use of service in outdoors or on the move. This section outlines these peripheral technologies.

Error Correction Technology

The error-correcting code is used for correcting transmission errors generated in the radio channels. Bit Selective Forward Error Correction (BS-FEC) or Unequal Error Protection (UEP) is used to perform error correction efficiently since they use correction codes with different capabilities depending on the error sensitivity of the speech coding information bit (the size of distortion given to the decoded voice when the bit is erroneous).

Error-Concealment Technology

If an error is not corrected with the aforementioned error-correcting code or information is lost, correct decoding cannot be performed with the received information. In such a case, speech signals of the erroneous part are generated with parameter interpolation using past speech information to minimize the deterioration of the speech quality. This is called the error-concealment technology. Parameters to be interpolated include linear prediction coefficient, pitch cycle, and gain, which have high temporal correlation.

Discontinuous Transmission

Discontinuous Transmission (DTX) sends no or very little information during a period when there is no speech, which is effective to save the battery of Mobile Stations (MSs) and to reduce interference. Voice Activity Detector (VAD) uses voice parameters to determine whether there is speech or not. In silent periods, background noise is generated on the basis of the background noise information that contains far less amount of information than speech information in order to reduce the user's discomfort caused by DTX.

Noise Suppression

As mentioned in Section 6.2.2.1, since the CELP algorithm uses the vocal model of human beings, the characteristics of other sounds such as street noises deteriorate. Therefore, suppressing noises other than human voice required for conversation improves speech quality.

6.2.2.3 IMT-2000 Speech Coding AMR

Standardization

With the establishment of the IMT-2000 Study Committee in Association of Radio Industries and Businesses (ARIB) in 1997, Japan became one of the first countries in the world to start the standardization of the CODEC for the third generation mobile communications system. The CODEC Working Group under the IMT-2000 Study Committee was assigned with the responsibility for selecting the CODEC for IMT-2000. Since several speech-coding schemes were proposed by member companies of the Working Group (WG), the evaluation procedures were drafted and evaluation tests were carried out. In the midst of testing, Third Generation Partnership Project (3GPP) was formed at the end of 1998 with the participation by ARIB, Telecommunication Technology Committee (TTC), Telecommunications Industry Association (TIA) and European Telecommunications Standards Institute (ETSI) and so on. It was therefore agreed to carry out the selection process at 3GPP Technical Specification Group-Services and System Aspects (TSG-SA) WG4 (CODEC) based on the evaluation results of ARIB. Consequently, AMR [8] was regarded superior to the other candidate technologies, and was thus adopted as the mandatory speech-coding algorithm of 3GPP.

Algorithm Overview

AMR is a multirate speech-coding method developed on the basis of Algebraic CELP (ACELP), adopted as a GSM speech-coding method in 1998. It provides eight coding modes ranging from 12.2 kbit/s to 4.75 kbit/s. Among them 12.2 kbit/s, 7.4 kbit/s and 6.7 kbit/s have common algorithm with the speech coding schemes standardized as in other regional standards.

Its algorithm is basically the same as G.729 with some innovations for multirate. Frame length is fixed to 20 ms in all modes. Multirate capability is provided by changing the number of subframes and the number of quantized bits (Table 6.1).

The linear prediction coefficients are analyzed twice per frame in 12.2 kbit/s. Prediction is performed in the LSP area on 2 by 2 elements sequentially divided at every 2 orders from the lowest order of the LSP coefficient and then the residual is vector-quantized. In other modes, analysis is performed once per frame and vector quantization is performed on divided elements after prediction is made in the LSP area.

The long-term prediction tap is searched at noninteger resolutions, 1/6 in the 12.2 kbit/s mode, 1/3 in the other modes and differentially quantized in the frame.

The algebraic codebook consists of 2 to 10 nonzero pulses of size 1. Also the pitch prefilter is applied to codebook exploration, a filter that has the same effect as Pitch Synchronous Innovation (PSI). In the 12.2 kbit/s and 7.95 kbit/s modes, codebook gain is quantized separately for the adaptive codebook and fixed codebook. In the other modes, they are vector-quantized. The decoder applies the Formant post filter and frequency tilt compensation filter to the synthesized voice to obtain the final decoded voice.

AMR also stipulates peripheral technologies required for mobile communications. Two options are provided as VAD algorithms required for DTX. Background noise information [Silence Insertion Description (SID)]: is transmitted at a certain interval with the short-term prediction coefficient and frame power quantized in 35 bits. Also requirements are defined for concealment in case of an error. For example, interpolation of coding parameters such as codebook gain and the short-term prediction coefficient is defined according to the status transition caused by errors.

Mode	Parameter	1st subframe	2nd subframe	3rd subframe	4th subframe	Frame total
12.2 kbit/s	LSP × 2 Pitch delay Pitch gain Algebraic code Codebook gain Total	9 4 35 5	6 4 35 5	9 4 35 5	6 4 35 5	38 30 16 140 20 244
10.2 kbit/s	LSP Pitch delay Algebraic code Gain Total	8 31 7	5 31 7	8 31 7	5 31 7	26 26 124 28 204
7.95 kbit/s	LSP Pitch delay Pitch gain Algebraic code Codebook gain Total	8 4 17 5	6 4 17 5	8 4 17 5	6 4 17 5	27 28 16 68 20 159
7.40 kbit/s	LSP Pitch delay Algebraic code Gain Total	8 17 7	5 17 7	8 17 7	5 17 7	26 26 68 28 148
6.70 kbit/s	LSP Pitch delay Algebraic code Gain Total	8 14 7	4 14 7	8 14 7	4 14 7	26 24 56 28 134
5.90 kbit/s	LSP Pitch delay Algebraic code Gain Total	8 11 6	4 11 6	8 11 6	4 11 6	26 24 44 24 118
5.15 kbit/s	LSP Pitch delay Algebraic code Gain Total	8 9 6	4 9 6	4 9 6	4 9 6	23 20 36 24 103
4.75 kbit/s	LSP Pitch delay Algebraic code Gain Total	8 9	4 9 8	4 9	4 9 8	23 20 36 16 95

Table 6.1 AMR bit distribution

Radio Access Network (RAN) of IMT-2000 is defined so that it can be designed flexibly as a toolbox. To enable this, classification of coding information is defined according to its significance so that RAN can apply UEP to the AMR coding information. Note that IMT-2000 Steering Group (ISG) defines radio parameters to meet this classification.

Quality

Figure 6.8 shows part of the subjective assessment of AMR, conducted by DoCoMo conforming to the ARIB testing procedure and submitted to 3GPP. The testing was conducted with Wideband Code Division Multiple Access (W-CDMA) Bit Error Rate (BER) set to 0.1% (but the radio transmission method slightly differs from the current one). The result shows that 12.2 kbit/s is better than any other coding method and that it is also superior to other coding methods with an equivalent bit rate.

In addition, the quality of AMR has been reported in 3GPP standard TR26.975 [9].

Uses Other than Telephone Nontelephony Applications

AMR is adopted as a mandatory speech-coding algorithm for 3G-324 M [10], that is, codecs for circuit-switched multimedia telephony services of 3GPP because of its unprecedented flexible structure and excellent quality. Internet Engineering Task Force (IETF) also specifies a Real-Time Protocol (RTP) payload format to apply AMR to Voice over Internet Protocol (VoIP). AMR is thus widely used in addition to the IMT-2000 speech services.

Future Trends

In March 2001, 3GPP approved AMR-Wide Band (AMR-WB), which is a wider bandwidth version (up to 7 kHz) of AMR. The selected algorithm was adapted as the ITU-T's wideband speech coding. ITU-T also is working on the standardization of 4 kbit/s speech coding with a quality equivalent to the public switched telephone lines.

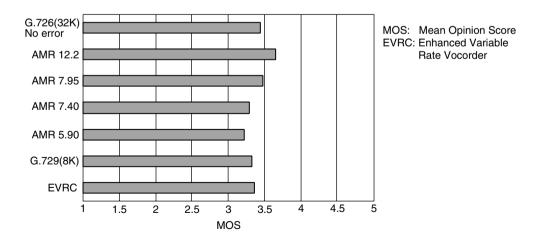


Figure 6.8 AMR subjective evaluation results

On the other hand, the possibility to apply VoIP or speech coding to streaming services is also actively discussed, in order to provide telephone services equivalent to circuit-switched networks on IP networks, given the fact that communication networks are becoming increasingly IP oriented. The standardization activities for VoIP are carried out mainly by such groups as the Telecommunication and Internet Protocol Harmonization over Network (TIPHON) project of ETSI, IETF's IP Telephony (IPTEL), and Audio/Video Transport (AVT). Meanwhile, 3GPP is proceeding with its standardization tasks cooperating with these organizations, with an aim to implement IP over mobile networks.

6.2.3 Multimedia Signal Processing Systems

6.2.3.1 History of Standardization

Figure 6.9 shows the history of the international standardization of audiovisual terminals. H.320 [11] is the recommendation for audiovisual terminals for N-ISDN prescribed by ITU-T in 1990. This recommendation was very successful in that it ensured interconnectivity among equipment from different vendors, having contributed to the spread of videoconference and videophone services. After this, B-ISDN, analog telephone networks [public switched telephone network (PSTN)] and IP network terminals and systems were studied, resulting in the development of recommendations H.310 [12], H.324 [13] and H.323 [14], respectively, in 1996.

With the explosive spread of mobile communications and the progress of the standardization activity of the third generation mobile communication system, ITU-T commenced studies on audiovisual terminals for mobile communications networks in 1995. Studies were made by extending the H.324 recommendation for PSTN, and led to the development of H.324 Annex C in February 1998. H.324 Annex C enhances error resilience against transmission over radio channels.

Since H.324 Annex C is designed as a general-purpose standard not specialized for a particular mobile communication method and defined as an extension of H.324, it includes

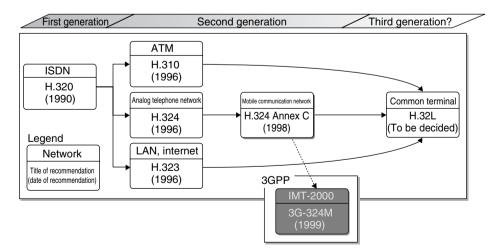


Figure 6.9 History of audiovisual terminal standardization

specifications that are not necessarily suitable for IMT-2000. To solve this problem, the 3GPP CODEC Work Group selected mandatory speech and video coding (CODEC) and operation mode optimized for IMT-2000 requirements, and prescribed 3GPP standard 3G-324M [15] in December 1999. CODECs optimal for 3G were selected in this process not restricted to that of the ITU-T standard. Visual phones to be used in W-CDMA service are compliant with 3G-324M.

6.2.3.2 3G-324M Terminal Configuration

3G-324M defines the specifications for the audiovisual communication terminal for IMT-2000, optimally combining ITU-T recommendations and other international standards. It stipulates functional elements for providing audiovisual communications as well as communication protocols that cover the entire flow of communication.

For transmission methods of multiplexing speech and video into one mobile communication channel and control messages exchanged in each communication phase, H.223 and H.245 are used. 3G-324M also stipulates efficient methods for transmitting control messages in the presence of transmission errors.

Figure 6.10 shows a 3G-324M terminal configuration. The 3G-324M standard is applied to speech/video CODEC, the communication control unit and multimedia-multiplexing unit. The speech CODEC requires AMR support as a mandatory function and video CODEC requires the H.263 baseline as a mandatory capability with MPEG-4 support recommended. The support of H.223 Annex B, which offers improved error resilience, is a mandatory requirement for the multimedia-multiplexing unit.

6.2.3.3 Media Coding

While various media coding schemes can be used in 3G-324M by exchanging the terminal capability through the use of communication control procedures, which is described in

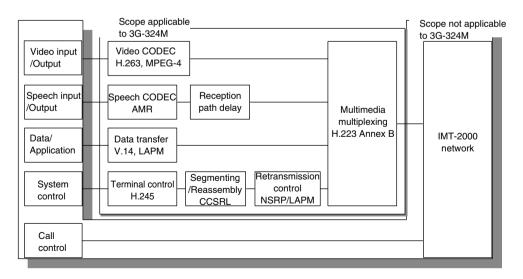


Figure 6.10 3G-324M terminal configuration

further detail later, and changing the CODEC setting upon the establishment of logical channels, 3G-324M defines a set of minimum mandatory CODECs to ensure interoperability between different terminals.

For speech CODEC, 3G-324M specifies Advanced MultiRate (AMR), which is the same CODEC as basic speech service, an mandatory requirement taking into account the ease of terminal implementation, and G.723.1 as a recommended optional CODEC, which is defined as a mandatory CODEC in H.324.

As video CODEC, 3G-324M specifies H.263 baseline (excluding the optional capabilities) as a mandatory CODEC, as is the case for H324. It also specifies in detail and recommends the use of MPEG-4 video to cope with transmission errors unique to mobile communications.

6.2.3.4 Multimedia Multiplexing

Speech, video, user data, and control messages are mapped onto one line of bit sequence by the multimedia MUltipleXer (MUX) (hereinafter called MUX) for transmission. The receiving side needs to accurately demultiplex information from the received bit sequence. The role of the MUX also includes the provision of transmission service according to the type of information [such as Quality of Service (QoS) and framing].

H.223 [16], the multimedia-multiplexing scheme for H.324, satisfies the above-mentioned requirements by adopting a two-layered structure consisting of an adaptation layer and a multiplexing layer. In mobile communication, strong error resilience is required for multimedia multiplexing in addition to the above-mentioned requirements. As such, H.324 Annex C includes extensions on H.223 for the support of mobile communications.

This extension enables error-resilience levels to be selected according to the transmission characteristics by adding error-resilience tools to H.223. At present, four levels, from level 0 to level 3, are defined. Level 1, 2 and 3 are defined in H.223 Annex A, B and C [17–19], respectively. To ensure interoperability, a terminal that supports a certain level has to support lower levels as well. In 3G-324M, the support of level 2 is a mandatory requirement. The following sections describe the characteristics of levels 0 to 2.

Level 0

H.223

Three adaptation layers are defined corresponding to the type of the higher layers:

- 1. *AL1*: For user data and control information. Error control is performed in the higher layer.
- 2. AL2: For speech. Error detection and sequence numbers can be added.
- 3. *AL3*: For video. Error detection and sequence numbers can be added. Automatic Repeat reQuest (ARQ) is applicable.

The multiplexing layer combines time division multiplexing and packet multiplexing to achieve efficiency and small delay. Packet multiplexing is used for media with varying information bit rate such as video. Time division multiplexing is used for media that requires low delay such as speech.

An 8-bit HDLC (High Level Data Link Control) flag is used as the synchronization flag in the multiplexing frame. "0" bits are inserted in the information data to prevent this flag pattern from occurring in information data. Since byte consistency cannot be maintained, synchronization search needs to be performed bitwise.

Level 1

To improve the frame synchronization characteristics in the multiplexing layer, the synchronization flag of the frame is changed from 8-bit HDLC flag to 16-bit PN (Pseudorandom Numerical) sequence. "0" bit insertion is abolished to maintain byte consistency in the frame, enabling synchronization search in units of bytes.

Level 2

Level 1 is modified to improve the synchronization characteristics and the error resilience of the header information by adding the payload length field and applying error-correction code in the frame header. In addition, option fields can also be added to improve the burst error resilience of header information.

6.2.3.5 Terminal Control

3G-324M uses H.245 [20] as the terminal control protocol as in H.324. H.245 is widely used in ITU-T multimedia terminal standards for various networks as well as in 3G-324M and H.324. Relatively easy implementation of gateways between different types of networks is also an advantage of H.245.

The functions offered by H.245 include

- 1. Decision of master and slave: Master and slave are decided at the start of communication.
- 2. *Capability Negotiation*: Negotiate capabilities supported by each terminal to obtain the information on the transmission mode and coding mode that can be received and decoded by the far end terminal.
- 3. *Logical channel signaling*: Opens and closes logical channels and sets parameters to be used. The relationship between logical channels can also be set.
- 4. *Initialization and modification of multiplexing table*: Adds and deletes entries to and from multiplexing table.
- 5. *Mode setting request for speech, video, and user data*: Controls the transmission mode of the far end terminal.
- 6. *Decision of round trip delay*: Enables the measurement of round trip delay. Can also be used to confirm the operation of the other terminal.
- 7. Loop back test.
- 8. *Command and notification*: Requests for communication mode and flow control, and reports the status of the protocol.

To provide these functions, H.245 defines the messages to be transmitted and specifies the control protocol using these messages.

Messages are defined using the Abstract Syntax Notation (ASN) 1 (ASN.1, ITU-T X.680|ISO/IEC IS 8824-1) [21], which is a representation method with excellent readability and extensibility and converted to a binary format using Packed Encoding Rules (PER) (PER, ITU-T X.691|ISO/IEC IS 8825-2) [22], thereby enabling efficient message transmission. And Specification and Description Language (SDL) is used as the control protocol to stipulate status transition including exception handling visually and comprehensively.

6.2.3.6 Multilink

One of the distinctive features of IMT-2000 is its multicall capability that enables multiple calls to be established at the same time. With this function, high-quality audiovisual communications can be performed by using multiple physical channels simultaneously. To implement this, multilink transmission is required, a transmission method that aggregates multiple physical channels and provides them as one logical channel.

To meet this requirement, standardization studies were carried out on multilink transmission in ITU-T H.324 Annex C, which resulted in the development of H.324 Annex H (mobile multilink protocol) in November 2000 [23]. This capability is also specified as an option in 3G-324M so that it can be used as a standard. H.324 Annex H allows up to eight channels of the same bit rate to be aggregated. It is also designed to tolerate bit errors generated in radio transmission lines.

H.324 Annex H specifies multilink communication procedures, control frame structure exchanged upon at the setup of communication, frame structure for data transmission and the method of data mapping onto multilink frames. Figure 6.11 shows the operations and characteristics of the mobile multilink.

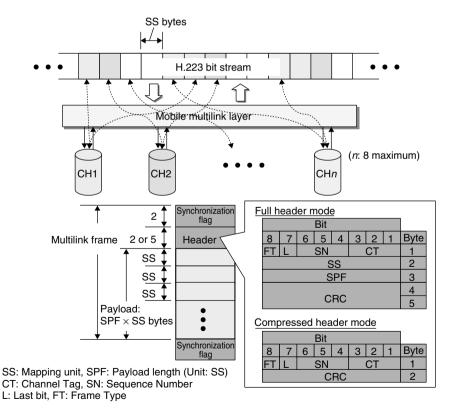


Figure 6.11 Operations of mobile multilink layer

Mobile multilink layer, located between physical channels and H.223 multiplexing layer, divides the output bit sequence of the multiplexing layer into Sample Size (SS) byte samples and distributes them to each channel. The order of distribution is fixed to the ascending order of the Channel Tag (CT) allocated for each channel. The receiving side reconfigures the original bit sequence based on the CT field contained in the header. A synchronization flag is inserted at every Sample Per Frame (SPF) sample and a multilink frame is structured.

Two types of data transfer modes are specified on the basis of the header structure, namely, full header mode and compressed header mode. Transition between the two modes is performed using the H.245 control message. The multilink frame length can be changed only in the full header mode and the change has to be notified using the H.245 control message. By applying these restrictions, frame synchronization errors are suppressed in the presence of erroneous transmission.

6.3 Mobile Information Service Provision Methods

6.3.1 Mobile ISP Services

6.3.1.1 Introduction

When accessing the Internet using a fixed telephone network, such as a PSTN or ISDN, the access is generally established by connecting to an ISP from the fixed telephone network. On the other hand, when accessing the Internet from a mobile network, the mechanism is basically the same in that the connection is made via an ISP. In both cases, ISPs provide various information services for users to exchange mails or information provided by Internet applications such as Web sites between mobile terminals or PCs and the Internet. The following sections describe in detail the types of services provided as part of the ISP services for connecting to the Internet through a mobile communications network (hereinafter called mobile ISP service), as well as the configuration and functions that are used to enable the provision of such services.

6.3.1.2 Information Services Provided by Mobile ISPs

Portal service is part of the information services provided by mobile ISPs, which function as an entry to access the Internet and search Web sites. Generally, some ISPs provide the portal service on their own and other ISPs use independent portal sites such as Yahoo. At present, however, very few independent portal sites provide portal service specially designed for mobile terminals. Providing portal service as part of mobile ISP service is therefore important to enhance the level of convenience offered to mobile phone users.

Another information service provided by mobile ISPs is the mail service. Mail service offered by the mobile ISPs support mail exchange between mobile terminals or a mobile terminal and a Packet Combining (PC) and so on, connected to a landline telephone. Such mail service embraces functions designed for improved convenience. For instance, when a mobile ISP receives a mail from the sender, the mobile phone is paged. If the mobile handset is ready to receive the mail, it will be transferred to the phone automatically.

The third service is interconnection with the Internet. This service enables the user to access general Web sites by designating the URL without visiting the above-mentioned portal.

The fourth is the bill collection service for premium content. This service manages subscribers' joining and quitting from premium Web sites, and collects the usage fee on behalf of the providers of premium Web sites.

6.3.1.3 Mobile ISP Configuration

Figure 6.12 shows the configuration of mobile ISP, which consists of the following:

Circuit Interface

An interface to connect with the access point of mobile communications network.

Firewall

- *Firewall for leased lines*: Performs access control from the Web site if connection to the provider is made with the leased line. It has the function to cache the access to Web sites from mobile ISP.
- *Firewall for Internet*: Performs access control from the Internet. This firewall also serves as the passage for mails coming via the Internet.

WWW Server

Displays menus for accessing various Web sites. The WWW server also provides My Portal feature that enables the user to customize the Web sites to be displayed on the menu.

Mail Server

Manages mail accounts. Attaches default values to mail accounts, and accepts mail account change requests.

Message Server

A message box for mail and message push (mentioned later). Sends an incoming mail notice to the mobile terminal when the server receives a mail. Accumulated messages are deleted when the specified time has elapsed or transmission is confirmed.

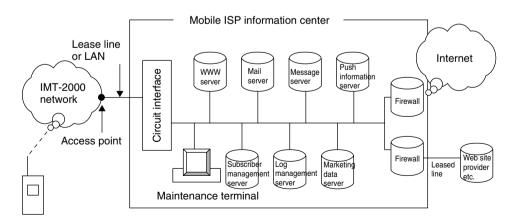


Figure 6.12 Mobile ISP configuration

Push Information Distribution Server

When information from the Web site provider is distributed simultaneously to multiple users such as message push (mentioned later), a single message received from the Web site provider is written into the message boxes of multiple users and the required processing is reduced thereby.

Maintenance Terminal

Sends and receives information necessary for monitoring and maintaining each server in the mobile ISP.

Subscriber Management Server

Manages the subscriber information of the mobile ISP. This server also manages the contract and cancellation information of premium Web sites.

Log Management Server

Collects the system log of each server for operation management.

6.3.1.4 Mobile ISP Functions

Functions for Implementing Portal Services

(1) Link Setup Function Between Portal Service and Web Sites

This function sets up links to various Web sites from the portal site screen provided by the mobile ISP.

This function registers the name of the Web site and URLs to be linked within the portal site menu held in the WWW server of the mobile ISP.

(2) Connection Function to Web Site

This function displays portal site pages provided by the mobile ISP service and enables the user to access various Web sites linked to the portal site.

The Hyper Text Transfer Protocol (HTTP) request issued from a mobile terminal is accepted by the WWW server via the circuit interface, and an HTTP response is returned to each mobile terminal to display the portal site page. If a link to a Web site is designated on the portal site page (in *i-mode*, various sites are displayed on the menu, to which links are connected), the Web site is accessed via a leased line or the Internet based on the URL whose anchor was designated.

(3) My Portal Registration Function

This function allows the user to customize the Web sites displayed on the portal page. In the case of premium Web sites, it also supports the registration to My Portal as well as subscription contract and manages the sites subject to bill collection on behalf of the provider. Furthermore, it also registers the conditions for distributing message push (mentioned later).

After an access to the Web sites is established through the connection procedures mentioned in the preceding text, the Web sites provide guidance on how to register the site in My Portal. (In the case of premium sites, the contractual conditions are presented at this juncture). Then, at the same time, while asking for the password for user authentication, an access is made again to the WWW server of the mobile ISP. The entered password is passed to the subscriber management server via the WWW server. The subscriber management server performs user authentication and other verification. If the data is authentic, a registration completion notice is sent to the mobile terminal via the WWW server and the circuit interface and at the same time the completion of authentication is reported to the Web site.

Mail Service Functions

(1) Mail Transfer Between Mobile Terminals

A mail transmission request from the sender mobile terminal is authenticated by the subscriber management server. After the mail account of the destination is confirmed by the mail server, the message is stored in the message server. The message server notifies the recipient mobile terminal of the reception of a message, and if the terminal is ready, the message is delivered. When the recipient mobile terminal sends a reception confirmation notice, the message is deleted from the message server. If the terminal is not ready to receive the message, the message server stores it temporarily and sends it together with other messages next time the recipient mobile terminal requests distribution.

(2) Mail Transmission to Internet from Mobile Terminals

This function forwards mail messages from mobile terminals to the Internet via the circuit interface and firewall (firewall for Internet).

(3) Mail Reception from Internet by Mobile Terminals

This function lets the mail server verify the destination mail account information of mail messages sent from the Internet via firewall (firewall for Internet) and stores them in the message server. The subsequent processing is the same as the "Mail Transfer between Mobile Terminals."

(4) Message Push Distribution

This function distributes only the messages that meet the conditions registered by the user in advance.

The subscriber management server verifies the destination of messages received from the Internet, after which the messages are distributed to the applicable message box in the message server by the push information distribution server.

6.3.1.5 Challenges for Mobile ISPs

Finally, challenges in implementing portal service will be discussed in the following text as part of the issues to be solved by mobile ISPs in the future.

One of the issues to be taken into account when mobile ISPs offer portal services is to enable users to access various Web sites comfortably, even from the limited screen size of mobile terminals. While portal services for PC-based Internet generally provide functions to display a list of Web sites through keyword search, the screen of a mobile terminal is too small to display all the searched results. Therefore, *i-mode*, for example, displays the menu in a hierarchical structure instead of keyword search to enable access to Web sites. However, if the number of Web sites linked to the menu is too large, the hierarchical structure of the menu becomes too complex for the user to find the desired Web site. One

of the future challenges to be solved is therefore to study a portal functionality unique to mobile terminals that allow the users to find the desired Web sites easily and quickly.

6.3.2 Multimedia Information Distribution Methods

6.3.2.1 Overview of Multimedia Information Distribution Server

In contrast with relatively small amount of information such as voice and text handled by conventional communications, large amount of digital information such as image and sound is called multimedia information. When multimedia information including text, image, and sound is organized and provided as a composed unit, it is called contents.

Contents are created and provided as shown in Figure 6.13. The following sections elaborate on this.

The first step is to create contents with the contents production system. This system consists of an encoder that digitizes and encodes images and sound and an authoring tool capable of creating contents by combining images and sound. The coding methods for images and sound are referred to in Section 6.2. The markup language, which instructs on how to organize multimedia information and express them as contents, is explained in Section 6.3.3.

The next step is to store the output files of the encoder and the authoring tool in the multimedia information distribution server and distribute them to the terminals based on the request from the terminals.

The terminal that received the content performs decoding in order to replay the images and sound in the format before encoding. The contents are then reconfigured and replayed.

There are two methods of distribution between the multimedia information distribution server and a mobile phone, namely, the download method and the streaming method. The download method downloads all the contents into the mobile phone before playing them. The streaming method plays the contents in a sequential manner while they are being sent to the mobile phone.

As shown in Figure 6.14, the download method takes a longer wait time since it downloads all the contents before playing them. In addition, because of the limitation in the terminal memory size, the length of the contents that can be distributed is limited. Since

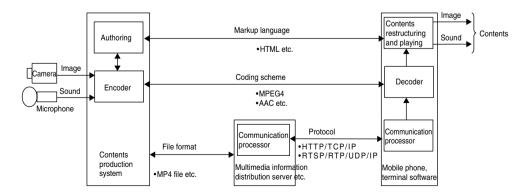


Figure 6.13 Configuration of multimedia information distribution server

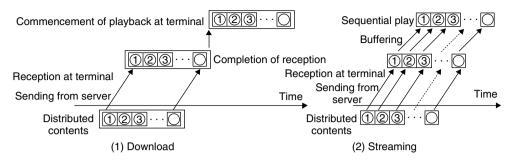


Figure 6.14 Download and streaming

the entire contents distributed can be stored, they can be reproduced if copyright protection is not applied. For copyright protection, refer to Section 6.3.2.2. On the other hand, it takes shorter time for the streaming method before the contents are replayed, as the contents are divided and sent in small units and replayed sequentially. The wait time is the sum of the transmission time and buffering time for each unit. However, this method is not suitable for storing or reproducing the distributed contents.

The download method requires reliable communication protocol between the multimedia information distribution server and a terminal even though transmission delay to some extent may be tolerable. Communication procedures that meet this requirement include HTTP [24, 25] on Transmission Control Protocol/Internet Protocol (TCP/IP) [26, 27] and File Transfer Protocol (FTP) [28], which are used widely on the Internet.

As shown in Figure 6.15, HTTP is a protocol structure implemented on TCP/IP. After the data losses caused by transmission errors are corrected by the functions of TCP/IP, downloading is performed with HTTP. The file designated by the terminal is downloaded from the server according to the sequence between the terminal and the server; the terminal sends a request with HTTP GET and the server responds with HTTP PUT.

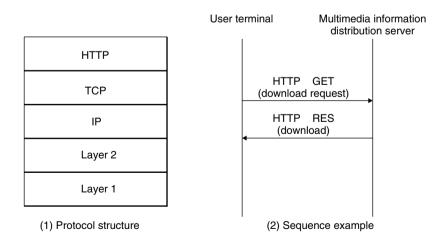


Figure 6.15 HTTP protocol structure and example of sequence

As for the streaming method, on the other hand, solutions from various vendors such as Microsoft's Windows Media Technology [29] and Realnetworks' Real System [30] are competing with one another to establish a *de facto* standard for the Internet. IETF has drawn up a Request For Comment (RFC) for Real-Time Streaming Protocol (RTSP) [31] as a streaming method.

RTSP is used with the protocol structure shown in Figure 6.16. Streaming requires a low transmission delay, while packet loss can be tolerated to some extent. To satisfy this requirement, RTSP is implemented on User Datagram Protocol (UDP) [32], which sends packets without assuring reliability through retransmission and on Real-time Transport Protocol (RTP) [33], which is designed for real-time transmission of images, audio and so on. RTP Control Protocol (RTCP), which reports to the sender the reception status of images and sound transmitted with RTP to control the service quality, is specified in addition to RTP. RTSP is a communication procedure that enables the control of multimedia sessions. With RTSP, it is possible to implement various requirements such as pausing the streaming play of images and sound, or fast forward and slow motion play. Streaming based on RTSP uses a sequence in which the server prepares transmission with SET UP issued by the terminal, starts transmission with PLAY and ends transmission with TEARDOWN.

6.3.2.2 Copyright Protection Method

If multimedia contents are stored, reproduced or distributed without the permission of the copyright holder, not only the holder's right is violated but also the creation and provision of high-quality content predicated upon royalty income may be hampered. To prevent such illegal reproduction, copyright protection methods based on cryptography and electronic watermarking technology have been developed.

Several cryptography-based copyright protection methods have been announced including IBM's Electronic Music Management System (EMMS) [34] and Sony's Open Magic-Gate (OpenMG) [35]. These products were developed with an objective to prevent illegal

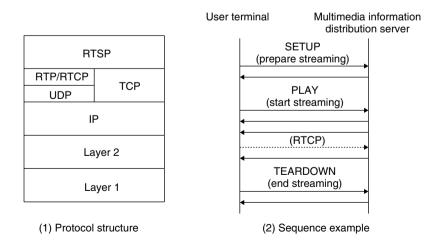


Figure 6.16 RTSP/RTP protocol structure and example of sequence

copying by delivering contents only to authorized users, and making it impossible to replay the distributed contents if they are copied. The method of encryption, encryption key distribution and of invalidating reproduced contents are not made public in most cases to maintain the security of such copyright protection methods.

Although the specific mechanism of these copyright protection schemes has not been disclosed, the basic concept is as follows (Figure 6.17). First, the server authenticates and bills the user, then sends the encrypted contents and decryption key to the users who have paid the fee. Symmetric cryptography is usually used for the encryption and decryption of large contents because it takes less processing time. In symmetric cryptography, if the encryption key is stolen, the contents can also be stolen. Therefore, the encryption key is transferred after it is further encrypted with asymmetric cryptography. Since the data size of encryption key is small, it takes only a short time to encrypt and decrypt the key with asymmetric cryptography. Since contents decrypted with the decryption key can be reproduced and played without another step, encryption specific to the storage media is performed so that the contents cannot be replayed if reproduced on other storage media.

Electronic watermark, on the other hand, controls the use of reproduced contents and detects illegally reproduced contents by embedding unnoticeable information in the contents taking advantage of the large data size of multimedia information such as images and audios. This technique controls reproduction by embedding information such as reproduction permitted, permitted once, or reproduction prohibited in the watermark so that the duplicating device can control reproduction by reading this information. Also, illegally reproduced contents can be identified by reading the rights management information, which contains information such as the copyright holder and the user, written in the electronic watermark.

The electronic watermarking technique has to ensure that it does not spoil the contents with data embedded in multimedia information, that it functions when part of the

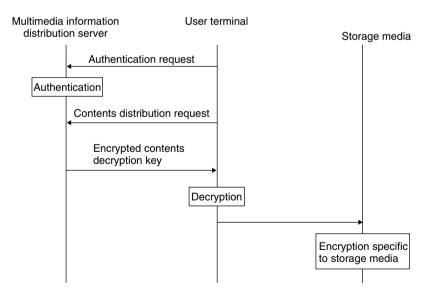


Figure 6.17 Basic concept of copyright protection method

multimedia information is retrieved and that embedding and detecting the hidden data is easily performed. Electronic watermarking techniques that meet these requirements are selected for each application. For example, Secure Digital Music Initiative (SDMI), established for the purpose of specifying the requirements for copyright protection in music distribution, adopts electronic watermarking developed by ARIS (currently known as Verance) and 4C (IBM, Intel, Matsushita, and Toshiba) [36–38].

6.3.2.3 Multimedia Information Storage Methods

As mentioned in Section 6.3.2.1., to distribute multimedia information, multimedia contents are first created with the contents production system, stored in the multimedia information distribution server and then distributed to users. The contents production system and the multimedia information distribution server transfer the multimedia contents in a specific file format. If the download method is used to transfer information between the multimedia information distribution server and the terminal, the terminal retrieves the contents from the file received with HTTP and replays them. As the file format for such multimedia information, Microsoft's Advanced Streaming Format (ASF) [38] and Apple's QuickTime [39] are often used. MPEG 4 specifies the MPEG-4 (MP4) file format as the standard [40].

Figure 6.18 shows an example of the MP4 file format. MP4 stores multimedia information such as images and music in the mdat area in a free format, and the time interval between multimedia information, data size, offset value from the beginning of the file and other information in the moov area. Each area consists of object-oriented structures called atom and each atom is identified with the tag and length.

6.3.3 Contents Markup Languages

6.3.3.1 Compact Hyper Text Markup Language (HTML)

Compact HTML, (hereinafter called CHTML) [41] is a page markup language designed for small information devices such as mobile phones and is a subset of HTML2.0, HTML3.2 and HTML4.0 [42]. CHTML does not support some functions provided by HTML, such as JPEG (Joint Photographics Expert Group) support, table, frame, image map, multiple fonts and styles, background color and image as well as style sheet.

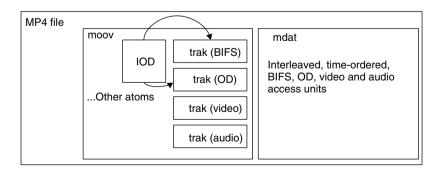


Figure 6.18 MP4 file format

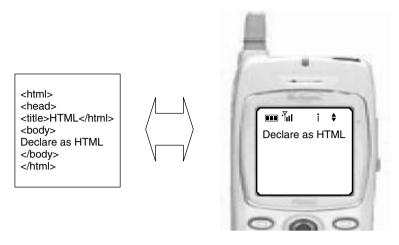


Figure 6.19 CHTML and screen image

A CHTML document is structured in the same way as HTML, with the <html> tag at the beginning and the </html> tag at the end, indicating that the document is written in CHTML. Header information (such as page tile and information for the server) is written between <head> and </head> and the contents displayed on the screen between <body> and </body> (Figure 6.19).

CHTML provides capabilities to link to a telephone number called Phoneto and to use numeric keys on the mobile phone called Easy Focus. Phoneto makes a call to a linked telephone number when it is selected. Link to a telephone number is written in the HREF attribute in the same way as URL or mail address.

< a href = "tel: 090-1234-5678" > Make a call < /a >

Easy Focus enables anchor selection by putting a numeric key in the accesskey attribute as shown in the following text.

```
< a href = "http://www.***.***" accesskey = "1" > Access homepage < /A >
```

In the case of the aforementioned example, the anchor can be selected by just pressing key "1" of the cellular phone once, making the operation quite simple and easy.

The page markup language for the *i-mode* service (*i-mode*-compatible HTML), which was launched by NTT DoCoMo in 1999, is based on CHTML.

6.3.3.2 WML

Wireless Markup Language (WML) [43] is a page markup language used in WAP version 1.0. WML is based on Phone.com's Handheld Device Markup Language (HDML) [44]. Upon the version upgrade from WAP 1.0 to 1.1, tag names specific to HDML were modified to be aligned with HTML.

One of the key features of WML is that contents are described with the concept called "card" and "deck" that enables multiple screens to be downloaded at a time. Figure 6.20

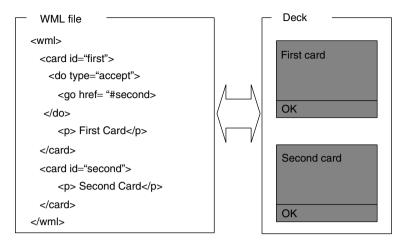


Figure 6.20 Concept of card and deck

shows that the portion between <wml> and </wml> (called deck) is downloaded at a time. The portion between <card> and </card> within the deck is called card constituting one screen. To include multiple screens, multiple cards have to be written in the deck. Each card is identified with the id attribute.

The WML card has four styles, namely, "view text," "enter text," "select" (one from the list), and "multiselect" (more than one from the list). With the \langle setvar \rangle tag, variables and their values can be specified. Switching between cards is described with the \langle do \rangle tag for using soft keys, the \langle a. \rangle tag for indicating hyper text link, and the ontimer attribute for auto switching with the timer.

WML provides a telephony APplication Interface (API) called Wireless Telephony API, enabling the user to call a displayed telephone number, namely, Phoneto in CHTML.

6.3.3.3 XHTML

Extensible HTML (XHTML) [45] is a new contents markup language to replace HTML recommended by W3C in January 2000, redefining HTML in Extensible Markup Language (XML) [46]. In XHTML, a flexible definition of element types and precise data structure description, which had been difficult with HTML, are enabled by XML, enhancing the flexibility and extensibility of contents description. In addition, XHTML can use vector graphics and Synchronized Multimedia Integration Language (SMIL) [47], which enables synchronization with video images, voice and text, enhancing the expressiveness of contents more than HTML.

XHTML uses HTML tags to describe the contents as a well-formed XML document. To turn HTML to well-formed XML, the following must be done; nest tags correctly, use lowercase for tags, enclose attribute values in quotation marks, put the end tag, use the /> tag for null elements and declare as XML. It is also necessary to describe the name space in the html element, a root element in the document, to indicate that tags in the document conform to XHTML as shown in the following text.

< html xmlns = "http://www.w3c.org/1999/xhtml" >

Modularization of XHTML is under way so that a new tag set can be designed easily. The idea is to divide XHTML tags into several modules and to create a new tag set by combining modules. Document Type Definition (DTD) of XHTML 1.1 is a collection of module declarations (called driver) to be used.

A typical XHTML tag set for compact information terminals is XHTML Basic [48] recommended by W3C in December 2000. XHTML Basic is composed of the following XHTML modules: Structure, Basic Text, Hypertext, List, Basic Forms, Basic Tables, Image, Object, Meta information, Link and Base. Table 6.2 lists the tags for each module.

In the next-generation WAP, the markup language will be structured using XHTML Basic as a core.

6.3.3.4 Java*

Java is an object-oriented programming language [49], its execution environment [50] announced by Sun Microsystems in 1995. The execution environment, called Java platform, is composed of Virtual Machine (VM) and class libraries. The Java platform was classified into the following three editions according to the target area of the application in 1999: Java 2 platform Enterprise Edition (J2EE), Java 2 platform Standard Edition (J2SE) and Java 2 Micro Edition (J2ME) [51].

The Java platform used for mobile phones is J2ME. To cope with the difference in Central Processing Unit (CPU) power and memory capacity of each device in which J2ME is implemented, J2ME is composed of core class libraries called configuration [Connected, Limited Device Configuration (CLDC] [47] and Connected Device Configuration (CDC)) [48] and class libraries called profile, provided according to the type of devices. As for VM, two types of VM, Kilo-byte VM (KVM) [52] and Compact VM (CVM) are defined.

Module	Tag			
Structure	Body, head, html, title			
Text	abbr, acronym, address,			
	blockquote, br, cite, code, dfn,			
	div, em, h1, h2, h3, h4, h5, h6,			
	kbd, p, pre, q samp, span,			
	strong, var			
Hypertext	A			
List	dl, dt, dd, ol, ul, li			
Basic forms	Form, input, label, select, option,			
	testarea			
Basic tables	Caption, table, td, th, tr			
Image	img			
Object	Object, param			
Metainformation	Meta			
Link	Link			
Base	Base			

Table 6.2 XHTML basic modules and tags

The Java platform for the mobile phone service launched in 2001 in Japan is composed of KVM or CLDC-compliant VM [called collectively as (K)VM hereinafter], CLDC and Mobile Information Device Profile (MIDP) [53] specified by the profile for mobile phones [Java Community Process (JCP)] or NTT DoCoMo *i-mode* extension libraries (hereinafter "*i-mode* Java") [54]. Contents are programmed using CLDC and the profile for mobile phones. The contents using *i-mode* java are called *i-appli*, and contents using MIDP are called Midlet. An *i-appli* or Midlet is compressed into Jar format files, downloaded to the mobile phone and loaded into (K)VM. MIDP is capable of bundling multiple Midlets in one Jar file to load into (K)VM. On the other hand, *i-mode* java can store only one *i-appli* in one Jar file in view of security.

As the Java platform for mobile phones does not have (K)VM incorporated in the browser, it uses the application management mechanism called Java Application Manager (JAM) to activate, select and delete *i-appli* or Midlet, as well as load into (K)VM and download from the server. *i-appli* is downloaded only if JAM determines that it is executable. Specifically, JAM first downloads the descriptor file that contains information such as the file size and source URL. JAM then determines whether the *i-appli* can be executed according to the information in the descriptor file. Download starts only when JAM determines that it can be executed (refer to Figure 6.21).

Because the available memory capacity of (K)VM is limited to the order of only several kilobytes, it does not support functions such as the user-defined class loader and reflection. In addition, security manager is not supported, either. The security model of (K)VM is a sandbox model [55] that does not allow access to resources other than those specified when (K)VM was implemented, regardless of the contents.

CLDC is composed of J2SE subsets and proprietary classes based on the Generic Connection framework. The Generic Connection framework generalizes datagram-oriented communication, such as serial IO and IrDa, and functionality call, such as socket, HTTP and file IO and handles them uniformly owing to the limitations of memory capacity.

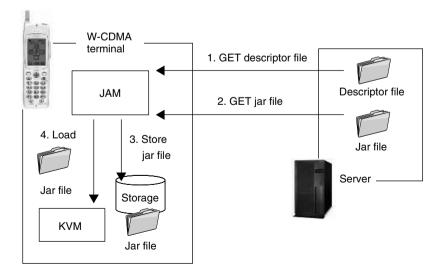


Figure 6.21 Download mechanism

For example, HTTP connection and file open can be handled uniformly using the open method as shown in the following text.

HTTP connection: Connector.open ("http://www.***.***");

File open: Connector.open ("file:/****.dat");

The CLDC data type does not support floating point because the target devices of CLDC are not considered to have a processor for floating point arithmetic.

The profile for mobile phones defines classes for the User Interface (UI), content's life cycle, HTTP communication, timer and so on. MIDP and *i-mode* java differ in class design and behavior. For UI, *i-mode* java adopts component-based class design to provide a look and feel almost the same as *i-mode*-compatible HTML, while MIDP adopts a screenbased class design to implement a look and feel similar to card and deck. For a life cycle, both have the following four states: Loaded, Paused, Active and Destroyed. In i-mode java, when there is an incoming call, the status changes to Paused immediately so that the user can pick up the phone without fail and *I-appli* is interrupted automatically. *i-mode* java is equipped not only with the interval timer and the one-shot timer like the MIDP but also has the short timer to meet the requirements of entertainment contents such as games. One of the MIDP-specific classes is the RMS class used to access data storage. *i-mode* java accesses data storage with the Connector class in compliance with the Generic Connection framework. The classes unique to *i-mode* java include the AudioPresenter class and VisualPresenter class for handling media data in the ui package. These classes are used to program media type contents with Java (*i-mode* offers *i-melody* and other media type contents using animation GIF (Graphics Interchange Format)). In addition, i-mode java contains classes that handle SJIS (Shift Japanese Industrial Standard) text processing to support Japanese language.

6.3.4 Mobile Internet Standardization (WAP)

6.3.4.1 Introduction

The WAP Forum was established in January 1998 by the four companies of Phone.com (now, Openwave), Nokia, Motorola and Ericsson. The number of member companies has increased to 641 as of January 2001 (full members: 251 companies, associate members: 390 companies). The primary objective of this forum is to create an open and globally standardized set of specifications for using Internet services from a wireless network. The WAP Forum cooperates with other standard bodies for Internet and telephony such as W3C, IETF, ECMA (European Computer Manufacturers Association), 3GPP1/2, ETSI and (Mobile Internet Access Forum) MITF to develop the specifications. DoCoMo joined the Forum immediately after its inauguration and has contributed to its activities as one of the 13 board members since October 1998.

In view of the launch of General Packet Radio Service (GPRS) and IMT-2000 services, requests to develop a new version of WAP specifications, compatible with the next-generation mobile networks, have been mounting. To respond to such demands, the WAP Forum initiated the development of new specifications for the next-generation WAP compatible with the next-generation mobile networks (WAP Next Generation: WAP-NG), with the primary focus to achieve convergence with the Internet.

This section reports the recent developments in the WAP Forum. First, the existing WAP specifications (WAP 1.X) and new capabilities planned for the future are introduced. Next, the requirements for the next-generation network are summarized, followed by some updates on the progress of next-generation WAP specifications development.

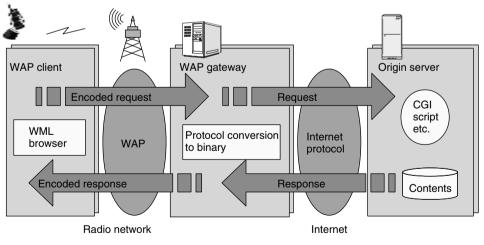
6.3.4.2 Overview of Existing WAP Specifications

WAP 1.0 was released in June 1998. With the publication of WAP 1.1, the basic architecture of WAP was determined. Since then the specifications were upgraded twice. To distinguish these existing WAP specifications from the next-generation WAP, the existing WAP is referred to as WAP1.X in general. Figure 6.22 illustrates the architecture model of WAP 1.X.

WAP 1.0 is composed of basic elements such as protocols, markup language and script. The shape of the standard was organized with the development of WAP 1.1, which made some improvements to the previous version including interoperability and bug fix. The version supported by many MSs today is WAP 1.1. WAP 1.2 released in January 2000 added new capabilities such as push and Wireless Identity Module (WIM). Subsequent to this version, releases are made twice a year (June and December), and the latest version today is WAP 2.0. These specifications have been made public on the Web site of the WAP Forum [56]. The WAP 1.X architecture has two characteristics: (1) the radio protocols are optimized and (2) WML is adopted. Figure 6.23 shows the WAP 1.X protocol stack.

6.3.4.3 Requirements for the Next Generation WAP

WAP 1.X optimized to connect the Internet and the wireless world under the constraints of wireless data transmission, for example, low speed, long delays and unstable connection.



CGI: Common Gateway Interface WAP: Wireless Application Protocol WML: Wireless Markup Language (markup language)

Figure 6.22 WAP 1.X architecture model

Internet	Wireless application protocol		
HTML JavaScript	Application environment (WAE)	Generic name of WAP application environment Services using WML, API	
HTTP	Session layer (WSP)	HTTP 1.1 base Push-type communication, session management	
	Transaction layer (WTP)	•Transaction service class(0,1,2) •Segmenting/reassembly, retransmission control	
TLS - SSL	Transport security layer (WTLS)	•TLS base •Security establishment, ciphering, authentication	
TCP/UDP	Transport layer (WDP)	•UDP base •Bearer adaptation	
IP	Bearer W-CDMA, PDC-P, SMS, CDMA, CDPD, IS-136, USSD etc.		

CDMA: Code Division Multiple Access CDPD: Cellular Digital Packet Data HTML: Hyper Text Markup Language IP: Internet Protocol PDC-P: PDC Mobile Packet Data Communication System SMS: Short Message Service SSL: Secure Socket Layer TCP: Transmission Control Protocol TLS: Transport Layer Security UDP: User Datagram Protocol USSD: Unstructured Supplementary Service Data WAE: Wireless Application Environment WAP: Wireless Application Protocol W-CDMA: Wideband Code Division Multiple Access WDP: Wireless Datagram Protocol WSP: Wireless Tanasport Layer Security WTP: Wireless Transport Layer Security WTP: Wireless Transaction Protocol

Figure 6.23 WAP 1.X protocol stack

This optimization was effective in a circuit-switched connection and a Short Message Service (SMS) to which WAP 1.X was first applied. However, the GPRS services were introduced in Europe in the winter of 2000 and IMT-2000 service was started in Japan in the spring of 2001. These next-generation networks are characterized by their high-speed transmission capabilities. IMT-2000 offers a data rate of up to 384 kbit/s, 40 times faster than GSM data communication (9.6 kbit/s).

Although it is recognized that reflecting the results of the rapid expansion of the Internet in WAP is the key to the future growth of WAP, its current protocols and applications are optimized only for wireless transmission, causing a time lag when incorporating Internet technologies or applications. Optimization for radio transmission was necessary when the transmission speed was limited to between 9.6 and 14.4 kbit/s, however, this is no longer required now, when higher transmission speeds are available. Rather, the convergence with the Internet bears greater importance.

In view of the above, NTT DoCoMo proposed a next-generation WAP, with advanced features for the convergence with the Internet, to the WAP Forum in December 1999 with the cooperation of Ericsson. The proposal triggered the studies on the requirements for WAP-NG in the following year, and the objectives described in the following text were agreed upon.

Convergence with the Internet

The existing Internet specifications defined by IETF/W3C will be incorporated in WAP as much as possible so as to facilitate convergence with the Internet. This should enable utilizing Internet applications and contents on wireless networks in much less time. On the other hand, the achievements of WAP can be reflected in the Internet to contribute to its development.

Convergence with Telephony

WAP stands in a position where three network technologies, namely, wireless data, telephony, and the Internet are integrated. As such, WAP and the telephony technologies provided by organizations such as ETSI, 3GPP, 3GPP2 need to be harmonized.

Smooth Migration from WAP 1.X

The next-generation WAP has to be an evolution from the existing WAP. To facilitate and secure a smooth path for the migration from WAP 1.X, deliberate attention must be paid to backward compatibility.

6.3.4.4 Overview of Next-Generation WAP

Development of the next-generation WAP specifications is currently under way with the cooperation of various manufacturers taking in many of the proposals made by NTT DoCoMo. To meet the two requirements for WAP-NG-to develop the specifications swiftly to respond to market demands and to avoid functional degradation from the existing specifications, NTT DoCoMo proposed to divide the process into several steps and started working on the core specifications in step 1. The following sections summarize the status of progress in each WG involved in the drafting of core specifications under step 1.

Architecture

Since WAP 1.X uses a proprietary protocol optimized for radio transmission, a gateway is installed between the server and clients for protocol conversion. Although the WAP-NG will be added with an IP stack, namely, HTTP, Transport Layer Security (TLS) and TCP/IP, the same gateway architecture as the existing one will be maintained because protocol tuning for radio transmission will remain effective. Figure 6.24 shows the architecture of WAP-NG.

Applications

The next-generation WAP will adopt XHTML, a version of HTML reformatted to the XML format. W3C is working on modularizing XHTML 1.1 and will adopt XHTML Basic as the core of the next-generation markup language. The functions of the existing *i-mode-compatible* HTML will also be supported in the next-generation markup language by adding the subset for mobile phones [Cascade Style Sheet (CSS) Mobile Profile] from CSS. Furthermore, in order to carry over the card and deck and other distinctive features

Client	WAP gateway		Server
WAE (XHTML)	WAE (XHTML)		WAE (XHTML)
HTTP + Push	HTTP + Push	HTTP + Push	HTTP + Push
TLS	TLS	TLS	TLS
Wireless profiled TCP	Wireless profiled TCP	ТСР	ТСР
IP	IP	IP	IP
Radio network (e.g.	In	ternet	

GPRS: General Packet Radio Service HTTP: Hyper Text Transport Protocol IP: Internet Protocol TCP: Transmission Control Protocol TLS: Transport Layer Security

WAE: Wireless Application Environment WAP: Wireless Application Protocol XHTML: eXtensible Hyper Text Markup Language

Figure 6.24 Next-generation WAP protocol

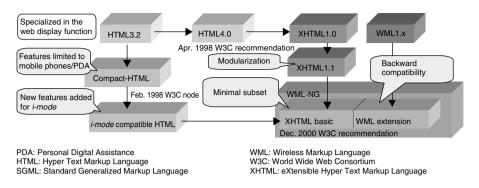


Figure 6.25 XHTML derivations

of WML to the next generation, WML Extension, which realizes the WML functions on XHTML, will be added as well. Figure 6.25 shows XHTML derivations.

Protocols

Internet standard protocols specified by IETF will be adopted. In the transport layer, Wireless Profiled TCP will be used in order to perform tuning on TCP for the wireless environment. Various tuning techniques, which guarantee the interconnectivity with TCP without tuning, have been proposed to IETF. The technique to be used is defined as a profile.

In the application layer, HTTP 1.1 used on the Web, will be adopted.

Security

In step 1, it is agreed that higher priority will be attached to end-to-end TLS, for which Internet standard TLS1.0 will be adopted. TLS is widely used for applications that require end-to-end security, such as banking systems. IETF has proposed several encryption methods for TLS. As in the case of TCP, the scheme to be used and the functions to be supported as a mandatory requirement will be defined as a profile.

Push

Push is a capability specific to WAP, adopted since WAP 1.X. In the packet communication environment, in particular, push services are effective since mobile phones are always connected to the network. The next-generation WAP will adopt two types of push, one that requires confirmation on TCP and one that does not.

These specifications are being drawn up as of January 2001 and are expected to be released in June 2001 after voting by the Forum.

6.4 Multimedia Messaging Methods

6.4.1 Overview

Multimedia messaging is a technology for transferring multimedia information using a store-and-forward type transmission technology called messaging. This technology is distinguished from real-time communication technologies such as videophone and remote conference in terms of immediacy of information. Multimedia information integrates multiple media information including text, video, image and voice into one unit according to a specific format. The MIME format [57–61] is a typical message format.

6.4.2 Trends of Standardization

Standardization of Multimedia Messaging Service (MMS) on the next-generation mobile communications network is under way in 3GPP and WAP Forum [62, 63]. This section outlines the trend of 3GPP.

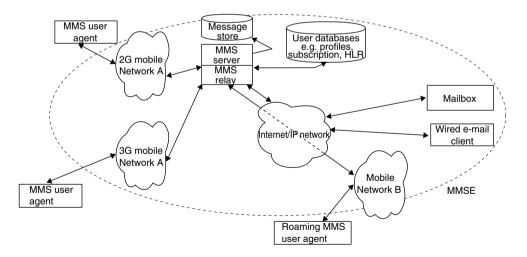
The 3GPP Services and System Aspects WG 1 (3GPP SA1) specified the MMS service requirements [63] in 1999. On the basis of this, the 3GPP Terminals WG 2 (3GPP T2) drafted the MMS functional specifications [64], which were approved in March 2000. As of January 2001, the group is drafting the next release of the MMS functional specifications [65].

The subsequent sections outline MMS based on these specifications.

6.4.3 Conceptual Model

The conceptual model of MMS architecture is illustrated in Figure 6.26.

The entire functional elements required for providing the MMS service are referred to as the MMS environment (MMSE). Main elements of the MMSE include the MMS server for storing and processing multimedia messages, the MMS relay for forwarding messages between messaging systems and the MMS user database for retaining userrelated information such as the profile of each user and the area in which each user is currently located. An application that resides on the mobile phone or equipment connected to the mobile phone to provide the user with capabilities such as reference to, sending, receiving and deleting multimedia messages is called the MMS user agent.



MMSE: Functional element group for providing MMS services MMS Server : Multimedia Message Storage and processing MMS Relay: Message forwarding between Messaging Systems MMS User Database: User-related information database MMS User Agent: Application on mobile phone

Figure 6.26 Conceptual model of MMS architecture

6.4.4 Implementation Model

The 3GPP MMS specifies two implementation models. One is based on the IP technology and the other on WAP technology.

IP-Based Implementation Model

The IP-based implementation model is based on the Internet standard protocols prescribed by IETF. The IP-based gateway, installed as required, performs conversion between the wireless protocol and the fixed network protocol as shown in Figure 6.27. Protocols such as Simple Mail Transfer Protocol (SMTP) [66], POP3 [67], IMAP4 [68] and HTTP [69] are specified to be used for the transfer between the MMS User Agent and MMS Relay, depending on the service.

WAP-Based Implementation Model

The message transfer protocol for the WAP-based implementation model in Figure 6.28 is based on the standards prescribed by the WAP Forum and the IETF. Wireless Session Protocol (WSP) [70] is used as the transfer protocol between the MMS user agent and the WAP gateway. HTTP is used as the protocol between the WAP gateway and the MMS relay.

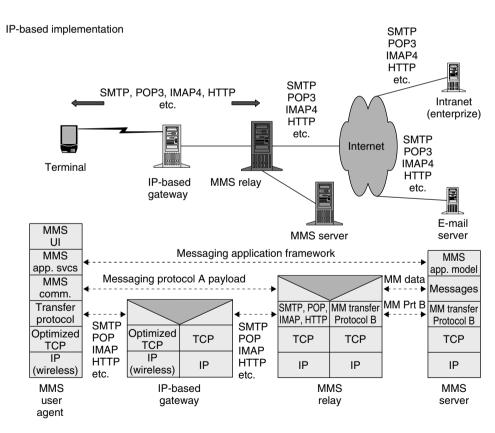


Figure 6.27 IP-based implementation model

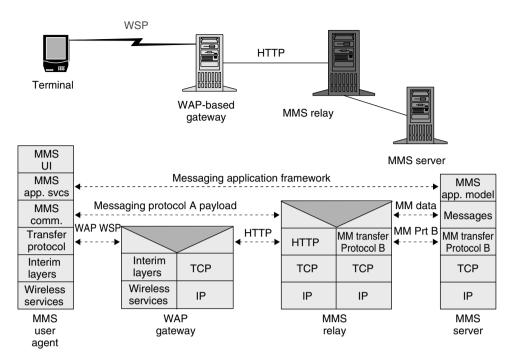


Figure 6.28 WAP-based implementation model

6.4.5 Push Technology

Core technologies of MMS include capability negotiation and push. This section explains push in brief. Push is the capability to notify the user agent of information from the MMSE without a request from the user agent. Push in MMS includes notification and multimedia message delivery. Notification is to notify the user of the arrival of a message when a message arrives at the MMS server. Multimedia message delivery is an automatic message delivery function to the user agent according to the user's settings. MMS Release 1999 prescribes WAP Push Protocol as the protocol for push [71].

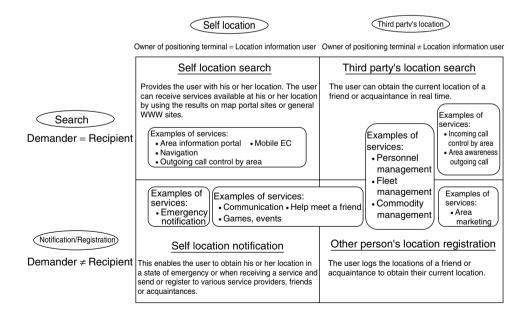
6.5 Location Information Processing Methods

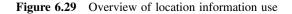
Among the sources of information unique to mobile multimedia, one of the most important is the location information of mobile phones. This information is originally managed by the mobile network in order to send signals to the area in which the mobile phone is located in the event of an incoming call. Recently, owing partly to regulations such as E911 [72] of the Federal Communications Commission (FCC) of the United States, applications that use the location information of mobile phones and various terminals connected to them have been drawing attention. These applications are likely to be more handy and used widely in various areas, as W-CDMA enables data transmission over a wider bandwidth and a more accurate location detection due to smaller cell radius, compared with the conventional Personal Digital Cellular (PDC) technology. This section explains the required location information distribution system in order to use the location information of mobile phones and various terminals connected to them as a mobile multimedia application.

6.5.1 Location Information Use Overview

Before explaining location information processing methods, this section classifies the use of location information to help understand the location information processing methods.

The way in which location information is used can be classified as shown in Figure 6.29, which can be roughly divided into two main categories-location search and notification/registration. The difference in these two categories is whether the person requesting location measurement is the same person who receives the result of the measurement. In the case of notification/registration services, the measurement results are normally received by the server. It is also possible to classify the way in which location information is used by the criteria whether the owner of the terminal to be measured is the same as the user of the measured result. By this criteria it can be divided into services that process the location of the owner's terminal and services that process a third person's location. According to these classifications, the location information use method can be divided into the following four: self-location search, self-location notification, third-party location search and third-party location registration. Figure 6.29 shows examples of services assumed for each category. Recently the use of mobile phones is restricted in some areas such as trains, hospitals and concert halls. In the future, therefore, incoming and outgoing calls may be automatically controlled depending on the location of the terminal or users may be able to switch between voice calls and mail according to the area.





A flexible location information processing system catering to various usage patterns is required.

6.5.2 Structure of the Location Information Processing System

The location information processing system is structured as shown in Figure 6.30. The positioning system is constructed on various mobile communication methods. Networkbased systems include the conventional positioning method used for PHS (Personal Handy-phone), which utilizes the cell location of the Base Station (BS) covering the mobile phone to be searched [7] and other positioning methods that use the time difference or angle of the electric signals sent from the BS to mobile phones, such as AOA (Angle of Arrival), TOA (Time of Arrival), and TDOA (Time Difference of Arrival) [73, 74]. In a broad sense, the method to transmit cell-based information in the mobile network also belongs to this layer. Autonomous GPS-based methods include positioning with ordinary stand-alone measurement by Global Positioning System (GPS), and corrected positioning with differential GPS information. Some network-assisted GPS [75, 76] receive information for positioning via the mobile communications network or use the calculation capability on the server. Besides the above-mentioned method, other methods have also been conceived, including a method to attach buildings with a tag or marker for sending and receiving weak electric waves, for example, infrared and Bluetooth [77, 78] and to obtain the location information with a device connected to the mobile phone.

A transmission method is required to send the location information obtained by the above-mentioned methods to outside the mobile network, that is, to the applications residing on the mobile handsets, to various devices connected with the mobile phones or to the location server in the center. Particularly, it is indispensable for this system to support seamless transmission over various mobile networks, various location-positioning schemes and various terminal devices used in the market. Details are described in Section 6.5.3.

Various information sources		WWW, maps, various databases, and various information services
Location information processing method	Location Information distribution and management technolog system	Method for appropriately coordinating the obtained location information and various information sources
	Location information distribution system	Formats used for describing location information and location-related information
	Transmission method outside the mobile communication networl	Method for transmitting location information seamlessly outside the mobile communications network
Po	sitioning method	Mobile network-based positioning, autonomous GPS- based method, network-assisted GPS-based method, tags and markers
	W-CDMA mobile commu	ication system Various mobile communication systems such as PDC and PHS



The location information distribution methods are used for exchanging the location information sent outside the mobile communications network with various systems via the Internet and for exchanging location-related information, including map information linked to the location information. Details are elaborated in Section 6.5.4.

In addition, a distribution platform is required in order to adequately link the location information with various other information sources such as the World Wide Web (WWW), map and various databases and to facilitate the distribution of location information. This system takes advantage of the so-called information processing technologies such as information search and image processing. Details are provided in Section 6.5.5.

6.5.3 Transmission System Outside the Mobile Communications Network

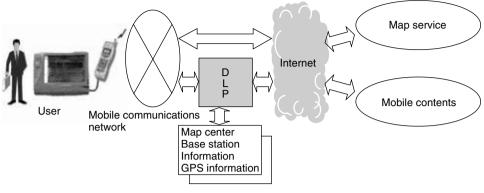
The transmission system outside the mobile communications network is used to transmit the obtained location information to devices outside the mobile communications network, such as applications residing on mobile phones, various terminals connected to the mobile phone and the information server at the center. As various mobile networks, positioning systems and terminals coexist in the market today, a scheme that is capable of transmitting the location information in a seamless manner is indispensable for this system. The following section introduces two notable organizations that study such schemes.

DoCoMo Location Platform Consortium (DLP Consortium)

This consortium consists of about 180 companies (as of the end of 2000) including NTT DoCoMo, Japanese mapmakers, content providers, GPS manufacturers, terminal manufacturers. The group is involved in the studies on a platform capable of providing location information seamlessly, regardless of the difference in the mobile communications networks, positioning methods and terminals. The consortium has been engaged in the specification development of a protocol called Location Information Service Application Protocol (LISAP), for requesting location positioning, processing various positioning sessions and transmitting measurement results between the DLP center facilities, GSP (Global Positioning System) positioning center, BS information server and the terminals and servers of the users of location information, as shown in Figure 6.31 [79, 80]. Studies on the protocol commenced in July 1999, resulted in the creation of Version 1.0 in May 2000 and Version 2.0 in November 2000. As of January 2001, the consortium is continuing its studies, releasing new versions of the specification from time to time. LISAP is a protocol defined mainly on TCP/IP. As recent terminals and servers are equipped with the TCP/IP protocol stack in many cases, LISAP is very easy to implement. LISAP is also developed with due care to security, attaching different passwords to authenticate the requestor of location information for each function in view of privacy protection.

LIF (Location Interoperability Forum)

This forum consists of approximately 80 companies (as of November 2000) including Motorola, Nokia, Ericsson, network operators, terminal manufacturers and service providers. LIF was organized in September 2000 with a goal to develop location positioning and acquisition methods that are independent of radio interfaces, specific positioning methods and network operators and the procedures for verifying these technologies [81]. LIF itself, basically does not propose any plans to be standardized. It rather intends to



Source: NTT DoCoMo News Release (July 29, 1999)

Figure 6.31 Overview of DoCoMo location platform

achieve its goals by making approaches to various standard bodies including ETSI, 3GPP, WAP, IETF, W3C, OpenGIS and ANSI.

6.5.4 Location Information Distribution Methods

Location information distribution methods are used for transferring the location information sent outside the mobile communications network over the Internet as well as location-related information such as town information and map information associated with location information. Various proposals have been presented for standardization, all of which are to be implemented on TCP/IP, on HTTP in particular. In short, these proposals describe the description format and can be classified as presented in Figure 6.32. When the basic information flow is assumed to be sending the measured location information from the terminal to the server and then sending back other location-related information to the terminal, there are various standardization proposals for each of these information flows, as shown in Figure 6.32. They can be classified into XML-based ones, URL-based ones, those based on extended HTML and others. An overview of typical location information distribution methods is presented in Table 6.3.

At present, however, none of them has achieved a decisive position as a proposal for standardization. Although POIX and NVML were submitted to W3C, W3C does not seem to take up individual DTD on XML as standardization proposal.

Under the present conditions, it may be better not to extend existing standard proposals but to adopt a general method, as in the case of *i*-navi link, because methods familiar to terminal software developers and content developers are considered to have a better chance for penetration. Figure 6.33 shows a sample description of location information and location-related information in the format of *i*-navi link.

In the long run, general-purpose and extensible formats such as XML-based formats are likely to be widely adopted. Thus, a generic scheme with highly secure protocols that gives considerations on privacy protection, such as SloP, is expected to gather attention.

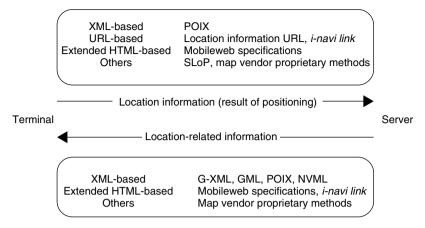


Figure 6.32 Classification of location information distribution system

6.5.5 Location Information Distribution Platform

The location information distribution platform is a platform to appropriately link location information with WWW, maps and various databases to facilitate the distribution of location information. This platform makes the most of information processing technology such as information retrieval and image processing and is composed of various technologies to deal with various objects. The following section introduces some of such technologies.

	Coverage		Proponent	Characteristic	
	Location information expression	Location-related information expression			
(POIX) [82] Point Of eXchange language	O (XML)	O (XML)	Toyota and others (proposed to W3C)	Since POIX was originally developed for use with car navigation, it can express moving direction, moving speed, means of transportation and route information in addition to a spot location.	
(NVML) [83] NaVigation Markup Language		O (XML)	Fujitsu (pro- posed to W3C)	Since NVML is used to describe navigational information from one point to another, it can express messages to be displayed on the way as well as voice messages.	

 Table 6.3
 Overview of location information distribution system

	Co	overage	Proponent	Characteristic
	Location information expression	Location-related information expression		
G-XML [84]		O (XML)	Database Promotion center, Japan	G-XML is designed with an aim to provide an open GIS contents distribution platform by describing map information and POI information to be overlaid in the XML format. Version 1 was made public in May 2000. The cabinet decided to that objects common to G-XML and GML be liaised and set as a JIS standard in 2001.
GML [85]		O (XML)	OpenGIS Consortium	GML is designed to describe GIS information such as map information and POI information to be overlaid in a unified XML format. Version 1.0 was made public in May 2000.
<i>i-navi link</i> specifica- tions [86]	(URL)	(HTML)	NTT DoCoMo	The <i>i-navi link</i> specifications are designed to enable search information on areas around the car in an <i>i-mode</i> -compatible car navigation system and display the searched location on the car navigation map using the obtained POI information. The Web linkage function of GPS-compatible Personal Digital Assistant (PDA) terminal <i>Naviewn</i> [87] also employs the URL-based location information expression specifications. The <i>i-navi link</i> specification is compatible with location information URL.

(continued overleaf)

	Coverage		Proponent	Characteristic	
	Location information expression	Location-related information expression			
Location Information URL [88]	O (URL)		Mobile Office Promotion Association	This format is not used for simply transmitting location information. It can also describe search conditions such as the type of information search and map information the user wants.	
Mobileweb Specifica- tions [89]	O (Extended HTML)	O (Extended HTML)	Mobileweb Promotion Association	This format is designed for use with Web browsers installed in car navigation systems. The user can transmit the current location as well as the destination as location information. A location in the location-related information can be set as the destination. Although extension of HTML tags is generally not welcomed, this specification was adopted because it applies only to the closed area of car navigation service even though it involves HTML tag extension.	
SLoP [90] Spatial Location Payload	(IP)		Nokia and Others (Internet Draft)	Several Internet drafts regarding SLoP have been submitted since around July 2000. This specification is being studied with a focus on IP protocols, security, and privacy protection for transmitting the location information to obtain location-related information with reliable, secure and scalable methods on the Internet.	

Table 6.3 (continued)

Note: JIS: Japanese Industrial Standard

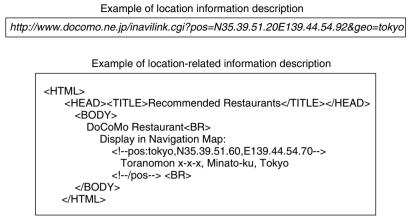


Figure 6.33 Example of description based on *i-navi link* specifications

Location Information Mutual Conversion Technology

Location information can be expressed not only in coordinates by latitude and longitude but also in addresses, zip codes, area codes of phone numbers, stations, landmark buildings and so forth. Even the coordinates can be expressed in latitude/longitude of various positioning systems or in plane rectangular coordinates, which are commonly used for maps. The location information distribution platform requires a technology that enables mutual conversion of various location information expressions. To carry out such conversion, an intermediate coordinate format is used. The coordinates used in the conversion are normally expressed as polygons instead of points. A polygonal search method is required to determine whether polygons overlap or are adjacent to one another. Although indexing is generally used for increasing the search speed in information retrieval techniques, popular indexing algorithms such as hash or Binary Tree do not support search by a range of information that may have two dimensions or more, which is necessary for searching polygons. Therefore, in polygonal search, an indexing algorithm called *r-tree* index [91] is often used. This algorithm forms the centerpiece of location information mutual conversion technology.

Location-Oriented Information Integration Technology

When a user tries to search location-related information using search engines available on the Internet, the user has to enter a letter string that matches the expression of the location information as a keyword. Furthermore, the user cannot designate a search condition that supports a wider geographic coverage. The location-oriented information integration technology solves such problems as shown in Figure 6.34 [92] [93]. This technology enables the location-oriented search engine to automatically extract the location information string in an HTML document on the Internet, converts it into latitude and longitude with the location information mutual conversion technology, automatically stores them as a pair of latitude/longitude and URL in the database and then searches this database using an arbitrary location information expression or polygon as the search key. For directory information composed of CGI (Common Gateway Interface) and so on, the location-oriented meta search engine functions as a mediator to automatically relay the search request.

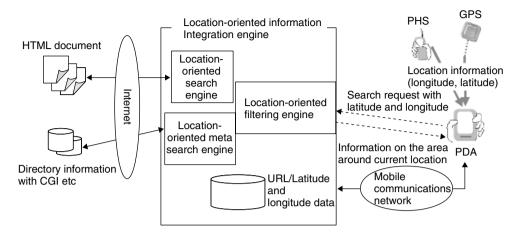


Figure 6.34 Overview of location-oriented information integration technology

In addition, the location-oriented information-filtering engine automatically extracts areaspecific information from the collected information. This technology enables full and flexible search for location-related information on the Internet.

Map Media Control Technology

Map information is regarded as one of the most important location-related information. However, when a map is displayed on the small screen of a mobile phone or a terminal connected to a mobile phone, if the map information is displayed without any modification, users will face difficulty in reading the map as it contains too much information. Map media control technology selects the necessary information depending on the circumstances and automatically converts the map into an optimal media to make it easier to understand [94]. Figure 6.35 illustrates an example of the operation flow of a system adopting this technology. No matter how screen resolution may improve in the future or even if holographic display becomes available, a technology that automatically selects only the map information required by the user depending on the circumstances and controls the appropriate media will continue to be an indispensable technology for providing an easy-to-read map to the users.

Augmented Reality Technology

While Virtual Reality (VR) artificially creates three-dimensional images for the user to feel as if he or she were in the real world, Augmented Reality (AR) overlays virtual images on the real-world images to create an environment in which the real world is expanded. This technology combines highly accurate three-dimensional location detection (in the order of centimeters), orientation detection, image processing and image recognition technologies to overlay appropriate visual information on the real world. In mobile multimedia services, users have more opportunities to interact with the real world rather than with a virtual space [95]. Therefore, there are many cases in which AR technology can be applied. One of the typical examples is the real-view navigation technology [96] and Figure 6.36 shows an example of its operation. In this example, on top of an image actually taken by a camera, information to identify the object is overlaid. By overlaying additional

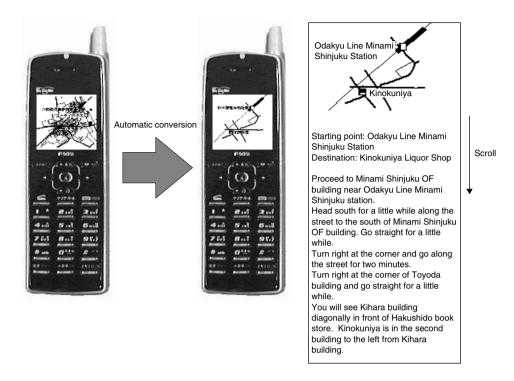


Figure 6.35 Operation of map media control technology (example)



Figure 6.36 Operation of real-view navigation technology (example)

information according to the situation, users will be able to enjoy a more sophisticated interaction with the real world in a mobile environment.

6.6 Mobile Electronic Authentication Methods

6.6.1 Electronic Authentication

Electronic authentication (hereinafter "authentication") is used to certify the authenticity of communication. Authenticity of communication is certified if the person at the remote end is identified as the real person and the communicated message is not altered. The former is called *user authentication* and the latter is called *message authentication*. Recently, authentication with electronic signature based on public key ciphering has become the mainstay on the Internet, because of various reasons including its security in delivering authenticated information, affinity with an open environment and its ability to serve as an evidence for authenticated information.

Figure 6.37 outlines the authentication method using electronic signature.

The signer obtains a message digest by applying the hash function, which is unidirectional and collision-free, to the document to be signed. Next, the signer attaches a message digest encrypted with the signature key to the document as an electronic signature. The electronic signature verifier obtains the message digest by decrypting the

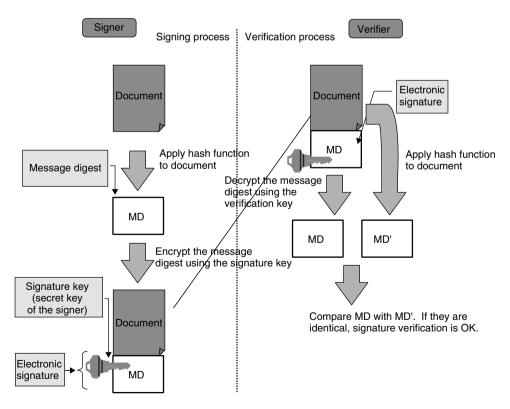


Figure 6.37 Authentication with electronic signature

electronic signature using a public verification key. By comparing this with the message digest of the document, the authenticity of the electronic signature can be verified.

The system that provides the infrastructure for electronic signature is referred to as Public Key Infrastructure (PKI). In authentication schemes based on electronic signature, it must be assured that the verification key belongs to the signer. The relationship between the signer and the verification key is certified by Certification Authority (CA), a main functionality offered by PKI. CA issues a certificate as a document certifying the relationship between the signer and the verification key. A certificate is a document with an electronic signature of the issuer CA, containing information to identify the signer as well as information relating to the corresponding verification key. The signer provides the verifier with a means of secure verification by presenting a certificate issued by the CA trusted by the verifier in addition to the document and electronic signature.

The following sections introduce authentication technologies used for mobile communication focusing on the activities for WAP.

6.6.2 WAP Authentication Model

The players of the WAP authentication model are terminals, server, gateway, PKI portal, CA and certificate database. Authentication is performed on the communications between the terminals and the server. This model is roughly divided into a WTLS-based model [97] and a model based on WML script Sign Text [98, 99]. The former is further divided into the two-phase security model and the end-to-end security model.

The two-phase security model (Figure 6.38) authenticates the communication between the server and the terminal by combining two steps of authentication, between the terminal and the gateway and between the gateway and the server. WTLS (Wireless Transport Layer Security protocol) is used for authentication between the terminal and the gateway. Secure Sockets Layer (SSL)/TLS is used for authentication between the gateway and the server. This model can be regarded as the mainstay model in WAP 1.2 for the following

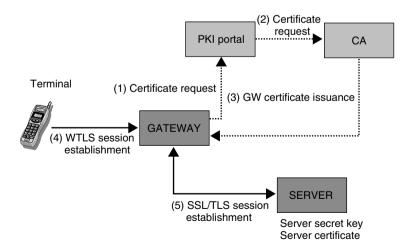


Figure 6.38 WAP electronic authentication model (WTLS two-phase security)

reasons: (1) The terminal only needs to authenticate the gateway, which simplifies the implementation of authentication processing. (2) Protocols catered to wireless transmission are adopted between the terminal and gateway, which results in an efficient utilization of radio spectrum. (3) The popular SSL/TLS protocol is used between the gateway and the server, ensuring excellent connectivity with existing servers on the Internet. In this model, however, it is a tacit understanding that the gateway is secure because it mediates authentication.

Generally, if a communication has to rely on a third entity other than the entities directly involved in communication, this entity tends to become a security hole. In the two-phase security model, the gateway may become such a security hole. This can be fatal when applying this model to services requiring high-level security such as financial services. In view of this, the WAP Forum has also proposed a model that is not predicated on the reliance on a gateway. This model is the end-to-end security model (Figure 6.39).

With the end-to-end security model, the terminal directly authenticates the server without using the gateway. Since both the terminal and the server need not rely on the gateway for authentication, this model can exclude other entities to be trusted (except for CA). This model, however, requires the implementation of WTLS protocols on the server and its connectivity with the Internet servers is relatively lower than the two-phase security model.

Figure 6.40 shows a model based on Sign Text.

While the WTLS-based authentication model authenticates the person to communicate with, the Sign Text model authenticates the authenticity of the author of the signed document. This model also certifies that the document is not altered after signed, consequently authenticating that the document is written by the author at the same time. Although the details of this model have not been standardized, this model is considered to be a major topic of discussion since electronic signature on a document may have a legally binding power with the development of electronic signatures and so on.

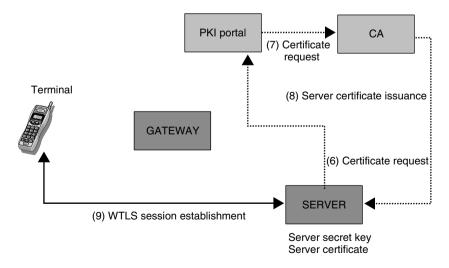


Figure 6.39 WAP electronic authentication model (WTLS end-to-end security)

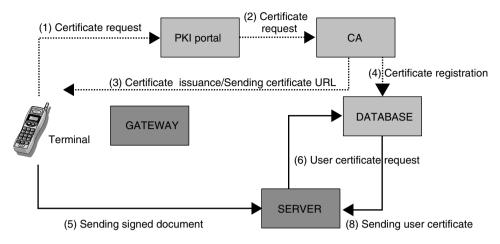


Figure 6.40 WAP electronic authentication model (sign text)

6.6.3 Electronic Certificate for Mobile Communications

In mobile environments in which terminal memory size and available radio bandwidth are limited, reducing the amount of data stored inside the terminals or transmitted over radio channels has always been an important matter. In electronic authentication, reducing the data size of the certificate is particularly important.

In the world of Internet, the X.509 format is commonly used for certificates [100-103]. Although the basic part of the X.509 certificate contains several hundred bytes of information, additional information such as user attribute and authentication policy can be entered as extension, so the entire size may possibly expand to 1 to 2 KB. It must be noted that reducing the certificate size without due attention may hamper the connectivity with the server on the Internet. This section introduces attempts of the WAP Forum to reduce the certificate size.

- 1. For certificates stored in the terminal (such as route CA certificates), decreasing the size is more important than connectivity with the external world. The WTLS certificate [97] used here adopts a proprietary format to reduce the size and save terminal resources.
- 2. For certificates stored in the terminal and transferred on radio channels during communication (such as client certificates), the X.509 certificate is used in view of the connectivity with the server. To ensure efficient radio bandwidth utilization, WAP proprietary profiling [104] is used to limit the certificate size.
- 3. For certificates stored on the server, a certificate compliant with the profile specified by IETF [103] is used taking into account compatibility with the Internet.

6.6.4 Transport Layer Security (TLS)

TLS has been standardized by IETF based on the Secure Socket Layer (SSL), a secure communication protocol developed by Netscape [105]. TLS resides just above the transport layer, and consists of two layers, namely, TLS handshake protocol and TLS record layer protocol (refer to Figure 6.41).

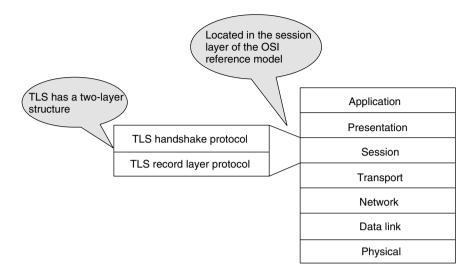


Figure 6.41 Transport layer security

The TLS handshake protocol establishes a secure session by performing user authentication and key sharing. The TLS record layer protocol creates records for actual secure communications by encryption and embedding message certificators.

The WAP Forum specifies WTLS [106], the mobile version of TLS. WTLS differs from TLS in that (1) WTLS is implemented on a datagram (WDP: Wireless Datagram Protocol) [107] in which transport is not guaranteed, while TLS is implemented on a protocol with a guaranteed transport, (2) WTLS can designate the URL that shows the location of the certificate, while TLS has to send the actual certificate to the other end for user authentication at the start of a session. The former, referred to as certificate URL [108], can reduce the data size in TLS handshaking.

6.6.5 Short-Lived Certificate

A certificate is revoked when the key is in jeopardy or the user to whom the certificate was issued is disqualified. CA lists the revoked certificates in the Certificate Revocation List (CRL), where it manages to indicate that these certificates are no longer valid. This information is retained in the CRL until the certificate expires (normally one to two years) and deleted when the certificate expires. The verifier can check the validity of the certificate attached to the electronic signature by referring to the CRL before verifying the electronic signature.

As CA grows in scale and the number of certificates issued increases, the size of CRL will also increase. In a mobile environment, since a terminal cannot store CRL of a large size, an alternative means is required.

The short-lived certificate [109, 110] is one of the means used to solve this problem. The certificate sent from the signer is verified at the gateway of its validity and then replaced by the short-lived certificate. Since the short-lived certificate is valid only for several minutes, it can be regarded as valid without the need to making reference to the CRL. Verifying the validity of the electronic signature without retaining the CRL in the terminal is thus made possible. Yet there are many problems to be solved such as increase in the number of certificates issued and the workload for installing certificates. Whether short-lived certificates be adopted or not will be decided considering the terminal capabilities and transmission speeds.

6.6.6 Future Challenges

As mentioned in the preceding sections, authentication technologies for the mobile environment have been developed mainly from the standpoint of reducing the size of authenticated information and traffic on the radio channels. This approach, however, caused an incompatibility problem with the Internet. With the introduction of W-CDMA, a wider bandwidth radio carrier will become available and improvements in the performance of mobile terminals can be expected. The necessity to develop authentication technology specially tuned for the mobile environment is thus fading. The emphasis has shifted to how to expand this market through convergence with the Internet.

References

- [1] Yasuda, H., 'Global Standard of MPEG/Multimedia Coding', Tokyo, Maruzen, 1994.
- [2] Fujiwara, H., 'Latest MPEG Textbook', ASCII Publishing, Tokyo, 1994.
- [3] Sullivan, G. and Wiegand, T., 'Rate-Distortion Optimization for Video Compression', *IEEE Signal Processing Magazine*, 15(6), 1998, 74–90.
- [4] ITU-T Recommendation H. 261-1993, Video Codec for Audiovisual Services at p × 64 kbit/s, 1993.
- [5] ISO/IEC 11172-2: 1993, Information Technology-Coding for Moving Pictures and Associated Audio for Digital Storage Media at up to About 1.5 Mbit/s-Part 2: Video, 1993.
- [6] ITU-T Recommendation H.262-1995||ISO/IEC 13818-2: 1996, Information Technology-Generic Coding of Moving Pictures and Associated Audio Information: Video, 1995, 1996.
- [7] ISO/IEC 14496-2: 1999, Information Technology-Very-Low Bitrate Audiovisual Coding-Part 2: Visual, 1999.
- [8] 3GPP 3G TS26.090, 'AMR Speech Codec; Transcoding Functions', December 1999.
- [9] 3GPP 3G TR 26.975, 'Performance Characterization of the AMR Speech Codec', January 2000.
- [10] 3GPP 3G TS 26.111, 'Codec for Circuit Switched Multimedia Telephony Services; Modifications to H.324', August 1999.
- [11] ITU-T Recommendation H.320, Narrow-Band Visual Telephone Systems and Terminal Equipment, July 1997.
- [12] ITU-T Recommendation H.310, Audiovisual and Multimedia Systems, September 1998.
- [13] ITU-T Recommendation H.324, Terminal for low bit-rate Multimedia Communication, February 1998.
- [14] ITU-T Recommendation H.323, Packet Based Multimedia Communications Systems, February 1998.
- [15] 3GPP TS 26.111, 'Codec for Circuit switched Multimedia Telephony Service; Modifications to H.324', 1999.
- [16] ITU-T Recommendation H.223, Multiplexing Protocol for low bit rate Multimedia Communication.
- [17] ITU-T Recommendation H.223, Annex A-Multiplexing Protocol for low bit rate Multimedia Mobile Communication over low Error-Prone Channels, February 1998.
- [18] ITU-T Recommendation H.223, Annex B-Multiplexing Protocol for low bit rate Multimedia Mobile communication Over Moderate Error-Prone Channels, February 1998.
- [19] ITU-T Recommendation H.223, Annex C-Multiplexing Protocol for low bit rate Multimedia Mobile Communication Over High Error-Prone Channels, February 1998.
- [20] ITU-T Recommendation H.245, Control Protocol for Multimedia Communication, February 1998.
- [21] ITU-T Recommendation X.680, Information Technology Abstract Syntax Notation One (ASN.1): Specification of Basic Notation, July 1994.

- [22] ITU-T Recommendation X.691, Information Technology ASN.1 Encoding Rules: Specification of Packed Encoding Rules (PER), April 1995.
- [23] ITU-T Recommendation H.324, Annex H-Mobile Multilink Operation, November, 2000.
- [24] Berners-Lee, T., Fielding, R. and Frystyk, H., 'Hyper Text Transfer Protocol', RFC1945, May 1996.
- [25] Fielding, R., Gettys, J., Mogul, J., Frystyk, H. and Berners-Lee, T., 'Hyper Text Transfer Protocol', RFC2068, January 1997.
- [26] 'Transmission Control Protocol', RFC793, September 1981.
- [27] 'Internet Protocol', RFC791, September 1981.
- [28] Postel, J. and Reynolds, J., 'File Transfer Protocol', RFC959, October 1985.
- [29] Microsoft Corporation, 'Inside Windows Media', QUE (ISBN 0-7897-2225-9), November 1999.
- [30] Real Networks, 'RealSystem G2 Overview', http://www.realnetworks.com/devzone/documentation/ wp_g2_overview.html?src=noref,nosrc.
- [31] Schulzrinne, H., Rao, A. and Lanphier, R., 'Real Time Streaming Protocol', RFC2326, April 1998.
- [32] Postel, J., 'User Datagram Protocol', RFC768, August 1980.
- [33] Schulzrinne, H., Casner, S., Frederick, R. and Jacobson, V., 'A Transport Protocol for Real-Time Applications', RFC1889, January 1996.
- [34] IBM, 'The Electronic Media Management System from IBM: A Solution for Digital Media Distribution and Rights Management', http://www-4.ibm.com/software/is/emms/pdfs/emms_brochure_in_english.pdf, July 2000.
- [35] Sony, 'What's Open MG', http://www.openmg.com/jp/what/index.html, November 1999.
- [36] SDMI Portable Device Specification Part1, Version 1.0, July 1999.
- [37] Amendment 1 to the SDMI Portable Device Specification Part1, Version 1.0, September 1999.
- [38] 4C entity, '4C 12 Bit Watermark Specification', http://www.4centity.com/4centity/data/tech/4cspec.pdf, October 1999.
- [39] Towner, G., Apple Computer, 'Discovering QuickTime: An Introduction for Windows and Macintosh Programmers', ISBN 0-12059-640-7, Morgan Kaufman, May 1999.
- [40] ISO/IEC 14496-1, 'Coding of Audio Visual Objects: Systems', 1999.
- [41] Kamada, T., 'Compact HTML for Small Information Appliances', http://www.w3.org/TR/1998/NOTEcompactHTML-19980209, February 1998.
- [42] Raggett, D., 'HTML 4.01 Specification', http://www.w3c.org/TR/html4, December 1999.
- [43] 'Wireless Application Protocol Wireless Markup Language Specification Version 1.2', http://www. wapforum.org/what/technical, November 1999.
- [44] King, P. and Hyland, T. 'Handheld Device Markup Language Specification', http://www.w3c.org/TR/ NOTE-Submission-HDML-spec.html, May 1997.
- [45] Pemberton, S., et al., 'XHTML TM 1.0: The Extensible HyperText Markup Language, A Reformation of HTML4 in XML 1.0', http://www.w3c.org/xhtml1, January 2000.
- [46] Bray, T., et al., 'Extensible Markup Language (XML) 1.0 (Second Edition)', October 2000.
- [47] W3C, 'Synchronized Multimedia Integration Language (SMIL2.0) Specification', http://www.w3c.org/ TR/smil20, September 2000.
- [48] Baker, M., et al., 'XHTML Basic', http://www.w3c.org/TR/xhtml-basic, December 2000.
- [49] Gosling, J., Joy, B. and Steele, G., 'The JavaTM Language Specification', Addison-Wesley, Reading, MA, 1996.
- [50] Lindholm, T. and Yellin, F. 'The Java Virtual Machine Specification', Addison-Wesley, Reading, MA, 1997.
- [51] http://java.sun.com/products.
- [52] Sun Microsystems, 'JavaTM 2 Platform Micro Edition (J2METM) Technology for Creating Mobile Devices', http://java.sun.com/products/cldc/wp/KVMwp.pdf, 2000.
- [53] Sun Microsystems, 'Mobile Information Device Profile (JSR-37)', May 2000, http://java.sun.com/about-Java/communityprocess/final/jsr037/index.html, September 2000.
- [54] NTT DoCoMo, '*i-mode* Supporting Java Contents Developer's Guide (Advanced)', http://www.nttdocomo. co.jp/i/java.html, November 2000.
- [55] Li Gong, 'Inside JavaTM 2 Platform Security', Addison-Wesley, Reading, MA, 1999.
- [56] http://www.wapforum.org/.
- [57] Freed, N. and Borenstein, N., 'Multipurpose Internet Mail Extensions (MIME) Part One: Format of Internet Message Bodies', RFC 2045, November 1996.

- [58] Freed, N. and Borenstein, N., 'Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types', RFC 2046 November 1996.
- [59] Moore, K., 'MIME (Multipurpose Internet Mail Extensions) Part Three: Message Header Extensions for Non-ASCII Text', RFC 2047, November 1996.
- [60] Freed, N., Klensin, J. and Postel, J., 'Multipurpose Internet Mail Extensions (MIME) Part Four: Registration Procedures', RFC 2048, November 1996.
- [61] Freed, N. and Borenstein, N., 'Multipurpose Internet Mail Extensions (MIME) Part Five: Conformance Criteria and Examples', RFC 2049, November 1996.
- [62] WAP Forum, 'WAP Multimedia Messaging Service Version 1.0 Message Encapsulation Draft 0.8', 17 February 2000.
- [63] 3GPP TSG-SA, 'Multimedia Messaging Service; Service Aspects; Stage 1 (Release 1999)', 3G TS 22.140 v3.0.0, 1999–2012.
- [64] 3GPP TSG-T, 'Multimedia Messaging Service (MMS); Functional description; Stage 2 (Release 1999)', 3GPP TS 23.140 v3.0.1, 2000–2003.
- [65] 3GPP TSG-T, 'Multimedia Messaging Service (MMS); Functional description; Stage2 (Release 4)', 3GPP TS 23.140 v4.1.0, 2000–2012.
- [66] Postel, J., 'Simple Mail Transfer Protocol', 1 August 1982.
- [67] Nelson, R., 'Some Observations on Implementations of the Post Office Protocol (POP3)', RFC 1957, June 1996.
- [68] Crispin, M., 'Internet Message Access Protocol-Version 4rev1', RFC 2060, December 1996.
- [69] Fielding, R., Gettys, J., Mogul, J., Frystyk, H., Masinter, L., Leach, P. and Berners-Lee, T., 'Hypertext Transfer Protocol-HTTP/1. 1', RFC 2616, June 1999.
- [70] WAP Forum, 'WAP Wireless Session Protocol', November 1999.
- [71] Wap Forum, 'WAP Push Access Protocol Specification', 8 November 1999.
- [72] United States Federal Communications Commission, 'Enhanced 911', http://www.fcc.gov/e911/.
- [73] Young, R., 'Wireless Positioning Techniques and Services', Office of the Manager National Communication System Technical Notes, http://www.ncs.gov/n6/content/technote/tnv5n2/tnv5n2.htm.
- [74] Birchler, M., 'E911 Phase 2 Location Solution Landscape', FCC Location Round Table, http://www. fcc.gov/e911/mottutorial.pdf.
- [75] SiRF Technology, 'SiRF Web Site', http://www.sirf.com/.
- [76] SnapTrack, 'SnapTrack Web Site', http://www.snaptrack.com/.
- [77] Suwa, K. et al., 'Proposal of Personal Location Information System in Mobile Environment', Multimedia, Dispersion, Cooperation, and Mobile (DICOMO) Symposium, July 1998, pp. 579–586.
- [78] Kato, 'Location Information Acquisition Technique using IC Tag and its Application for Pedestrian Information Guidance System', Information Processing Society of Japan, 58th National Meeting, First term in 1999, 4S-01, pp. 54, 55.
- [79] Toriyama, K. and Yamamoto, H., 'Location Information Service for Pedestrians', NTT Technical Journal, 8(3), 2000–2010, 27–35.
- [80] DLP Consortium, 'DLP Consortium Web Site', http://www.dlp.gr.jp/.
- [81] LIF, 'Location Inter-Operability Forum', http://www.locationforum.org/.
- [82] Kanematsu, H. and Kamada, T., 'POIX: Point Of Interest eXchange Language Specification', W3C Note, http://www.w3.org/TR/poix/.
- [83] Sekiguchi, M., et al., 'NaVigation Markup Language (NVML)', W3C Note, http://www.w3.org/TR/ NVML/.
- [84] Database Promotion Center, Japan, 'GXML Web Site', http://gisclh01.dpc.or.jp/gxml/contents/index.htm.
- [85] Open GIS Consortium, 'Geography Markup Language (GML) Version 1.0', http://www.opengis.org/ techno/specs/00-029/GML.html.
- [86] Katayama, M. Kawabata, H., Koudo, T. and Yamauchi, S. et al., 'Car Navigation System Supporting *i-mode*', NTT DoCoMo Technical Journal, 8(3), 2000, pp. 13–17.
- [87] Yamamori, O. and Igarashi, K., 'Enhanced GPS Built-in Terminal "Naviewn" Development of New GPS Positioning Method', NTT DoCoMo Technical Journal, 8(3), 2000, pp. 36–41.
- [88] Mobile Office Promotion Association, 'Location Information URL Specifications for Mobile Tools', http://www.mopa.or.jp/japanese/specification/url1999.pdf.
- [89] Mobileweb Promotion Association, 'Mobileweb Promotion Association Web Site', http://www.mobileweb.gr.jp/.

- [90] Rosen, B., et al., 'Spatial Location Payload Requirements with Protocol Recommendations', IETF Internet Draft, http://search.ietf.org/internet-drafts/draft-rosen-spatial-requirements-00.txt.
- [91] Guttman, A., 'R-trees: A Dynamic Index Structure for Spatial Searching', Proceedings of the 1984 ACM SIGMOD International Conference on Mgmt of Data, pp. 45–57.
- [92] Yokomichi, S., et al., 'Techniques for Collecting, Structuring, and Searching for Location-Oriented Information', *Information Processing Society of Japan, Journal*, 41(7), 1987–1998.
- [93] Miura, N., et al., 'Structuralization and Classification of Location-Oriented Information Three Experiments of Mobile Information Search', *Information Processing Society of Japan, IPSJ SIG Notes, 99-MBL-11*, **99**(97), 39–44.
- [94] Machida, M., 'A Study on the Presentation of Pedestrian Navigation Information on Tiny Screen of Mobile Phones', Information Processing Society of Japan, IPSJ SIG Notes, 99-MBL-11, 99(97), 57–62.
- [95] Rekimoto, J., 'Research Trend in Real World Oriented Interface', Computer Software, 13(3), 1996, 4–18.
- [96] Katagiri, M., et al., 'Attempt of Mobile Navigation Using Live Movie', Technical Report of PRMU98-169, The Institute of Electronics, Information and Communication Engineers, 1998–2012, p. 149–156.
- [97] WAP Forum, 'Wireless Application Protocol Wireless Transport Layer Security Specification', 18 February 2000.
- [98] WAP Forum, 'WML Script Crypto Library Specification', 5 November 1999.
- [99] WAP Forum, 'Wireless Application Protocol Public Key Infrastructure Definition', Proposed Version 3 March 2000.
- [100] ITU-T Recommendation X. 509-1997 IISO/IEC 9594-8:1997, 'Information Technology Open Systems Interconnection – The Directory: Authentication Framework', 1997.
- [101] ITU-T Recommendation X.509-1997, ISO/IEC 9594-7:1997 DTC 7, 'Technical Corrigendum 7', 1997.
- [102] American National Standard X9.55-1997, 'Public Key Cryptography For The Financial Services Industry: Extensions to Public Key Certificates and Certificate Revocation Lists', 1997.
- [103] Housley, R., Ford, W., Polk, W. and Solo, D., 'Internet X.509 Public Key Infrastructure Certificate and CRL Profile', IETF RFC 2459, January 1999.
- [104] WAP Forum, 'Wireless Application Protocol Certificate and CRL Profiles', 9 March 2000.
- [105] Dierks, T. and Allen, C., 'The TLS Protocol', IETF RFC 2246, January 1999.
- [106] WAP Forum, 'Wireless Application Protocol Transport Layer E2E Security Specification', 11 July 2000.
- [107] WAP Forum, 'Wireless Datagram Protocol Specification', 19 February 2000.
- [108] WAP Forum, 'WAP Identity Module Specification', Part: Security, Version 18, February 2000.
- [109] Ellison, C., Frantz, B., Lampson, B., Rivest, R., Thomas, B. and Ylonen, T., 'SPKI Certificate Theory', RFC 2693, September 1999.
- [110] Hsu, Y.-K. and Seymour, S.P., 'An Intranet Security Framework Based on Short-Lived Certificates', *IEEE Internet Computing*, 2(2), 1998.
- [111] Enoki, K. and Matsunaga, M. et al., 'Special Report on *i-mode Services'*, NTT Technical Journal, 7(2), 1999, 6–32.
- [112] Sun Microsystems, 'Connected, Limited Device Configuration (JSR-30)', http://java.sun.com/aboutJava/ communityprocess/final/jsr030/index.html, July 2000.
- [113] Sun Microsystems, 'J2ME CDC Specification Version 0.2', http://java.sun.com/aboutJava/communityprocess/first/jsr036/index.html, 2000.

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

7 Future Prospects

Yoshiyuki Yasuda, Takchiro Nakamura, Shinji Uebayashi, Hiroshi Fujiya and Tomoyuki Oya

7.1 Overview

As discussed in the previous chapters, the International Mobile Telecommunications-2000 (IMT-2000) system is now an up-and-running system after its studies commenced in 1985 in pursuit of a future mobile communications system. IMT-2000 is expected to develop further into a more advanced and diversified system in response to growing demand and need. Efforts to make the IMT-2000 system more sophisticated are continuing at the International Telecommunication Union (ITU) and at various other organizations. Under the International Telecommunication Union-Telecommunication standardization sector (ITU-T), a new organization called IMT-SSG (Special Study Group) has started working on the future prospects of IMT-2000. In the International Telecommunication Union-Radio communication sector (ITU-R), Working Party (WP) 8F is engaged in studies on the development and sophistication of IMT-2000 after Task Group (TG) 8/1 completed its tasks. The 3rd Generation Partnership Project (3GPP) is working on Release 4/5 with an aim to achieve convergence with Internet Protocol (IP) technologies and the provision of IP multimedia services, building on Release99, 3GPP's first version of IMT-2000 specifications.

In particular, technologies geared to faster and higher-quality packet communications with IP communications in mind have been attracting a great deal of attention in a wide range of fields. Some of them are already about to undergo development for implementation, with standard specifications agreed upon and frequency bands assigned, such as the Time Division Duplex (TDD) transmission scheme suitable for asymmetric traffic. On the other hand, intensive efforts are being made by organizations to improve the properties and the qualities of IMT-2000, including radio transmission technologies for high-speed packet transmissions, IP-oriented network technologies, and signal processing technologies that take into account high-definition speech/acoustic CODEC and packet transmissions.

This chapter reviews some promising technologies for the future that are currently under study for the further advancement of IMT-2000, with reference to a number of topics.

7.2 Prospects of Radio Technologies

7.2.1 TDD Scheme

IMT-2000 CDMA TDD was approved by ITU as one of the radio transmission technologies for IMT-2000, as with Wideband Code Division Multiple Access (W-CDMA) Frequency Division Duplex (FDD) mode. Its standardization is in progress at 3GPP in parallel with W-CDMA. The introduction of TDD is expected to gain momentum after the introduction of W-CDMA, especially in Europe where the frequency band for TDD has already been assigned to carriers.

Figure 7.1 compares the principles of CDMA TDD with FDD. Whereas FDD divides the uplink and downlink channels by frequency, TDD divides each frame (10 msec) into 15 slots on the time axis and assigns an uplink–downlink channel to each slot. FDD and TDD are the same in that the channels are code-multiplexed by spreading codes.

Accordingly, TDD has the following characteristics [1].

- 1. FDD requires a pair of frequency bands for uplink and downlink. In contrast, TDD can be applied to an unpaired frequency band, which means that the conditions for the frequency bands to be used are more relaxed.
- 2. As slots can be assigned freely to uplink and downlink, efficient transmission can be assured when the information volume in uplink is unbalanced with that of downlink.

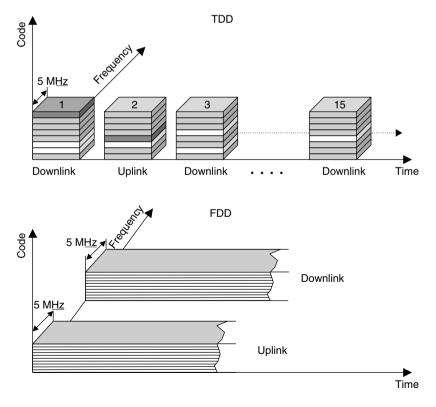


Figure 7.1 Principles of CDMA FDD and TDD

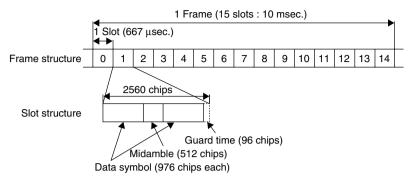


Figure 7.2 Example of IMT-2000 CDMA TDD frame structure

- 3. FDD can be operated with asynchronous Base Stations (BSs). TDD requires inter-BS synchronization in order to avoid interference.
- 4. FDD can suppress transmission power at a low level because of continuous transmission. On the other hand, TDD has a high peak transmission power because of bursty transmissions. Also, the propagation delay must not exceed the guard time between slots, which makes it difficult to cover a wide area.
- 5. The specifications of CDMA TDD and W-CDMA (FDD) of 3GPP have much in common. (Layer 2 and higher layers are the same. Layer 1 has also been made the same to the greatest extent possible, e.g. common chip rate, frame structure etc.)

Owing to these characteristics, TDD is suitable for a system that primarily supports data communications in a small area, rather than a basic cellular system that horizontally covers the entire nation. As it has many things in common with W-CDMA, it may be applied as a dual system with W-CDMA, possibly with a system configuration that complements W-CDMA.

Figure 7.2 shows an example of the frame structure of 3GPP's CDMA TDD [2]. Each frame is divided into 15 slots, and a midamble signal is inserted in the middle of each slot for synchronization and demodulation. At least 1 slot is assigned to a downlink channel including the Synchronization Channel (SCH) and the Common Control Channel (CCCH). Other slots may either be assigned to uplink or downlink.

Table 7.1 shows the basic CDMA TDD specifications of 3GPP. Uplink adopts variable Spreading Factor (SF) to suppress the peak factor of transmission signals, whereas downlink adopts multicode transmission, partly because it allows the simplification of reception. In order to achieve a data rate of 8 kbit/s or so with 1 slot and 1 code, the SF is no more than 16. The chip rate, frame structure, Forward Error Correction (FEC), speech CODEC and so on are the same as in W-CDMA.

CDMA TDD can adopt unique radio transmission technologies by exploiting the small SF and the same frequency applied to uplink and downlink. The main technologies of CDMA TDD are as follows.

7.2.1.1 Interference Cancelation Technologies

In CDMA TDD, it is relatively easy to implement interference cancellation technologies on mobile phones because of the small SF. While 3GPP sets a slot structure assuming

Item	Specifications
Chip rate	3.84 Mcps
Time slot	15 slots/frame
Spreading factor	Uplink: (1), 2, 4, 8, 16
	Downlink: (1), 16
Midamble length	512, 256 chips
Forward error correction	Combination of turbo codes and convolutional codes
Speech CODEC	AMR

 Table 7.1
 Basic specifications of IMT-2000 CDMA TDD

Note: AMR: Adaptive MultiRate

the use of joint detection [3] as the interference cancellation technology, multipath canceller [4] and so on are also considered promising for downlink.

7.2.1.2 Open-Loop Transmit Power Control (TPC)

In FDD, closed-loop TPC, which controls the transmit power of Mobile Stations (MSs) on the basis of the instruction from the BS, is an essential requirement for the MSs because the propagation paths differ between uplink and downlink as different frequencies are assigned. In TDD, mobile phones may use open-loop TPC, which decides the uplink transmission power based on the downlink received power, because the same frequency is used in uplink and downlink, resulting in the same propagation path. Open-loop TPC can control the reception level of uplink signals but cannot estimate the quality in terms of Signal to Interference Power Ratio (SIR), Bit Error Rate (BER) and so on. While it may depend on the reception technology, it is thus believed that mechanisms such as outer-loop control and so on are required as in the case of FDD.

7.2.1.3 Transmission Diversity

Open-loop transmission diversity would be effective because it is possible to take advantage of the same uplink and downlink propagation paths. 3GPP standardizes transmission diversity technologies such as Selective Transmit Diversity (STD) and Transmit Adaptive Antennas (TxAA) [5].

7.2.2 High-Speed Downlink Packet Access (HSPDA)

HSDPA is being studied as a faster packet transmission scheme for IMT-2000, aimed at providing faster peak downlink speed, smaller transmission delays and higher throughput. The main technical characteristics of HSPDA are as follows.

7.2.2.1 Channel Structure

Basically, one physical channel is shared by multiple mobile phones by time division, as in the case of the existing Physical Downlink Shared CHannel (PDSCH). Various

algorithms may be considered in deciding the mobile phone to which information should be transmitted at a certain time: it may be the mobile phone that requests the highest transmission rate based on Adaptive Modulation and Coding (AMC) described in the following text, or it may be decided in consideration of fairness among mobile phones.

The frame structure is basically compliant with the existing frame structure, which is a layered structure consisting of time slots and radio frames. In consideration of low transmission delay and transmission properties, a structure that allows a shorter interleave length than the existing frame length (10 ms) and assumes an interleave length between 1 slot and several slots is being studied.

7.2.2.2 Adaptive Modulation and Coding (AMC)

AMC is a transmission scheme that adaptively and swiftly changes the modulation scheme and the FEC rate according to fluctuations in the propagation environment. In a favorable propagation environment, a faster modulation scheme is applied and the FEC rate is increased to accelerate the transmission rate. Specifically, the mobile phone (or BS) measures the downlink propagation status of each MS from time to time. On the basis of the measurement results, BS determines the mobile phone to which the information should be transmitted and the optimal transmission rate at each interleave length and sends the information. To support higher transmission rates, modulation schemes under study include not only the existing Quadrature Phase Shift Keying (QPSK) but also 8 Phase Shift Keying (8PSK), 16 Quadrature Amplitude Modulation (QAM) and even 64QAM. The coding rate being considered is between 1/4 and 3/4. Figure 7.3 illustrates an example

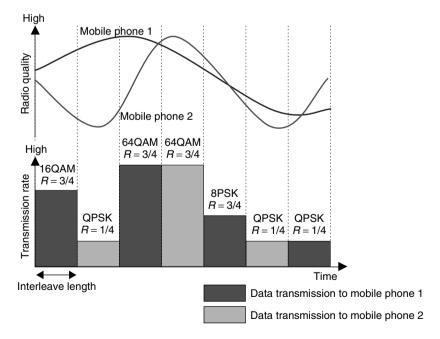


Figure 7.3 Example of AMC application

of AMC operation, showing how AMC is applied when information is transmitted to two mobile phones equally often.

The transmission rate should be controlled as often as slot units, as in the case of fast TPC, in order to track the fluctuations in the propagation environment. However, transmission rate control at regular intervals is being considered because the accuracy of measuring the propagation environment by the mobile phone or BS severely affects the performance of this transmission scheme.

7.2.2.3 Hybrid Automatic Repeat reQuest (H-ARQ)

Studies are being conducted on the application of Hybrid Automatic Repeat reQuest (H-ARQ; refer to Section 2.2.4.1), which is a transmission scheme that combines ARQ with FEC process.

A potential termination node on the UMTS Terrestrial Radio Access Network (UTRAN) side for H-ARQ is Node B and the Radio Network Controller (RNC). However, in consideration of the impact on transmission quality and the memory size of the mobile phone, studies are leaning toward Node B as the termination node because of its ability to shorten transmission delays.

7.2.2.4 Fast Cell Selection (FCS)

For HSDPA, studies are being conducted toward the application of a hard-handover-like transmission scheme that transmits downlink information from only one cell at a certain time, rather than transmitting the same information simultaneously from multiple cells as in the case of soft handover. BSs will be switched quite often to transmit downlink information, and the optimal BS will always be selected by tracking fluctuations in the propagation environment at high speed. Specifically, MS will measure the propagation status in each cell and inform BS of the optimal cell. Downlink information will be transmitted only from the cell that has been informed by the mobile phone.

7.3 Prospects of Network Technologies

7.3.1 IP Packet Communications in Mobile Communication Networks

Both Circuit-Switched (CS) and Packet-Switched (PS) communication technologies supported by existing mobile networks are based on mobility control performed with reference to the same mobile terminal phone number and routing technologies. Accordingly, packet communications merely function as a means to access the Internet, corporate Local Area Network (LAN) and other external IP networks by tunneling the packets inside the network rather than directly routing the users' IP packets (Figure 7.4) [6].

Recently, however, IP communications is becoming increasingly dominant in terms of traffic. Therefore, it is believed that the direct routing and mobility control of IP packets will be an effective way to accomplish a transport scheme that has a high level of affinity with IP communications and suitable for coordinating with and providing various IP applications inside mobile networks. By aligning the basic transport mechanism with the Internet, rapidly progressing IP technologies can smoothly be introduced, which should increase the potential of creating new services in alliance with the Internet. Against this

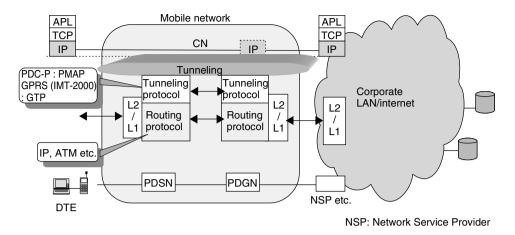


Figure 7.4 Mobile packet communications in PDC-P and IMT-2000

background, efforts are being made to build a network on the basis of the following aims and functions in order to integrate all sorts of communications including voice with IP and to swiftly develop, provide and roll out services.

- 1. Assuming that all terminals would be IP terminals in the future following the progress in IP communications, the network must have functions to execute routing and mobility control directly by the user's IP address.
- 2. Hardware-based switching and transport functions aimed at achieving faster speed and larger capacity must be separated from software-based control functions aimed at diversity and flexibility. By this arrangement, equipment can be dispersed and allocated adequately according to the capabilities of each, and addition and expansion of functions need be done only to the equipment that require them.
- 3. Considering the need to provide various services in the future, it is important to provide and develop services swiftly. To achieve this, Open Application Interface (Open API) must be applied.

7.3.2 Technology Trends in IP Networks

The following technical issues need to be tackled to build an IP network with the aims and functions mentioned in the previous section.

- 1. Study adequate network architecture that is "all IP" from end to end, not necessarily adhering to the conventional configuration based on Radio Access Network (RAN) and Core Network (CN).
- 2. Establish an IP mobility control scheme based on IP addresses without using the phone number of mobile phones.
- 3. Implement end-to-end IP-Quality of Service (QoS) control and real-time communications including speech and streaming video on the basis of such application technologies.
- 4. Establish a signaling scheme over IP for connection control of Voice over IP (VoIP) and so on.

- 5. Study how to apply an architecture that separates the system and control system, and measure the effects of separation.
- 6. Apply Open API to the control system, and study IP multimedia services in coordination with the Internet on the basis of that arrangement.

The following sections refer to IP mobility technologies for item 2 mentioned above, VoIP technologies for items 3 and 4, and Open API for items 5 and 6.

7.3.2.1 IP Mobility Technologies

The current mobile packet communications network performs mobility control based on the phone numbers using the same Location Register (LR) as in circuit switching. In other words, the movement of the MS is tracked and registered on the LR, and the IP addresses of incoming calls from external IP networks are converted into telephone numbers at the gateway, which is forwarded to the location identified by the LR. In contrast, an IP-based network would require mobility control functions using IP addresses.

One way to implement IP mobility is using the mobile IP [7] advocated by the Internet Engineering Task Force (IETF). However, mobile IP is intended to realize portability of IP addresses, and performs mobility control in a dispersed manner. As continuous, fast mobility control must be assured when it is applied to a mobile network, in handover for instance, it is necessary to establish an IP mobility scheme that is suitable for mobile networks, integrating functions as such.

7.3.2.2 Voice Over IP (VoIP)

The progress in IP communications gives rise to the need to support not only PC data but also speech, video and other real-time media handled by telephone networks and broadcasters. QoS control is a technology geared to make this possible, and prominent technologies include Integrated Services (Intserv) and Differentiated Services (Diffserv). It is necessary to verify whether they can effectively work in large-scale mobile communication networks and whether they can assure reliable communications even in the event of congestion and under abnormal conditions. An important challenge to be solved is the establishment of a QoS technology that will allow IP communications to take over circuit-switched communications, which has traditionally been used for voice communications.

To realize VoIP, a signaling scheme that enables capability exchange between the terminals and the network and verifies a secure connection with guaranteed QoS is required. Protocols that help achieve this include H.323 and Session Initiation Protocol (SIP) [8]. Currently, 3GPP is working on studies in the direction of adopting SIP.

7.3.2.3 Architecture with Separate Transmission and Control Systems and Open API

Other than the introduction of IP into networks, another important technology trend is to pursue an architecture that separates the transmission system and the control system. The aim is to adequately disperse and allocate devices according to the capabilities of each, add or deploy functions only on devices that are regarded necessary, swiftly introduce new services and improve the productivity of software, including the application of Open API.

Standardized Open APIs include Parlay [9] and Java API for Integrated Network (JAIN), and attention should be paid to their applications and effects in the future.

7.3.3 All IP Network Configuration and Deployment

Figure 7.5 shows an All-IP network architecture specified in R4/R5 of 3GPP. Although R4/R5 attempts to carry out all transmission functions through IP transport, the CS domain and the PS domain remain separated, and the packet-switching system is still based on the General Packet Radio Service (GPRS). Its mobility function is based on the existing mobility control scheme, and there is much room left to study in regard to the introduction of IP mobility, such as mobile IP. One of the most noteworthy characteristics is the mechanism for providing IP multimedia services in coordination with the Internet called the *IP Multimedia Subsystem* (IMS).

Figure 7.6 shows an example of the configuration of an All-IP routing network that incorporates IP technologies explained hitherto. The Media Gateway (MG) provides the function to connect the fixed phone network or the existing RAN with the IP CN (IP packet conversion, coding etc.). The IP CN consists of an IP router called *Core Router* (CR). The IP router network is equipped with a node with Home Agent (HA) and Foreign Agent (FA) functions, and provides mobile IP functions. The control system consists of Certification Authority (CA), FS and so on, and is separated from the transport system in terms of architecture.

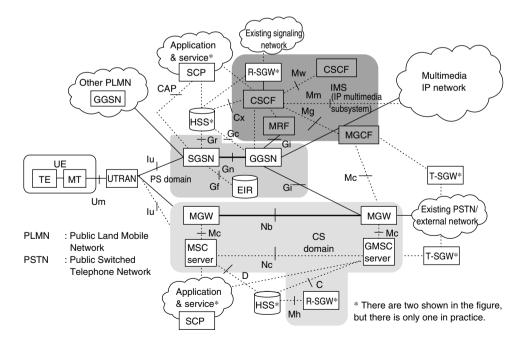


Figure 7.5 All-IP architecture under 3GPP (R4/5 reference architecture)

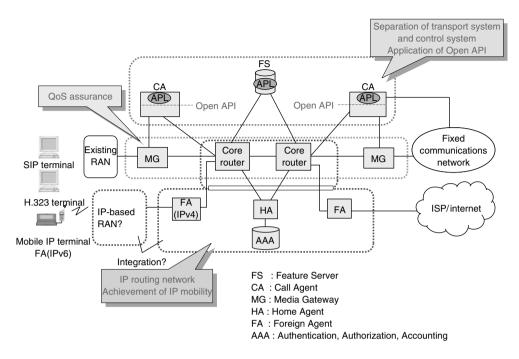


Figure 7.6 Overview of IP-based mobile network configuration (example)

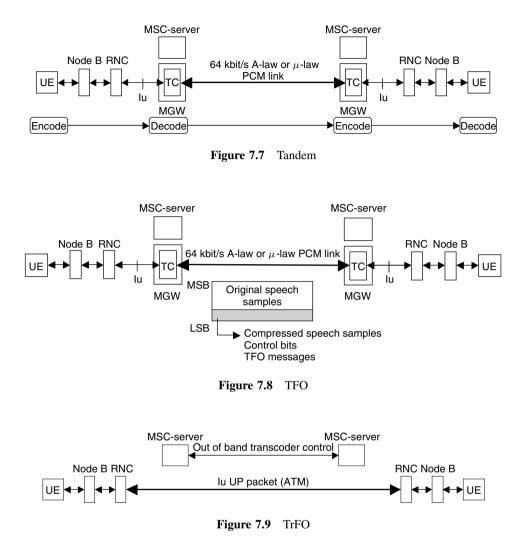
The important task in the future is to assess the adequacy of applying IP in mobile networks and to migrate to a full-fledged IP network from 3GPP R5 onward.

7.4 Prospects of Signal Processing Technologies

As discussed in Chapter 6, there are two types of CODEC specified by 3GPP Release99: (1) CODEC for basic speech services and (2) CODEC for videophone. These specifications are developed primarily by the 3GPP Technical Specification Group-Service and System Aspect (TSG-SA) Working Group (WG) 4 (CODEC). Currently, studies are being conducted to improve the quality and enhance the functions for future 3GPP releases. The main technology topics are as follows.

7.4.1 Tandem Connection Avoidance Technologies

Connections like the one illustrated in Figure 7.7, which occur in mobile-to-mobile connection, are referred to as the tandem connection of CODEC. As reported in literature [10], when there is a tandem connection, coding and decoding take place two or more times, which inevitably leads to deterioration in quality because of quantization distortion in CODEC. The deterioration in quality is particularly acute in low bit rate coding. Tandem-Free Operation (TFO) [11] and Transcoder-Free Operation (TrFO) [12], which are standardized in 3GPP Release 4, are technologies to avoid in tandem connections, which may be applied when the same CODEC is used. Other than preventing the quality from



deteriorating, TFO and TrFO are expected to help effectively use network resources and suppress increases in delays.

The difference between TFO and TrFO depends on whether there is a TransCoder (TC) in the communication path. In TFO, the interfacing TCs negotiate the CODEC using the least significant bit in the 64 kbit/s Pulse Code Modulation (PCM) link, and in a TFO state, maps the coded information in the least significant bits (Figure 7.8). In contrast, in TrFO, the interfacing Mobile Switching Center-Servers (MSC-Server) negotiate the CODEC and execute routing by excluding TC from the communication path so as to transmit the Iu UP packet [13] that transmits coded information directly to RNC on the other side (Figure 7.9).

As TFO and TrFO are controlled by the network, the user does not have to be aware of them.

7.4.2 Adaptive MultiRate-WideBand (AMR-WB)

Services for transmitting high-quality audio data are rapidly spreading, including music distribution over the Internet. At present, international standards for coding are specified to achieve playback quality equivalent to Compact Disc (CD), including the International Organization for Standardization/International Electrotechnical Commission (ISO/IEC) Moving Picture Experts Group (MPEG) 1, 2, 4 and so on [14–18]. These specifications are mainly applicable to 48 kHz sampling (24 kHz playback band), for high bit rate, high-quality audio playback of approximately 48 kbit/s–128 kbit/s (Figure 7.10). On the other hand, Adaptive MultiRate-Narrow Band (AMR-NB), which is an AMR CODEC for speech telephony services referred to in Chapter 6, is applicable to 3.4 kHz playback band, and specializes in encoding speech between 4.75 kbit/s and 12.2 kbit/s.

With the aim to bridge the gap between the areas to which these two standards are applicable, the standardization of AMR-WB is under progress. It is studied as a wideband speech encoding scheme (7 kHz playback band) that can be commonly used among the 3G UTRAN channel, the Global System for Mobile Communications (GSM) full-rate channel (22.8 kbit/s), the EDGE phase II channel and the GSM multislot channel (n*22.8 kbit/s). The requirements including quality are as shown in Table 7.2, and an algorithm with a bit rate of approximately 6.6 to 23.85 kbit/s is due to be approved by 3GPP in March 2001.

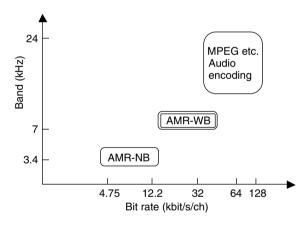


Figure 7.10 Areas to apply speech/acoustic CODEC

	Table 7.2	Requirements	for	AMR-	WB
--	-----------	--------------	-----	------	----

	Requirements	Remarks
Coding processing volume	40wMOPS	Approx. 2.4 times of AMR-NB
Required memory	15 kword RAM 18 kword ROM	Approx. 1.2 to 2.8 times of AMR-NB
Quality	Must exceed G.722-48 k to G.722-56 k assuming error free to C/I 13dB	

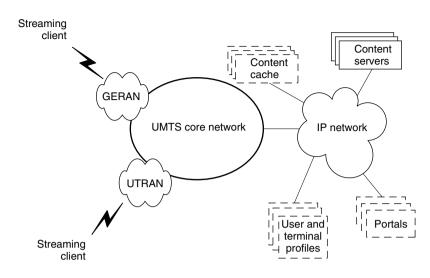


Figure 7.11 Packet streaming configuration

7.4.3 Packet-Transmitted Multimedia

As referred to in Chapter 6, IMT-2000 can provide various multimedia applications, such as videophones. Because of constraints in circuit utilization efficiency, multimedia will mainly be provided on CS connection that has limited header overhead, especially during the early days after service launch. However, the use of multimedia on IP protocol is expected to gain momentum in the future, in consideration of their compatibility with multimedia applications on the Internet. 3GPP is working on multimedia protocols and CODECs with the introduction of various QoS assurance technologies in mind, including IM Subsystem in CN.

Various activities are under way for the standardization of multimedia based on IP protocols, such as the standardization of the transport protocol by IETF [19] and the specification of service implementation provisions by Wireless Multimedia Forum (WMF) [20]. 3GPP activities are mainly concentrating on coding that is adapted to the transmission properties of 3G systems.

Two types of packet multimedia CODECs are being studied by 3GPP, namely, (1) a packet multimedia CODEC for interactive, real-time speech and (2) a CODEC for packet streaming. The latter CODEC specifies audiovisual streaming assuming the system illustrated in Figure 7.11.

As shown in Figure 7.12, the scope of activities to define the CODECs and profiles include not only speech, video and acoustic CODECs but also provisions for still pictures and text transmission, Synchronized Multimedia Integration Language (SMIL) [21], other scene description languages and provisions for inter-terminal protocols incorporating them. In regard to provisions for the transport layer, a great deal of importance is attached to the compatibility with IETF standards such as Real-time Transport Protocol (RTP), Real-time Transport Control Protocol (RTCP), Real-time Streaming Protocol (RTSP). In the future, multimedia communications between a wide range of terminals including the Internet is expected to become a reality.

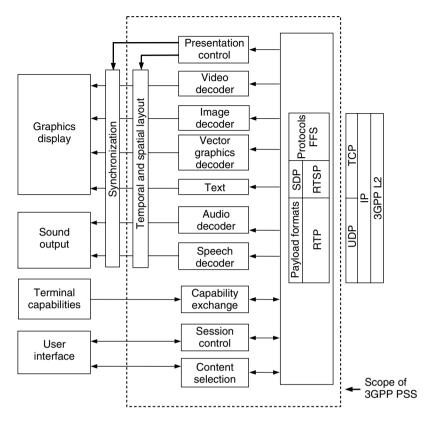


Figure 7.12 Specifications of packet streaming CODEC

References

- Futakata, T. and Uebayashi, S., 'Examination of IMT-2000 TDD System Configuration Scheme', Proceeding of the IEICE General Conference, B-5-156, 1997–1999, p. 144.
- [2] 3GPP, 'Physical Channels and Mapping of Transport Channels onto Physical Channels (TDD)', 3G TS 25.221 v3.4.0, September 2000.
- [3] Klein, A., Kalch, G.K. and Baier, P.W., 'Zero Forcing and Minimum Mean-Square-Error Equalization for Multiuset Detection in Code-Division Multiple-Access Channels', *IEEE Transaction on Vehicular Technology*, 45(2), 1996, 276–287.
- [4] Higuchi, K., Fujiwara, A. and Sawahashi, M., 'Throughput Performance of High-Speed Packet Transmission with Adaptive Modulation and Coding Scheme Using Multipath Interference Canceller in W-CDMA Forward Link', to appear in IEEE Vehicular Technology Conference 2001, May 2001.
- [5] 3GPP, 'Physical Layer Procedures (TDD)', 3G TS 25.224 v3.4.0, September 2000.
- [6] Fujitani, H. 'Mobile Communications Network and IP Communications Technology', *IEICE Journal*, 83(4), 327–333, Apr. 2000.
- [7] Perkins, C., editor, 'IP Mobility Support', IETF RFC2002, October 1996.
- [8] 'SIP: Session Initiation Protocol', IETF RFC2543, March 1999.
- [9] Parlay APIs 2.1 Specifications, http://www.parlay.org.
- [10] 3GPP TR 26.975, 'Performance characterization of the AMR Speech Codec'.
- [11] 3GPP TS 28.062, 'Inband Tandem Free Operation (TFO) of Speech Codecs; Stage 3-Service Description'.
- [12] 3GPP TS 23.153, 'Out of Band Transcoder Control-Stage 2'.
- [13] 3GPP TS 26.102, 'AMR speech Codec; Interface to Iu and Uu'.

- [14] ISO/IEC 11172-3, 'Coding of Moving Pictures and Associated Audio for Digital Storage Media at up to about 1.5 Mb/s-Part 3: Audio', August 1993.
- [15] ISO/IEC 13818-3, 'Generic Coding of Moving Pictures and Associated Audio Information-Part 3: Audio', May 1995.
- [16] ISO/IEC 13818-7, 'MPEG-2 Advanced Audio Coding (AAC)', December 1997.
- [17] ISO/IEC JTC1/SC29/WG11 N2503, 'Text of ISO/IEC FDIS 14496-3', October 1998.
- [18] ISO/IEC JTC1/SC29/WG11 N3058, 'Text of ISO/IEC14496-3 FDAM1', December 1999.
- [19] http://www.ietf.org.
- [20] http://www.wmmforum.com/.
- [21] http://www.w3.org/AudioVideo/.

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

Appendix

Interface Specifications

Appendix Table 1 shows the documentation and the list of items contained in 3GPP specifications. The table shows the 21-series and the subsequent series relating to IMT-2000 DS-CDMA. GSM specifications of 01 to 12 series are excluded. For further information, refer to Appendix Table 2, which lists the specification number, title and the relevant TSG-WG name. For reference, the corresponding ARIB/TTC standard numbers are also indicated. It should be noted that this volume was compiled on the basis of the information available as of September 2000. As there may have been changes since then depending on the progress of standardization, the latest information should be obtained from the following Web sites.

3GPP-related information: *http://www.3gpp.org/* ARIB-related information: *http://www.arib.or.jp/* TTC-related information: *http://www.ttc.or.jp/*

Series number	Item
21-series	Requirements
	Specifies requirements relating to the entire system.
	 Release '99 requirements, USIM requirements
	- Security requirements etc.
22-series	Service specifications
	- Specifies service requirements and guidelines.
23-series	Technical implementation-related
	- Specifies basic functions required for implementing service requirements.
24-series	Signal protocol (UE-CN)
	- Specifies signal protocols for each node required by each service.
25-series	UTRA radio access specifications
	 Specifies UTRA radio access (Layers 1, 2 and 3) and UTRA network specifications.

Appendix Table 1 3GPP document series numbers and items

(continued overleaf)

Series number	Item
25.1XX-series	UTRA radio characteristics
	Defines specifications relating to radio characteristics of UTRA
	 Mobile station transmit/receive characteristics
	 Base station transmit/receive characteristics etc.
25.2XX-series	UTRA radio access; Layer 1 specifications
	 Physical channel structure
	 Multiplexing, channel coding and interleave process
	 Spreading and modulation process
	 Procedures for establishing synchronization and transmit power
	control etc.
25.3XX-series	UTRA radio architecture; Layers 2 and 3 specifications
	 Overview of radio interface protocols
	 Services and functions of Layers 2 and 3
	 Radio resource management procedures
	 MS cell selection procedures etc.
25.4XX-series	UTRA network specifications
	 Overview of UTRAN architecture
	 Uu interface synchronization mechanism
	– Iur interface
	– Iub interface
25.8XX	UTRA technical material
25.9XX-series	 Technical materials related to 25-series
	– RF introduction
	– UTRAN vocabulary etc.
26-series	CODEC-related
	 Defines speech/video CODEC specifications etc.
27-series	Data applications-related
	 Defines specifications for implementing data applications.
28-series	Signal protocol (relating to RNS – network)
	- Defines specifications of signal protocol between RNS and network.
29-series	Core network
	 Defines specifications of signal protocol in core network.
30-series	3GPP work program
	- Project management documents relating to work programs of 3G mobile
	communications system defined by 3GPP.
31-series	USIM-related
	 Defines specifications of Universal Subscriber Identity Module (USIM).
32-series	Operation and maintenance
	 Defines specifications relating to operation and maintenance.
33-series	Security
	- Defines specifications relating to security.
34-series	Test specifications
	 Defines specifications relating to tests involving mobile phones.
35-series	Algorithms
	 Defines specifications relating to algorithms.

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	21.101	3rd generation mobile system Release '99 specifications	S	ARIB/ TTC	ARIB STD-T63-21.101 JP-3GA-21.101
TS	21.102	3rd generation mobile system Release 2000 specifications	S	ARIB/ TTC	
TS	21.111	USIM and IC card requirements	T3	ARIB	ARIB STD-T63-21.111
TS	21.133	Security threats and requirements	S 3	ARIB	ARIB STD-T63-21.133
TR	21.810	Report on multimode UE issues; ongoing work and identified additional work	T2	ARIB	ARIB STD-T63-21.810
TR	21.900	3GPP working methods	S	_	
TR	21.904	UE Capability Requirements (UCR)	T2	ARIB	ARIB TR-T12-21.904
TR	21.905	3G vocabulary	S 1	ARIB/ TTC	ARIB TR-T12-21.905 JP-3GA-21.905
TR	21.910	Multimode UE issues; categories, principles and procedures	T2	ARIB	ARIB TR-T12-21.910
TR	21.978	Feasibility technical report – CAMEL control of VoIP services	N2	TTC	Not published
TS	22.001	Principles of circuit telecommunication services supported by a Public Land Mobile Network (PLMN)	S1	ARIB	ARIB STD-T63-22.001
TS	22.002	Circuit bearer services supported by a PLMN	S 1	ARIB	ARIB STD-T63-22.002
TS	22.003	Circuit teleservices supported by a Public Land Mobile Network (PLMN)	S1	ARIB	ARIB STD-T63-22.003
TS	22.004	General on supplementary services	S 1	TTC	JP-3GA-22.004
TS	22.011	Service accessibility	S 1	TTC	JP-3GA-22.011

Appendix Table 2 List of 3GPP ARIB/TTC specifications (Release 99)

(continued overleaf)

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	22.016	International Mobile Equipment Identities (IMEI)	S 1	ARIB	ARIB STD-T63-22.016
TS	22.022	Personalization of GSM ME mobile functionality specification; Stage 1	S 3	ARIB	ARIB STD-T63-22.022
TS	22.024	Description of Charge Advice Information (CAI)	S 1	TTC	JP-3GA-22.024
TS	22.030	Man-Machine Interface (MMI) of the Mobile Station (MS)	S 1	ARIB	ARIB STD-T63-22.030
TS	22.034	High-Speed Circuit-Switched Data (HSCSD); Stage 1	S1	ARIB	ARIB STD-T63-22.034
TS	22.038	SIM Application Tool kit (SAT); Stage 1	S 1	ARIB	ARIB STD-T63-22.038
TS	22.041	Operator-determined call barring	S 1	TTC	JP-3GA-22.041
TS	22.042	Network Identity and Time Zone (NITZ); Stage 1	S 1	TTC	JP-3GA-22.042
TS	22.043	Support of Localized Service Area (SoLSA); Stage 1	S 1	TTC	JP-3GA-22.043
TS	22.057	Mobile station application Execution Environment (MExE); Stage 1	S1	ARIB	ARIB STD-T63-22.057
TS	22.060	General Packet Radio Service (GPRS); Stage 1	S 1	TTC	JP-3GA-22.060
TS	22.066	Support of Mobile Number Portability (MNP); Stage 1	S1	TTC	JP-3GA-22.066
TS	22.067	enhanced MultiLevel Precedence and Preemption service (eMLPP); Stage 1	S1	ARIB	ARIB STD-T63-22.067
TS	22.071	LoCation Services (LCS); Stage 1	S 1	ARIB	ARIB STD-T63-22.071

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	22.072	Call Deflection (CD); Stage 1	S 1	TTC	JP-3GA-22.072
TS	22.078	CAMEL; Stage 1	S 1	TTC	JP-3GA-22.078
TS	22.079	Support of optimal routing; Stage 1	S1	TTC	JP-3GA-22.079
TS	22.081	Line identification supplementary services; Stage 1	S1	TTC	JP-3GA-22.081
TS	22.082	Call Forwarding (CF) supplementary services; Stage 1	S 1	TTC	JP-3GA-22.082
TS	22.083	Call Waiting (CW) and call Hold (HOLD) supplementary services; Stage 1	S1	TTC	JP-3GA-22.083
TS	22.084	MultiParTY (MPTY) supplementary service; Stage 1	S1	TTC	JP-3GA-22.084
TS	22.085	Closed User Group (CUG) supplementary services; Stage 1	S 1	TTC	JP-3GA-22.085
TS	22.086	Advice of Charge (AoC) supplementary services; Stage 1	S1	TTC	JP-3GA-22.086
TS	22.087	User-to-User Signaling (UUS); Stage 1	S 1	TTC	JP-3GA-22.087
TS	22.088	Call Barring (CB) supplementary services; Stage 1	S1	TTC	JP-3GA-22.088
TS	22.090	Unstructured Supplementary Service Data (USSD); Stage 1	S 1	TTC	JP-3GA-22.090
TS	22.091	Explicit Call Transfer (ECT) supplementary service; Stage 1	S1	TTC	JP-3GA-22.091
TS	22.093	Call Completion to Busy Subscriber (CCBS); Stage 1	S1	TTC	JP-3GA-22.093
TS	22.094	Follow me; Stage 1	S 1	TTC	JP-3GA-22.094
TS	22.096	Calling NAme Presentation (CNAP); Stage 1 (T1P1)	S 1	TTC	JP-3GA-22.096

(continued overleaf)

ppoint	IX Table 2	(communed)			
TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	22.097	Multiple Subscriber Profile (MSP); Stage 1	S 1	TTC	JP-3GA-22.097
TS	22.100	UMTS Phase 1	S1	ARIB	ARIB STD-T63-22.100
TS	22.101	UMTS service principles	S 1	ARIB	ARIB STD-T63-22.101
TS	22.105	Services & service capabilities	S1	ARIB	ARIB STD-T63-22.105
TS	22.115	Service aspects charging and billing	S1	TTC	JP-3GA-22.115
TS	22.121	Provision of services in UMTS – the virtual home environment; Stage 1	S1	TTC	JP-3GA-22.121
TS	22.129	Handover requirements between UMTS and GSM or other radio systems	S 1	TTC	JP-3GA-22.129
TS	22.135	Multicall; Stage 1	S 1	TTC	JP-3GA-22.135
TS	22.140	Multimedia messaging service; Stage 1	S1	ARIB	ARIB STD-T63-22.140
TR	22.924	Charging and accounting mechanisms	S1	TTC	TD-3GA-22.924
TR	22.925	Quality of service and network performance	S1	TTC	TD-3GA-22.925
TR	22.945	Study of provision of fax service in GSM and UMTS	T2	ARIB	ARIB STD-T63-22.945
TR	22.960	Mobile multimedia services	S 1	TTC	TD-3GA-22.960
TR	22.970	Virtual home environment report	S1	TTC	TD-3GA-22.970
TR	22.971	Automatic establishment of roaming relationships	S1	TTC	TD-3GA-22.971
TR	22.975	Advanced addressing	S 1	TTC	TD-3GA-22.975
TS	23.002	Network architecture	S2	TTC	JP-3GA-23.002
TS	23.003	numbering, addressing and identification	N4	TTC	JP-3GA-23.003
TS	23.007	Restoration procedures	N4	TTC	JP-3GA-23.007
TS	23.008	Organization of subscriber data	N4	TTC	JP-3GA-23.008
TS	23.009	Handover procedures	N1	TTC	JP-3GA-23.009

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	23.011	Technical realization of supplementary services – general aspects	N4	TTC	JP-3GA-23.011
TS	23.012	Location management procedures	N4	TTC	JP-3GA-23.012
TS	23.014	Support of Dual Tone MultiFrequency (DTMF) signaling	N1	TTC	JP-3GA-23.014
TS	23.015	Technical realization of Operator-Determined Barring (ODB)	N4	TTC	JP-3GA-23.015
TS	23.016	Subscriber data management; Stage 2	N4	TTC	JP-3GA-23.016
TS	23.018	Basic call handling – technical realization	N4	TTC	JP-3GA-23.018
TS	23.032	Universal Geographical Area Description (GAD)	S2	TTC	JP-3GA-23.032
TS	23.034	High-Speed Circuit-Switched Data (HSCSD); Stage 2	N1	TTC	JP-3GA-23.034
TS	23.038	Alphabets & language	T2	ARIB	ARIB STD-T63-23.038
TR	23.039	Interface protocols for the connection of Short Message Service Centers (SMSCs) to Short Message Entities (SMEs)	T2	ARIB	ARIB TR-T12-23.039
TS	23.040	Technical realization of short message service	T2	ARIB	ARIB STD-T63-23.040
TS	23.041	Technical realization of cell broadcast service	T2	ARIB	ARIB STD-T63-23.041
TS	23.042	Compression algorithm for SMS	T2	ARIB	ARIB STD-T63-23.042
TS	23.054	Shared interworking functions; Stage 2	N3	TTC	23.054
TS	23.057	Mobile station application Execution Environment (MExE)	T2	ARIB	ARIB STD-T63-23.057

(continued overleaf)

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	23.060	General Packet Radio Service (GPRS) service description; Stage 2	\$2	TTC	JP-3GA-23.060
TS	23.066	Support of GSM Mobile Number Portability (MNP); Stage 2	N4	TTC	JP-3GA-23.066
TS	23.067	Enhanced MultiLevel Precedence and Preemption (EMLPP) service; Stage 2	N4	TTC	JP-3GA-23.067
TS	23.072	Call deflection supplementary service; Stage 2	N4	TTC	JP-3GA-23.072
TS	23.073	Support of Localized Service Area (SoLSA); Stage 2	N4	TTC	JP-3GA-23.073
TS	23.078	CAMEL; Stage 2	N2	TTC	JP-3GA-23.078
ТS	23.079	Support of optimal routing; Phase 1; Stage 2	N4	TTC	JP-3GA-23.079
TS	23.081	Line identification supplementary services; Stage 2	N4	TTC	JP-3GA-23.081
TS	23.082	Call Forwarding (CF) supplementary services; Stage 2	N4	TTC	JP-3GA-23.082
TS	23.083	Call Waiting (CW) and call Hold (HOLD) supplementary service; Stage 2	N4	TTC	JP-3GA-23.083
TS	23.084	MultiParTY (MPTY) supplementary service; Stage 2	N4	TTC	JP-3GA-23.084
TS	23.085	Closed User Group (CUG) supplementary service; Stage 2	N4	TTC	JP-3GA-23.085
TS	23.086	Advice of Charge (AoC) supplementary service; Stage 2	N4	TTC	JP-3GA-23.086
TS	23.087	User-to-User Signaling (UUS); Stage 2	N4	TTC	JP-3GA-23.087

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	23.088	Call Barring (CB) supplementary service; Stage 2	N4	TTC	JP-3GA-23.088
TS	23.090	Unstructured Supplementary Service Data (USSD); Stage 2	N4	TTC	JP-3GA-23.090
TS	23.091	Explicit Call Transfer (ECT) supplementary service; Stage 2	N4	TTC	JP-3GA-23.091
TS	23.093	Call Completion to Busy Subscriber (CCBS); Stage 2	N4	TTC	JP-3GA-23.093
TS	23.094	Follow me; Stage 2	N4	TTC	TD-3GA-23.093
TS	23.096	Name identification supplementary service; Stage 2	N4	TTC	JP-3GA-23.096
TS	23.097	Multiple Subscriber Profile (MSP); Stage 2	N4	TTC	JP-3GA-23.097
TS	23.101	General UMTS architecture	S2	TTC	JP-3GA-23.101
TS	23.107	Quality of service, concept and architecture	S2	ARIB	ARIB STD-T63-23.107
TS	23.108	Mobile radio interface Layer 3 specification core network protocols; Stage 2 (structured procedures)	N1	TTC	JP-3GA-23.108
TS	23.110	UMTS access stratum services and functions	S2	TTC	JP-3GA-23.110
TS	23.116	Super charger; Stage 2	N4	TTC	JP-3GA-23.116
TS	23.119	Gateway Location Register (GLR); Stage2	N4	TTC	JP-3GA-23.119
TS	23.121	Architecture requirements for Release 99	S 2	TTC	JP-3GA-23.121
TS	23.122	Non-access-stratum functions related to Mobile Station (MS) in idle mode	N1	TTC	JP-3GA-23.122

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	23.127	Virtual home environment; Stage 2	S 2	TTC	JP-3GA-23.127
TS	23.135	Multicall; Stage 2	N4	TTC	JP-3GA-23.135
TS	23.140	Multimedia Messaging Service (MMS)	T2	ARIB	ARIB STD-T63-23.140
TS	23.153	Out of band transcoder control; Stage 2	N4	TTC	TD-3GA-23.153
TS	23.171	Functional Stage 2 description of location services in UMTS	S2	ARIB	ARIB STD-T63-23.171
TR	23.814	Separating RR- and MM-specific parts of the MS classmark	N1	TTC	TD-3GA-23.814
TR	23.908	Technical report on prepaging	N4	TTC	TD-3GA-23.908
TR	23.909	Technical report on the gateway location register	N4	TTC	TD-3GA-23.909
TR	23.910	Circuit-switched data bearer services	N3	TTC	TD-3GA-23.910
TR	23.911	Technical report on out-of-band transcoder control	N4	TTC	TD-3GA-23.911
TR	23.912	Technical report on super-charger	N4	TTC	TD-3GA-23.912
TR	23.922	Architecture for an all IP network	S2	TTC	Not published
TR	23.923	Combined GSM and mobile IP mobility handling in UMTS IP CN	S2	TTC	TD-3GA-23.923
TR	23.925	UMTS core network-based ATM transport	S 2	TTC	TD-3GA-23.925
TR	23.930	Iu principles	S2	TTC	TD-3GA-23.930
TR	23.972	Circuit-switched multimedia telephony	N1	TTC	TD-3GA-23.972
TS	24.002	GSM-UMTS Public Land Mobile Network (PLMN) access reference configuration	N1	TTC	JP-3GA-24.002

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	24.007	Mobile radio interface signaling Layer 3-general aspects	N1	TTC	JP-3GA-24.007
TS	24.008	Mobile radio interface Layer 3 specification; core network protocols; Stage 3	N1	TTC	JP-3GA-24.008
TS	24.010	Mobile radio interface Layer 3-supplementary services specification – general aspects	N4	TTC	JP-3GA-24.010
TS	24.011	Point-to-Point (PP) Short Message Service (SMS) support on mobile radio interface	N1,T2	ARIB	ARIB STD-T63-24.011
TS	24.012	Short Message Service Cell Broadcast (SMSCB) support on the mobile radio interface	N4,T2	ARIB	ARIB STD-T63-24.012
TS	24.022	Radio Link Protocol (RLP) for data and telematic services on the MS-BSS interface and the Base Station System–Mobile- services Switching Center (BSS-MSC) interface	N3	TTC	Not published
TS	24.030	LoCation Services (LCS); Stage 3 SS (MO-LR)	N4	TTC	JP-3GA-24.030
TS	24.067	enhanced MultiLevel Precedence and Preemption service (eMLPP); Stage 3	N4	TTC	JP-3GA-24.067
TS	24.072	Call deflection supplementary service; Stage 3	N4	TTC	JP-3GA-24.072

391

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	24.080	Mobile radio Layer 3 supplementary service specification – formats and coding	N4	TTC	JP-3GA-24.080
TS	24.081	Line identification supplementary service; Stage 3	N4	TTC	JP-3GA-24.081
TS	24.082	Call forwarding supplementary service; Stage 3	N4	TTC	JP-3GA-24.082
TS	24.083	Call Waiting (CW) and call Hold (HOLD) supplementary service; Stage 3	N4	TTC	JP-3GA-24.083
TS	24.084	MultiParTY (MPTY) supplementary service; Stage 3	N4	TTC	JP-3GA-24.084
TS	24.085	Closed User Group (CUG) supplementary service; Stage 3	N4	TTC	JP-3GA-24.085
TS	24.086	Advice of Charge (AoC) supplementary service; Stage 3	N4	TTC	JP-3GA-24.086
TS	24.087	User-to-User Signaling (UUS); Stage 3	N4	TTC	JP-3GA-24.087
TS	24.088	Call Barring (CB) supplementary service; Stage 3	N4	TTC	JP-3GA-24.088
TS	24.090	Unstructured Supplementary Service Data (USSD); Stage 3	N4	TTC	JP-3GA-24.090
TS	24.091	Explicit Call Transfer (ECT) supplementary service; Stage 3	N4	TTC	JP-3GA-24.091
TS	24.093	Call Completion to Busy Subscriber (CCBS); Stage 3	N4	TTC	JP-3GA-24.093
TS	24.096	Name identification supplementary service; Stage 3	N4	TTC	JP-3GA-24.096
TS	24.135	Multicall; Stage 3	N4	TTC	JP-3GA-24.135
TS	25.101	UE radio transmission and reception (FDD)	R4	ARIB	ARIB STD-T63-25.101

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	25.102	UE radio transmission and reception (TDD)	R4	ARIB	(ARIB STD-T65-25.102)
TS	25.104	UTRA (BS) FDD: radio transmission and reception	R4	ARIB	ARIB STD-T63-25.104
TS	25.105	UTRA (BS) TDD: radio transmission and reception	R4	ARIB	(ARIB STD-T65-25.105)
TS	25.113	Base station EMC	R4	ARIB	ARIB STD-T63-25.113
TS	25.123	Requirements for support of radio resource management (TDD)	R4	ARIB	(ARIB STD-T65-25.123)
TS	25.133	Requirements for support of radio resource management (FDD)	R4	ARIB	ARIB STD-T63-25.133
TS	25.141	Base station conformance testing (FDD)	R4	ARIB	ARIB STD-T63-25.141
TS	25.142	Base station conformance testing (TDD)	R4	ARIB	(ARIB STD-T65-25.142)
TS	25.201	Physical layer – general description	R1	ARIB	ARIB STD-T63-25.201
TS	25.211	Physical channels and mapping of transport channels onto physical channels (FDD)	R1	ARIB	ARIB STD-T63-25.211
TS	25.212	Multiplexing and channel coding (FDD)	R1	ARIB	ARIB STD-T63-25.212
TS	25.213	Spreading and modulation (FDD)	R1	ARIB	ARIB STD-T63-25.213
TS	25.214	Physical layer procedures (FDD)	R1	ARIB	ARIB STD-T63-25.214
TS	25.215	Physical layer; measurements (FDD)	R1	ARIB	ARIB STD-T63-25.215
TS	25.221	Physical channels and mapping of transport channels onto physical channels (TDD)	R1	ARIB	(ARIB STD-T65-25.221)
TS	25.222	Multiplexing and channel coding (TDD)	R1	ARIB	(ARIB STD-T65-25.222)

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	25.223	Spreading and modulation (TDD)	R1	ARIB	(ARIB STD-T65-25.223)
TS	25.224	Physical layer procedures (TDD)	R1	ARIB	(ARIB STD-T65-25.224)
ГS	25.225	Physical layer; measurements (TDD)	R1	ARIB	(ARIB STD-T65-25.225)
ГS	25.301	Radio interface protocol architecture	R2	ARIB	ARIB STD-T63-25.301
ГS	25.302	Services provided by the physical layer	R2	ARIB	ARIB STD-T63-25.302
ГS	25.303	UE functions and interlayer procedures in connected mode	R2	ARIB	ARIB STD-T63-25.303
ГS	25.304	UE procedures in idle Mode and procedures for cell reselection in connected mode	R2	ARIB	ARIB STD-T63-25.304
ΓS	25.305	Stage 2 functional specification of LoCation Services (LCS) in UTRAN	R2	ARIB	ARIB STD-T63-25.305
ГS	25.321	Medium Access Control (MAC) protocol specification	R2	ARIB	ARIB STD-T63-25.321
ГS	25.322	Radio Link Control (RLC) protocol specification	R2	TTC	JP-3GA-25.322
ГS	25.323	Packet Data Convergence Protocol (PDCP)	R2	TTC	JP-3GA-25.323
ΓS	25.324	Radio interface for broadcast/multicast services	R2	TTC	JP-3GA-25.324
ГS	25.331	Radio Resource Control (RRC) protocol specification	R2	TTC	JP-3GA-25.331
TS	25.401	UTRAN overall description	R3	ARIB	ARIB STD-T63-25.401
ГS	25.402	Synchronization in UTRAN; Stage 2	R3	ARIB	ARIB STD-T63-25.402
ГS	25.410	UTRAN Iu interface: general aspects and principles	R3	TTC	JP-3GA-25.410

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	25.411	UTRAN Iu interface Layer 1	R3	TTC	JP-3GA-25.411
TS	25.412	UTRAN Iu interface signaling transport	R3	TTC	JP-3GA-25.412
TS	25.413	UTRAN Iu interface RANAP signaling	R3	TTC	JP-3GA-25.413
TS	25.414	UTRAN Iu interface data transport & transport signaling	R3	TTC	JP-3GA-25.414
TS	25.415	UTRAN Iu interface user plane protocols	R3	TTC	JP-3GA-25.415
TS	25.419	UTRAN Iu interface: cell broadcast protocols between SMS-CBC and RNC	R3	TTC	JP-3GA-25.419
TS	25.420	UTRAN Iur interface: general aspects and principles	R3	ARIB	ARIB STD-T63-25.420
TS	25.421	UTRAN Iur interface Layer 1	R3	ARIB	ARIB STD-T63-25.421
TS	25.422	UTRAN Iur interface signaling transport	R3	ARIB	ARIB STD-T63-25.422
TS	25.423	UTRAN Iur interface RNSAP signaling	R3	ARIB	ARIB STD-T63-25.423
TS	25.424	UTRAN Iur interface data transport & transport signaling for CCH data streams	R3	ARIB	ARIB STD-T63-25.424
TS	25.425	UTRAN Iur interface user plane protocols for CCH data streams	R3	ARIB	ARIB STD-T63-25.425
TS	25.426	UTRAN Iur and Iub interface data transport & transport signaling for DCH data streams	R3	ARIB	ARIB STD-T63-25.426
TS	25.427	UTRAN Iur and Iub interface user plane protocols for DCH data streams	R3	ARIB	ARIB STD-T63-25.427
TS	25.430	UTRAN lub interface: general aspects and principles	R3	ARIB	ARIB STD-T63-25.430

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	25.431	UTRAN lub interface Layer 1	R3	ARIB	ARIB STD-T63-25.431
TS	25.432	UTRAN Iub interface signaling transport	R3	ARIB	ARIB STD-T63-25.432
TS	25.433	UTRAN Iub interface NBAP signaling	R3	ARIB	ARIB STD-T63-25.433
TS	25.434	UTRAN lub interface data transport & transport signaling for CCH data streams	R3	ARIB	ARIB STD-T63-25.434
TS	25.435	UTRAN Iub interface user plane protocols for CCH data streams	R3	ARIB	ARIB STD-T63-25.435
TS	25.442	UTRAN implementation specific O&M transport	R3	ARIB	ARIB STD-T63-25.442
TR	25.831	Study items for future release	R3	TTC	Not published
TR	25.832	Manifestations of handover and SRNS relocation	R3	TTC	TD-3GA-25.832
TR	25.833	Physical layer items not for inclusion in Release 99	R1	ARIB	ARIB TR-T12-25.833
TR	25.921	Guidelines and principles for protocol description and error handling	R2	TTC	TD-3GA-25.921
TR	25.922	Radio resource management strategies	R2	TTC	TD-3GA-25.922
TR	25.925	Radio interface for broadcast/multicast services	R2	TTC	TD-3GA-25.925
TR	25.926	UE radio access capabilities definition	R2	ARIB	ARIB TR-T12-25.926
TR	25.931	UTRAN functions, examples on signaling procedures	R3	TTC	TD-3GA-25.931
TR	25.941	Document structure	R4	ARIB	ARIB TR-T12-25.941
TR	25.942	RF system scenarios	R4	ARIB	ARIB TR-T12-25.942
TR TR	25.943 25.944	Deployment aspects Channel coding and multiplexing examples	R4 R1	ARIB ARIB	ARIB TR-T12-25.943 ARIB TR-T12-25.944

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TR	25.990	Vocabulary for UTRAN	R4	ARIB	ARIB TR-T12-25.990
TS	26.071	AMR speech CODEC; general description	S4	ARIB	ARIB STD-T63-26.071
TS	26.073	AMR speech CODEC; C-source code	S4	ARIB	ARIB STD-T63-26.073
TS	26.074	AMR speech CODEC; test sequences	S4	ARIB	ARIB STD-T63-26.074
TS	26.090	AMR speech CODEC; transcoding functions	S4	ARIB	ARIB STD-T63-26.090
TS	26.091	AMR speech CODEC; error concealment of lost frames	S4	ARIB	ARIB STD-T63-26.091
TS	26.092	AMR speech CODEC; comfort noise for AMR speech traffic channels	S4	ARIB	ARIB STD-T63-26.092
TS	26.093	AMR speech CODEC; source-controlled rate operation	S4	ARIB	ARIB STD-T63-26.093
TS	26.094	AMR speech CODEC; voice activity detector for AMR speech traffic channels	S4	ARIB	ARIB STD-T63-26.094
TS	26.101	AMR speech CODEC; frame structure	S4	ARIB	ARIB STD-T63-26.101
TS	26.102	AMR speech CODEC; interface to Iu and Uu	S4	ARIB	ARIB STD-T63-26.102
TS	26.103	CODEC lists	S4	ARIB	ARIB STD-T63-26.103
TS	26.104	AMR speech CODEC; floating point C-code	S4	ARIB	ARIB STD-T63-26.104
TS	26.110	CODEC for circuit-switched multimedia telephony service; general description	S4	ARIB	ARIB STD-T63-26.110
TS	26.111	CODEC for circuit-switched multimedia telephony service; modifications to H.324	S4	ARIB	ARIB STD-T63-26.111

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	26.131	Narrowband (3,1 kHz) speech & video telephony terminal acoustic characteristics	S 4	ARIB	ARIB STD-T63-26.131
TS	26.132	Narrowband (3,1 kHz) speech & video telephony terminal acoustic test specification	S4	ARIB	ARIB STD-T63-26.132
TR	26.911	CODEC for circuit-switched multimedia telephony service; terminal implementor's guide	S4	ARIB	ARIB TR-T12-26.911
TR	26.912	CODEC for circuit-switched multimedia telephony service; quantitative performance evaluation of H.324 Annex C over 3G	S4	ARIB	ARIB TR-T12-26.912
TR	26.913	Quantitative performance evaluation of real-time packet-switched multimedia services over 3G	S4	ARIB	ARIB TR-T12-26.913
TR	26.915	QoS for speech and multimedia CODEC; quantitative performance evaluation of real-time packet-switched multimedia services over 3G	S4	ARIB	ARIB TR-T12-26.915
TR	26.975	Performance characterization of the AMR speech CODEC	S4	ARIB	ARIB TR-T12-26.975
TS	27.001	General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)	N3	TTC	JP-3GA-27.001

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	27.002	Terminal Adaptation Functions (TAF) for services using asynchronous bearer capabilities	N3	TTC	JP-3GA-27.002
TS	27.003	Terminal Adaptation Functions (TAF) for services using synchronous bearer capabilities	N3	TTC	JP-3GA-27.003
TS	27.005	Use of Data Terminal Equipment–Data Circuit terminating Equipment (DTE-DCE) interface for Short Message Service (SMS) and Cell Broadcast Service (CBS)	Τ2	ARIB	ARIB STD-T63-27.005
TS	27.007	AT command set for 3G User Equipment (UE)	T2	ARIB	ARIB STD-T63-27.007
TS	27.010	Terminal Equipment to User Equipment (TE-UE) multiplexer protocol User Equipment (UE)	T2	ARIB	ARIB STD-T63-27.010
TS	27.060	GPRS mobile stations supporting GPRS	N3	TTC	JP-3GA-27.06
TS	27.103	Wide area network synchronization	T2	ARIB	ARIB STD-T63-27.103
TR	27.901	Report on terminal interfaces – an overview	T2	ARIB	ARIB TR-T12-27.901
TR	27.903	Discussion of synchronization standards	T2	ARIB	ARIB TR-T12-27.903
TR	27.A01	Report on external interfaces connector		ARIB	ARIB TR-T12-27.A01
TR	27.A02	MA-TA interface description		ARIB	ARIB TR-T12-27.A02
TS	29.002	Mobile Application Part (MAP)	N4	TTC	JP-3GA-29.002

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	29.007	General requirements on interworking between the PLMN and the ISDN or PSTN	N3	TTC	JP-3GA-29.007
TS	29.010	Information element mapping between Mobile Station-Base Station System (MS-BSS) and Base Station System-Mobile- services Switching Center (BSS-MCS) signaling procedures and the Mobile Application Part (MAP)	N4	TTC	JP-3GA-29.010
TS	29.011	Signaling interworking for supplementary services	N4	TTC	JP-3GA-29.011
TS	29.013	Signaling interworking between ISDN supplementary services Application Service Element (ASE) and Mobile Application Part (MAP) protocols	N4	TTC	JP-3GA-29.013
TS	29.016	Serving GPRS Support Node (SGSN) – Visitors Location Register (VLR); Gs interface network service specification	N1	TTC	JP-3GA-29.016
TS	29.018	Serving GPRS Support Node (SGSN) – Visitors Location Register (VLR); Gs interface Layer 3 specification	N1	TTC	JP-3GA-29.018

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	29.060	GPRS Tunneling Protocol (GTP) across the Gn and Gp interface	N4	TTC	JP-3GA-29.060
TS	29.061	General Packet Radio Service (GPRS); interworking between the Public Land Mobile Network (PLMN) supporting GPRS and packet	N3	TTC	JP-3GA-29.061
TS	29.078	CAMEL; Stage 3	N2	TTC	JP-3GA-29.078
TS	29.108	Application of the Radio Access Network Application Part (RANAP) on the E interface	R3	TTC	JP-3GA-29.108
TS	29.119	GPRS Tunneling Protocol (GTP) specification for Gateway Location Register (GLR)	N4	TTC	JP-3GA-29.119
TS	29.120	Mobile Application Part (MAP) specification for Gateway Location Register (GLR); Stage 3	N4	TTC	JP-3GA-29.120
TS	29.198	Open services architecture API; Part 1	N5	TTC	JP-3GA-29.198
TR	29.998	Open services architecture API; Part 2	N5	TTC	TD-3GA-29.998
TS	30.504	Work plan and study items – RAN WG4	R4	ARIB	ARIB STD T63-30.504
TS	30.531	Work plan and study items-RAN WG3	R3	TTC	Not published
TS	31.101	UICC-terminal interface; physical and logical characteristics	Т3	ARIB	ARIB STD-T63-31.101
TS	31.102	Characteristics of the USIM application	Т3	ARIB	ARIB STD-T63-31.102

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	31.110	Numbering system for telecommunication IC card applications	Т3	ARIB	ARIB STD-T63-31.110
TS	31.111	USIM Application Tool kit (USAT)	T3	ARIB	ARIB STD-T63-31.111
TS	31.120	Terminal tests for the UICC interface; Part 1	Т3	ARIB	ARIB STD-T63-31.120
TS	31.121	Terminal tests for the UICC interface; Part 2	T3	ARIB	ARIB STD-T63-31.121
TS	31.122	UICC test specification	T3	ARIB	ARIB STD-T63-31.122
TS	32.005	Telecommunications management; charging and billing; 3G call and event data for the Circuit-Switched (CS) domain	85	TTC	JP-3GA-32.005
TS	32.015	Telecommunications management; charging and billing; 3G call and event data for the Packet-Switched (PS) domain	S5	TTC	JP-3GA-32.015
TS	32.101	3G telecom management principles and high-level requirements	S5	TTC	JP-3GA-32.101
TS	32.102	3G telecom management architecture	S5	TTC	JP-3GA-32.102
TS	32.104	3G performance management	S5	TTC	JP-3GA-32.104
TS	32.105	Charging & billing; GSM call and event data for the Circuit-Switched (CS) domain	S5	TTC	TD-3GA-32.105
TS	32.106-1	Telecommunication management; configuration management; Part 1: 3G configuration management; concept and requirements	S5	TTC	JP-3GA-32.106-1

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	32.106-2	Telecommunication management; configuration management; Part 2: notification integration reference point; information service version 1	S5	TTC	JP-3GA-32.106-2
TS	32.106-3	Telecommunication management; configuration management; Part 3: notification integration reference point; CORBA solution set version 1:1	85	TTC	JP-3GA-32.106-3
TS	32.106-4	Telecommunication management; configuration management; Part 4: notification integration reference point: CMIP solution set version 1:1	S5	TTC	JP-3GA-32.106-4
TS	32.106-5	Telecommunication management; configuration management; Part 5: basic configuration management IRP information model (including NRM) version 1	85	TTC	JP-3GA-32.106-5
TS	32.106-6		\$5	TTC	JP-3GA-32.106-6

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	32.106-7	Telecommunication management; configuration management; Part 7: basic configuration management IRP CMIP solution set version 1:1	S5	TTC	JP-3GA-32.106-7
TS	32.106-8	Telecommunication management; configuration management; Part 8: name convention for managed objects	S5	TTC	JP-3GA-32.106-8
TS	32.111-1	Telecommunication management; fault management; Part 1: 3G fault-management requirements	S5	TTC	JP-3GA-32.111-1
TS	32.111-2	Telecommunication management; fault management; Part 2: alarm integration reference point: information service	S5	TTC	JP-3GA-32.111-2
TS	32.111-3	Telecommunication management; fault management; Part 3: alarm integration reference point: CORBA solution set version 1:1	\$5	TTC	JP-3GA-32.111-3
TS	32.111-4	Telecommunication management; fault management; Part 4: alarm integration reference point: CMIP solution set	S5	TTC	JP-3GA-32.111-4
TS	33.102	Security architecture	S 3	ARIB	ARIB STD-T63-33.102
TS	33.103	Security integration guidelines	S 3	ARIB	ARIB STD-T63-33.103

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	33.105	Cryptographic algorithm requirements	S 3	ARIB	ARIB STD-T63-33.105
TS	33.106	Lawful interception requirements	S 3	ARIB	ARIB STD-T63-33.106
TS	33.107	Lawful interception architecture and functions	\$3	ARIB	ARIB STD-T63-33.107
TS	33.120	Security objectives and principles	S 3	ARIB	ARIB STD-T63-33.120
TR	33.900	Guide to 3G security	S 3	ARIB	ARIB TR-T12-33.900
TR	33.901	Criteria for cryptographic algorithm design process	S3	ARIB	ARIB TR-T12-33.901
TR	33.902	Formal analysis of the 3G authentication protocol	S 3	ARIB	ARIB TR-T12-33.902
TR	33.908	Security Algorithms Group of Experts (SAGE); general report on the design, specification and evaluation of 3GPP standard confidentiality and integrity algorithms	\$3	ARIB	ARIB TR-T12-33.908
TR	33.909	ETSI SAGE 3GPP standards algorithms task force: report on the evaluation of 3GPP standard confidentiality and integrity algorithms	83	ARIB	Not published
TS	34.108	Common test environments for User Equipment (UE) conformance testing	T1	ARIB	ARIB STD-T63-34.108
TS	34.109	Logical test interface (TDD and FDD)	R2	ARIB	ARIB STD-T63-34.109
TS	34.121	Terminal conformance specification, radio transmission and reception (FDD)	T1	ARIB	ARIB STD-T63-34.121

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
ГS	34.122	Terminal conformance specification, radio transmission and reception (TDD)	T1	ARIB	ARIB STD-T63-34.122
ГS	34.123-1	UE conformance specification, Part 1–conformance specification	T1	ARIB	ARIB STD-T63-34.123-1
ГS	34.123-2	UE conformance specification, Part 2–ICS	T1	ARIB	ARIB STD-T63-34.123-2
TS	34.123-3	UE conformance specification, Part 3-abstract test suites	T1	ARIB	ARIB STD-T63-34.123-3
ΓS	34.124	ElectroMagnetic Compatibility (EMC) requirements for mobile terminals and ancillary equipment	T1	ARIB	ARIB STD-T63-34.124
ΓR	34.907	Report on electrical safety requirements and regulations	Τ2	ARIB	ARIB TR-T12-34.907
ΓR	34.925	Specific Absorption Rate (SAR) requirements and regulations in different regions	Τ2	ARIB	ARIB TR-T12-34.925
ΓS	35.201	Specification of the 3GPP confidentiality and integrity algorithms; Document 1: f8 and f9 specifications	\$3	ARIB	ARIB STD-T63-35.201
TS	35.202	Specification of the 3GPP confidentiality and integrity algorithms; Document 2: Kasumi algorithm specification	\$3	ARIB	ARIB STD-T63-35.202

TS/TR	3GPP spec. number	Technical specifications/ reports (TS/TR) title	3GPP TSG/WG	ARIB/TTC	ARIB/TTC standard number
TS	35.203	Specification of the 3GPP confidentiality and integrity algorithms; Document 3: implementors' test data	S3	ARIB	ARIB STD-T63-35.203
TS	35.204	Specification of the 3GPP confidentiality and integrity algorithms; Document 4: design conformance test data	\$3	ARIB	ARIB STD-T63-35.204

W-CDMA: Mobile Communications System. Edited by Keiji Tachikawa Copyright © 2002 John Wiley & Sons, Ltd. ISBN: 0-470-84761-1

Index

3.1kHz audio Multimedia, 242, 244 3G-324M, 321 3G-H.324/M, 224, 242 3GPP, 11, 15, 17, 215, 224, 258, 338, 343, 372 3GPP2, 15, 215, 338 AAL, 219 AAL type 2, 226 AAL1, 219 AAL2, 219 AAL2 Signaling Protocol CS.1, 226 AAL3/4, 219 AAL5, 219, 220, 226 ABR, 222 Access Service Class, 117, 118, 128 Access Service Class Access Slot Access Slots, 96, 117-119, 128 Access Stratum, 153 Acknowledge Mode, 132, 134, 135, 136, 138-141 Acquisition, 25 Acquisition Acquisition Indicator, 108 Adaptive Antenna Array (AAA) Diversity, 71 Adaptive Intra-Refresh, 313 Adaptive Intra-Refresh (AIR) Adjacent Channel Selectivity (ACS), 197 Admission Control. 146 Advanced Billing Scheme, 274 AI, 101 All IP, 373 Alternative Scrambling Code, 112

AMC, 369 AMD PDU, 136, 138-140 AMP, 7, 185 AMPlifier (AMP), 27 AMPS, 1, 3, 9, 185 AMR, 224, 317 AMR-WB, 319, 376 Analog Cellular System, 1 ANSI, 13, 15, 349 Antenna Verification, 64, 123 APN, 247, 251, 252 ARIB, 16, 17, 18, 209 AS, 145, 153 ASC, 42, 117, 118, 128 Association of Radio Industries and Businesses (ARIB), 4, 84, 317 ATM, 182, 186, 187, 216–223, 226, 248-249, 287, 291 ATM Adaptation Layer, 186, 219 ATM Adaptation Layer type 2, 226 Attach, 11, 227, 228, 347 Attachment/Detachment, 227 Augmented Reality Technology, 354 Augmented Reality Technology Automatic Congestion Control Algorithm, 291 Automatic Repeat reQuest (ARQ), 42, 332

Base Station (BS), 6, 283, 347 Base station Control Equipment (BCE), 7 Base Transceiver Station (BTS), 85, 296 Base Transceiver Station (BTS) Basic Call State Model, 254 Battery Saving, 201 BCSM, 254, 255, 257 Bearer Capability Information Element, 242 Bearer Independent Call Control Protocol, 237 BICC, 237, 238 B-ISUP, 159, 226 Bluetooth, 194, 208 BMC, 86 Browser, 8, 134, 206, 337 Browser Software, 8 BS, 4, 23, 85, 283, 347, 368 C Plane, 86, 238 CA, 357-360, 373 CAC, 223, 291 Call Termination Notice, 247 CAMEL, 226, 259 CAP, 226 Capability Negotiation, 323, 345 Capacity, 2, 25, 82, 222, 283, 336, 371 Carrier Frequency Calibration, 72–74 CBR, 222-223 CC, 14, 86, 186, 224 CDC, 336 cdma2000, 15, 18 CDPD, 9, 243 Cell, 25, 82, 85, 218, 345, 370 Cell Breathing, 174, 181 Cell Search, 25, 31–35, 87 CELL_DCH, 154 CELL_FACH, 154 CELL_PCH, 154 Cellular System, 2, 27, 367 CELP, 313, 314, 316, 317 CEPT, 3 Certificate, 357, 359, 360 Certificate Certification Authority, 357, 373 Channel Coding, 57, 103, 104, 106, 121, 196 Channel Estimation, 24, 92 Channel Estimation Filter, 51, 64 Channelization Code, 29, 88

Channelization code Chip Rate, 13, 18, 21, 108, 367 CIF, 310, 312 Ciphering Key, 140, 143, 239 Circuit-switching Function, 14 CLDC, 336-338 Closed-loop Transmission Diversity, 62, 63, 65, 121-123 CN, 14-15, 85-86, 145, 182, 188, 215-218, 220-225, 227, 231-232, 238, 240, 248, 250, 252, 258, 264-269, 273-274, 283, 284, 285, 371, 373, 377 CN Strata, 285, 287 Code Excited Linear Prediction, 5, 313 Code Excited Linear Prediction (CELP) Codec, 7, 194, 236-238, 313, 317, 321, 322, 365, 367, 374–377 Codec Through (Codec Bypass), 236 Common Intermediate Format, 310 Common Part Sublayer Packet, 220 Common Platform, 1, 270, 307 Communications Industry Association of Japan (CIAJ) (TIA), 3 Compact HTML (CHTML), 333 Compressed Mode, 96–97, 111–112, 116, 125-126 Compressed mode Conference of European Postal and Telecommunications Administration (CEPT), 3 Congestion Pattern, 290 Connection Admission Control (CAC), 223, 291 Content, 8-10, 86, 203, 207, 217, 267, 307, 326, 329, 331, 348-349 Contract Parameter Monitor and Control(UPC), 223 Convolutional Code, 213 **CORBA**, 187 Core Network (CN), 283, 371 CPS Packet, 220 CSS, 341 Cyclic Redundancy Check (CRC), 26, 129

Data Partition, 312 Data Partition Detach, 227-228 Detection Point, 254 Device, 182, 188, 198, 200, 203, 208-209, 268, 332, 334, 336, 337, 347 Diffserv, 372 Digital Cellular System, 3 Direct Spread, 13, 18, 81 Discontinuous Transmission (DTX), 46, 108, 316 Discrete Cosine Transform (DCT), 308 Diversity Hand Over (DHO), 172 **DLP** Consortium, 348 DoPa, 8 DoPa Download Method, 329-330, 333 DP, 254, 257 DS-CDMA, 18, 21, 26–28, 49, 91, 113

E.164, 15, 230 ECMA, 338 EIA, 3 Electro Luminescence, 205 Electronic Authentication, 307, 356 - 359Electronic Industries Association (EIA), 3 Electronic Signature, 356–358, 360-361 Electronic Wallet, 208 Electronic Watermark, 332 End-to-end Security Model, 357–358 Equivalent Traffic, 171 Error Control, 42–43, 332 ETSI, 3, 12, 16–18, 84–85, 226, 317, 320, 338, 340, 349 European Telecommunications Standards Institute (ETSI), 3, 84, 226, 317

Family Concept, 14 Family Member, 15 Fast Cell Selection (FCS), 370 Fast TPC, 45, 47, 53–55, 57–59, 64-66, 75, 91, 113, 178, 370 FBI, 31, 62–65, 95–96, 115–116, 121 - 123FDD, 13, 16–17, 72, 92, 156, 183, 366 - 368FDMA, 13, 21, 82-83, 169, 180 Feed Forward, 7, 185 FPLMTS, 11–12, 16 G.711, 236 gain factor, 107 Gateway, 5-6, 9, 14, 215, 217, 226, 247, 264–274, 292, 294, 341, 344, 353, 357–360, 372–373 Gateway Location Register (GLR), 5–6 Gateway Mobile Switching Center (G-MSC), 5 Geometry, 173-174, 176-177, 180 GGSN, 216, 226, 247, 249, 252–254, 265 - 268Global, 1, 10, 14, 17, 18, 28, 81, 146, 215, 224, 233, 309 Global Roaming, 216, 226, 250 GLR, 5–6, 226, 233, 235 GML, 350-351 GMM, 225 G-MSC, 5, 14 GMSK, 5 Gold Sequence, 24–25, 111–112 GPRS, 9, 154, 215–216, 225–226, 243, 247-248, 250, 338, 340, 373 GPS, 28, 347-348 GSM, 9, 13, 15, 146–147, 154, 202, 215, 224-227, 232, 236, 238-240, 254, 258, 294, 317, 319, 340, 376 GTP, 225–226, 250 G-XML, 350-351 H.223, 321-322, 325 H.245, 321, 323, 325 H.261, 309-311 H.263, 309, 311-312, 321-322

H.323, 320, 372, 374

H.324, 224, 242, 320, 322, 324

HFN, 140 HLR, 5-6, 14, 226-228, 230, 233-235, 246-247, 253-257, 260, 291-292, 305 HLR Congestion, 294 Home Location Register (HLR), 5-6, 226, 291 HSDPA, 368, 370 HTML, 8-9, 206, 333-335, 338, 341, 349, 353 HTTP, 266, 327, 330, 333, 337-338, 341, 342, 344, 349 Hybrid ARQ 43-44 Hybrid ARQ (HARQ) Hybrid Coding, 313 Hybrid Phase Shift Keying (HPSK), 31, 85 i-appli, 337-338 IEEE802.11b, 208-209 IETF, 269, 319-320, 331, 338, 340, 342, 344, 349, 359, 372, 377 i-mode, 8-9, 194, 204, 206, 210, 216, 245-246, 265, 266, 290, 295, 327-328, 334, 337-338, 341 *i-mode* Java, 337, 338 *i-mode* Server, 9 IMS, 373 IMSI, 153, 228, 231, 252, 253 IMT-2000, 1, 10, 11, 12, 14, 15, 16, 17, 18, 19, 81, 189, 190, 191, 192, 202, 203, 209, 210, 215, 217, 219, 220, 221, 223, 224, 225, 226, 227, 230, 231, 232, 233, 236, 238, 239, 240, 242, 246, 247, 250, 251, 265, 266, 268, 270, 277, 280, 283, 289, 294, 296, 303, 307, 308, 312, 317, 319, 321, 324, 338, 340, 365, 366, 368, 377 IN, 216, 217, 226, 254, 255, 257, 258, 259, 296 IN Scheme, 254, 258 *i-navi link* Specifications, 351 Infra-red Data Association, 208 Infrared Interface, 207, 208 Integrity Protection, 148, 240 Intelligent Transport System (ITS), 217 Interference Canceller, 111

Interference from Other Users, 24, 51, 75.179 Interference Margin, 170, 171, 172, 174, 175, 176, 177 Interframe Motion Compensated Prediction, 308, 309, 312 Intermittent Reception, 32, 40, 41, 83, 87, 101, 154, 155, 198, 200, 201, 202Intermittent Reception Control, 198, 201 International Mobile Telecommunications-2000 (IMT-2000), 215, 277, 307, 365 Intserv, 372 IP Mobility, 371, 372, 373 **IP** Packet Communications, 370 IP-QoS, 371 IrDA, 208, 209, 337 IS-54, 3 IS-95, 4, 5, 9 ISM Band, 208 ISP, 8, 225, 247, 251, 252, 264, 267, 268, 270, 272, 273, 274, 307, 325, 326, 327 ISUP, 159, 226, 237 IT, 211 ITS, 11, 217 ITU, 10, 11, 12, 14, 15, 16, 17, 18, 19, 81, 178, 196, 202, 230, 277, 309, 311, 319, 320, 321, 323, 324, 365, 366 ITU-R, 11, 12, 15, 16, 17, 18, 19, 178, 196, 365 ITU-T, 11, 14, 15, 230, 277, 309, 311, 319, 320, 321, 323, 324 Iu, 85, 221, 254, 375

JAIN, 373 JAM, 337 Java, 9, 336, 337, 338 JDC, 4 Joint Detection, 368 J-TACS, 3

KVM, 336, 337

LAI, 230, 231 Language (SMIL), 335, 377 Large Capacity System, 2 Large City Car Phone System Length Indicator, 135, 136, 138, 139 Link Budget, 177, 178, 179, 181 Link Level Simulation, 170, 176, 177, 179 LISAP, 348 Location Information, 255, 274, 307, 345, 346, 347, 348, 349, 350, 353 Location Information Distribution and Management Platform, 350, 353 Location Information Distribution Method, 348, 349 Location Information Mutual Conversion Technology, 353 Location Information Processing Method, 345, 346 Location Information Service, 374 Location Information URL, 346 Location Information Use, 348 Location Interoperability Forum (LIF), 348 Location-oriented Information Filtering Engine, 353 Location-oriented Information Integration Technology, 353 Location-oriented Meta Search Engine, 353 Location-oriented Search Engine, 353 Long scrambling code, 111 MAC, 26, 45, 46, 86, 87, 92, 118, 126, 127, 128, 129, 130, 131, 133, 141, 142, 158, 240 MAC-I, 240 Macro Block, 308, 309 Macro Cell, 181 MAP, 43, 226, 236, 294 Maximal Ratio Combining (MRC), 75, 120 Maximum Ratio Combining (MRC), 51 MBS, 222 MCC, 7

MCR, 222

MDE, 43, 226, 236, 294 Measurement, 37, 39, 43, 47, 48, 49, 53, 54, 55, 56, 58, 59, 65, 75, 113, 115, 125, 126, 127, 147, 156, 157, 158, 196, 197, 323, 346, 347, 348, 369 Message Delivery, 345 MGW, 272, 273, 274 Micro Cell, 181 Midlet, 337 MIDP, 337, 338 MIL, 43, 55, 103 **MITF**, 338 MM, 86, 224, 225, 226, 227 Mobile Communications Control Center (MCC), 7 Mobile Internet, 7, 8, 11, 204, 205, 210, 307, 338 Mobile IP, 372, 373 Mobile Multi-link, 324, 325 Mobile Packet Tunneling Protocol, 27 Mobile Phone Type Mobile Station, 6, 23, 75, 76, 82, 83, 230, 246 Mobile Station Roaming Number (MSRN), 230 Mobility Specifications, 224, 225, 227 Mobility Management Control, 7, 183 Modulation and Demodulation Equipment (MDE), 7, 183 MP4 File Format, 333 MPEG-1, 309, 310, 311 MPEG-2, 309, 310, 311, 312 MPEG-4, 307, 308, 311, 312, 313, 321, 322, 333 MPT, 270 MS, 6, 7, 23, 27, 29, 31, 32, 34, 36, 37, 39, 40, 41, 42, 47, 49, 50, 53, 55, 56, 57, 58, 59, 60, 61, 62, 63, 64, 65, 66, 74, 75, 76, 77, 83, 85, 116, 170, 172, 173, 175, 176, 177, 202, 230, 369, 370, 372 MSA, 272, 273, 274 MSM, 272, 273 MSRN, 230, 235 Multi-call, 165

- Multicarrier, 13, 18, 185
- Multimedia, 1, 10, 11, 18, 81, 89, 103, 182, 194, 195, 210, 215, 217, 224, 227, 240, 242, 244, 246, 265, 266, 270, 271, 273, 274, 296, 307, 308,
 - 312, 319, 320, 321, 322, 323, 329,
 - 330, 331, 332, 333, 335, 342, 343,
 - 345, 346, 354, 365, 372, 373, 377
- Multimedia Gateway, 272
- Multimedia Information Distribution Server, 329, 330, 333
- Multimedia Messaging Service, 189, 343
- Multimedia Messaging Service (MMS), 343
- Multimedia Multiplexing, 321, 322
- Multimedia Service Agent, 273
- Multimedia Service Management, 273
- Multimedia Service Platform, 266, 270, 273, 274
- Multipath Interference, 24, 30, 47, 51, 66, 113
- Multipath Interference (MPI), 24, 113
- Multiple Access Interference (MAI), 24
- Multistage Interference Canceller, 66, 67
- Multistage Interleaver, 43, 103

NAS, 142, 145, 153

- Network Authentication, 225, 239
- Next generation WAP, 336, 338, 339, 340, 341, 342
- NF, 170, 173, 185, 196, 199
- Node B, 23, 85, 86, 156, 157, 159, 167, 182, 220, 370 Non-Access Stratum, 142, 145
- Not Reachable, 236
- nrt-VBR, 222, 223
- N-TACS, 3
- NTT System, 1, 2
- Numbering Plan, 15, 230
- NVML, 349
- OA-RA, 183, 185 Open Air Receive Amplifier (OA-RA), 183

- Open API, 371
- Open-loop Transmission Diversity, 368
- Operation System (OpS), 277
- OpS, 277, 280, 281, 283, 292, 296,
- 298, 299, 304, 305
- Organic EL, 206
- Organic EL Display, 206
- Orthogonal Variable Spreading Factor (OVSF), 29, 85
- Orthogonality Factor, 173, 179
- OSI Management, 278
- Other Person Search (Third-party location search), 346
- Other Person's Location Registration, 346
- Out of Band Transcoder Control, 236, 238
- Outage, 173, 176
- Outer Loop, 45, 47, 49, 53, 114, 115, 147, 368
- Overlay, 181
- OVSF (Orthogonal Variable Spreading Factor), 29, 85

Packet Mobile Communications System, 7 Packet multimedia CODEC for interactive, real-time speech, 377 Packet Network, 7 Packet One, 9 Packet-streaming CODEC, 377 Packet-switching Function, 182, 373 Paging, 40, 87, 101, 145, 147, 153, 156, 159, 160, 169, 173, 225, 226, 227, 228, 230, 231, 235 Paging Indicator, 87, 101 Parlay, 373 Path Search, 25, 50, 56 PC Card Type, 194 PCR, 222 PDA Type, 194 PDCP, 86, 142, 143, 144, 145 PDC-P, 8, 243, 245, 247, 248, 270, 289, 294, 295 Peripheral Cell Search During Communications, 39

Personal Information Manager (PIM), 207 Pico Cell, 181 Piggybacked PDU, 140 Piggybacked STATUS PDU, 136, 137 PIN, 203 PKI, 357 POIX, 349 Pole Capacity, 170 Positioning Method, 347 Power Allocation, 172, 173, 174, 176, 177, 179, 202 Power Ramping, 42, 116, 117, 118 Preamble, 42, 96, 101, 112, 116, 117, 118, 119 Preamble Pre-distortion, 185 Pre-paging, 226, 230, 235 Primary Synchronizatoin Code (PSC), 32, 33, 34, 36, 37 Profile, 25, 37, 39, 49, 50, 55, 57, 58, 273, 274, 311, 312, 313, 336, 337, 338, 341, 342, 343, 359 Protocol Conversion Gateway, 266, 267 Provider, 8, 225, 307, 326, 327 P-SCH, 100, 101, 109, 113 P-TMSI, 153 Public Key Infrastructure (PKI), 357 Puncture, 116 Push, 326, 327, 328, 339, 342, 345 PVC, 219, 226 Q.AAL2, 237, 238

QCIF, 310, 312 QoS, 216, 217, 221, 223, 225, 246, 247, 248, 249, 250, 252 QPSK, 5, 7, 23, 31, 37, 51, 62, 72, 85, 107, 109, 112, 369 Quarter CIF, 310

R4/R5, 373
Radio Access Bearer, 162, 164, 165, 230, 254
Radio Access Network (RAN), 14, 82, 215, 319, 371

Radio LAN, 209 Radio Link Design, 169, 171, 172, 174, 175, 177 Radio Network Controller (RNC), 14, 57, 85, 159, 282, 370 Radio Network Controller (RNC) Radio Network Temporary Identity, 154 RAN, 14, 15, 82, 85, 86, 183, 188, 215, 218, 219, 221, 225, 230, 231, 240, 252, 283, 284, 285, 289, 296 RAN Strata, 285, 289 RANAP, 225, 226 Random Access, 31, 41, 42, 87, 96, 101, 116, 117, 118, 119 Random Access Subchannel, 117, 118 Rate Matching, 45, 46, 47 Rate matching Rate Matching Attribute, 46 RCR, 4 Real View Navigation Technology, 354 Reference Sensitivity Level, 196 Relocation, 142, 144, 145, 159, 232, 250, 251, 253, 254 Repetition, 42, 45, 70, 106, 291 Research and Development Center for Radio Systems (RCR), 4 **RESET ACK PDU, 137, 140** RESET PDU, 137, 140 **RESET/RESET ACK PDU, 137** Resynchronization, 312 Resynchronization Reversible Variable Length Code (RVLC), 313 Reversible Variable Length Coding, 313 RF Circuit Calibration, 72 RLC, 43, 86, 127–129, 131–146, 157, 162 **RNTI**, 154 Roaming 5-6, 15, 216, 226, 230, 233, 235, 250 Route Map Information Summarizing Technology, 354 Routing, 5–6, 218–219, 230, 261, 262, 370-371, 373, 375

RRC, 86, 117, 127–128, 131–134, 140–141, 143, 145–154, 157, 159–166 RS-232C, 209 RSCP, 115 r-tree index, 353 RTSP, 331, 377 rt-VBR, 222–223

SAW Filter Technology, 199 SCP, 14, 226, 301 SCR, 222 Scrambling Code, 25, 29, 31–37, 39-41, 88, 98-100, 108, 110-113, 116–117, 158, 202 Scrambling Code Secondary Synchronization Channel, 32 Secondary Synchronization Code (SSC), 32, 100 Sectoring, 180-181, 190 Self-location Notification, 346 Self-location Search, 346 Service Access Class, 117–118, 131 Service Control Point (SCP), 14, 226 SGSN, 215, 224-226, 233, 346-247, 249, 252 - 254Shadowing, 32, 53, 59, 171 Shaping Control, 291 Short Message Service (SMS), 203, 217, 340 Short scrambling code, 109, 111-112 Short-lived Certificate, 360 SI, 240-241 Sign Text, 357-359 Signature, 42, 96, 101, 112, 117–118, 128, 356-361 SIM, 202 Simple Profile, 312–313 SIP, 372 SIR, 24, 27, 39, 45, 47–51, 56–57, 59, 60-61, 64-65, 67-68, 71, 74-76,91, 114–116, 169, 181, 368 Site Diversity, 28, 56–60, 115, SloP, 349 SM, 196, 264 SM/GMM, 225

Small/Medium-sized City Car Phone System SM-MO, 260-262 SM-MT, 260-262, 264 SMS, 203, 217, 225, 260-264, 340 Soft/softer Handover, 56, 156 SOHO Terminal, 194 Sound Articulation, 319 Spatial Location Payload, 352 Speech Processing Equipment (SPE), 7 Spreading Factor (SF), 21, 29, 83, 85, 96, 110, 367 Spreading Factor (SF) Spreading Modulation Process, 30 Spurious Emission, 196 SRNC, 141, 145, 159 SRNS, 142, 144-145 SS, 225, 233, 256, 325 SSDT, 60, 96, 115 STATUS PDU, 136-137, 139-140 Status Report, 261 Streaming, 219, 221, 223, 249, 320, 329-331, 333, 371, 377 Streaming Method, 329-331 STTD, 61-62, 92, 120-121 Subscriber Information, 202–203, 226-228, 246, 252, 254 Super Twisted Nematic (STN) Liquid Crystal Display, 204–206 SVC, 219, 223, 226 Synchronized Multimedia Integration, 335, 377 System Level Simulation, 179 TACS, 3 Tandem Connection, 236, 374 Tandem Free Operation, 236, 374

TCP Gateway, 266, 267, 268, 269 TD-CDMA, 16, 17 TDD, 13, 16, 17, 92, 156, 365–368 TDMA, 3, 4, 7, 13, 16, 21, 25, 27, 82, 83, 169, 178, 180, 199, 247 TF, 46, 47, 95, 97, 128, 131, 157, 158 TFC, 46, 47, 95, 128, 131, 157, 158 TFCI, 30, 47, 95, 96, 97, 99, 100, 131 TFCS, 46, 47, 128, 131, 158

TFD-type Liquid Crystal Display, 205 TFI, 131 TFO, 236, 374, 375 TFS, 46, 126, 128, 131 TFT-type Liquid Crystal Display, 205 TG8/1, 12 Thin-Film-Diode-type Liquid Crystal Display, 205 Thin-Film-Transistor-type Liquid Crystal Display, 205 Three-step Cell Search, 25, 32, 34, 37, 39, 40, 54, 65 TIA, 3, 17, 18, 317 Tilting, 181, 192 Tilting Angle, 181 TLS, 341, 342, 357, 358, 359, 360 TMN, 277, 280 TMSI, 153, 228, 231 TOM, 278 Tracking, 25, 74, 75, 115, 370 Tracking Traffic, 6, 7, 43, 90, 127, 128, 157, 128, 166, 167, 169, 171, 174, 175, 180, 181, 186, 216, 218, 219, 220, 221, 222, 223, 224, 226, 240, 248, 249, 293, 294, 295, 297, 298, 302, 303, 361, 365, 370 Transcoder, 230, 236, 238, 375 Transcoder Free Operation, 374 Transmission Diversity, 31, 61, 62, 63, 65, 66, 91, 96, 98, 119, 120, 121, 123, 185, 368 Transmission Method Outside the Mobile Communications Network, 347 Transmission Time Interval (TTI) 46, 130 Transmit Power Control Transmit Power Control (TPC), 27, 60, 82, 368 Transparent Mode, 128, 131, 133, 137 Transport Block, 26, 45, 118, 129 Transport Block Set, 130, 131 Transport Block Set Size, 130, 131 Transport Format, 46, 95, 117, 118, 131 Transport Format Combination, 46, 131 **Transport Format Combination** Indicator, 30, 131 Transport Format Combination Set, 46, 128, 131 Transport Format Indicator, 131 Transport Format Set, 131 TrD PDU, 135, 137, 138 TrFO, 374, 375 TRX, 183, 185, 186 TSTD, 34, 37, 61 TTC, 5, 17, 224, 226, 317 Tunneling, 225, 249, 253, 245, 266, 269, 270, 370 Tunneling Gateway, 266, 269, 270 Turbo Code, 84, 105 Two-phase Security Model, 357, 358 U Plane, 86, 238 UBR, 222, 223, 248 UDI Multimedia, 242 UE, 23, 198, 201, 203, 224, 225, 227, 228, 230, 231, 235, 236, 238, 239, 240, 242, 244, 246, 247, 248, 249, 252, 253, 254, 259, 260, 261, 263, 264, 265, 266, 270, 274 UE Maximum Output Power, 196 UIM, 14, 202, 203 UMD PDU, 135, 138 Unacknowledge Mode, 131, 133, 138 Universal Serial Bus (USB), 209 UPC, 223 URA_PCH, 154 User Authentication, 239, 252, 327, 328, 356, 360 UTRA, 17, 92, 183 UTRAN, 41, 85, 125, 131, 140, 144, 145, 146, 147, 154, 155, 156, 157, 158, 159, 370, 376 VAD, 316, 317 Variable Length Coding (VLC), 308, 309, 310, 311 VC, 219, 220 Vehicle Terminals, 194 Video Distribution, 11, 195, 205, 217, 221, 308, 312

Videophone Type, 196
Virtual Home Environment (VHE), 216
Visited Mobile Switching Center (V-MSC), 5
VLR, 228, 230, 231, 233, 235
VLR Number, 228, 233
V-MSC, 5, 6
Voice Activity Detector, 316
VoIP, 319, 320, 371, 372
VP, 219, 287

W3C, 335, 336, 338, 340, 341, 349
WAE, 340
Walsh Code, 25, 29
WAP Forum, 307, 338, 343, 358
WAP1.X, 339
WARC, 18
W-CDMA, 1, 13, 21, 28, 50, 51, 56, 66, 82, 109, 169, 181, 182, 190, 192, 194, 216, 247, 254, 319, 321, 345, 361, 366

WDP, 360 Weighted Multi-Slot Averaging (WMSA), 51, 52, 64 Wideband Speech Encoding, 376 WIM, 339 Wireless PPP, 270 Wireless Profiled TCP, 342 Wireless TCP (W-TCP), 226 WML, 334, 335, 339, 342, 357 WML Extension, 342 W-PPP, 270 WRC, 19 WSP, 344 WTLS, 357, 358, 359, 360 WTLS Certificate, 359 WTP, 340

X.509 Certificate, 359 XHTML, 335, 336, 341, 342 XHTML Basic, 336, 341 XML, 335, 341, 349