Digital Audio Broadcasting

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

Digital Audio Broadcasting

Principles and Applications of DAB, DAB + and DMB

THIRD EDITION

Editors

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WILEY

A John Wiley and Sons, Ltd, Publication

This edition first published 2009

Registered office
John Wiley & Sons Ltd, The Atrium, Southern Gate, Chichester, West Sussex, PO19 8SQ, United Kingdom

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The authors and publishers would like to acknowledge the European Telecommunications Standards Institute (ETSI) for granting permission to reproduce figures and tables from the four ETSI Standards referred to in the book

Library of Congress Cataloging-in-Publication Data
Digital audio broadcasting : principles and applications of DAB, DAB+ and DMB / edited by
Wolfgang Hoeg, Thomas Lauterbach. — 3rd ed.
p. cm.
Includes bibliographical references and index.
ISBN 978-0-470-51037-7 (cloth)
1. Digital audio broadcasting. I. Hoeg, Wolfgang. II. Lauterbach, Thomas.
TK6562.D54D54 2009
621.384—dc22
2009001879

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

ISBN 978-0-470-51037-7 (Hbk)

Set in 10/12pt Times by Integra Software Services Pvt. Ltd., Pondicherry, India
Printed and bound in Great Britain by Antony Rowe Ltd, Chippenham, Wiltshire.

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Thomas Lauterbach, Prof. Dr. rer. nat., Nürnberg, Germany, received his Diplom-Physiker degree and PhD from Erlangen University. In 1992 he joined Robert Bosch GmbH, where he became involved with the development of DAB. In 1997 he became head of a multimedia systems development department. He was with several Eureka 147/DAB Working Groups, the German DAB platform and Euro-DAB/WorldDAB and contributed to ETSI. He also coordinated the MEMO (ACTS) project. Since 1997 he has been with Georg-Simon-Ohm-Hochschule Nürnberg – University of Applied Sciences as a Professor of Physics. He is currently involved in the Digital Radio Mondiale (DRM) project. In 1996 he edited one of the first books on DAB in German. (Contributed to all of the editions, Chapters 1, 2, 4 and 7)

Contributors

Stephen Baily, MA (Cantab), London, United Kingdom, joined the British Broadcasting Corporation in 1985. For the last few years he has worked for BBC Research and Development on various aspects of digital broadcasting, with a particular focus on the transmission aspects. He has designed experimental and operational transmission equipment for DAB, and has also worked in the areas of satellite delivery, spectrum planning and the design of broadcast systems. He has been a contributor to several Eureka 147/DAB Working Groups, Task Forces and the Executive Committee. (Contributed to the 2nd edition, Chapter 7)

Birgit Bartel-Kurz, Dipl.-Math., Erlangen, Germany, studied mathematics at University of Applied Science, Darmstadt and at University of Namur and earned her degree in 1987. She worked for Robert Bosch GmbH in the telecommunications sector and lectured at the University of Cooperative Education, Mannheim. Currently Birgit is with the Fraunhofer Institute for Integrated Circuits IIS, working in the area of digital broadcast applications and multimedia security technologies. Birgit was engaged in several technical working groups of WorldDMB. There she chaired the Task Force Conditional Access Framework. (Contributed to the 3rd edition, Chapter 2)

Michael Bolle, Dr.-Ing., Hildesheim, Germany, received his Dr.-Ing. degree with honors from Ruhr-Universität Bochum. In 1992 he joined Robert Bosch GmbH, working in the field of DAB receiver development. Since 1997 he has been Head of the Department 'Advanced Development Multimedia Systems', in which the Bosch DAB VLSI project took place. Michael Bolle left Bosch in 1999, and was co-founder and executive VP of engineering of Systemonic AG in Dresden. In 2002 he returned to Robert Bosch GmbH, and is now with Bosch Car Multimedia. He holds more than 10 patents in the field of DAB receiver technology. (Contributed to the 1st and 2nd editions, Chapter 8)

Thomas Boltze, Dr.-Ing., AES, SMPTE and IEEE, Philips Research Shanghai, received his PhD in electrical engineering from the University of Technology Hamburg. In 1995 he joined the Advanced Systems and Applications Lab of Philips Consumer Electro-nics in The Netherlands, where he developed audio compression systems as Senior Systems Designer and contributed to ITU, Eureka 147/DAB and the DVB project. At Visible World, New York, he was dealing with systems architectures for DTV. He is currently a Principal Scientist with Philips Research in Shanghai, taking care of DTV platforms and applications. (Contributed to the 1st and 2nd editions, Chapter 3)

Roland Brugger, Dr. rer. nat., München, Germany, received his Diplom-Physiker degree and PhD from Ludwig-Maximilians-Universität, München. In 1987 he joined the Institut für Rundfunktechnik (IRT). Since 1992 he has been deeply involved in the planning and development of digital terrestrial broadcasting (T-DAB and DVB-T). He participated in the international planning conferences since 1995 up to 2004/2006, for which he contributed to the development of planning methods. Since 2004 he has been head of the frequency planning section of IRT. At present he is involved in the implementation of the GE06 frequency plan for digital terrestrial broadcasting. (Contributed to the 3rd edition, Chapter 7)

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Detlef Clawin, Dr.-Ing., Pacifica, USA, received his Dr.-Ing. degree from Ruhr-Universität Bochum. He worked in the fields of high speed circuit design for fibre optics and RFICs for DAB receivers at Bell Communications Research in Redbank, NJ, USA, and in Germany at Fraunhofer-Gesellschaft IMS in Duisburg, MICRAM Microelectronics in Bochum, and Robert Bosch GmbH, Corporate Research in Hildesheim, responsible for the development for a single chip DAB tuner IC. Presently, he is with NXP. (Contributed to the 1st and 2nd editions, Chapter 8)

Bernhard Feiten, Dr.-Ing., AES, Berlin, Germany, received his doctor degree in electronics at the Technische Universität Berlin in the field of psycho-acoustic and audio bit-rate reduction. He worked as an assistant professor at the Technische Universität Berlin in communication science. Since 1996 he has been with Deutsche Telekom. At the research company T-Nova Deutsche Telekom Berkom, he was head of the Audio systems division since 1999, responsible for current research on DAB matters. He is a member of the ISO/MPEG Audio group. (Contributed to the 1st and 2nd editions, Chapter 4)

Neil H. C. Gilchrist, BSc, C.Eng., MIET, AES Fellow, Ashtead, United Kingdom, graduated from Manchester University in 1965 with a BSc honours degree in physics and electronic engineering, and joined the BBC Research Department. He worked on broadcast audio, including NICAM and digital audio broadcasting. Towards the end of his BBC career he became involved in a number of European RACE, Eureka 147/DAB and ACTS projects, and in international standardisation with CCIR (ITU-R), AES and EBU groups. After retiring in 2002 he has become an audio consultant. He was President of the Audio Engineering Society (AES) in 2005/2006. (Contributed to the 2nd and 3rd editions, Chapter 3)

Chris Gould, BMus (Tonmeister), London, United Kingdom, graduated from the University of Surrey in 1994 with a BMus honours degree in Music and Sound Recording (Tonmeister). He joined the UBC Media Group in 1994, working within the audio facilities company, Unique Facilities Limited. He has been actively involved with DAB Digital Radio since 1999, and chaired the DAB EPG Task force within WorldDMB. Chris now runs his own digital radio technology company, All In Media, and is providing digital radio data technology to leading broadcasters across the world. (Contributed to the 2nd and 3rd editions, Chapter 5)

Frank Herrmann, Dipl.-Ing., Frankfurt, Germany, studied electrical engineering at the Technical University of Braunschweig and then joined the European R&D centre of Clarion. In 1994 he led his company into the Eureka 147 DAB Project. In 1996 he changed to Panasonic R&D Europe and remained responsible for DAB. Frank chaired the WorldDAB/WorldDMB Technical Committee from 2004 to 2007 and received the Per-Erik-Selemark Award in 2005. In 2006 he became Panasonic's European R&D project leader for the second generation of DVB. (Contributed to the 3rd edition, Chapters 2 and 9)

Frank Hofmann, Dr.-Ing., Hildesheim, Germany, received the Dipl.-Ing. degree in electrical engineering from the University of Karlsruhe and his PhD from the university Hanover, 2004. Since 1996 he has been with Robert Bosch GmbH, Corporate Research, Hildesheim. His main focus is in the field of digital radio system design as well as RF and baseband receiver development. Currently he works as project manager. He leads the receiver working group of the DRM consortium and is a member of its Steering Board. (Contributed to the 3rd edition, Chapter 8)

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Egon Meier-Engelen, Dipl.-Ing., Cologne, Germany, received his Dipl.-Ing. degree in communications engineering from the Technische Hochschule Aachen in 1963. Since then he has worked in R&D with different firms on various fields of communications technologies. In 1985 he joined the German Aerospace Center (DLR), Cologne, where he headed a section managing research grants for information technology by the Federal Ministry for Education and Research (BMBF). He directed the Eureka 147 DAB project from 1986 to 1998. In 2001 he retired from DLR. (Contributed to the 1st edition, Chapter 1)

Torsten Mlasko, Dipl.-Ing., Pattensen, Germany, received the Dipl.-Ing. degree in electrical engineering from the University of Hanover in 1995. Since 1995, he has been with Robert Bosch GmbH, Department Advanced Development Multimedia Systems, Hildesheim, Germany. He is now with Bosch Car Multimedia. He was an active member of Eureka 147 Working Group A and of the ISO/MPEG Audio group, and is presently an active member of WorldDAB and DRM. (Contributed to the 1st and 3rd editions, Chapter 8)

Hans-Jörg Nowottne, Dr.-Ing., Dresden, Germany, studied electrical engineering at the University of Technology in Dresden, and worked for a long time in R&D in the electronics industry, focused on computer-aided design. Since 1992 he has been with the Fraunhofer-Institute of Integrated Circuits, Dresden. He has dealt with prototype development in the field of telecommunications and digital broadcasting (DAB, DVB) since 2001 as a head of department. In the Eureka 147/DAB project, he was involved in the definition of the Service Transport Interface (STI) and was with Working Group B. Now he is retired. (Contributed to the 1st and 2nd editions, Chapter 6)

Roland Plankenbühler, Dr.-Ing., Nürnberg, Germany, studied electrical engineering at the University of Erlangen. After his Diploma Degree he worked there at the 'Lab for Technical Electronics'. Having obtained his PhD he joined Fraunhofer IIS-A in 1990, and became manager of the 'Data Services' group involved with DAB. Since 1997 he has managed the Department 'Terminal Devices', which is mainly focused on the definition and implementation of data services and terminals. He is currently Head of the 'IT Services' department. He was a member of several working groups of Eureka 147/DAB, as well as WorldDAB Module A. (Contributed to the 1st and 2nd editions, Chapter 4)

Markus Prosch, Dipl.-Inf., Erlangen, Germany, studied computer sciences at the University of Erlangen. In 1995 he joined the Fraunhofer Institute for Integrated Circuits IIS. He is responsible for the development of the Fraunhofer DAB/DMB ContentServer. Since 1996, he has been deeply involved in several DAB standards. He chaired the WorldDMB task force MOT and the task force audio coding for DAB+. In 2007, WorldDMB conferred the Per Erik Selemark Award for his services to DAB+. Starting in 2001, he has also been involved in the specification of Digital Radio Mondiale (DRM) standards. (Contributed to the 3rd edition, Chapters 2 and 4)

Thomas Schierbaum, Munich, Germany, after technical education, he studied media marketing at the 'Bayerische Akademie für Werbung und Marketing'. He is with the Institut für Rundfunktechnik (IRT), working as a project manager for data service controlling and ancillary data transmission in DAB. Among other things, he was responsible for the infrastructure of the DAB-Program 'Bayern Mobil' of the Bayerischer Rundfunk, DAB product management and consulting. He is with several working groups of the German public broadcasters (ARD/ZDF). (Contributed to all of the editions, Chapter 5)

Henrik Schulze, Prof. Dr. rer. nat., Meschede, Germany, received the masters degree (Dipl.-Phys.) in 1983 and his PhD in 1987 from the University of Göttingen, Germany. From 1987 to 1993 he was with Robert Bosch GmbH at the Research Institute in Hildesheim, where he worked on the development of modulation and channel coding for the DAB system. Since 1993 he has been Professor for Communication Theory at Fachhochschule Südwestfalen – University of Applied Sciences, Division Meschede. He is currently involved in the Digital Radio Mondiale (DRM) project. (Contributed to all of the editions, Chapters 1 and 2)

Ralf Schwalbe, Dipl.-Ing., Berlin, Germany, studied systems engineering at the Technical University in Chemnitz, and joined the research and development centre RFZ of Deutsche Post, Berlin. Since the 1990s he has been with Deutsche Telekom, and is now at T-Systems, dealing with research and development for audio and data services with DAB and DMB. He was a member of several working groups of the Eureka 147/DAB project and is now with WorldDMB. (Contributed to the 1st and 2nd editions, Chapter 4)

Gerhard Stoll, Dipl.-Ing., AES Fellow, München, Germany, studied electrical engineering at the universities of Stuttgart and Munich. In 1984 he joined the Institut für Rundfunktechnik (IRT), and became head of the psycho-acoustics and digital audio group. There he was responsible for the development of the MPEG-Audio Layer II standard. Mr Stoll was also a member of different international standardisation groups, such as MPEG, Eureka 147/DAB, DVB and EBU. As a senior engineer at the IRT, he is now in charge of advanced multimedia broadcasting and information technology services. (Contributed to the 1st edition, Chapters 1 and 3)

Wolfram Titze, Dr., C.Eng., MIEE, Berlin, Germany, graduated from Friedrich-Alexander University Erlangen and received his PhD in electronic engineering from University College London in 1993. He joined Robert Bosch GmbH, and later became head of the department multimedia systems development. He was with the Eureka147/DAB Executive Committee, worked on standardisation of DAB distribution interfaces at ETSI, and was vice chairman of the technical module of WorldDAB. He also worked as Eureka147 promotion engineer. Dr Titze is now head of amplifier systems with the Broadcasting Division of Rhode & Schwarz. (Contributed to all of the editions, Chapter 7 and Appendix 2)

Lothar Tümpfel, Dipl.-Ing., Berlin, Germany, studied telecommunication engineering in Leipzig and information technology in Dresden. He worked for many years in the field of the development of studio audio equipment. Since the 1990s he has been with Deutsche Telekom, later at T-Nova Deutsche Telekom Berkom, dealing with research and development for DAB. He was a member of several working groups and of the Steering Committee of Eureka 147/DAB, and contributed to the EBU Specialist Group V4/RSM. In 2001 he retired from Deutsche Telekom. (Contributed to the 1st and 2nd editions, Chapter 6)

Herman Van Velthoven†, Ing., AES, Antwerpen, Belgium, joined Pioneer Europe in 1973, where he became successively Manager, Product Planning General Audio, and Manager, Engineering Department. In 1996 he changed to Pioneer's R&D Division. When in 1993 Pioneer joined the Eureka 147/DAB project as the first non-European company, he participated in several working groups and task forces. He then worked in the WorldDAB Module A, chaired EACEM PT1.1, and was an ETSI rapporteur and a member of CENELEC TC-206. He retired in 2002. Unfortunately, just after revising the manuscript for the 2nd edition, he suddenly died in early 2003. (Contributed to the 1st and 2nd editions, Chapters 2 and 5)

Alexander Zink, MBA, Dipl.-Ing., Erlangen, Germany, graduated from Friedrich-Alexander-University Erlangen-Nuremberg in electrical engineering/microelectronics and joined Fraunhofer Institute for Integrated Circuits in Erlangen, Germany, in 2000 as a vice group leader 'Broadcast Applications'. He is project director for the information system 'UMIS', the broadcast system 'Fraunhofer DRM ContentServer' and the DAB information service 'Journaline'. For the Digital Radio Mondiale consortium (DRM) he serves as vice-chairman of the Technical Committee and as vice president of the DRM Association. In addition, he is member of the WorldDMB Technical Committee and the European HD Radio Alliance (EHDRA). (Contributed to the 3rd edition, Chapter 4)

Preface

Preface to the first edition

The new digital radio system DAB (Digital Audio Broadcasting), developed within the Eureka 147 project in close cooperation with the EBU, is a very innovative and universal multimedia broadcast system that is just being introduced, and which has the potential to replace existing AM and FM audio broadcast services in many parts of the world in the near future. In several countries in Europe and overseas, broadcasting organisations, network providers and receiver manufacturers are already implementing digital broadcasting services using the DAB system.

DAB is very different from conventional analogue broadcasting systems. Most of the system components such as perceptual audio coding (MPEG-1/2), OFDM channel coding and modulation, the provision of a multiplex of several services and data transmission protocols (MOT), are new concepts typical of digital broadcasting. Even experts in analogue transmission systems will feel less familiar with these new elements of broadcasting technology. Therefore, the aim of this book is to inform the expert reader about the basic concepts of the DAB system.

Besides introducing the basics, the focus of the book is on the practical implications of service provision and the new infrastructure required in broadcasting houses, for multiplex and network management, and for coverage planning. Also, some elements of up-to-date receiver concepts are described. The level of standardisation of the DAB system is quite advanced, and the relevant recent international standards and related documents are introduced and referred to for easy access for the reader seeking technical details. An extended bibliography is also provided.

The book is designed as a well-structured technical guide by a team of expert authors closely involved in the development and standardisation of DAB. This ensures competent presentation and interpretation of the facts based on the latest state-of-the-art. The book is primarily aimed at professional users such as developers and manufacturers of professional devices for distribution networks or consumer receivers, planning engineers and operational staff with broadcasters, network providers, service and content providers. For other technically minded people who wish to become acquainted with the concepts of digital broadcasting, the book will serve as a comprehensive introduction to the field, since it contains all the information needed for further study. The book may also serve for academic or educational use, because it is based on the latest versions of the relevant international standards and publications, as well as actual experience with pilot applications and first implementation of services.

The editors wish to take this opportunity to express their thanks to all the contributors for the enjoyable cooperation and their excellent work, which most of them had to complete in addition to their demanding jobs. Many thanks also to Mrs Helga Schön, who was kind enough to design a portion of the drawings for the book. The editors also wish to thank the publishing team at John Wiley & Sons Ltd. for their interest, understanding and patience during the writing and production period.

May this book help to introduce the DAB system worldwide.

The Editors
Berlin/Nürnberg, Autumn 2000

Preface to the second edition

The first edition of this book was nearly sold out within a period of less than eighteen months. Considering that DAB (now often called Digital Radio) was still in a growing state of introduction and penetration worldwide, and that this book was still the only comprehensive publication in the English language on DAB, the publisher offered to issue a revised edition. This was the chance for the editors and contributors to prepare not only a simple revision of the book, but also to add some newer results of development and standardisation, and to further complete the book by adding sections on several issues. The editors appreciated the many detailed reviews of the first edition (in particular the one of Mr Franc Kozamernik, in EBU Techn. Review), which pointed their attention to items that were not appropriately covered, and made suggestions for topics to be included in the second edition.

Here, it is only possible to address the most important changes and additions that were made to the book: Chapter 1 (Introduction) was updated in several respects considering the newest developments worldwide. Also Chapter 2 (System aspects) was revised in many details. Chapter 3 (Audio services) was completely rearranged, newer developments concerning audio coding and several additional aspects were included. Chapter 4 (Data services) was also completed by adding text on new features such as IP tunneling applications. Chapter 5 (Provision of services) now includes the very new technology for a DAB Electronic Programme Guide (EPG) and other new features. Chapter 7 (Broadcast side) was completed by an extended part on propagation aspects. Chapter 8 (Receiver side) takes up the latest developments in IC technology and receiver design.

Appendix 2 (Introduction of DAB) was updated to the state reached in early 2003. A new Appendix 3 (DAB Frequencies) was added, which provides the current CEPT frequency tables valid for DAB service implementations in Europe, Canada and Australia. Last but not least, the Bibliography was updated and completed to the latest state of standardisation and other literature.

Covering such a huge number of additional subjects was only possible because several new expert authors from the international scene of DAB development (Gerald Chouinard, Neil H. C. Gilchrist, Chris Gould and Ralf Schwalbe) could be persuaded to contribute. The current team of authors (including the editors) now comprises more than 20 outstanding experts in the field of DAB. The editors and authors also appreciated comments and support from many other experts – in particular, the authors of Chapter 4 wish to thank Markus Prosch, FhG-IIS, and Uwe Feindt, Robert Bosch GmbH for their valuable contributions and comments on this chapter.

The editors wish to express their thanks to all the contributors, who made an enormous effort to provide an up-to-date picture of the DAB development and implementation.

The editors and contributors express their deepest regret for the death of their colleague Herman Van Velthoven, who died suddenly a few days after finishing the revision of his contributions to Chapters 2 and 5.

The editors also wish to thank the publishing team at John Wiley & Sons Ltd. for their interest, their understanding and patience during the revision and production period.

May this book help to further support the introduction of Eureka 147 DAB worldwide.

The Editors
Berlin/Nürnberg, Spring 2003

Preface to the third edition

Similarly, the second edition of this book was practically sold out within a short time. Considering that Digital Audio Broadcasting (DAB) was in a growing state of introduction and penetration worldwide, and that this book was still the only comprehensive publication in the English language on DAB, the publisher offered to issue a revised edition. This was the chance for the editors and contributors to prepare not only a simple revision of the book, but also to include the latest results in international development and standardisation, Especially, further developments which led to the DAB system family comprising DAB and the newer DAB+ and DMB systems, are considered through completely revised respectively new chapters.

At this point, it is only possible to address some of the most important changes and additions that were made to the book: Chapter 1 (Introduction) was updated considering the newest developments. Also Chapter 2 (System Concepts) was updated with respect to the newly added feature of concatenated coding; the sections on Conditional Access and Service Information were revised. Chapter 3 (Audio Services) was completely rearranged again, new developments concerning audio coding for DAB+ and multichannel audio programmes were included. Chapter 4 (Data Services) was also updated, in particular extended sections on advanced text features were added. Chapter 5 (Provision of Services) now includes the latest technology for the DAB Electronic Programme Guide (EPG). Chapter 7 (Broadcast Side) was completed by an extended part on propagation aspects and latest results from planning conferences. Chapter 8 (The Receiving Side) takes up the latest developments in IC technology and receiver design. An additional Chapter 9 (Mobile Television and Multimedia) now covers DMB (Digital Multimedia Broadcasting) and IPDC (IP Datacast) as rather new components of the DAB system family.

Whilst some of the authors who contributed to the former editions were no longer available, a number of new expert authors were persuaded to contribute to the third edition (Birgit Bartel-Kurz, Roland Brugger, Frank Herrmann, Frank Hofmann, David Marston, Markus Prosch, Alexander Zink). Most of them were (or still are) active in different DAB related standardisation bodies, such as WorldDAB/WorldDMB, CEPT, DRM etc. - sometimes at higher levels.

The editors wish to express their thanks to all the contributors, who again made much effort to provide an up-to-date picture of the DAB system family's development and implementation. They also wish to thank again the publishing team at John Wiley & Sons Ltd. for their interest, their support and patience during the revision and production period of the third edition.

May this third edition of the book help to further support the implementation of the DAB system family and its applications worldwide.

The Editors
Berlin/Nürnberg, Autumn 2008

Foreword

Radio must surely be the most enduring and reliable of all electronic media. Not only did it survive the introduction of television, it thrived and grew. More recently it has faced competition from the Internet and mobile entertainment on mobile phones and MP3 players. And still, radio survives and flourishes! This does not surprise me at all. Why? Because radio is a simple proposition, it is available everywhere, it costs nothing, all you do is turn it on and it just works. No licences, no downloading, no complicated choices, it is live, always relevant and engaging.

Will it go on surviving? My view – as a radio engineer, as a programmer and as a broadcaster – is a very definite 'yes'. But it must go digital to offer a richer experience to those listeners who want more, and work as well or better for those who are happy with radio as they know it. Going digital has not been an easy path, and undoubtedly some countries have struggled whilst others have had great success. However, this is not a reflection of the technology (otherwise it would not have worked in the UK, Denmark or Korea, for example) but is down to the local regulation, politics and business models, all of which have to be right for radio to thrive.

The DAB family is now well established with the underlying technology having proven to be robust and flexible. But technology does not stand still and the family of standards, collectively referred to as 'Eureka 147', now includes DMB mobile TV and DAB+ with highly efficient AAC audio coding, not to mention new multimedia technologies and updated metadata options. This has enabled digital radio to keep pace with changing consumer expectations and the impact of media convergence.

The traditional experience of one type of media being consumed on a specific device has been disappearing for some time to the extent that we now expect to view television content on almost anything with a screen, whether it is a TV, games console, MP4 player or a computer. This shift in how people expect to consume media is having a profound effect on broadcasters and the communications industry, and one that WorldDMB and its members have been able to address.

DAB and DAB+ offer an identical consumer experience centred on radio, with text, slideshow, EPG and other multimedia features. DAB offers eight to ten radio services within a 1.5 MHz multiplex whilst DAB+ (using HE AACv2) can accommodate 20–30 radio services in the same spectrum. DMB is primarily a mobile TV platform, sharing the same multiplex and carrier structure as DAB, but it can be used for a visual radio service with similar multimedia properties to DAB and DAB+.

Since any combination of DAB, DAB+ and DMB services can co-exist on one or more multiplexes, the Eureka 147 Family is now seen as a highly cost effective solution for many broadcasters who want to enhance their services with multimedia content. Services are now commercially available in Belgium, China, Denmark, Germany, Malta, Norway, Singapore, South Korea, Spain, Switzerland and the UK. Meanwhile Australia, the Czech Republic, France, Ghana, Hong Kong, Hungary, Indonesia, Ireland, Israel, Italy, Kuwait, Malaysia and Vietnam are already broadcasting or about to start full time licensed services, with many other countries engaged in successful trial phases. This endorsement of the technology by so many countries is testament to the strength of the technology and the compelling advantages it offers over other options. Other means of getting radio to consumers also exist, of course, including the Internet and audio channels over digital TV, but for a mass market free-to-air platform, the 'Eureka 147 Family' remains the system of choice.

Perhaps the most exciting development since this handbook was last published is the remarkable growth in consumer receivers for DAB and DMB, which are now plentiful and cheap. The market already boasts around 250 DMB different receiver models, in all shapes and sized from simple USB sticks to mobile phones and even navigation systems. There are 1,000 different digital radio receivers from hundreds of manufacturing names with consumer prices from as low as $25. Digital radios offering new features such as 'pause and rewind', electronic programme guides, pictures and text to complement the audio are all becoming popular, and there are no barriers to building the technology into other devices such as PMPs and mobile phones thanks to the advanced low-power chipsets available. This is always the hallmark of a successful and mature technology.

The drive behind the growth of the 'Eureka 147 Family' has been the consumer-led demand for interactive and innovative multimedia content. No other technology can offer the unique combination of content flexibility required by broadcasters, cost advantage and scalability of transmission networks, and the maturity of receivers as reflected in the choice available and low device cost. Broadcasters and manufacturers are responding with new features and services, exciting new ways to use the radio and enjoy its content. Radio is rapidly evolving to become a blend of passive and active content, audio and multimedia, listen live or listen later, even a social networking hub where listeners can share the same experience together!

Any broadcaster who really wants to engage with his or her audience, whatever their age, expectations and media demands, will find digital radio remains, like traditional radio, one of the most powerful ways to reach people whatever they are doing, wherever they may be. And the great thing about the Eureka 147 family is . . . 'it just works'.

Quentin Howard
President, WorldDMB Forum

Abbreviations

A/D	analogue/digital conversion
AAC	Advanced Audio Coding
ACI	adjacent channel interference
ACTS	Advanced Communications Technologies and Services
ADC	analogue to digital converter
ADR	Astra Digital Radio
ADSL	asymmetric digital subscriber line
AES	Audio Engineering Society
AES	Advanced Encryption Standard
AFC	automatic frequency control
AGC	automatic gain control
AGL	above ground level
AIC	Auxiliary Information Channel
AM	amplitude modulation
AMPS	Advanced Mobile Phone System
AMSS	AM Signalling System
API	application programming interface
ASCII	American Standard Code for Information Interchange
ASCTy	Audio Service Component Type
Asu	announcement support
ATM	asynchronous transfer mode
ATSC	Advanced Televison Systems Committee
AU	Access Unit (MPEG-4)
AV	Audio-visual
AWGN	Additive white Gaussian noise
BAL	Bit Allocation
BER	bit error ratio
BST	Band Segmented Transmission
BWS	Broadcast Web Site
C/N	carrier-to-noise ratio
CA	Conditional access
CAT	Conditional Access Table
CAZAC	constant amplitude zero auto-correlation

CCITT	Comité Consultatif International Téléphonique et Télégraphique (ITU-T)
CD	Compact Disk
CDF	cumulative distribution function
CDMA	code division multiple access
CENELEC	European Committee for Electrotechnical Standardisation, Brussels
CEPT	Conférence des Administrations Européenes des Postes etTelecommunications
CFB	Cipher Feedback Block Chaining
CIF	Common Interleaved Frame
CMAC	Cipher-based Message Authentication Mode
CMOS	complementary metal oxide semiconductor
COFDM	Coded Orthogonal Frequency Division Multiplex
CPU	Central processing unit
CRC	cyclic redundancy check
CTS	Composition Time Stamp
CU	capacity unit
CW	control word
CW	continous wave
D/A	digital /analogue conversion
DAB	Digital Audio Broadcasting
DAC	digital-to-analogue converter
DAD	destination address
DAT	Digital Audio Tape recording format
dBFS	relative signal level in Decibel (Full scale), related to the digital coding limit 0 dBFS
dBm	absolute signal level in Decibel, relative to 1 mW
dBrel	relative signal level in Decibel, related to a pre-defined reference level
dBu	absolute signal level in Decibel, related to a voltage of 0 dBu = 0.775 V
DC	Direct current
DCA	digitally controlled amplifier
DE	Downstream Entity
DECT	Digital European Cordless Telephone
DGPS	differential GPS
DIQ	digital in-phase and quadrature
DL	Dynamic Label
DMB	Digital Multimedia Broadcasting
DQPSK	differential quadrature phase shift keying
DRB	Digital Radio Broadcasting (Canada)
DRC	Dynamic Range Control
DRCD	DRC data
DRM	Digital Radio Mondiale
DRM	Digital Rights Management
DSCTy	Data Service Component Type
DSP	digital signal processor

DSR	Digital Satellite Radio
DTP	desktop publishing system
DTV	Digital television
DVB(-T) (-H)	Digital Video Broadcasting (-Terrestrial) (-Handheld)
DVD	Digital Video (or Versatile) Disk
e.i.r.p.	effective isotropic radiated power
EACEM	European Association of Consumer Electronics Manufacturers
EBU	European Broadcasting Union
EC	European Community
ECC	Extended Country Code
ECM	Entitlement Checking Message
EEP	Equal Error Protection
EId	Ensemble Identifier
EIT	Event Information Table
EMC	electromagnetic compatibility
EMM	Entitlement Management Message
EN	European Telecommunication Standard (ETSI document type; normative)
EOH	End of Header
EPG	Electronic Programme Guide
EPM	Enhanced Packet Mode
EPP	Ensemble Provider Profile
ERP, e.r.p.	effective radiated power
ES	European Standard (ETSI document type; normative)
ETI	Ensemble Transport Interface
ETS	European Telecommunications Standard (ETSI document type; normative)
ETSI	European Telecommunications Standards Institute
FBAR	film bulk acoustic resonator
FCC	Federal Communications Commission
FEC	forward error correction
FFT	fast Fourier transformation
FI	Frequency Information
FIB	Fast Information Block
FIC	Fast Information Channel
FIDC	Fast Information Data Channel
FIG	Fast Information Group
FM	frequency modulation
F-PAD	Fixed Programme Associated Data
FTP	File Transfer Protocol
GE06	Geneva 2006 frequency plan
GIF	Graphics Interchange Format
GPS	Global Positioning System
GSM	Global System for Mobile
HAAT	height above average terrain
HE-AAC	High Efficiency AAC

HECA	High Efficient Conditional Access
HF	high frequency
HiFi, hi-fi	High Fidelity
HMI	human machine interface
HTML	Hyper Text Markup Language
HuMIDAB	Human Machine Interface for DAB
I/Q	in-phase and quadrature
IBAC	In-Band-Adjacent-Channel
IBOC	In-Band-On-Channel
Id	identifier
IF	intermediate frequency
IFA	Internationale Funkausstellung (World of Consumer Electronics), Berlin
IFFT	inverse fast Fourier transformation
IM	intermodulation
IMDCT	inverse modified discrete cosine transformation
INT	IP/MAC Notification Table
IOD	Initial Object Description
IP	Internet Protocol
IP3	3rd order intercept point
IPDC	Internet Protocol Data Casting
IS	International Standard (ISO document type; normative)
ISDB	Integrated Services Digital Broadcasting
ISDN	Integrated Services Digital Network
ISO	International Standards Organisation
ITU(-R)	International Telecommunications Union, Radiocommunications Sector
ITU-T	International Telecommunications Union, Telecommunications Standardisation Sector
JESSI	Joint European Submicron Silicon Initiative
JML	Journaline Markup Language
JPEG	Joint Photographic Experts Group
kbps	kilobit per second
kW	kilowatt
LAN	local area network
LFE	Low Frequency Effect/Enhancement (channel/signal)
LI	logical interface
LNA	low noise amplifier
LSB	least significant bit
LTO	local time offset
LU	Loudness Unit
M/S	music/speech
M4M	Multimedia for Mobiles
MA02	Maastricht 2002 frequency plan
MAC	multiple accumulate
Mbps	Megabit per second

MCI	Multiplex Configuration Information
MEMO	Multimedia Environment for Mobiles
MFN	multiple frequency network
MJD	Modified Julian Date
MNSC	Multiplex Network Signalling Channel
MOT	Multimedia Object Transfer
MPE	multi-protocol encapsulation
MPEG	Moving Pictures Experts Group
MPS	MPEG Surround
MSB	most significant bit
MSC	Main Service Channel
MST	Main Stream Data
MTBF	mean time between failures
MUSHRA	MUltiple Stimuli with Hidden Reference and Anchor (test)
MUSICAM	Masking pattern-adapted Universal Sub-band Integrated Coding and Multiplexing
MW	medium wave
NAB	National Association of Broadcasters (USA)
NASC	Network Adapted Signalling Channel
NCO	numerically controlled oscillator
NICAM	Near Instantaneously Companded Audio
NIT	Network Information Table
NRSC	National Radio Systems Committee
OD	Object Description
ODA	Open Data Application
ODG	Objective Difference Grade (of quality assessment)
OFDM	Orthogonal Frequency Division Multiplex
OLED	organic light-emitting diode
OMA	Open Mobile Alliance
OOI	onset of impairment point
OSI	Open Systems Interconnections
PAD	Programme-associated Data
PAL	Phase Alternating Line
PAT	Programme Association Table
PC	parity check
PCM	pulse code modulation
PCR	Programme Clock Reference
PDH	plesiochronous digital hierarchy
PEAQ	Perceptual Evaluation of Audio Quality
PES	Packetized Elementary Stream
PI	Programme Identifier
PID	Packet Identifier
PL	Protection Level
PLL	phase locked loop
PMC	Production technology Management Committee (EBU)
PML	permitted maximum level

PMT	Programme Map Table
PNA	personal navigation assistant
PNum	Programme Number
POF	point of failure
PPM	peak programme level meter
pps	pulse per second
PRBS	pseudo random binary sequence
PRC	Prime Rate Channel
PS	Parametric Stereo (coding)
PSI/SI	Program Service Information/Service Information
PSK	phase shift keying
PSTN	Public Switched Telephone Network
PTy	Programme Type
QAM	quadrature amplitude modulation
QMF	quadrature mirror filter
QPSK	quaternary phase shift keying
RAM	random access memory
RBDS	Radio Broadcast Data System
RCPC	rate compatible punctured convolutional codes
RDI	Receiver Data Interface
RDS	Radio Data System
RF	radio frequency
RISC	reduced instruction set computer
RMS	root mean square value
RRC	Regional Radiocommunication Conference
RS	Reed-Solomon
RSS	Rich Site Summary *or* Really Simple Syndication
SAC	Spatial Audio Coding
SAD	Start Address
SAOC	Spatial Audio Object Coding
SAT	Sub-channel Assignment Table
SAW	surface acoustic wave
SBR	Spectral Band Replication (audio coding scheme addendum)
SC	Synchronisation Channel
SC Lang	Service Component Language
SCF-CRC	Scale Factor CRC
SCFSI	Scale Factor Select Information
SCT	Service Component Trigger
SD	Secure Digital
S-DAB	Satellite-based DAB
SDG	Subjective Difference Grade (of quality assessment)
SDH	synchronous digital hierarchy
SDT	Service Description Table
SES	Société Européenne des Satellites
SFN	single frequency network
SI	Service Information

SId	Service Identifier
SIM	Subscriber Identification Module
SL	Synchronisation Layer
SMPTE	Society of Motion Pictures & Television Engineers
SNR	signal-to-noise ratio
SPP	Service Provider Profile
SSTC	Single Stream Characterisation
STC-C(TA)	STI – Control – Transport Adapted Interface
STI	Service Transport Interface
STI(PI,X)	STI – Physical Interface
STI-C	STI – Control
STI-C(LI)	STI – Control – Logical Interface
STI-D	STI – Data
STI-D(LI)	STI – Data – Logical Interface
TA	Traffic Announcement
TCP	Transfer Control Protocol
T-DAB	terrestrial DAB
TDC	Transparent Data Channel
TDM	time-domain multiplex
T-DMB	Terrestrial Digital Multimedia Broadcasting
TFPR	Time-Frequency-Phase-Reference
TII	Transmitter Identification Information
TM	Transmission Mode
TMC	Traffic Message Channel
TNS	Temporal Noise Shaping
TP	Traffic Programme
TPEG	Transport Protocol Experts Group
TR	Technical Report (ETSI document type; informative)
TS	Technical Specification (ETSI document type; normative), Transport stream (MPEG)
TSDT	Transport Stream Description Table
TTI	Traffic and Travelers Information
TV, tv	television
UCS	Universal Character Set
UE	Upstream Entity
UEP	Unequal Error Protection
UHF	ultra high frequency
UMTS	Universal Mobile Telecommunication System
UTC	Coordinated Universal Time
UTF	Unicode Transformation Format
VCA	voltage controlled amplifier
VCO	voltage controlled oscillator
VHF	very high frequency
VLSI	very large scale integration
V_{pp}	peak-to-peak voltage
VU	Volume Unit

WARC	World Administrative Radio Conference
WCDMA	Wide Band Code Division Multiple Access
WFA	wave digital filter
WI95	Wiesbaden 1995 frequency plan
WLAN	wireless local area network
WMA	Windows Media Audio
XML	Extensible Markup Language
X-PAD	Extended Programme-associated Data

1

Introduction

1.1 General

The new digital radio system DAB (Digital Audio Broadcasting, nowadays often called *Digital Radio*) is a very innovative and universal multimedia broadcast system which will replace the existing AM and FM audio broadcast services in many parts of the world in the future. It was developed in the 1990s by the Eureka 147/DAB project. DAB is very well suited for mobile reception and provides very high robustness against multipath reception. It allows use of single frequency networks (SFNs) for high frequency efficiency.

Besides high-quality digital audio services (mono, two-channel or multichannel stereophonic), DAB is able to transmit programme-associated data and a multiplex of other data services (e.g. travel and traffic information, still and moving pictures, etc.). A dynamic multiplex management on the network side opens up possibilities for flexible programming.

In several countries in Europe and overseas broadcast organisations, network providers and receiver manufacturers are going to implement digital broadcasting services using DAB system in pilot projects and public services.

DAB works very differently from conventional broadcasting systems. Most of the system components such as perceptual audio coding, channel coding and modulation, multiplex management or data transmission protocols are new solutions and typically not so familiar to the expert in existing analogue or digital broadcast systems.

DAB was developed to a DAB system family, comprising the DAB (Digital Audio Broadcasting), DAB+ (extended DAB system for new audio coding schemes) and DMB (Digital Multimedia Broadcasting) systems.

The level of standardisation of the DAB system is rather advanced and the various recent international standards and related documents are introduced and referred to for easy access for the reader seeking technical details.

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

1.2 Radio in the Digital Age

Radio broadcasting is one of the most widespread electronic mass media comprising hundreds of programme providers, thousands of HF transmitters and billions of receivers worldwide. Since the beginning of broadcasting in the early 1920s, the market has been widely covered by the AM and FM audio broadcasting services.

Today we live in a world of digital communication systems and services. Essential parts of the production processes in radio houses were changed to digital ones in recent times, beginning with the change from conventional analogue audio tape to digital recording on magnetic tape or hard disk, digital signal processing in mixing desks and digital transmission links in distribution processes. In addition, there are also other digital distribution or storage media in a growing music market such as several digital tape or disk formats (CD, MiniDisk or DVD), or streaming and download formats (such as MP3) for distribution via the Internet (see also section 1.6.4).

Consequently, broadcast transmission systems now tend to change from conventional analogue transmission to digital. The first steps in the introduction of digital broadcasting services were taken by the systems NICAM 728 (Near Instantaneously Companded Audio Multiplex, developed by the BBC for stereo television sound in the VHF/UHF bands), DSR (Digital Satellite Radio, which was already shut down), or ADR (Astra Digital Radio), see section 1.6.1, but none were suited to replace the existing conventional services completely, especially for mobile reception. For that reason, the universal digital multimedia broadcasting system Eureka 147 DAB was developed and is now being introduced worldwide. In parallel, other digital broadcasting systems such as DRM (Digital Radio Mondiale, see section 1.6.3) or DVB-T (Digital Video Broadcasting, see section 1.6.2) are going to complement digital radio and television.

Normally, it takes a period of a human generation (or at least a period in the life of a receiver type generation, i.e. approximately 10 years) to replace an existing broadcasting system with a new one. Therefore, strong reasons and very convincing advantages are required to justify the introduction of such a new system.

1.3 Benefits of the Eureka 147 DAB Systems Family

1.3.1 The Original DAB System

As expected, there will always be some problems, or additional effort will be needed, when replacing an existing technology with a new one, such as:

- lack of transmission frequencies;
- costs for development and investment;
- looking for providers for new non-conventional services (e.g. data services);
- solving the chicken and egg problem (who will be first – the service provider or the receiver manufacturer?).

Nevertheless, the Eureka 147 DAB system family provides a wealth of advantages over conventional audio broadcast systems such as analogue VHF/FM or

AM radio, and also partly over other existing digital broadcast systems such as DSR (no longer available), ADR, etc. The following list will only highlight some key advantages; many more details will be explained in the corresponding sections of the book.

1.3.1.1 Quality of Service

DAB uses all the possibilities of modern digital communication technologies and can thus provide a much higher level of quality of service, such as:

- *Superior sound quality*: DAB users can enjoy pure undistorted sound close to CD quality. New features such as Dynamic Range Control (DRC) or Music/Speech Control can be used individually by customers to match the audio quality to their needs.
- *Usability*: Rather than searching wavebands, users can select all available stations or preferred formats from a simple text menu.
- *Perfect reception conditions*: With just a simple, non-directional whip antenna, DAB eliminates interference and the problem of multipath while in a car. It covers wide geographical areas with an even, uninterrupted signal. Once full services are up and running, a driver will be able to cross an entire country and stay tuned to the same station with no signal fade and without altering frequency.

1.3.1.2 Wide Range of Value-added Services

DAB is quite unique in that both music and data services can be received using the same receiver. One receiver does it all, such as:

- *Typical audio broadcasting (main service)*: Music, drama, news, information, etc., can be received in monophonic or stereophonic form as is well known from conventional radio programmes; there is also the potential to transmit multichannel (5.1 format) audio programmes as well.
- *Programme-associated data (PAD)*: DAB broadcast receivers can display text information in far greater detail than RDS, such as programme background facts, a menu of future broadcasts and complementary advertising information. Receivers attached to a small screen will display visual information such as weather maps or CD cover images.
- *Information services*: Services from sources other than the broadcasting station are included within the same channel for the user to access at will. These include news headlines, detailed weather information or even the latest stock market prices.
- *Targeted music or data services*: Because digital technology can carry a massive amount of information, specific user groups can be targeted with great accuracy because each receiver can be addressable.
- *Still or moving pictures*: Data can also appear as still or moving photographic pictures, accompanied by an audio service or as separate information.

1.3.1.3 Universal System Layout

The DAB system has a fairly universal and well-standardised system layout which allows applications for all known transmission media and receiving situations.

- *Standardisation*: The level of international standardisation of all basic principles and transmission tools for the new DAB system is very high (much more than 50 international standards cover all the necessary details).
- *Unique system design*: DAB services will be available mainly on terrestrial, but are also suited for cable and satellite networks, and the same receiver could be used to provide radio programmes and/or data services for national, regional, local and international coverage.
- *Wide choice of receivers*: It is possible to access DAB services on a wide range of receiving equipment including fixed (stationary), mobile and portable radio receivers, optionally completed with displays or screens, and even personal computers.

1.3.1.4 Flexibility of Multiplex Configuration

DAB services are transmitted in a flexible multiplex configuration, which can be easily changed instantaneously to the actual needs of the content providers.

- *Multiplex configuration*: The arrangement of services in a DAB multiplex may be changed instantaneously to match the needs of the providers of programmes or data services, without interrupting ongoing services.
- *Bit rate flexibility*: The programme provider can choose an appropriate bit rate for a certain audio programme according to its quality, for instance less than 100 kbps for a pure speech programme, 128 kbps for monophonic or 256 kbps for stereophonic music; also half sampling frequency can be used for lower quality services. So the available bit rate can be split optimally between different services of a DAB ensemble.

1.3.1.5 Transmission Efficiency

Compared to conventional broadcast systems much less economic effort in investment and operation is needed for a DAB transmission system.

- *Lower transmission costs for broadcasters*: DAB allows broadcasters to provide a wide range of programme material simultaneously on the same frequency. This not only makes room for a vastly increased number of programmes to increase user choice, but also has important broadcast cost-cutting implications.
- *Lower transmission costs for transmitter network providers*: For digital transmission a DAB transmitter needs only a fraction of the electrical energy compared to a conventional AM or FM transmitter.
- *Frequency efficiency*: DAB transmitter networks can be designed as Single Frequency Networks (SFNs), which saves a lot of transmission frequencies and thus transmission capacity on air.

These advantages of DAB (and there are more if we look further into the details) justify the introduction of DAB into the media world in order to replace the existing conventional radio systems step by step over a longer period.

1.3.2 Benefits of the Upgraded System DAB+

In the meantime, the WorldDMB (WorldDAB) Forum have developed an upgrade of the Eureka 147 DAB system called DAB+ in order to improve the audio coding efficiency using the new coding schemes of MPEG-4 HE AAC v2, see Chapter 3. This will provide additional advantages over the mentioned benefits of the original DAB system

- Latest MPEG-4 audio codec delivers *exceptional performance* efficiency.
- More *stations* can be broadcast on a multiplex, greater station choice for consumers available.
- Higher *frequency efficiency* of radio spectrum than with conventional DAB.
- Lower *transmission costs* for digital stations than with conventional DAB.
- New receivers are *backwards compatible* with existing MPEG Audio Layer II broadcasts, including scrolling text and multimedia services.
- Current MPEG Audio *Layer II services* remain unaffected.
- More *robust* audio delivery than with conventional DAB.
- Optimised for *live broadcast* radio.
- Broadcasters/regulators can select *different audio standards* – either MPEG-2 Audio Layer II, or the new MPEG-4 audio coding, or both, to suit their country.
- Fast re-tuning *response time* of the receiver (low zapping delay).
- Transmission of *MPEG Surround* is possible, with lower bit-rates than with conventional DAB.

Most of these benefits are also true for DMB (Digital Multimedia Broadcasting), see Chapter 9.

1.4 History of the Origins of DAB

1.4.1 Steps of Development

In the early 1980s the first digital sound broadcasting systems providing CD-like audio quality were developed for satellite delivery. These systems made use of the broadcasting bands in the 10 to 12 GHz region, employed very little sound data compression and were not aimed at mobile reception. Thus, it was not possible to serve a great majority of listeners, such as those travelling in cars. Also, another feature of the well-established FM radio could not be provided by satellite delivery, namely ‘local services’. Consequently terrestrial digital sound broadcasting was considered as an essential delivery method for reaching all listeners.

At first, investigations were initiated by radio research institutes looking into the feasibility of applying digital modulation schemes in the FM bands. However, the straightforward use of pulse code modulation (PCM) in the upper portions of the FM band generated intolerable interference in most existing FM receivers and was spectrally very inefficient. Mobile reception was never tried and would not have succeeded. A much more sophisticated approach was definitely necessary.

In Germany the Federal Ministry for Research and Technology (BMFT, now BMBF) launched a research initiative to assess the feasibility of terrestrial digital sound broadcasting comprising more effective methods of sound data compression and efficient use of the radio spectrum. A study completed in 1984 indicated that promising results could be expected from highly demanding research activities. As a new digital sound broadcasting system could only be implemented successfully by wide international agreement, BMFT set the task for its Project Management Agency at DLR (German Aerospace Centre) to form a European consortium of industry, broadcasters, network providers, research centres and academia for the development of a new digital audio broadcasting system. Towards the end of 1986 a consortium of 19 organisations from France, Germany, The Netherlands and the United Kingdom had signed a co-operation agreement and applied for notification as a Eureka project. At the meeting in December 1986 of the High Level Representatives of the Eureka partner states in Stockholm the project, now called 'Digital Audio Broadcasting, DAB', was notified as the Eureka 147 project. National research grants were awarded to that project in France, Germany and The Netherlands. However, owing to granting procedures official work on the project could not start before the beginning of 1988 and was supposed to run for four years.

Credit must also be given to the European Broadcasting Union (EBU), which had launched work on the satellite delivery of digital sound broadcasting to mobiles in the frequency range between 1 and 3 GHz, by awarding a research contract to the Centre Commun d'Etudes de Télédiffusion et Télécommunications (CCETT) in Rennes, France, prior to the forming of the DAB consortium. As the CCETT also joined the DAB project, the work already begun for the EBU became part of the DAB activities and the EBU a close ally and active promoter for DAB. Later, this proved very important and helpful in relations with the International Telecommunications Union (ITU-R) and the standardisation process with the European Telecommunications Standards Institute (ETSI).

From the beginning the goals set for the project were very demanding and difficult to achieve. Perfect mobile reception was the overall aim. In detail the list of requirements to be met included the following items:

- audio quality comparable to that of the CD;
- unimpaired mobile reception in a car, even at high speeds;
- efficient frequency spectrum utilisation;
- transmission capacity for ancillary data;
- low transmitting power;
- terrestrial, cable and satellite delivery options;
- easy-to-operate receivers;
- European or better worldwide standardisation.

The first system approach considered at least 16 stereo programmes of CD audio quality plus ancillary data to be transmitted in the 7 MHz bandwidth of a television channel. This definitely cannot be achieved by simply transmitting the combined net bit rates of 16 CD-like programme channels, which are around 1.4 Mbps each, over the TV channel. So a high degree of audio data compression without any perceptible loss of audio quality was mandatory. Data rates below 200 kbps per stereo channel had to be achieved.

Unimpaired mobile reception was also required to overcome the adverse effects of multipath signal propagation with the associated frequency selective fading.

Audio data compression and the transmission method became the central efforts of the research project. Both tasks were addressed in a broad and comprehensive manner. For audio coding four different approaches were investigated: two sub-band coding systems competed with two transform coding systems. Similarly, for the transmission method four different schemes were proposed:

- one narrow-band system;
- one single carrier spread-spectrum system;
- one multicarrier OFDM system;
- and one frequency-hopping system.

All approaches were developed to an extent where – either through experimental evidence or at least by thorough simulation – a fair and valid comparison of the performance of the proposed solutions became possible. The period of selection of and decision for the best suited audio coding system and the most appropriate transmission scheme was a crucial moment in the history of the Eureka 147 consortium. For audio coding the greatest part of the selection process happened externally to the consortium. All four coding schemes previously had been within the activities of the ISO/IEC Moving Pictures Experts Group (MPEG), which worked on standardisation for data compressed video and audio coding. There the solutions offered by Eureka 147 competed against 10 other entries from other countries, worldwide. The MPEG Audio Group set up a very elaborate and qualified audio quality assessment campaign that was strongly supported by Swedish Radio, the British Broadcasting Corporation and the Communications Research Centre, Canada, among others. The very thorough subjective audio quality tests revealed that the audio coding systems submitted by Eureka 147 showed superior performance and consequently were standardised by ISO/IEC as MPEG Audio Layers I, II and III. Within the Eureka 147 consortium, after long consideration Layer II, also known as MUSICAM, was selected for the DAB specification.

The process of choosing the most appropriate transmission method took place within the Eureka 147 consortium alone. In simulations performed according to rules worked out by the members the four approaches were put to the test. This showed that the broadband solutions performed better than the narrow-band proposal. Among the broadband versions the spread-spectrum approach had a slight advantage over the OFDM approach, while the frequency-hopping solution was considered too demanding with respect to network organisation. However, the OFDM system was the only one that was already available in hardware with field-test experience – in the form of the coded Orthogonal Frequency Division Multiplex (COFDM) system, while the spread-spectrum proposal by then was not developed as hardware at all and was estimated to be very complex. So, the choice fell on COFDM, which has since proven to be an excellent performer.

A further important decision had to be made relating to the bandwidth of the DAB system. From a network and service area planning point of view as well as obtainable frequency spectrum perspectives, an ensemble of 16 programmes on one transmitter with the 7 MHz bandwidth of a TV channel proved to be much too inflexible, although in experiments it had provided very good performance in a multipath environment. Therefore, a considerable but reasonable reduction in transmission bandwidth was necessary. In Canada experiments with the COFDM system revealed that substantial performance degradation begins around 1.3 MHz and lower. So, a reasonable bandwidth for a DAB channel or 'DAB block' was defined as 1.5 MHz. This allows several possibilities, as follows.

One 7 MHz TV channel can be divided into four DAB blocks, each carrying ensembles of five to seven programmes. With four blocks fitting into 7 MHz service area planning is possible with only one TV channel, without having adjacent areas using the same DAB block. Furthermore, 1.5 MHz bandwidth is sufficient to transport one MPEG coded audio/video bit stream.

After the above-mentioned important decisions had been made the members of the consortium all turned their efforts from their individual approaches to the commonly defined system architecture and with rapid progress developed the details of the complete basic DAB specification to be submitted to the international standardisation bodies. By that time, another European research project, the JESSI project AE-14, was awaiting the DAB specification to begin the development of chip-sets for DAB. Also, the standardisation bodies like ETSI were well aware of the submission of the specification since Eureka 147 members had been very active in testing and presenting the results of their research and development on many important occasions, together with or organised by the EBU.

The first official presentation of DAB took place at the World Administrative Radio Conference 1988 (WARC'88) in Geneva for the delegates of this conference. As the issue of a frequency allocation for digital sound broadcasting in the L-band around 1.5 GHz was up for decision, a demonstration simulating satellite reception was presented. A transmitter on Mont Salève radiated the DAB signal in the UHF TV band, giving in downtown Geneva an angle of signal incidence similar to that from a geostationary satellite. Mobile reception was perfect and the delegates to the conference were highly impressed. In consequence the conference assigned spectrum in the L-band for satellite sound broadcasting with terrestrial augmentation permitted.

One year later DAB was presented at the ITU-COM exhibition in Geneva. In 1990 tests with mobile demonstrations were run by Canadian broadcasters in Toronto, Ottawa, Montreal and Vancouver. DAB demonstrations and exhibits have been shown at all International Radio Shows (IFA) in Berlin and the UK Radio Festivals in Birmingham since 1991. Four International DAB Symposia have been held up to now: 1992 in Montreux, 1994 in Toronto, 1996 again in Montreux and 1999 in Singapore. In 1994 a mobile DAB demonstration was presented at the Arab States Broadcasting Union Conference in Tunis. From all these activities worldwide recognition and appreciation was gained for DAB.

The efforts and results of finding acceptance for DAB in the United States deserve an extra paragraph. As early as 1990 – by invitation of the National Association of Broadcasters (NAB) – the consortium presented DAB in a low key fashion at the NAB

Convention in Atlanta, Georgia. This led to a very elaborate mobile demonstration and exhibition at the next NAB Convention in 1991 in Las Vegas, Nevada. Several of the NAB officials by that time were very interested in reaching a cooperation and licence agreement with the Eureka 147 consortium. However, strong opposition against that system – requiring new spectrum and the bundling of several programmes onto one transmitter – also arose. The US radio industry – that is, the broadcasters – feared new competition from the licensing of new spectrum. They preferred the idea of a system approach named 'In-Band-On-Channel (IBOC)', where the digital presentation of their analogue programmes is transmitted together and within the spectrum mask of their licensed FM channel (see also section 1.6).

This, of course, would avoid the need of new licensing for digital broadcasting and thus keep new competition away. However appealing and spectrum efficient this concept may be, the realisation might prove to be a very formidable technical task. No feasible development was available at that time, but fast development of an IBOC system was promised. Eureka 147 DAB performed flawlessly at Las Vegas, but those opposing the system claimed that the topography around Las Vegas was much too favourable to put DAB to a real test. So it was requested that DAB should next come to the 1991 NAB Radio Show in San Francisco to be tested in a very difficult propagation environment. One main transmitter and one gap filler were set up and mobile reception in downtown San Francisco was impressively demonstrated. An announced demonstration of IBOC broadcasting did not take place as the equipment was not ready. In spite of the good results the opposition to DAB gained momentum and the NAB officially announced its preference for an IBOC system. This was not in line with the intentions of the Consumer Electronics Manufacturers Association (CEMA) of the Electronics Industry Association (EIA) which came to an agreement with the NAB to run very strictly monitored laboratory tests in Cleveland and field trials in San Francisco comparing the Eureka 147 DAB with several US IBOC and IBAC (In-Band-Adjacent Channel) systems. The results were to be presented to the Federal Communications Commission (FCC) for rule-making relating to digital sound broadcasting. While Eureka 147 soon had equipment ready for the tests, the US proponents for a long time could not provide equipment and delayed the tests until 1995. In the laboratory tests DAB outperformed all competing systems by far. Claiming to have been unfairly treated in the evaluation of the laboratory tests, all but one of the US proponents of terrestrial systems withdrew from the field tests in San Francisco.

For Eureka 147 the Canadian partner Digital Radio Research Inc. (DRRI) installed on contract a single frequency network of one main transmitter and two gap fillers to provide coverage for the area designated for mobile testing. Again DAB provided excellent performance as was documented in the final reports of the CEMA. Nevertheless, the United States still pursued the concept of IBOC although several generations of IBOC equipment and redesigns have only produced marginal performance. Even though numerous Americans now admit that DAB is a superior system they claim that it is not suited for US broadcasters. Finally, in October 2002, the Federal Communications Commission FCC approved In-Band On-Channel (IBOC) systems for the AM and FM band developed by the company iBiquity. Stations may implement digital transmissions immediately; however, AM stations may send the IBOC signal during the day only.

Standardisation in Europe and worldwide occurred at a better pace. The first DAB-related standard was achieved for audio coding in 1993, when the International Organisation for Standardisation/International Electrical Commission (ISO/IEC) released the International Standard IS 11172-3 comprising MPEG/Audio Layers I, II and III [IS 11172]. Also in 1993 ETSI adopted the basic DAB standard ETS 300 401, with several additional standards following later (now replaced by [EN 300401]). The ITU-R in 1994 issued Recommendations [BS.1114] and [BO.1130] relating to satellite and terrestrial digital audio broadcasting, recommending the use of Eureka 147 DAB mentioned as 'Digital System A'. Consumer equipment manufacturers have also achieved several standards concerning the basic requirements for DAB receivers issued by the Comité Européen de Normalisation Electrotechnique (CENELEC) (for more detailed information see section 1.5).

Even though the technology, the norms and standards had been developed, the most critical issue was still not resolved: provision of frequency spectrum for DAB. WARC'88 had allocated 40 MHz of spectrum in the L-band to satellite sound broadcasting, allowing also terrestrial augmentation. Through the intense intervention of several national delegations WARC'92 conceded primary terrestrial use of a portion of that allocated spectrum, which for several countries is the only frequency resource available. The L-band – very well suited for satellite delivery of DAB – on the other hand becomes very costly for terrestrial network implementation. VHF and UHF are much more cost efficient than the terrestrial L-band. There was no hope of acquiring any additional spectrum below 1 GHz outside of the bands already allocated to broadcasting.

So, in 1995 the Conference Europeénne des Administrations des Postes et des Télécommunications (CEPT) convened a spectrum planning conference for terrestrial DAB in Wiesbaden, Germany, that worked out an allotment plan for DAB in VHF Band III (mostly former TV Channel 12) and in the L-band from 1452 to 1467 MHz, allowing for all CEPT member states two coverages of DAB, one in the VHF range, the other in the L-band. This decision made possible the installation of experimental DAB pilot services in many countries of Europe and the beginning of regular services starting in 1997 with Sweden and the United Kingdom. More and more countries – also outside of Europe – are following suit. As the available spectrum was not sufficient to move all existing FM and future programmes to DAB, new frequency bands have been opened for DAB in a later step (see also Appendix 2).

1.4.2 Organisations and Platforms

A few of the organisations and bodies were or still are busy promoting and supporting the development and introduction of DAB system worldwide:

1.4.2.1 Eureka 147 Consortium

As mentioned above the Eureka 147 consortium was the driving force in the development of DAB. It formed a Programme Board, dealing with strategic planning, contractual and legal affairs, membership and promotion, a Steering Committee, planning the tasks of the working groups and making all technical decisions, and four working groups of varying task assignments. DLR in Cologne, Germany, was chosen to act as the managing agency and as the project office for Eureka affairs.

While the original consortium did not accept new members at the beginning, this policy was changed when the basic specification of DAB [EN 300401] was ready to be released to the public for standardisation in 1992. The Eureka 147 Programme Board established rules for the entry of new partners into the consortium, requiring, for example, an entrance fee of DM 150,000 from industrial companies while entry for broadcasters and network operators was free of charge. The money earned from the entrance fees was mainly reserved for system promotion and to a smaller extent for organisational expenses. In 1993 the first new members were admitted and soon the consortium grew to 54 members from 14 countries.

The research work of the original partners had led to a number of basic patents for DAB, individually held by several members. Consequently the consortium came to the conclusion to offer licenses as a package to all necessary intellectual property rights (IPR) through authorised agents, one for receiver matters and another for transmitter and measuring equipment.

In 1997 the idea of merging Eureka 147 with the promoting organisation World-DAB (see below) was discussed with the result that a gradual merger was adopted and the final merger completed by the beginning of the year 2000. The activities of the EU-147 Project Office were transferred to the WorldDAB Project Office by the end of 1998. With the complete merger at the end of 1999 Eureka 147 ceased to exist. The members of the consortium now cooperate in the Technical Committee of WorldDMB, where technical issues of DAB, DAB+ and DMB are handled (see below).

1.4.2.2 National DAB Platforms

The first national DAB platform *DAB Plattform e.V.* was initiated by the German Ministry for Education and Research in 1991. Founded as a national platform it soon accepted members from Austria and Switzerland and consequently dropped the word 'national' from its name. It soon reached a membership of 52 organisations. The main objective of this platform was the promotion of the DAB system in German-speaking countries. The organisation of and activities for DAB presentations at the IFA events in Berlin were highlights in the promotion programme of the German platform. The members decided to end the existence of DAB Plattform e.V. by autumn 1998 after considering it to have achieved its objectives.

Likewise, national platforms were established in several other European countries. France launched the *Club DAB France*, The Netherlands the Dutch *DAB Foundation* and the United Kingdom the UK DAB Forum. Promotional activities for the public and information to governmental agencies were again the objectives.

1.4.2.3 EuroDAB/WorldDAB/WorldDMB Forum

Strongly stimulated and supported by the EBU, an organised cooperation of national platforms and other interested bodies lead to the founding of the *EuroDAB Forum in* 1995. A EuroDAB Project Office was set up at EBU Headquarters in Geneva.

Membership rose quickly, bringing together interested organisations not only from Europe but from many parts of the world. Accordingly the General Assembly of EuroDAB decided to change its name to the *WorldDAB Forum* in 1997. Promotion of DAB is again the main objective. Formerly EuroDAB and now WorldDMB issue a quarterly *DAB Newsletter*. The Forum has organised all but the first International DAB Symposia mentioned above. Technical, legal and regulatory as well as promotional issues are dealt with in several modules of the Forum. Since January 2000 the Eureka 147 consortium has fully merged with the WorldDAB Forum. The WorldDAB Project Office moved from Geneva to London in 1998 and has also acted for Eureka 147 since 1999. An extensive web-site of WorldDAB and actual DAB or DMB information is maintained at this office [www.WorldDAB]. Later on the WorldDAB Forum changed its name to WorldDMB Forum, an international non-government organisation whose objective it is to coordinate the implementation of all Eureka147-based technologies, such as DAB, DAB+ and DMB. It has expert committees on technical, regulatory and marketing issues, as two regional committees with emphasis to Europe and Asia. At the time of writing, more than 70 companies and organisations from more than 25 countries worldwide were organised with the WorldDMB Forum.

1.4.2.4 WorldDMB Asia Committee

The Asian DAB Committee of the former WorldDAB Forum was formally established in June 2000 in Singapore. The main focus of the former AsiaDAB Committee was to raise awareness of DAB in Asia and to speed up the roll-out of DAB in this important part of the world. Nowadays, the WorldDMB Asia committee encourages cooperation and sharing of experiences on DAB matters as a part of the WorldDMB Forum. It is a valuable platform for industry players and regulators in the Asian marketplace. For more details see [www.WorldDAB].

1.4.3 Milestones of Introduction

At the time of writing (autumn 2008) DAB and DMB – after almost 20 years' development and standardisation – are operational radio broadcast systems in many countries of the world. However, in several countries still not all services are available on DAB and the receiver market is rather small. Although DAB was primarily developed in the Eureka 147 project as a European broadcast system, it can be shown already in this phase that most of the industrially developed regions worldwide (i.e. Asia, Australia, and parts of America such as Canada and South America) are now going to introduce this highly sophisticated universal broadcast system, too. For example, DMB providing video services to mobile and portable receivers has found much attraction in South Korea, where operational networks have been established. The worldwide penetration of DAB is mainly supported by the WorldDAB respectively WorldDMB Forum.

The actual status of introduction can always be found at the web-site given below. At the time of writing, about 16 countries worldwide operate regular services in DAB or DMB, and about 30 additional countries are dealing with trial or test services. Although the DAB system is still in a status of introduction, at this time over 500 million people around the world can receive more than 1000 different DAB services using one of more than 1000 different types of receivers available. (It is not practical to list any details here, because this will change significantly in a short time. For up-to-date information see [www.WorldDAB]).

Eureka147 DAB is also introduced in Canada, using the trade mark Digital Radio Broadcasting (DRB). Only in the important market places of United States and Japan have proprietary digital radio systems been developed, see section 1.6.

1.5 International Standardisation

The DAB system family shows a very high level of standardisation in its principles and applications, which is rather unusual for a new broadcasting or multimedia system. (There are more than 50 corresponding international standards and related documents.)

After the initial development by some broadcasters and related research institutes, the first international standards were passed by the ITU-R (International Telecommunications Union, Radiocommunications Sector). In the period following, the system has been developed within the Eureka 147 project, supported by a wide representation of telecommunication network providers, broadcasters, receiver manufacturers and related research institutes, in close cooperation with the EBU (European Broadcasting Union).

In the following some basic standards of the DAB system are listed. A more complete listing of the corresponding standards and related documents can be found in the Bibliography, which is referred to within various parts of the text.

1.5.1 Basic Requirements and System Standards

The basic ITU-R Recommendation [BS.774] shortly specifies as '*Digital system A*' the main requirements to the new broadcasting system DAB. Other ITU-R Recommendations [BS.789] and [BO.1130] regulate conditions needed for additional frequency ranges for emission.

Several ITU World Administrative Conferences (WARC'79, WARC'85, WARC'88, WARC'92) dealt with digital radio and allocated frequency bands for satellite and terrestrial digital sound broadcasting. Detailed frequency plans for Europe and Asia were developed at several CEPT planning meetings [CEPT, 1995], [CEPT, 2002a], [CEPT, 2007a] and an ITU-R Regional Radiocommunication Conference [RRC06, 2006], see Chapter 7 and Appendix 2.

As a main result of developments within the Eureka 147 project, the *DAB Standard or DAB Specification* [EN 300401] was approved by the ETSI European Telecommunications Standards Institute, which defines the characteristics of the DAB transmission signal, including audio coding, data services, signal and service multiplexing, channel coding and modulation.

1.5.2 Audio Coding

DAB represents one of the most important applications of the generic MPEG-1 Layer II audio coding scheme [IS 11172]. The use of this ISO/IEC coding standard is recommended by the ITU-R in [BS.1115], and certainly in the DAB Specification [EN 300401]. DAB is also designed to transmit MPEG-2 Layer II audio [IS 13818], for instance for lower quality half-sample-rate transmission or multichannel audio programmes, see also [ES 201755]. For the use with the DAB+ or DMB systems newer coding schemes such as HE-AAC or SBR were standardised, see Chapter 3.

Other standards specify procedures and test bit-streams for DAB audio conformance testing [TS 101757], or audio interfaces for transmission within the studio region [IEC 60958], [IEC 61937].

1.5.3 Data and Multimedia Services

Owing to the special needs for data services in DAB a new standard for Multimedia Object Transfer (MOT) was created defining the specific transport encoding of data types not specified in [EN 300401] and ensuring interoperability between different data services and application types or equipment of different providers or manufacturers. Additional guidelines are given in the 'MOT Rules of operation' [TS 101497], 'Broadcast Web-site application' [TS 101498] or 'Slide show application' [TS 101499]. An important bridge between the well-known radio data service RDS for FM [EN 50067] and the data services in DAB is provided by the European Norm [EN 301700] which defines service referencing from FM-RDS and the use of RDS-ODA (open data applications). A transparent data channel for DAB transmission is described in [TS 101759]. Late standards were given for Internet Protocol (IP) datagram tunnelling [TS 101735] and for the DAB Electronic Programme Guide (EPG) [TS 102818]. Of course, DAB is also considered an important bearer for newer developments in traffic and traveller information such as TMC and TPEG.

After first trials and implementation in the 1990s, the DAB transmission system became an attractive candidate for mobile multimedia services – both MPEG and IP based – in the first decade of the new millennium. Consequently, the standard was extended to carry MPEG transport streams and IP packets. A short historical summary focusing on these developments is provided in the introduction to Chapter 9.

1.5.4 Network and Transmission Standards

Based on the DAB main standard [EN 300401] additional standards are given to define the DAB multiplex signal formats for distribution (networking) and transmission. This is the so-called Service Transport Interface (STI) [EN 300797] and [TS 101860] for contribution networks between service providers and broadcast studios, the Ensemble Transport Interface (ETI) [EN 300799], and the Digital baseband In-phase and Quadrature (DIQ) interface [EN 300798] for DAB channel coding using OFDM modulation.

To provide interactive services, transport mechanisms for IP Datagram Tunnelling [ES 201735], Network Independent Protocols [ES 201736] and Interaction Channel Through GSM/PSTN/ISDN/DECT [ES 201737] are defined.

1.5.5 Receiver Requirements

Based on the DAB specification [EN 300401] additional standards are given to define the implementation of DAB receivers. [EN 50248] describes DAB receiver characteristics for consumer equipment for terrestrial, cable and satellite reception. Details are given in Chapter 8. EMC parameters for receivers are identified in EACEM TR-004. [EN 50255] specifies the Receiver Data Interface (RDI) between DAB receivers and peripheral data equipment. A special command set to control receivers is described in [EN 50320]. The ETSI Technical Report [TR 101758] lists general field strength considerations for a DAB system. More general requirements to radio receivers concerning EMC are given in [EN 55013] and [EN 55020].

1.5.6 Guidelines for Implementation and Operation

In addition to the given standards, more detailed guidelines and rules for implementation and operation are compiled in [TR 101296] as a main guideline document for service providers and manufacturers. A guide to standards, guidelines and bibliography is given in the ETSI Technical Report [TR 101495]. A broadcaster's introduction to the implementation of some key DAB system features [BPN 007] is provided by the EBU.

1.6 Relations to Other Digital Broadcasting Systems

In addition to the European DAB system which is mainly covered by this book, several other digital sound broadcasting services exist which have been or are being developed and (partly) introduced.

These systems differ in many aspects (specifications, service complexity, parameters) from the DAB concept. Some are focused more strongly on a single application or service (e.g. stationary reception) or provide lower audio quality levels (such as WorldSpace or DRM). Except for ISDB-T, which uses technologies very similar to DAB, all other systems are not expected to be able to comply with the audio quality and quality of service in mobile reception provided by DAB. Nevertheless the basic concepts and limitations of some of these systems will be introduced briefly.

Also, there are a few cable-based digital radio services, which are mainly derived from existing satellite or terrestrial radio systems (for instance, ADR, DVB, etc.) so it may not be necessary to describe them separately. In general, those systems use QAM schemes in order to achieve high data rates in the limited bandwidth of the cable. This is possible because of the high signal-to-noise ratio available. However, these services have only local importance, depending on the extension of the broadband cable distribution network used.

1.6.1 Satellite-based Digital Radio Systems

Although the DAB system can also be used for satellite transmissions, a number of different proprietary systems have been designed by companies providing direct broadcasting satellite services. Of course, many radio programmes are transmitted via telecommunication satellites to feed relay transmitters or cable head stations. These systems are beyond the scope of this section.

Generally, the system layout in the direct broadcasting systems focuses either on stationary reception with highly directive antennas (e.g. 'dishes') in the 11 GHz range (e.g. ADR) or on portable and possibly mobile reception in the L-band (1.5 GHz) and S-band (2.3/2.5/2.6 GHz) allocations (e.g. WorldSpace, XM and Sirius).

Historically, the first system for the direct satellite broadcasting of digital radio services was Digital Satellite Radio (DSR). In contrast to later systems, no sound compression was used but only a slight reduction in data rate as compared to CD by using a scale factor. This service has recently been closed down because of the end of the lifetime of the satellite used (Kopernikus). Therefore, it is not described here. More details can be found in [Schambeck, 1987].

The basic building blocks of modern satellite systems are audio coding, some kind of multiplexing, and modulation. While there are things in common with DAB with respect to the first two aspects, these systems do not use OFDM modulation because of the non-linear travelling wave tube amplifiers on satellites and the low spectral efficiency. Instead, often relatively simple PSK schemes are used, which are considered sufficient because, when using directional receiving antennas, multipath propagation does not occur in satellite channels.

Satellite systems face severe problems when the service is to be extended to mobile reception. Because only line-of-sight operation is possible owing to the limited power available in satellite transmissions, at higher latitudes, where geostationary satellites have low angles of elevation, either terrestrial retransmission in cities and mountainous regions is necessary, or the service has to be provided by several satellites at different azimuths in parallel to fill shaded areas. Another approach is to use satellites in highly inclined elliptical orbits which always show high elevation angles. However, this requires several satellites in the same orbit to provide continuous service and difficult switching procedures have to be performed at hand-over from the descending to the ascending satellite.

1.6.1.1 Astra Digital Radio

The Astra Digital Radio (ADR) system was designed by Société Européenne des Satellites (SES), Luxembourg, to provide digital radio on its geostationary direct TV broadcasting satellites. This co-positioned family of satellites labelled 'ASTRA' covers Central Europe operating in the 11 GHz range. Brief system overviews are given by [Hofmeir, 1995] or [Kleine, 1995]. A detailed description of the system is available at the ASTRA web-site [www.ASTRA].

The system design is compatible with the sub-carrier scheme for analogue sound transmission within TV transponders. It uses MPEG Layer II sound coding (as does DAB) at a fixed bit rate of 192 kbps for a stereo signal. Forward error correction is applied using a punctured convolutional code with code rate 3/4. The data are differentially encoded to provide easy synchronisation. QPSK modulation is used for each of up to 12 ADR sub-carriers which are combined with the analogue TV signal and are FM modulated. The baseband bandwidth of the digital signal is 130 kHz which is the same as for an analogue sound sub-carrier. This allows existing analogue radio services to be replaced by the digital ADR service one by one. It is also possible to use a whole transponder for ADR only. In this case, 48 channels can be accommodated in one transponder.

Additional data can be sent together with the audio signal. ADR uses the auxiliary data field of the MPEG audio frame to do this, but the coding is different from the DAB Programme Associated Data (PAD; see Chapter 2). ADR uses 252 bits per audio frame for this purpose, which results in a net data rate of about 6 kbps, because a (7,4) block code is used for error correction. This capacity is flexibly split into 'RDS data', 'Ancillary Data' and 'Control Data'. RDS data are coded in the format of the RDS [EN 50067] which is used on a digital sub-carrier on FM sound broadcasts and provides service-related information such as labels and Programme Type (see also section 2.5). The reason for using this format on ADR is that several broadcasters use ADR to feed their terrestrial FM networks. Hence these data can be extracted from the ADR signal and fed to the RDS encoder of the FM transmitter. Ancillary data are used by the broadcaster for in-house purposes. The control data field contains a list of all ADR channels to provide easy tuning and allows transmission of parameters for conditional access if applied.

With respect to DAB, ADR in principle provides the same quality of audio, because the same coding mechanism is used. The number of channels available, however, is much larger. While DAB currently only provides six to seven services in each network, ADR can provide 12 digital radio channels on each transponder in addition to a TV channel. Therefore, several hundred channels are available and in use by public and private broadcasters from all over Europe. ADR decoders are integrated in high-end satellite TV receivers and are therefore widespread. The most important drawback of ADR is that it can only be received with stationary receivers using a dish antenna. The flexibility of the system with respect to data transmission and future multimedia extensions is considerably lower than that of DAB.

1.6.1.2 WorldSpace

The WorldSpace digital radio system was designed by the US company WorldSpace Inc., to provide digital radio to developing countries on its two geostationary satellites 'AfriStar' covering Africa, Southern Europe and the Near and Middle East, and 'AsiaStar' covering India, China, Japan and South East Asia. A third satellite named 'CaribStar' is intended to cover Central and South America but has not yet been launched. The system downlink operates in the L-band. The standard is not public. A description is given by [Sachdev, 1997], for example.

The basic building blocks of the systems are 'Prime Rate Channels' (PRCs) which each transmit 16 kbps of data. Several of these (typically up to eight, resulting in 128 kbps) can be combined to provide a stereo sound channel or data channels. For audio coding, MPEG-1 Layer III (MP3) is used.

Each satellite serves three coverage areas (spots). For each spot, two multiplex signals containing 96 PRCs (i.e. 12 stereo channels) each in a time domain multiplex (TDM) are available. Each of these data streams is QPSK modulated. One multiplex is completely assembled in a ground station and linked up to the satellite from which it is retransmitted without further processing. The other multiplex is processed on board the satellites by assembling individual PRC uplinks to the TDM signal. The advantage of this is that there are no additional costs to transport the signal from the broadcaster to the uplink station. Moreover, a signal can be broadcast in several spots with only one uplink.

Owing to the lower path loss of the L-band signals as compared to 11 GHz, the WorldSpace system is able to provide significant field strength within the coverage area. Therefore reception is possible with portable receivers outdoors and inside buildings using a low-gain patch or helical antenna at a window where line-of-sight conditions apply. Minor attenuation by foliage or trees may be tolerable.

With respect to DAB, WorldSpace addresses a completely different broadcasting environment. Clearly, in developing countries digital satellite radio will considerably improve the number and quality of broadcast receptions, because in many parts up to now there has only been short-wave coverage. Mobile reception is probably not an urgent need and hence the system is not particularly designed for it.

From the point of view of audio quality, by using MPEG Layer III at 128 kbps the quality will be comparable to that of DAB at 160 to 192 kbps. At this bit rate, however, the number of channels per spot is restricted to 24. Owing to the large coverage areas this number may not be sufficient to serve the multilingual and multiethnic audience. Hence, the audio bit rate will have to be reduced in order to increase the number of channels available. Nevertheless the sound quality will in many cases still be comparable at least to mono FM, which is still a major improvement on short-wave AM.

Utilising the concept of PRCs the system is very flexible and can be extended to data transmission and multimedia contents. This is important because the lifetime of the satellites is approximately 15 years.

1.6.1.3 Satellite Systems in the United States

In the United States there is no allocation for digital radio in the L-band, but in the S-band. Originally, two companies have established satellite services targeted at mobile reception and covering the whole of the continental United States (i.e. except Alaska and Hawaii). Sirius Satellite Radio uses three satellites on inclined orbits to achieve high elevation angles and terrestrial repeaters in metropolitan areas, see [www.Siriusradio]. The other company, XM Satellite Radio, uses two high-powered geostationary satellites at 85°W and 115°W which both cover the entire continental United States and thus provide spatial diversity to overcome shading, see [www.XMRadio]. Nevertheless terrestrial retransmitters are also widely used, in particular in metropolitan and mountainous areas. In 2007, both companies merged.

The systems provide approximately 100 audio channels in a quality comparable to CD. The services are pay-radio based on a monthly subscription fee. With respect to sound quality these systems should be comparable to DAB, and with respect to reliability of service these systems should be comparable to WorldSpace; however, because of the measures taken they should show better performance in mobile and portable reception. The number of subscribers to these systems has reached almost 20 million at the time of writing.

1.6.2 Terrestrial Digital Broadcasting Systems

1.6.2.1 Digital Video Broadcasting – Terrestrial (DVB-T)

The DVB system was developed a few years after the DAB system. On the audio and video coding level as well as on the multiplexing level, it is completely based on the MPEG-2 standard. This is in contrast to the DAB standard, where the audio coding is MPEG, but the multiplex control is independent and specially adapted for the requirements. There are three DVB standards that differ very much in the transmission scheme: DVB-S for satellite [EN 300421], DVB-C for cable [EN 300429], and DVB-T for terrestrial broadcasting [EN 300744]. Here we consider only the last one. For a detailed description of all three standards, see [Reimers, 1995].

The DVB-T standard has been designed for stationary and portable terrestrial reception with multipath fading. In contrast to DAB, mobile reception was not required. On the coding and modulation level, many methods have been adopted from the DAB system, where they had already been implemented successfully.

Like DAB, DVB-T uses OFDM with a guard interval. There are two transmission modes. The first is called 8K mode (because it uses an 8192-point FFT) and has a symbol length of the same order as in DAB transmission mode I. It is suited for single frequency networks. The second is called 2K mode (because it uses a 2048-point FFT) and has a symbol length of the same order as in DAB transmission mode II. It is not suited for single frequency networks. The guard interval can be chosen to be 25% of the total symbol length, as for DAB, but also shorter guard intervals are possible. The total bandwidth of about 7.6 MHz is suited for terrestrial 8 MHz TV channels. The system parameters can be scaled for 7 MHz TV channels. A major difference from the DAB system is that DVB-T uses coherent modulation. Different signal constellations between QPSK and 64-QAM are possible. At the receiving side, the channel amplitude and phase have to be estimated. For channel estimation, nearly 10% of the total bandwidth is needed for pilot symbols. The coherent modulation scheme with channel estimation by a Wiener filter is more advanced than the differential demodulation scheme of DAB. It is even more robust against fast fading in mobile reception situations [Hoeher, 1991a], [Schulze, 1998], [Schulze, 1999].

The OFDM transmission scheme for DVB-T includes a frequency interleaver that consists of a simple pseudo-random permutation of the sub-carrier indices. It is very similar to the DAB frequency interleaver. In contrast to DAB, the DVB-T system has no time interleaving. Time interleaving only makes sense for mobile reception, which was not a requirement for the design of the system.

The channel coding is based on the same convolutional codes like the DAB system. Code rates between 1/2 and 7/8 are possible. Unequal error protection (UEP) is not possible for the DVB system, since the data and transmission level are completely separated. As a further consequence, in contrast to DAB, the whole multiplex has to be coded with the same code rate. To reach the very low bit error rates that are required for the video codec, an outer Reed–Solomon (RS) code is applied: blocks of 188 bytes of useful data are coded into blocks of 204 bytes. Between the outer RS code and the inner convolutional code, a (relatively) short byte interleaver has been inserted to break up the burst errors that are produced by the Viterbi decoder.

The data rates that can be carried by the DVB-T multiplex vary from about 5 Mbps to about 30 Mbps, depending on the modulation (QPSK, 16-QAM or 64-QAM), the code rate (1/2, 2/3, 3/4, 5/6 or 7/8) and the guard interval.

In the meantime DVB-T2 has been developed [EN 302 755]. It includes further options for the configuration of the OFDM parameters (including a 1k, 4k, 16k and 32k Mode in addition to the 2k and 8k Mode of DVB-T and additional choices for guard interval length) and uses advanced error correction codes.

Mobile reception of DVB-T, even though not a specific design feature, has been widely discussed and gave rise to many theoretical investigations and practical experiments. Furthermore, there seems to be a controversy between DAB and DVB adherents that concentrates very much on this question. Unfortunately, even though both systems are very close together, the chance has been missed to design a universal common system. From the technical point of view, we will list some comments on the question of mobile reception:

1. Unlike as one might expect, the differential modulation of DAB is not more robust than the coherent modulation for DVB even for severely fast-fading mobile reception. The coherent scheme is more robust and spectrally more efficient, see [Schulze, 1999].
2. For mobile reception in a frequency flat fading situation, the frequency interleaving even over a relatively wide band of 8 MHz is not sufficient. As a consequence, the lack of time interleaving may severely degrade the system in some mobile reception situations, see [Schulze, 1998].
3. Synchronisation losses may sometimes occur for mobile receivers. In contrast to DVB-T, this was taken into account in the design of the DAB system: the null symbol allows a rough frame and symbol synchronisation every 24 ms or a maximum of 96 ms. The FIC (see section 2.2.2.) can then immediately be decoded and provides the information about the multiplex structure. And, in the (typical) stream mode, the data stream does not need separate synchronisation because it is synchronised to the physical frame structure.
4. Bit errors in the scale factors of the MPEG audio data (*birdies*) are very annoying for the human ear. For a subjective impression this is much worse than, say, block error in a picture. The DAB system uses an extra CRC (that is not part of the MPEG bit-stream) to detect and to conceal these errors. This is not possible for DVB-T.

As a consequence, the DVB-T system allows mobile reception in many typical environments, but it may fail in some situations where more care has been taken in the DAB system.

To remedy the shortcomings of DVB-T with respect to portable and mobile reception of television and multimedia services within the DVB system family, DVB-H was developed [EN 302304]. Besides improvements in the transmission system itself (additional interleaving and forward error correction) it allows for time slicing of the applications to reduce the power consumption of portable devices. It can co-exist with DVB-T-services in one channel and hence allows services targeted to portable and mobile devices to be offered and – with higher data rate – for fixed receivers in the same channel. A comparison of DVB-H and DAB in particular with respect to features relevant to mobile reception is given by [Sieber, 2004]. Economic aspects of this comparison are covered by [Skiöld, 2006] and [Weck, 2006].

1.6.2.2 In-Band-On-Channel Systems

In the United States the broadcasting environment is considerably different from the European one. Technical parameters of the stations (AM or FM, power, antenna height) resulting in different coverage areas and quality of audio service appear to be important economic factors that are vital to the broadcasters and which many of them think have to be preserved in the transition to a digital system. Although the DAB system can provide such features to a certain extent by applying different audio data rates and coding profiles, the idea of multiplexing several stations into one transmitting channel does not seem to be very attractive to some US broadcasters.

In the early 1990s, a number of US companies therefore started to work on alternative approaches to digital audio radio (DAR) broadcasting known as IBOC (In-Band-On-Channel) and IBAC (In-Band-Adjacent-Channel). Systems were developed both for AM and FM stations in MW and VHF Band II. The basic principle is to use low-power digital sideband signals within the spectrum mask of the channel allocated to the station.

Later in the 1990s a number of IBOC/IBAC systems and the Eureka 147 DAB system and a satellite-based system (VOA/JPL) were tested in both the laboratory and field in the San Francisco area. The results of these tests were not encouraging for the IBOC/IBAC approaches, see [Culver, 1996], stating explicitly that 'of all systems tested, only the Eureka 147/DAB system offers the audio quality and signal robustness performance that listeners would expect from a new DAR service'. Since then work on IBOC systems has nevertheless continued. At the time of writing (2008), standards both for FM and AM IBOC systems have been issued and the system is in the market in the US.

The system designed for Medium Wave ('AM IBOC') has two basic configurations [Johnson, 2003]. The first is designed to be used in parallel to the analogue AM signal. It has OFDM blocks on either side of the AM carrier (even in the frequency band used by the AM side bands). However, the modulation and power level is different for the subcarriers depending on the frequency offset from the carrier. The carriers near to the AM carrier are QPSK modulated, those more than about 4 kHz from the carrier use 16-QAM and those more than 9.5 kHz from the carrier use 64-QAM. These have the highest power and carry most of the data. The total bandwidth of the signal is 29.4 kHz. This means that the digital transmission is not really on-channel but uses both adjacent channels for the original transmission. Therefore, the digital signal can only be broadcast during daytime, when there is no sky wave propagation. In a situation where no analogue AM transmission is required, the whole channel can be used for the digital transmission. In this case, the unmodulated AM carrier is still transmitted but the AM channel is filled with the 64-QAM modulated OFDM carriers.

The FM IBOC system [Peyla, 2003] is designed in a similar way. There is a hybrid mode where two OFDM blocks are sent on either side of the band used by the analogue FM signal. They extend from ±129 kHz to ±198 kHz, respectively and both carry the same digital information. The digital part of the signal can be extended by adding further OFDM carriers between ± 101 kHz and ± 129 kHz. If there is no analogue FM signal, the inner part of the channel can also be filled with the digital transmission resulting in an OFDM block consisting of 1092 carriers in a band 396 kHz wide. In all cases, the two digital side bands carry identical data to increase the robustness of the system.

Both the FM and AM IBOC systems use PAC as the audio coding system. However, independent of the individual system details there are two principle problems with the FM IBOC approach which are difficult to overcome.

Owing to the fact that the FM IBOC signal is limited in bandwidth to the FCC channel spectrum mask, the bandwidth is not sufficient to overcome frequency-selective fading in the VHF frequency range. This means that in portable or mobile reception a fade of the whole signal can occur. In this case there is hardly a chance to reconstruct the signal even if strong error correction codes and 'long' interleaving are used (think of stopping at a traffic light). Therefore, from the point of view of coverage, IBOC systems are not expected to be superior to FM, because they will fail in the same locations as FM does. In the FM IBOC system it is proposed to delay the analogue signal against the digital one and store it in the receiver. Whenever the digital transmission fades, the receiver would blend to the analogue signal.

Every IBOC signal will to a certain extent impair the analogue signal in the same channel and in adjacent channels. The effect of this is additional noise on the analogue audio signal and this may affect different analogue receiver circuits in different ways. Therefore the broadcaster has no control of what the digital signal will do to all the different analogue receivers being used. To keep this noise sufficiently low, the digital signal level must be kept far below the level of the analogue signal, e.g. 23 dB below the power of the FM carrier in iBiquity's FB IBOC system. Although the digital system requires less signal-to-noise ratio than FM, this means that the coverage of IBOC will be very limited.

The system is well introduced in the US and field trials are under way also in other countries [Joachim, 2008].

1.6.2.3 Integrated Services Digital Broadcasting (Japan)

In Japan, the NHK Science and Technical Research Laboratories have proposed a concept called Integrated Services Digital Broadcasting (ISDB). From this approach a system was created which can be configured for terrestrial and satellite broadcasts of radio, TV and multimedia services, see [Kuroda, 1997], [Nakahara, 1996], [ARIB, 1999]. For the terrestrial system (ISDB-T) the most important aims are rugged reception with portable and mobile receivers, use of Single Frequency Networks (SFNs) to achieve frequency efficient coverage and to have a flexible scheme in order to be future-proof. To achieve this, a variant of OFDM (see section 2.2) was developed which is called Band Segmented Transmission (BST)-OFDM and means that each signal consists of a number of basic OFDM building blocks called segments with a bandwidth of 571, 500 or 428 kHz (in TV channels with 8, 7 and 6 MHz bandwidth) which are spread over the band where frequencies are available. For the OFDM parameters of the system there are three sets of parameters that are similar to transmission modes I, II and IV of the DAB system. The system may be configured for channel bandwidth of 6, 7 or 8 MHz resulting in a segment bandwidth.

A single segment is sufficient to broadcast audio programmes and data, but it is also possible to combine three segments for this purpose. For TV signals 13 segments are combined to form a 7.4/6.5/5.6 MHz wide signal. In all cases the basic modulation and coding modes of the segments are (D)QPSK, 16QAM and 64QAM, and several codes rates between 1/2 and 7/8 are available. In the 3 and 13 segment modes the individual segments may be coded and modulated in different ways to achieve hierarchical transmission capacity. For the audio and video coding the MPEG-2 standard will be used including AAC for sound transmission.

In 1998 draft standards on digital terrestrial television and radio based on the ISDB-T concept were established, see [STRL, 1999], and field tests are being performed. Test transmissions took place in Tokyo and Osaka in late 2003.

From the point of view of the relation to DAB, ISDB-T basically uses the same principles and hence provides similar features as do DAB and DVB-T, respectively. By using the MPEG-2 standard for coding, the audio and video quality will be comparable. Due to the use of AAC for audio coding, however, the number of audio services which can be provided in a given bandwidth will be approximately twice that provided by DAB. Owing to the similarity of the OFDM parameters used, and similar bandwidth (at least when three segments are combined for audio), the coverage properties will also be similar to those of DAB. The advantage of ISDB-T is that the radio and TV systems are based on the same basic building blocks (although the coding and modulation will be different), which allows for reuse of some circuits in receivers, whereas DAB and DVB-T differ in many details. Another advantage of ISDB-T is the availability of a narrow-band mode (i.e. a single segment) which can deliver one or two radio programmes to small regions or communities, although potentially with some reception problems due to flat fading. With this segmentation ISDB-T can also flexibly use the frequencies not occupied by analogue services in the introductory simulcast phase. Such possibilities are not present in DAB.

1.6.3 Digital Radio Mondiale (DRM)

1.6.3.1 DRM in the Broadcasting Bands below 30 MHz

The frequency range below 30 MHz (short, medium, long wave) offers the possibility to provide a very large coverage area for broadcasting with only one high-power transmitter and very simple receivers with no directional antennas. Using this frequency range is the easiest – and often practically the only possible – way to cover large countries with low technical infrastructure. On the other hand, traditional broadcasting in this frequency range uses the most antiquated and simplest possible transmission scheme: double-sideband AM with a carrier. From one point of view this is a waste of power and bandwidth efficiency in such a valuable frequency range at a time when every year more and more efficient communication systems are being designed. From a practical point of view, this old technique provides only very poor audio quality and thus finds acceptance only in regions where no other service is available.

The obvious need for a more advanced system working in this frequency range gave rise to a consortium called *Digital Radio Mondiale* (DRM) to develop a worldwide standard for such a system.

The DRM standard [ES 201980] was first published as a European Telecommunications Standard and has now been approved by ITU-R and IEC. Overviews of the basic system features are given e.g. by [Stott 2001] and [Hofmann, 2003], latest information is available from [www.DRM].

The available bandwidth in the frequency range below 30 MHz is typically very small: the usual channel spacing is 5, 9 or 10 kHz. For the new system, it should be possible to bundle two 5 kHz channels to one 10 kHz channel. For such narrow-band channels, much more data reduction is required than for the DAB system. The *Advanced Audio Coding* (AAC) data reduction scheme of MPEG in conjunction with a newer technique called *Spectral Band Replication* (SBR) (see also section 3.2.4) allows for an audio quality comparable to FM at a data rate between 20 and 25 kbps, see for example [Dietz, 2000]. This is a great improvement compared to conventional AM quality. But bearing in mind that the DAB system typically transmits little more than 1 Mbps (e.g. six audio programmes of 192 kbps) in a 1.5 MHz bandwidth, even this low data rate is relatively high compared to the extremely small bandwidth of 10 kHz. This means that the DRM transmission scheme must provide more than twice the bandwidth efficiency compared to DAB.

The physical channel depends on the type of wave propagation, which is very special in this frequency range. It differs from night to day and it depends on the solar activity. Also interference caused by human noise plays an important role. In situations where only ground-wave propagation is present, there is a simple white Gaussian noise channel that is relatively easy to handle. But typically the wave propagation is dominated by ionospheric scattering (sky wave). This means that the channel is time and frequency variant – just like the mobile DAB channel (see section 2.1). Travel time differences of the wave of the order of several ms may occur which cause severe frequency-selective fading. The motion of the ionosphere causes a time variance in the channel with a Doppler spectrum with a typical bandwidth in the order of 1 Hz.

The DRM channel has similar problems in time-and frequency-selective fading as the DAB channel, so the DRM group decided on the same solution: OFDM (see section 2.1). The parameters, however, are totally different from the ones used in DAB because the channel characteristics are quite different. For sky wave propagation there are several modes with the OFDM symbol durations of 26.67 ms, 20 ms and 16.67 ms and guard intervals of 5.33 ms, and 7.33 ms, respectively. With this symbol length, the OFDM signal consists of approximately 200 sub-carriers in a 10 kHz wide channel. The requirement of a very high-bandwidth efficiency leads to a much higher signal constellation for each sub-carrier than used for DAB. 64-QAM or 16-QAM will be used depending on the propagation situation. Because of the very large time variance of the channel, a large amount of the total bandwidth (up to about 1/6) is needed for channel estimation pilots. The overall code rate is between 1/2 and 2/3 for the required data rate. It can be shown [Schulze, 1999] that such a coded 64-QAM OFDM system with a good channel estimator is surprisingly robust in a fading channel even with conventional 64-QAM symbol mapping. For DRM it has been decided to use the multilevel coding approach first proposed by [Imai, 1977]. This approach is very similar to that described by [Wörz, 1993] for a coded 8-PSK system. With multilevel 64-QAM, an additional gain of up to 1 or 2 dB can be reached compared to conventional 64-QAM. DRM is the first system that implements such a coded modulation scheme; see also [Dietz, 2000]. The same convolutional codes as DAB will be used here.

For ground mode propagation, another transmission mode is defined, because less channel estimation is needed, the guard interval is only 2.33 ms long, and even weaker codes can be used in a Gaussian channel to allow for a higher data rate.

Channel coding must be supported by interleaving to work in a fading channel. Frequency interleaving alone is not sufficient, especially if the echoes are too short or if only a two-path propagation situation occurs. Time interleaving is restricted by the delay that is allowed by the application: after the receiver has been switched on the listener cannot wait too long for a signal. A delay of 2 seconds is used for sky wave modes, and a delay of 0.5 s is used in ground wave. Since the time variance is very slow (1 Hz Doppler spread), the interleaving will not be sufficient in many situations.

1.6.3.2 DRM in the VHF Bands

In the course of the development of DRM it became clear that the system developed for the bands below 30 MHz could not be used in the VHF range. However, a demand was seen for a VHF capable system sharing many features with DRM to allow for a combined introduction of a digital system for all traditional sound broadcasting bands. Within the DRM consortium a system specification called DRM+ was developed which is in the process of field testing [Steil, 2008] and standardisation within the DRM standard as a new mode 'E' at the time of writing.

With DRM it shares MPEG-4 AAC+ audio coding, the multiplexing structure, service signalling etc. The most prominent difference of course is in the OFDM parameters. DRM+ has a bandwidth of 96 kHz and a symbol duration of 2.5 ms, with a 0.25 ms guard interval. This choice of parameters allows operation of single transmitters and single frequency networks in all VHF broadcasting bands upto 108 MHz. With net data rates of up to 286 kbps DRM+ is capable to carry up to four high quality audio services, the main focus, however, is on a single service per frequency.

1.6.4 Internet Radio (Web-casting)

A completely different way to provide radio services is the use of the Internet as a universal multimedia platform. That means, Internet radio – nowadays often called Web-casting – is a multimedia extension of the Internet. Audio and video signals can be digitised into a file or stream of data, which can easily be distributed over the existing Internet structures. The term Web-casting means the publishing of audio and video files or streams on web-sites, for live and/or on-demand delivery to the general public.

A main function of broadcaster's web-sites is to support the core radio and TV services in order to make them more attractive to the customer. The basic equipment for an Internet radio customer is a PC, complete with a sound adapter, a rather fast modem and Internet access, possibly via ISDN or ADSL. But also special 'Internet radio players' in the shape of a normal radio receiver are available that connect to Internet radio stations by Wireless LAN and high speed Internet access.

The availability of improved and scaleable audio coding schemes, on the other hand, is creating the opportunity for a better definition of audio quality obtainable via the Web. The Internet's impact on broadcasters has changed from its initial position as a site for posting information related to the station and programme to an additional medium to reach new customers through streaming audio, live or on-demand, still and moving pictures, and new interactive concepts. The importance of Internet audio services, which are using only a small part of the capacity of the Net, is already significantly increasing audio information delivered via IP. Several software companies have produced the necessary tools to distribute audio and video via the Internet with the goal to develop the Web as a new mass medium, similar to the traditional audio and video broadcasting services.

For both, either for live streaming or downloading of audio clips, there are several audio players (software codecs) in use which can provide sufficient audio quality. The most common way to distribute Internet radio is via streaming technology using a lossy audio codec. Popular streaming audio formats use MP3 (close to MPEG-1/2 Layer III), Ogg Vorbis, Windows Media Audio, RealAudio and MPEG-2/4 HE-AAC (sometimes called aacPlus, see also section 3.2.3). The bits are 'streamed' (transported) over the network in TCP or UDP packets, then reassembled and played within seconds.

The advent of such a large number of audio codecs as mentioned above has brought a radically new approach to standardisation: standards have become less important for the Internet transmission technology, since decoders (which are normally simple and do not require a lot of processing power) are downloadable to the client machine along with the content.

Therefore, in the Internet environment there is no longer a need for a single coding system as is the case in conventional broadcasting. Service providers decide which coding scheme to use. One of the advantages of this 'deregulated' approach is that decoders can be regularly updated as the technology advances, so the user can have the latest version of the decoder all the time.

Simulcast is a portmanteau of 'simultaneous broadcast', and refers to programmes or events broadcast across more than one medium, or more than one service on the same medium, at the same time. For example, Virgin radio is simulcast on both AM and on satellite radio, and the BBC's Prom concerts are often simulcast on both BBC radio and BBC television. Another application is the transmission of the original-language soundtrack of movies or TV series over local or Internet radio, with the television broadcast having been dubbed into a local language.

Apart from Internet radio, a so-called podcast is a series of audio or video digital-media files which is distributed over the Internet by download, through Web feeds, to portable media players or simply to a PC. Podcast is distinguished from other digital-media formats by its ability to be subscribed to, and downloaded automatically when new content is added. The term is a portmanteau of the words 'iPod' and 'broadcast', as the Apple iPod being the brand of portable media player for which the first podcasting script was developed.

The business model of audio streaming is likely to change due to the advent of multicasting. Today, ISPs charge per audio stream. In multicasting situations, however, a single stream will be delivered to several users. The user will then be charged according to the occupancy of the servers used. Due to the huge competition in the audio decoder market, audio streamers will be increasingly available for free.

With current audio codecs and streaming techniques high audio quality can be achieved and many broadcasters provide streams of up to approximately 100 kbps which provide almost CD quality. Even high quality 5.1 surround sound was already demonstrated [Kozamernik, 2006]. Internet radio, respectively web-casting, has conquered its own place in the radio world because of the access to hundreds of radio programmes worldwide. The drawback is that high speed links with flat rate marketing models are currently only available for fixed installations. Mobile usage of high speed Internet access using 3G mobile communication networks is still limited and expensive. It can be expected that Internet radio will become increasingly popular when Internet access will become cheaper on portable and mobile devices.

More detailed information on web-casting and its audio quality expectations is contained in [BPN 022], [BPN 035] and [Stoll, 2000]. The necessary Internet Transport protocols and delivery technologies are described in [Kozamernik, 2002], [Kozamernik, 2005].

2

System Concept

2.1 The Physical Channel

Mobile reception without disturbance was the basic requirement for the development of the DAB system.

The special problems of mobile reception are caused by multipath propagation: the electromagnetic wave will be scattered, diffracted, reflected and reaches the antenna in various ways as an incoherent superposition of many signals with different travel times. This leads to an interference pattern that depends on the frequency and the location or – for a mobile receiver – the time.

The mobile receiver moves through an interference pattern that changes within milliseconds and that varies over the transmission bandwidth. One says that the mobile radio channel is characterised by time variance and frequency selectivity.

The time variance is determined by the vehicle speed v and the wavelength $\lambda = c/f_0$ where f_0 is the transmission frequency and c the velocity of light. The relevant physical quantity is the maximum Doppler frequency shift:

$$f_{D\max} = \frac{v}{c} f_0 \approx \frac{1}{1080} \frac{f_0}{MHz} \frac{v}{km/h} Hz \tag{2.1}$$

Table 2.1 shows some practically relevant figures for $f_{D\max}$.

The actual Doppler shift of a wave with angle α relative to the vector of the speed of the vehicle is given by

$$f_D = f_{D\max} \cos(\alpha). \tag{2.2}$$

Typically, the received signal is a superposition of many scattered and reflected signals from different directions so that we may speak not of a Doppler shift but of a Doppler spectrum. The most popular Doppler spectrum is the Jakes spectrum that corresponds to the isotropic distribution of α (see e.g. [Schulze, 2005], p. 58). Figure 2.1 shows an example of a received VHF signal level for a fast-moving car ($v = 192$ km/h) as a function of the time for a carrier wave of fixed frequency $f_0 = 225$ MHz.

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

Table 2.1 Examples for Doppler frequencies

$f_{D\max}$	$v = 48$ km/h	$v = 96$ km/h	$v = 192$ km/h
$f_0 = 225$ MHz	10 Hz	20 Hz	40 Hz
$f_0 = 900$ MHz	40 Hz	80 Hz	160 Hz
$f_0 = 1500$ MHz	67 Hz	133 Hz	267 Hz

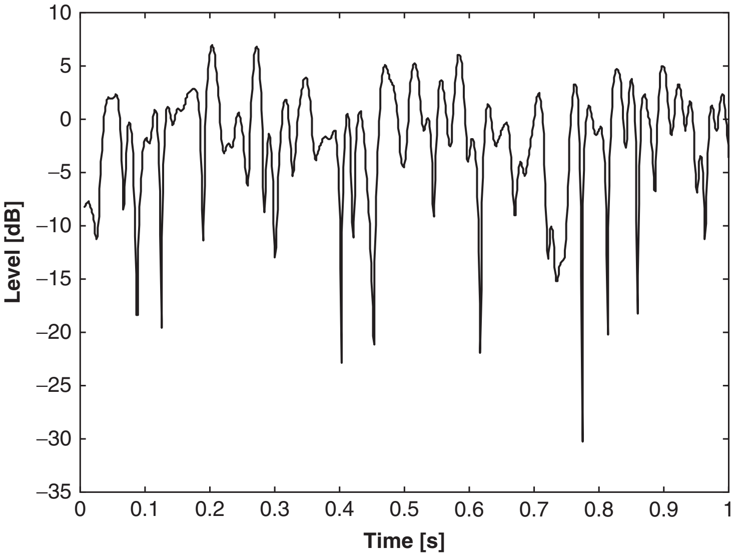

Figure 2.1 Time variance of multipath fading

The superposition of Doppler-shifted carrier waves leads to a fluctuation of the carrier amplitude *and* the phase. This means the received signal has been amplitude and phase modulated by the *channel*. Figure 2.2 shows the trace of the phasor in the complex plane. For digital phase modulation, these rapid phase fluctuations cause severe problems if the carrier phase changes too much during the time T_S that is needed to transmit one digitally modulated symbol. Amplitude and phase fluctuate randomly. The typical frequency of the variation is of the order of $f_{D\max}$. Consequently, digital transmission with symbol time T_S is only possible if

$$f_{\mathrm{Dmax}}\ T_s << 1 \tag{2.3}$$

The frequency selectivity of the channel is determined by the different travel times of the signals. They can be calculated as the ratio between the travelling distances and the velocity of light. Table 2.2 shows some typical figures.

Travel time differences of some microseconds are typical for cellular mobile radio. For a broadcasting system for a large area, echoes up to 100 μs are possible in a hilly or

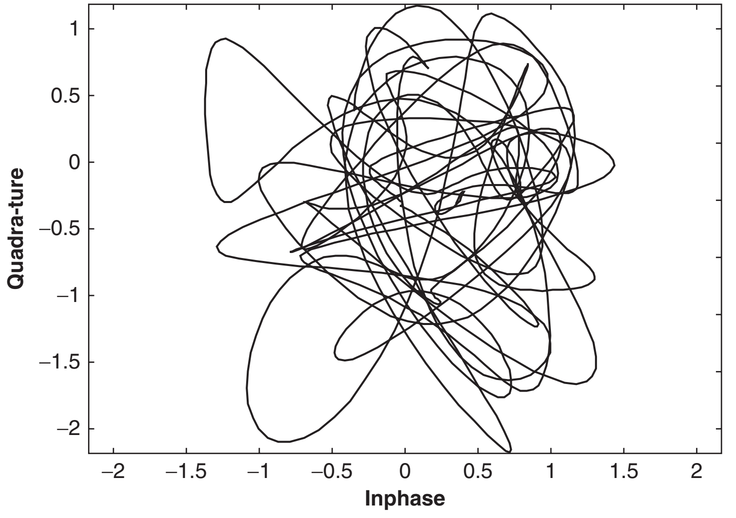

Figure 2.2 Time variance as a curve in the complex plane

Table 2.2 Examples for travel times of the signal

Distance	300 m	3 km	30 km
Time	1 μs	10 μs	100 μs

mountainous region. In so-called single frequency networks (see Chapter 7), the system must cope with even longer echoes. Longer echoes correspond to more fades inside the transmission bandwidth. Figure 2.3 shows an example of a received signal level as a function of the frequency at a fixed location where the travel time differences of the signals correspond to several kilometres. In the time domain, intersymbol interference disturbs the transmission if the travel time differences are not much smaller than the symbol duration T_S. A data rate of 200 kbps, for example, leads to $T_S = 10$ μs for the QPSK (Quadrature Phase-Shift Keying) modulation. This is of the same order as the echoes. This means that digital transmission of that data rate is not possible without using more sophisticated methods. Known techniques are equalisers, spread spectrum and multicarrier modulation. Equalisers are used in the GSM standard. The data rate for DAB is much higher than for GSM and the echoes in a broadcasting scenario are much longer than in a cellular network. This would lead to a higher complexity for the equaliser. Spread spectrum is spectrally efficient only for cellular networks where it is used as multiple access (CDMA), as in the UMTS standard, see e.g. [Schulze, 2005]. For DAB it was therefore decided to use multicarrier modulation, because it is able to cope with very long echoes and it is easy to implement.

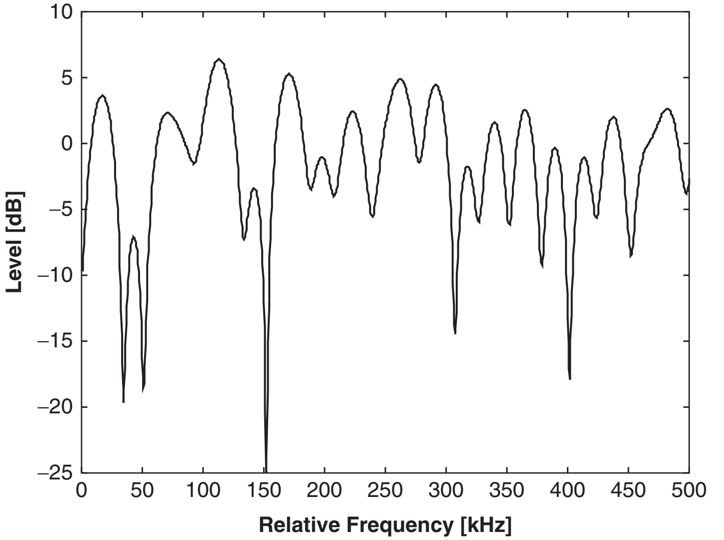

Figure 2.3 Frequency selectivity of multipath fading

Further propagation related aspects which are relevant for DAB are covered in Chapter 7. For a more detailed treatment of the mobile radio channel, we refer to textbooks like [Schulze, 2005], [Benedetto, 1999], [Proakis, 2008], [Kammeyer, 2008] and [David, 1996].

2.2 The DAB Transmission System

In the DAB transmission system several advanced techniques are implemented, such as OFDM multicarrier modulation, rate-compatible punctured convolutional (RCPC) codes, and time-frequency interleaving. In the context of the development of DAB+ (see section 3.4.2) and DMB (see Chapter 9) further the technique of concatenating the RCPC codes with Reed-Solomon (RS) – Codes was adopted. Here we only give a brief overview of the techniques applied in the DAB transmission system. A more detailed discussion of these topics with the emphasis on OFDM systems (including DAB) can be found in [Schulze, 2005]. We further refer to the comprehensive textbooks about digital transmission [Benedetto, 1999], [Proakis, 2008], [Kammeyer, 2008].

2.2.1 Multicarrier Modulation

To cope with the problem of intersymbol interference caused by long echoes, DAB uses the type of multicarrier modulation known as OFDM (Orthogonal Frequency Division Multiplex). The simple idea behind multicarrier modulation is to split up the high-rate data stream into K parallel data streams of low data rate and to modulate each of them separately on its own (sub-)carrier. This leads to an increase of the symbol duration T_S by a factor of K. For sufficiently high K, it is possible to keep T_S significantly longer than the echo duration and to make the system less sensitive to intersymbol interference.

OFDM is a spectrally very efficient kind of multicarrier modulation, because it minimises the frequency separation between the individual carriers by allowing some controlled spectral overlap between the carriers, without causing adjacent channel interference (ACI). This goes back to the mathematical property of orthogonality that gave the name to OFDM.

It is easy to understand an OFDM signal $s(t)$ as a kind of signal synthesis by a finite Fourier series defined by

$$s(t) = \sum_{k=-K/2}^{K/2} z_k \cdot e^{j2\pi kt/T}. \tag{2.4}$$

It is defined on an interval (Fourier period) of length T. The complex Fourier coefficients z_k carry the digitally coded information. For each time interval of length T, another set of $K+1$ information carrying coefficients can be transmitted. In many practical systems including DAB, the DC coefficient for $k = 0$ will not be used (i.e. is set to zero) for reasons of hardware implementation. The Fourier synthesis can be interpreted as a modulation of each complex modulation symbol z_k on a complex carrier wave exp ($j2\pi kt/T$) with frequency k/T ($k = \pm 1, \pm 2, \ldots, \pm K/2$). The signal $s(t)$ is the complex baseband signal and has to be converted to an RF signal by means of a quadrature modulator. At the receiver side, Fourier analysis of the downconverted complex baseband signal will produce the complex symbols using the well-known formula

$$z_k = \frac{1}{T}\int_0^T e^{-j2\pi kt/T} s(t)dt, \tag{2.5}$$

which results from the orthogonality of the carrier waves. Both Fourier analysis and synthesis will be implemented digitally by the FFT (Fast Fourier Transform) and IFFT (Inverse FFT) algorithms. The transmission chain is shown in Figure 2.4.

The part of the OFDM signal that transmits the K complex coefficients z_k is called the OFDM symbol.

To make the transmission more robust against long echoes, the OFDM symbol period T_S will be made longer than the Fourier period T by a so-called cyclic prefix or guard interval of length Δ simply by cyclic continuation of the signal. A synchronisation error smaller than Δ will then only lead to a frequency-dependent but constant phase shift. Echoes are superpositions of ill-synchronised signals and will cause no intersymbol interference, but a constant phasor, as long as the delays are smaller than Δ. For DAB, differential quadrature phase shift keying (DQPSK) is used so that this constant phase cancels out at the demodulator.

The length of T_S is limited by the requirement that the phase fluctuations must be small (Equation 2.3).

On the other hand, long echoes require a long guard interval and a long T_S. To keep the system flexible for different physical situations, four Transmission Modes (TMs) with different parameter sets have been defined, see Table 2.3.

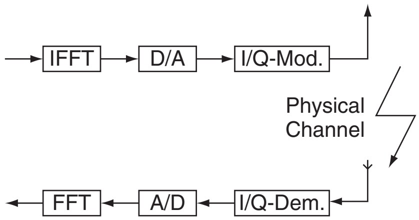

Figure 2.4 FFT implementation of OFDM

Table 2.3 The OFDM parameters for the four DAB transmission modes

Mode	K	$1/T$	T_s	Δ	max. frequency
TM I	1536	1 kHz	≈1246 μs	≈246 μs	≈375 MHz
TM II	384	4 kHz	≈312 μs	≈62 μs	≈1.5 GHz
TM III	192	8 kHz	≈156 μs	≈31 μs	≈3 GHz
TM IV	768	2 kHz	≈623 μs	≈123 μs	≈750 MHz

The product of the number of sub-carriers K and the spacing $1/T$ between them is the same for all transmission modes and determines the total signal bandwidth of approximately 1.5 MHz. The parameters of all transmission modes can be easily scaled into each other. The ratio Δ/T is always the same. The last column in Table 2.3 gives a rule of thumb for the maximum transmission frequency due to the phase fluctuation caused by the Doppler effect. A car speed of 120 km/h and a physical channel with no line of sight (the so-called isotropic Rayleigh channel with Jakes Doppler spectrum, see [Schulze, 2005], [David, 1996]) has been assumed. These values correspond to $f_{\mathrm{Dmax}}\ T_S = 0.05$, which is a rule-of-thumb upper limit where the DAB system begins to fail. This is based on theoretical curves as well as on simulations. For a more detailed treatment of that topic, we refer to [Schulze, 2005].

Transmission mode I with the very long guard interval of nearly 250 μs has been designed for large-area coverage, where long echoes are possible. It is suited for single frequency networks with long artificial echoes; 200 μs correspond to a distance of 60 km, which is a typical distance between transmitters. If all transmitters of the same coverage area are exactly synchronised and send exactly the same OFDM signal, no signal of relevant level and delay longer than the guard interval will be received. Since the OFDM symbol length T_S is very long, transmission mode I is sensitive against rapid phase fluctuations and should only be used in the VHF region.

Transmission mode II can cope with echoes that are typical of most topographical situations. However, in mountainous regions, problems may occur. This mode is suited for the transmission in the L-band at 1.5 GHz.

Transmission mode III has been designed for satellite transmission. It may be suited also for terrestrial coverage, if no long echoes are expected.

The parameters of TM IV lie just between mode I and II. It was included later in the specification to allow for more flexibility in broadcasting planning.

2.2.2 The Frame Structure of DAB

For each transmission mode, a transmission frame is defined on the physical signal level as a periodically repeating structure of OFDM symbols that fulfil certain tasks for the data stream. It is an important feature of the DAB system (and in contrast to the DVB system) that the time periods on the physical level and on the logical (data) level are matched for audio transmission. The period T_F of the transmission frame is either the same as the MPEG-1 and MPEG-2 Audio Layer II frame length of 24 ms or an integer multiple of it. As a consequence, the audio data stream does not need its own synchronisation. This ensures a better synchronisation stability especially for mobile reception.

The structure for TM II is the simplest and will thus be described first. The transmission frame length is 24 ms. Its first two OFDM symbols build up the Synchronisation Channel (SC). The next three OFDM symbols carry the data of the Fast Information Channel (FIC) that contains information about the multiplex structure and transmitted programmes. The next 72 OFDM symbols carry the data of the Main Service Channel (MSC). The MSC carries useful information, such as audio data or other services. Figure 2.5 shows the transmission frame structure. It is also valid for TMs I and IV.

All these OFDM symbols in a transmission frame of TM II have the same duration $T_S \approx 312$ µs, except for the first one. This so-called null symbol of length $T_{Null} \approx 324$ µs is to be used for rough time synchronisation. The signal is set to zero (or nearly to zero) during this time to indicate physically the beginning of a transmission frame. The second OFDM symbol of the SC is called the TFPR (Time–Frequency–Phase Reference) symbol. The complex Fourier coefficients z_k have been chosen in a sophisticated way so that this symbol serves as a frequency reference as well as for channel estimation for the fine tuning of the time synchronisation. Furthermore, it is the start phase for the differential phase modulation. Each of the following OFDM symbols carries 384 DQPSK symbols corresponding to 768 bits (including redundancy for error protection, see below). The three OFDM symbols of the FIC carry 2304 bits. Because they are highly protected with a rate 1/3 code, only 768 data bits remain. The FIC data of each transmission frame can be decoded immediately without reference to the data of other transmission frames, because this most important information must not be delayed. The 72 OFDM symbols of the MSC carry 55 296 bits, including error protection. This corresponds to a (gross) data rate of 2.304 Mbps. The data capacity of 55 296 bits in each 24 ms time period is organised in so-called Capacity Units (CUs) of 64 bits. In the MSC many audio programmes and other useful data services are multiplexed together. Since each of them has its own error protection, it is not possible to define a fixed net data rate of the DAB system.

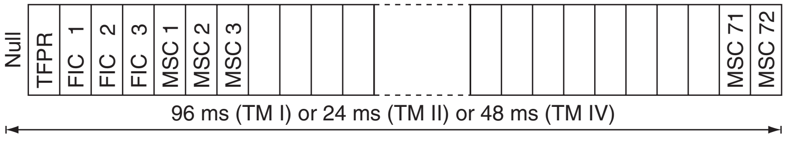

Figure 2.5 Transmission frame structure

The transmission frames of TMs I and IV have exactly the same structure. Since the OFDM symbols are longer by a factor of 4 or 2, respectively, the transmission frame length is 96 ms or 48 ms. The number of bits in the FIC and MSC increases by the same factor, but the data rate is always the same.

For TM III, the frame duration is $T_F = 24$ ms. Eight OFDM symbols carry the FIC, and 144 OFDM symbols carry the MSC. The data rate of the FIC is higher by a factor of 4/3 compared to the other modes. The MSC always has the same data rate.

For all four transmission modes, the MSC transports 864 CUs in 24 ms. There is a data frame of 864 CUs = 55 296 bits common for all transmission modes that is called the Common Interleaved Frame (CIF). For TMs II and III, there is exactly one CIF inside the transmission frame. For TM I, there are four CIFs inside one transmission frame of 96 ms. Each of them occupies 18 subsequent OFDM symbols of the MSC. The first is located in the first 18 symbols, and so on. For TM IV, there are two CIFs inside one transmission frame of 48 ms. Each of them occupies 36 subsequent OFDM symbols of the MSC.

2.2.3 Channel Coding

2.2.3.1 Convolutional Coding of the Logical Frames

The DAB system allows great flexibility in the choice of the proper error protection for different applications and for different physical transmission channels. Using rate compatible punctured convolutional (RCPC) codes introduced by [Hagenauer, 1988], it is possible to use codes of different redundancy without the necessity for different decoders. One has a family of RCPC codes originated by a convolutional code of low rate that is called the mother code. The daughter codes will be generated by omitting specific redundancy bits. This procedure is called puncturing. The receiver must know which bits have been punctured. Only one Viterbi decoder for the mother code is necessary. The mother code used in the DAB system is defined by the generators (133,171,145,133) in octal notation. The encoder is shown as a shift-register diagram in Figure 2.6.

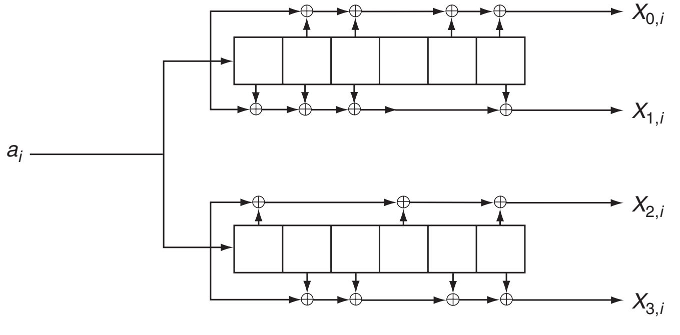

Figure 2.6 Encoder for the DAB mother code

The mother code has the code rate $R_c = 1/4$, that is for each data bit a_i the encoder produces four coded bits $x_{0,i}$, $x_{1,i}$, $x_{2,i}$ and $x_{3,i}$. As an example, the encoder output corresponding to the first eight data bits may be given by four parallel bit streams written in the following matrix (first bit on the left hand side):

1	0	1	1	0	1	1	0
1	1	1	1	0	0	1	0
1	1	0	0	1	0	1	0
1	0	1	1	0	1	1	0

A code of rate 1/3 or 1/2, respectively, can be obtained by omitting the last one or two rows of the matrix. A code of rate 2/3 (= 8/16) can be obtained by omitting the last two columns and every second bit in the second column. If we shade every omitted (punctured) bit, we get the matrix:

1	0	1	1	0	1	1	0
1	1	1	1	0	0	1	0
1	1	0	0	1	0	1	0
1	0	1	1	0	1	1	0

For 8 data bits now only 12 encoded bits will be transmitted: the code has rate 8/12. Using this method, one can generate code rates 8/9, 8/10, 8/11, ... 8/31, 8/32. The puncturing pattern can even be changed during the data stream, if the condition of rate compatibility is taken into account [Hagenauer, 1988].

RCPC codes offer the possibility of Unequal Error Protection (UEP) of a data stream: some bits in the data stream may require a very low bit error rate (BER), others may be less sensitive against errors. Using RCPC codes, it is possible to save capacity and add just as much redundancy as necessary.

UEP is especially useful for MPEG-1 and MPEG-2 Audio Layer II data. They are organised in frames of 24 ms. The first bits are the header, the bit allocation (BAL) table, and the scale factor select information (SCFSI). An error in this group would make the whole frame useless. Thus it is necessary to use a strong (low-rate) code here. The next group consists (mainly) of scale factors. Errors will cause annoying sounds ('birdies'), but these can be concealed up to a certain amount on the audio level. The third group is the least sensitive one. It consists of sub-band samples. A last group consists of Programme-associated Data (PAD) and the Cyclic Redundancy Check (CRC) for error detection in the scale factor (of the following frame). This group requires approximately the same

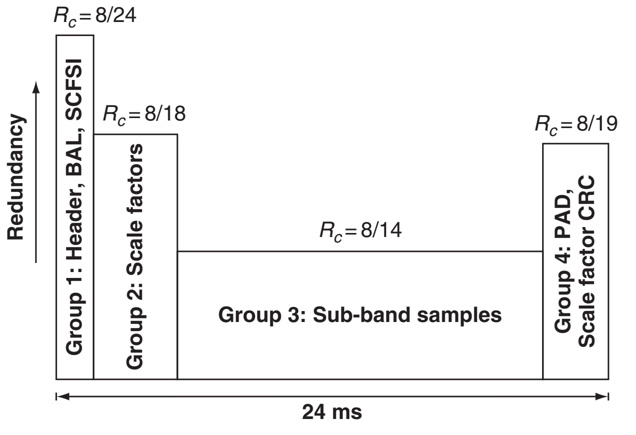

Figure 2.7 Example of an audio UEP profile

protection as the second one. The distribution of the redundancy over the audio frame defines a protection profile. An example is shown in Figure 2.7.

The PAD may be extended to the so-called X-PAD. In this case, the PAD group size increases and the audio sub-band sample group decreases. It is important to note that the error protection does not take this into account. The X-PAD is thus worse protected (see section 2.3.3.2).

For audio data with a sampling frequency of 48 kHz, the DAB system allows 14 different data rates between 32 and 384 kbps. The protection profiles for all these date rates are grouped into five Protection Levels PL1 to PL5. Inside each protection level different data rates are possible, but the robustness against errors is the same. This means, for example, that if a broadcaster switches between 192 and 256 kbps, the audio quality will change, but not the coverage area. PL1 is the most robust protection level, PL5 the least robust one. All protection levels except PL5 are designed for mobile reception; 14 data rates and five protection levels lead to 70 possible combinations. For 64 of them, a protection profile is defined. Table 2.4 shows the possible combinations and the required number of capacity units.

The DAB system allows eight protection levels for Equal Error Protection (EEP). They are intended for data transmission and also used for DAB+ and DMB. For the so-called A-profiles 1-A, 2-A, 3-A, 4-A, all data rates are possible that are integer multiples of 8 kbps. For the B-profiles the data rate must be a multiple of 32 kbps. Table 2.6 shows the eight protection levels and their code rates. The third column shows the number of CUs required for a 64 kbps data stream. The fourth column shows the required SNR to reach a BER of 2×10^{-4} for TM II in a Rayleigh fading channel with $f_{D\max} = 40$ Hz. The fifth column shows the same for $f_{D\max} = 125$ Hz.

The protection levels 4-A and 4-B are very sensitive to fast fading. They should not be used for mobile applications.

All the channel coding is based on a frame structure of 24 ms. These frames are called logical frames. They are synchronised with the transmission frames, and (for audio) with

Table 2.4 Capacity needed for the possible combinations of audio data rates and protection levels

Data Rate	PL1	PL2	PL3	PL3	PL5
32 kbit/s	35 CUs	29 CUs	24 CUs	21 CUs	16 CUs
48 kbit/s	52 CUs	42 CUs	35 CUs	29 CUs	24 CUs
56 kbit/s	X	52 CUs	42 CUs	35 CUs	29 CUs
64 kbit/s	70 CUs	58 CUs	48 CUs	42 CUs	32 CUs
80 kbit/s	84 CUs	70 CUs	58 CUs	52 CUs	40 CUs
96 kbit/s	104 CUs	84 CUs	70 CUs	58 CUs	48 CUs
112 kbit/s	X	104 CUs	84 CUs	70 CUs	58 CUs
128 kbit/s	140 CUs	116 CUs	96 CUs	84 CUs	64 CUs
160 kbit/s	168 CUs	140 CUs	116 CUs	104 CUs	80 CUs
192 kbit/s	208 CUs	168 CUs	140CUs	116 CUs	96 CUs
224 kbit/s	232 CUs	208 CUs	168 CUs	140CUs	116 CUs
256 kbit/s	280 CUs	232 Cus	192 CUs	168 CUs	128 CUs
320 kbit/s	X	280 CUs	X	208 CUs	160 CUs
384 kbit/s	41 6 Cus	X	280 CUs	X	192 CUs

Note: It can be seen from the figures in Table 2.4 that the coding strategy supports many possible changes of configuration. For example, if a 256kbit/s audio channel is split up into two 128 kbit/s channels at the same protection level, they will require the same capacity. Furthermore, in most cases one can increase the protection to the next better level and lower the audio data rate by one step without changing the required capacity. Such a diagonal of constant capacity of 140 CUs has been marked by shading in Table 2.4. It is possible to multiplex several audio channels of different size together, as long as their total size does not exceed 864 CUs. Table 2.5 shows as an example the number of 192 kbit/s audio programmes that can be transmitted for the different protection levels and the signal-to-noise ratio (SNR) that is needed at the receiver in a typical (not fast) fading channel [Schulze, 1995]. A small capacity for data services is always left.

Table 2.5 Number of 192 kbit/s audio programmes and required SNR

Protection Level	Number of Programmes	SNR
PL1	4	7.4 dB
PL2	5	9.0 dB
PL3	6	11.0 dB
PL4	7	12.7 dB
PL5	8	16.5 dB

the audio frames. At the beginning of one logical frame the coding starts with the shift registers in the all-zero state. At the end, the shift register will be forced back to the all-zero state by appending six additional bits (so-called tail bits) to the useful data to help the Viterbi decoder. After encoding this 24 ms logical frame builds up a punctured codeword. It always contains an integer multiple of 64 bits, that is an integer number of CUs. Whenever necessary, some additional puncturing is done to achieve this. A data

Table 2.6 EEP levels: code rate, 64 kbit/s channel size, and required SNR

Protection Level	$R_c =$	Size of 64 kbit/s	SNR (40Hz)	SNR (125Hz)
1-A	1/4	96 CUs	5.0 dB	5.4 dB
2-A	3/8	64 CUs	7.1 dB	7.6 dB
1-B	4/9	54 CUs	8.4 dB	8.8 dB
3-A	1/2	48 CUs	9.3 dB	10.0 dB
2-B	4/7	42 CUs	10.6 dB	11.5 dB
3-B	4/6	36 CUs	12.3 dB	13.9 dB
4-A	3/4	32 CUs	15.6 dB	19.0 dB
4-B	4/5	30 CUs	16.2 dB	21.5 dB

stream of subsequent logical frames that is coded independently of other data streams is called a sub-channel. For example, an audio data stream of 192 kbps is such a possible sub-channel. A PAD data stream is always only a part of an audio sub-channel. After the channel encoder, each sub-channel will be time interleaved independently as described in the next subsection. After time interleaving, all sub-channels are multiplexed together into the common interleaved frame (CIF).

Convolutional codes and their decoding by the Viterbi algorithm are treated in textbooks about coding, see for example [Clark, 1988], [Bossert, 1999], [Proakis, 2008]. Hoeher [1991b] gives some insight into how the channel coding for DAB audio has been developed. His work reflects the state of the research work on this topic a few months before the parameters were fixed. Bit error curves of the final DAB coding scheme in a mobile fading channel and a discussion of the limits of the system can be found in Schulze [1995].

2.2.3.2 Concatenated Coding for Enhanced Packet Mode Coding, DAB+ and DMB

For the transmission of video or AAC audio bitstreams, the convolutional code used in DAB does not guarantee a sufficiently low residual bit error rate in the data. Therefore additional outer Reed-Solomon (RS) coding is necessary to remove these residual errors. The same holds for some packet mode applications.

Reed Solomon (RS) codes may be regarded as the most important *block codes* because of their extremely high relevance for many practical applications. These include deep space communications, digital storage media, and the digital video broadcasting system DVB. The theory of RS codes can be found in many textbooks, see, e.g. [Bossert, 1999] and references therein. In this section we restrict ourselves to the discussion of some basic properties of RS codes as far as they are important for the practical application.

The most common RS codes are based on *byte arithmetics* rather than on bit arithmetics. Thus, RS codes correct *byte errors* instead of bit errors. As a consequence, RS codes are favourable for channels with bursts of bit errors as produced by the Viterbi decoder in the DAB system. They lead to a very low output residual error rate if the input

K data bytes	$2t$ PC bytes

Figure 2.8 A systematic RS code word

error rate is moderate. Thus, an inner convolutional code concatenated with an outer RS code is a suitable setup if a very low residual error rate is needed in case of a mobile radio transmission (see section 2.2.7.2).

RS codes based on byte arithmetics always have the code word length $N = 2^8 - 1 = 255$. For an RS(N,K,t) code, K data bytes are encoded to a code word of N bytes, and the code can correct up to t byte errors. Figure 2.8 shows the structure of a systematic RS code word with an even number $N - K = 2t$ of redundancy bytes called parity check (PC) bytes. In that example, the parity check bytes are placed at the end of the code word.

In practice, the fixed code word length $N = 255$ is an undesirable restriction. One can get more flexibility by using a simple trick. For an RS(N,K,t) code with $N = 255$, we want to encode only $K_1 < K$ data bytes and set the first $K - K_1$ bytes of the data word to zero. We then encode the K bytes (including the zeros) with the RS(N,K,t) systematic encoder to obtain a code word of length N whose first $K - K_1$ code words are equal to zero. These bytes contain no information and need not be transmitted. By this method we have obtained a *shortened* RS(N_1,K_1,t) code word with $N_1 = N - (K - K_1)$. Figure 2.9 shows the code word of a shortened RS(204, 188, $t = 8$) code obtained from an RS(255, 239, $t = 8$) code. Before decoding, at the receiver, the $K - K_1$ zero bytes must be appended at the beginning of the code word and a RS(255, 239, $t = 8$) decoder will be used. This shortened RS code used as the outer code for the DVB-T system is adopted to DAB for the enhanced stream and packet mode encoding (see section 2.3.3.3).

In DAB+ (see section 3.4.2) a shortened RS(120, 110, $t = 5$) code is used that is obtained from the original RS(255, 245, $t = 5$) code and hence differs from the one used in enhanced packet mode.

It may happen that the decoder detects errors which cannot be corrected. In that case of decoding failure an error flag can be set to indicate that the data are in error. The application may then take benefit from this information. If it is known that some received bytes are very unreliable (e.g. from an inner decoder that provides such reliability information), the decoder can make use of this fact and perform so-called 'erasure-based decoding', since these bytes are called *erasures*.

51 zero bytes	188 data bytes	16 PC bytes

Figure 2.9 A shortened RS code word

2.2.4 Interleaving and PSK Mapping

For an efficient error correction with a convolutional code, a uniform distribution of channel bit errors (before the decoder) is necessary. A mobile radio channel produces burst errors, since many adjacent bits will be disturbed by one deep fade. For OFDM, this holds in time and in the frequency direction. To reach a more uniform distribution of badly received bits in the data stream before the decoder, the encoded bits will be spread over a larger time-frequency area before being passed to the physical channel. This procedure is called (time and frequency) interleaving. At the receiver, this spreading has to be inverted by the deinterleaver to restore the proper order of the bit stream before the decoder.

2.2.5 Time Interleaving and Overall Delay

To spread the coded bits over a wider time span, a time interleaving will be applied for each sub-channel. It is based on a so-called convolutional interleaver. First, the codeword (i.e. the bits of one logical frame) will be split up into M small groups of $B = 16$ bits. The bits with number 0 to 15 of each group will be permuted according to the bit reverse law (i.e. $0 \rightarrow 0$, $1 \rightarrow 8$, $2 \rightarrow 4$, $3 \rightarrow 12, \ldots, 14 \rightarrow 7$, $15 \rightarrow 15$). Then, in each group, bit no. 0 will be transmitted without delay, bit no. 1 will be transmitted with a delay of 24 ms, bit no. 2 will be transmitted with a delay of 2×24 ms, and so on, until bit no. 15 will be transmitted with a delay of 15×24 ms. At the receiver side, the deinterleaver works as follows. In each group bit no. 0 will be delayed by 15×24 ms, bit no. 1 will be delayed by 14×24 ms, and so on, bit no. 14 will be delayed by 24 ms, and bit number 15 will not be delayed. Afterwards, the bit reverse permutation will be inverted. Obviously, the deinterleaver restores the bit stream in the proper order, but the whole interleaving and deinterleaving procedure results in an overall decoding delay of 15×24 ms $= 360$ ms. This is a price that has to be paid for a better distribution of errors. A burst error on the physical channel will be broken up by the deinterleaver, because a long burst of adjacent (unreliable) bits before the deinterleaver will be broken up, so that two bits of a burst have a distance of at least 16 after the deinterleaver and before the decoder.

The block diagram of a general convolutional interleaver (sometimes referred to as 'Forney-Interleaver') with parameters (B,M) is shown in Figure 2.10. The serial stream of symbols is converted into B parallel streams. Each parallel stream experiences a delay of $i \times M$ symbol clocks with $i = 0,1,\ldots,B-1$. The figure also shows the corresponding deinterleaver.

The time interleaving will only be applied to the data of the MSC. The FIC has to be decoded without delay and will therefore only be frequency interleaved.

It should be mentioned that other components of the DAB signal chain will cause additional time delay of the signal. This is caused, for example, by the source encoding and decoding, convolutional coding and decoding, additional delays in the distribution network to the transmitter sites needed for synchronisation in single frequency networks (see section 7.6.3) and path delays from the transmitter to the receiver. Typically, an MPEG-1/MPEG-2 Audio Layer II encoder needs a processing time of about 80 ms. The same delay is again required in the audio decoder. Generation of DRC data will also need a processing time of 24 ms to 48 ms, but it can be done in parallel to the source coding

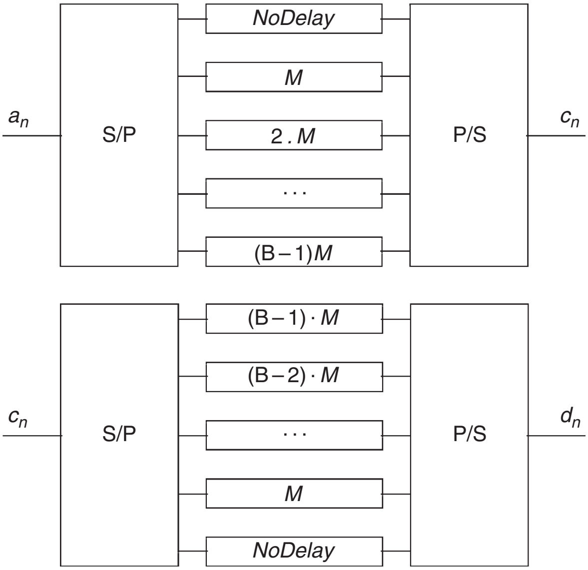

Figure 2.10 Block diagram of convolutional interleaver (top) and deinterleaver (bottom)

process. If several audio codecs are cascaded in the distribution chain, the time delay will increase according to the number of cascading steps.

Table 2.7 gives an overview on the orders of magnitude of the different contributions to the overall system delay.

Summing all these delay portions, the overall time delay of a DAB audio programme will result in more than 500 ms for terrestrial transmission, and more than 750 ms for networks using satellite distribution or transmission.

These values are significantly higher than for traditional analogue audio broadcasting, such as FM-radio. This has to be considered for any time-critical programme parts, such as time announcements (gong), or interactive programme items such as phone-ins.

Table 2.7 Contributions to the DAB overall system delay

Mechanism	Delay	Remarks
Audio coding and decoding	~ 160 ms	depending on implementation
MSC Time Interleaving	360 ms	prescribed by DAB standard
Convolutional coding and decoding	~ 1 ms	depending on implementation
Network (terrestrial distribution)	~ 2 ms	
Network (satellite distribution)	~ 275 ms	Geosynchronous orbit
Transmission (terrestrial)	~ 0.2 ms	
Transmission (satellite)	~ 135 ms	Geosynchronous and highly inclinded elliptical orbit

2.2.6 DQPSK Modulation and Frequency Interleaving

Because the fading amplitudes of adjacent OFDM sub-carriers are highly correlated, the modulated complex symbols will be interleaved in frequency direction. This will be done with the QPSK symbols before differential modulation. We explain the mechanism by the example of TM II: a block of 768 encoded bits has to be mapped onto the 384 complex coefficients for one OFDM symbol of duration T_S. The first 384 bits will be mapped to the real parts of the 384 QPSK symbols, the last 384 bits will be mapped to the imaginary parts. To write it down formally, the bits of the lth block $p_{i,l}$ ($i = 0, 1, \ldots, 2K - 1$) will be mapped to the QPSK symbols $q_{i,l}$ (i = 0, 1, ..., $K - 1$) according to the rule

$$q_{i,l} = \frac{1}{\sqrt{2}}\left[(1 - 2p_{i,l}) + j(1 - 2p_{i+K,l})\right],\ i = 0, 1, \ldots K - 1.$$

The frequency interleaver is simply a renumbering of the QPSK symbols according to a fixed pseudo-random permutation $F(i)$, as shown in Table 2.8. The QPSK symbols after renumbering are denoted by $y_{k,l}$ ($k = \pm1, \pm2, \pm3, \ldots, \pm K/2$).

The frequency interleaved QPSK symbols will be differentially modulated according to the law

$$z_{k,l} = z_{k,l-1} \cdot y_{k,l}.$$

The complex numbers $z_{k,l}$ are the Fourier coefficients of the OFDM symbol no. l in the frame.

2.2.7 Performance Considerations

2.2.7.1 Degradation Due to Failure of Interleaving

Sufficient interleaving is indispensable for a coded system in a mobile radio channel. Error bursts during deep fades will cause the Viterbi decoder to fail. OFDM is very well suited for coded transmission over fading channels because it allows time and frequency interleaving. Both interleaving mechanisms work together. An efficient interleaving requires some incoherency of the channel to get uncorrelated or weakly correlated errors at the input of the Viterbi decoder. This is in contrast to the requirement of the demodulation. A fast channel makes the time interleaving more efficient, but causes degradations due to fast phase fluctuations. The benefit of time interleaving is very small for $f_{D\max} < 40$ Hz. On the other hand, this is already the upper limit for the DQPSK demodulation for TM I. For even lower Doppler frequencies corresponding to moderate or low car speeds and VHF transmission, the time interleaving alone does not help very

Table 2.8 Permutation for frequency interleaving (TM II)

i	0	1	2	3	4	5	...	380	381	382	383
$k = F(i)$	−129	−14	−55	−76	163	141	...	−116	155	94	−187

much. In this case, the performance can be saved by an efficient frequency interleaving. Long echoes ensure efficient frequency interleaving. As a consequence, SFNs support the frequency interleaving mechanism. If, on the other hand, the channel is slowly *and* frequency flat fading, severe degradations may occur even for a seemingly sufficient reception power level. A more detailed discussion of these items can be found in Schulze [1995].

2.2.7.2 Performance Aspects of Concatenated Coding

The original audio coding algorithms based on MPEG-1 and MPEG-2 were quite robust and error-tolerant. The more efficient state-of-the-art source coding algorithms used for DAB+ and DMB are more sensitive to transmission errors, especially when for reasons of compatibility the advanced 'Error Resilience Tools' of AAC are not used. Therefore the concept of a virtually error-free channel was introduced, as was first done in DVB-T.

The virtually error free channel can be achieved by concatenated coding using a convolutional code ('inner code') together with a Reed-Solomon-Code ('outer code'). To achieve best possible performance of this concatenation, proper byte interleaving is necessary between the two stages of coding (Fig. 2.11). For DMB this is done by a convolutional interleaver, see section 2.3.3.2, for DAB+ and enhanced packet mode, a virtual interleaver is applied, see section 2.3.3.5.

If the outer interleaver is ideal, the burst errors at the output of the Viterbi decoder will lead to uniformly distributed byte errors at the input of the RS decoder. The RS code is able to correct a certain number of bytes in each block. It is therefore very efficient for burst error decoding as long as the bursts are not too long, because it takes advantage from the fact that generally there is more than one bit error inside one erroneaus byte. This is illustrated by the following example: Let P be the byte error probability and P_b the bit error probability at the output of the Viterbi decoder after decoding the inner convolutional code. The worst case of only one average bit error in one erroneous byte corresponds to $P = 8\,P_b$. An upper bound for the residual bit error rate and block error rate after RS decoding can be calculated from theoretical assumptions, see [Schulze, 2005,

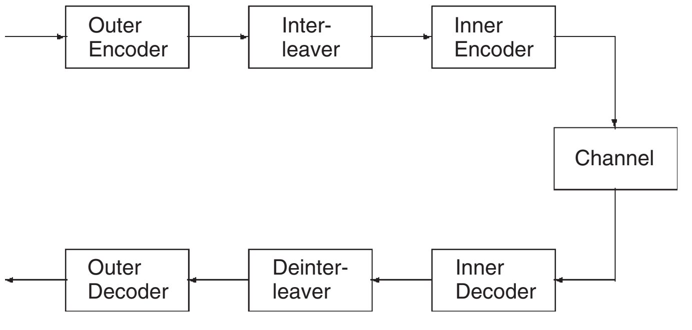

Figure 2.11 Block diagram of concatenated coding setup

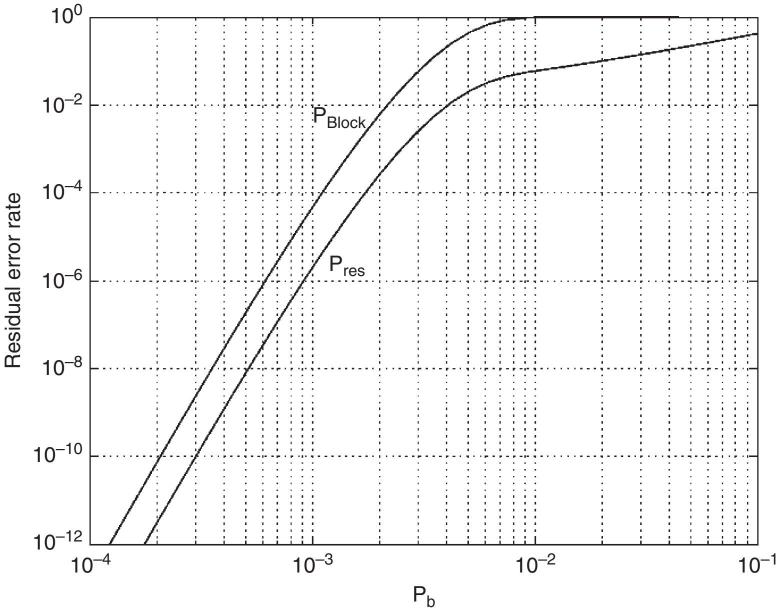

Figure 2.12 Block error rate and residual bit error rate for the RS code

p. 256]. For the RS (204, 188, $t = 8$) code, which can correct up to eight erroneous bytes in a block, this is displayed in Fig. 2.11. If ideal interleaving can be assumed for all interleaving mechanisms, the curve of for P_{Block} can be combined with simulated bit error curves for the convolutional codes for DAB [BS. 1114], [Schulze, 1995]. For example, to achieve a BER of $2 \cdot 10^{-4}$ at the output of the Viterbi decoder in a rural channel at a vehicle speed of 130 km/h at 226 MHz an average S/N of 14.7 dB is needed if code rate 0.5 is used. If we take into account some loss that is due to implementation, 16 dB may be a reasonable figure. From Figure 2.12 we infer a block error rate of 10^{-10} after RS decoding.

This estimation is confirmed by simulation results for the concatenated coding scheme presented in Figure 2.13. In this simulation the vehicle speed is 167 km/h, the other parameters are the same as for the estimation. From the simulated curves an average S/N of 17–18 dB is estimated to be required to achieve a BER of 10^{-10}. It can also be seen from this figure that the additional outer interleaver is crucial for achieving a significant reduction of BER by concatenated coding.

To interpret these results, we assume as an example a low video rate of approximately 500 kbps typical for DMB (see Chapter 9). Since each block has 188 useful bytes corresponding to 1504 useful bits, at this bit rate approximately 330 blocks are transmitted per second. For $P_{block} = 10^{-10}$ the average time between two error events is $30 \cdot 10^{6}$ seconds or almost one year. It is therefore justified to speak of error-free reception once the required S/N is provided.

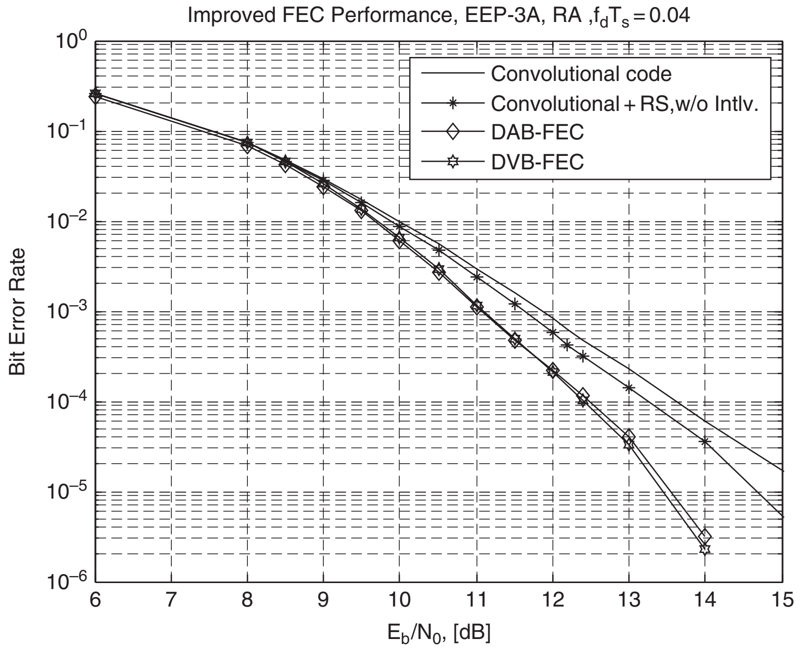

Figure 2.13 Performance of different coding schemes in a rural area Rayleigh channel for DAB mode I at vehicle speed 167 km/h. Reproduced with permission of © Communications Research Centre, Canada (CRC)

2.3 The DAB Multiplex

2.3.1 Mode-Independent Description of the Multiplex

The DAB system is designed for broadcasting to mobiles in the frequency range from 30 MHz to 3 GHz. This cannot be achieved by a single OFDM parameter set, so four different transmission modes are defined (see section 2.2.1). The DAB multiplex, however, can be described independently of the transmission mode. To achieve this, containers of information are defined which are used to transmit the data of applications (audio and data services, service information, etc.) to the receivers. Figure 2.14 shows the generation of the DAB multiplex.

The data of audio components and other applications are carried in what is called the Main Service Channel (MSC). Every 24 ms the data of all applications are gathered in sequences, called Common Interleaved Frames (CIFs). Multiplex and service-related information is mainly carried in the Fast Information Channel (FIC). Similar to the MSC, FIC data are combined into Fast Information Blocks (FIBs).

Depending on the transmission mode, a number of CIFs and FIBs are grouped together into one transmission frame which is mapped to a number of OFDM symbols (see section 2.2.2).

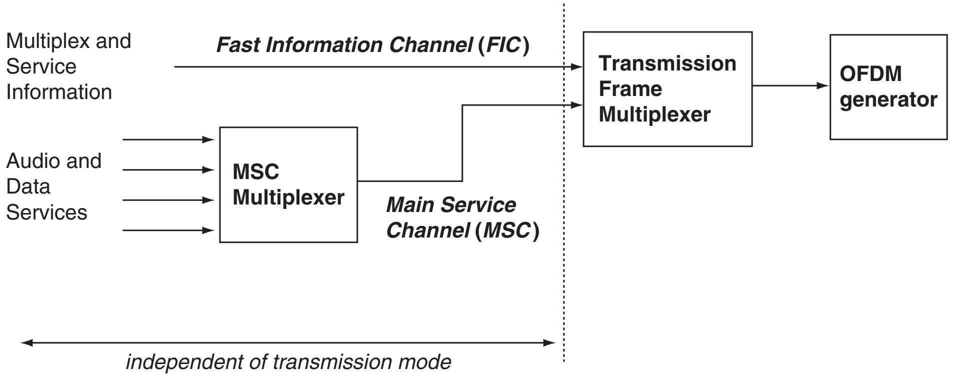

Figure 2.14 Generation of the DAB multiplex

2.3.2 The Main Service Channel

The MSC of the DAB system has a gross capacity of 2.304 Mbps. Depending on the convolutional code rate, the net bit rate ranges from approximately 0.6 to 1.8 Mbps. Single applications do not normally consume this overall capacity. The MSC is therefore divided into sub-channels. Data carried in a sub-channel are convolutionally encoded and time interleaved. Figure 2.15 shows the conceptual multiplexing scheme of the DAB/DMB system.

The code rate can differ from one application to another. The data rates available for individual sub-channels are given by integer multiples of 8 kbps (of 32 kbps for some protection schemes). Figure 2.16 shows an example for a multiplex configuration. Each sub-channel can be organised in stream mode or packet mode.

The division of the MSC into sub-channels and their individual coding profiles are referred to as the Multiplex Configuration. The configuration is not fixed but may be different for different DAB transmissions or may vary from time to time for the same transmission. Therefore the multiplex configuration must be signalled to the receivers. This is done through the FIC.

2.3.3 Transport Mechanisms

The DAB system provides several mechanisms to transport data to receivers which are each tailored to specific purposes. An overview is given in the figure of the inside back cover of the book.

The Stream Mode is designed for continuous and synchronous data streams such as coded audio. In this case every 24 ms the same number of bits is transmitted transparently, that is, there is no mechanism inside the DAB system to provide synchronisation or addressing except by the frame structure. All the necessary signalling has to be provided by the application itself. Stream mode sub-channels may contain audio encoded according to MPEG-1/MPEG-2 Audio Layer II, audio encoded according to the DAB+ scheme (see

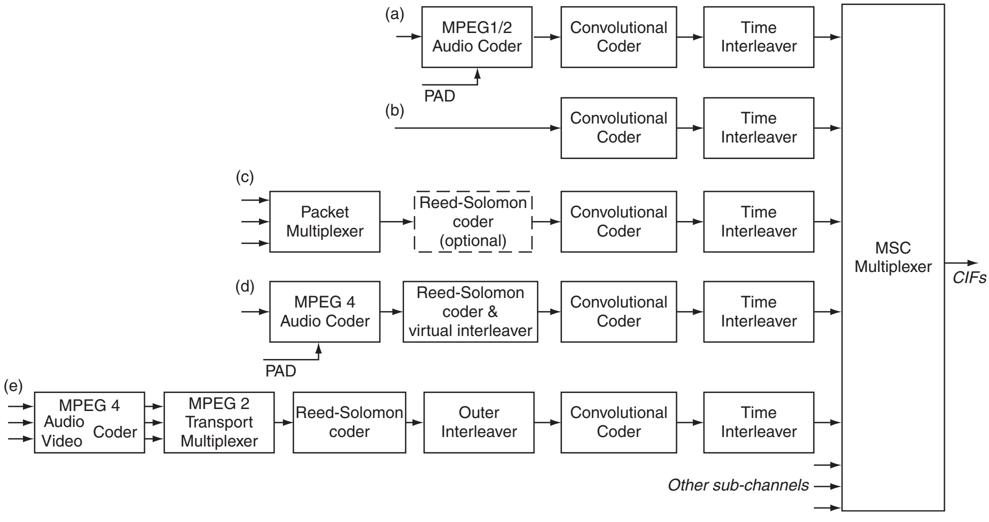

Figure 2.15 Generation of the common interleaved frames (CIFs) from a) an audio sub-channel using MPEG-1/ MPEG-2 Audio Layer II, b) a general data stream mode sub-channel, c) a packet mode sub-channel or enhanced packet mode channel, d) an audio sub-channel using MPEG-4 HE AAC V2 audio coding (DAB+), e) an enhanced stream mode sub-channel using MPEG-4 Audio (BSAC or AAC) and AVC video coding and formation of an MPEG-2 Transport Stream (DMB)

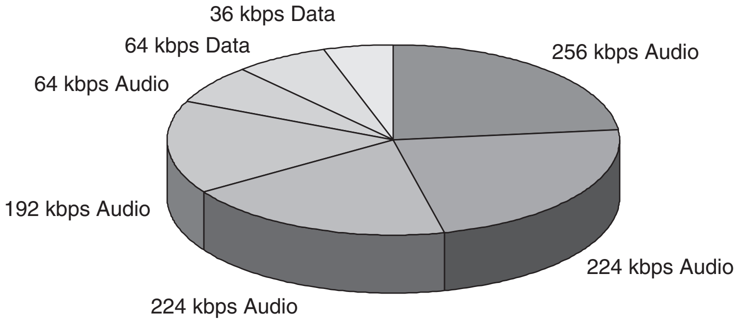

Figure 2.16 An example of how the overall capacity is assigned to individual applications (capacity pie)

section 3.4.2), an MPEG-2 Transport Stream containing audio and video data encoded according to the DMB specification (see Chapter 9) or general data at fixed bit rate. In the case of DAB+ or DMB, additional outer Reed-Solomon codes described in section 2.2.3.2 are used to increase the robustness against transmission errors (see section 2.2.7.2).

For asynchronous data there is Packet Mode, which provides a protocol for conveying single data groups through a packetised channel. The packet protocol allows repetition of data to be handled and the creation of a multiplex of several parallel applications, to which the capacity can be flexibly assigned. It is possible to use the additional outer code also in packet mode (see section 2.3.3.2).

A special way of transport is provided for Programme-associated Data (PAD) which is inserted into the MPEG-1/MPEG-2 Audio Layer II stream by defining a structure of the Auxiliary Data field of the MPEG audio frame specific to DAB. PAD can also be used in DAB+ subchannels. It provides a number of functions related to the contents of the audio programme and can be inserted at the place where the audio is produced. Therefore PAD is considered to be a part of the audio and not really a separate transport mechanism for data.

2.3.3.1 Stream Mode

Stream mode is used for applications which can provide a constant data rate of a multiple of 8 kbps (32 kbps for the B coding profiles, see section 2.2.3). For example, at a sampling rate of 48 kHz, the MPEG-1/MPEG-2 Audio Layer II encoder generates a data frame every 24 ms which exactly meets this requirement. When DAB+ audio is used, the audio data corresponding to 120 ms is organised in an audio super frame which is spread across five 24 ms – frames [TS 102563]. When transmitting general data, the data stream can be divided into "logical frames" containing the data corresponding to a time interval of 24 ms. These logical frames can be transmitted one after the other in the same manner as MPEG audio frames.

When stream mode is used there are two options for error protection. Unequal error protection (UEP) is used with MPEG-1/MPEG-2 Audio Layer II and provides error correction capabilities which are tailored to the sensitivity of the audio frame to bit errors (see section 2.2.3). For DAB+, DMB and for general data, equal error protection (EEP) is used, where all bits are protected in the same way.

2.3.3.2 Stream Mode Sub-Channels Using Concatenated Coding (DAB+ and DMB)

This Transport Mode is an evolution of the Stream Mode to allow for the use of concatenated coding. Although the principle is similar, the details of the concatenated coding are different for DMB and DAB+.

In the case of DAB+ [TS 102563], the audio super frames are split into portions which each contain 110 bytes. These portions are coded using a RS (120, 110, $t = 5$) code. Due to the arrangement of the data bytes and the order of transmission, virtual interleaving is achieved during the RS coding process similar to the enhanced packet mode (see section 2.3.3.5).

In the case of DMB, MPEG-2 Transport Stream packets are carried in the DAB subchannel. These 188-byte long blocks are coded with the shortened RS (204, 188, $t = 8$) code. It should be noted that this RS code and the one of DAB+ use the same Galois field to allow for easy hardware implementation. Furthermore a convolutional interleaver with $M = 17$ and $B = 12$ ($17 \times 12 = 204$, see Figure 2.10) is applied to those RS encoded 204-byte long packets. This means that the bytes of each code word are transmitted in 12 consecutive blocks each of length 204 bytes. This structure is similar to the one used in the DVB system. This mode is used for DMB with the MPEG-2 Transport Stream

[TS 102427] and for DAB-IPDC [TS 102978]. With a limited additional overhead of 7.8 % the required quasi error-free reception of TV content even in a highly mobile environment is achieved (see section 2.7.2.2).

2.3.3.3 Programme-Associated Data (PAD)

Although DAB provides mechanisms to carry general data in stream or packet mode, there is a need for an additional method for transmitting data which is closely linked to an MPEG-1 / MPEG-2 Audio Layer II or DAB+ service. This is referred to as PAD.

At the end of the MPEG-1/ MPEG-2 Audio Layer II frame there is an auxiliary data field which is not further specified in MPEG. For the DAB audio frame, however, this field is used to carry the CRC for the scale factors (see section 3.3.3), and two PAD fields, see Figure 2.17. In case of DAB+, the PAD is carried in a syntax element of the AAC frame called 'Data Stream Element'. The DAB system is transparent for the PAD. This means that the PAD can be produced and inserted at the time when the audio signal is coded, normally at the studio (see Chapters 5 and 6) and will be retrieved only when decoding the audio signal in the receiver.

The fixed PAD (F-PAD) field consists of two bytes at the end of the DAB audio frame, which are referred to as Byte L (last) and Byte L–1. It enjoys the same protection level as the SCF-CRC field and hence is well protected. Byte L–1 is used for signalling the contents of Byte L and the presence of extended PAD, and for transmitting serial commands, music/speech indication, and data related to the audio source (International Standard Recording Code or Universal Product Code/Euro-pean Article Number). The only applications standardised for Byte L up to now is to signal Dynamic Range Control data (DRC, see section 3.4.1) and in-house-information.

The extended PAD (X-PAD) field can be used to send larger amounts of data (up to 64 kbps), for example text messages or multimedia objects (see Chapter 4). To create a

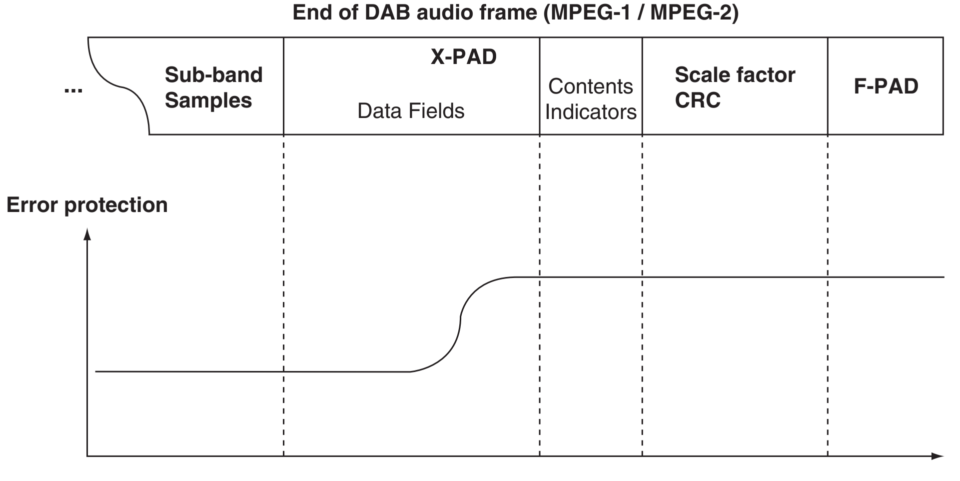

Figure 2.17 The PAD fields at the end of the DAB audio frame

flexible multiplex structure within the PAD, a special packet structure was developed. The data are arranged in X-PAD-Data groups which each consist of a data field and a Contents Indicator which signals the kind of data carried in the corresponding data field and in some cases the number of bytes of the data field. Data fields of X-PAD-Data Groups may be larger than the X-PAD data field and hence be sent in the X-PAD-fields of several audio frames. It is therefore possible to send large amounts of data although the length of the X-PAD-field is restricted in each audio frame to provide sufficient data rate for the coded audio.

The extended PAD (X-PAD) field partly (4 bytes) enjoys the protection level of the SCF-CRC; the larger part, however, only enjoys the protection level of the sub-band samples. In 'short X-PAD' mode only the four better protected bytes are used for X-PAD; in this case there are restrictions on the applications that are carried. The one-byte contents indicator signals the type of data which is contained in the other three bytes of short X-PAD.

If more than four bytes are being used, this is referred to as 'variable size X-PAD'. In this case, data of more than one data group and of several applications can be contained in the X-PAD-field. To signal this, a list of contents indicators is included in the X-PAD-field which signals the type of the data and the number of bytes for each application. From this information, the number of bytes used in total for X-PAD can be calculated. The contents indicators are separated from the data fields and are sent collectively in the better protected part to ensure that the important information on the contents and amount of data can be received with high probability.

For further details the reader is referred to section 7.4 of [EN 300401] and to section 5.4 of [TS 102563].

2.3.3.4 Packet Mode

For packet mode transmission, the data is organised in MSC data groups which consist of a header, a data field of up to 8191 bytes and optionally a cyclic redundancy check (CRC) for error detection. The data group header allows identification of different data group types such that, for instance, scrambled data and the parameters to access them can be carried in the same packet stream. There are also counters which signal if and how often the data group will be repeated. An extension field offers the possibility to address end user terminals or user groups. Data groups are transmitted in one or several packets.

The packets themselves constitute the logical frames in packet mode similar to the audio frames in stream mode. To fit into the DAB structure requiring a multiple of 8 kbps, that is a multiple of 24 bytes per 24 ms, packet lengths of 24, 48, 72 and 96 bytes are defined in the DAB standard. The packets consist of a 5 byte header, a data field, padding if necessary, and a CRC for error detection. The packets may be arranged in any order in the transmission. The header of each packet signals its length so that the decoder can find the next packet. An important feature of packet mode is that padding packets can be inserted if no useful data are available to fill the data capacity. Packet mode is therefore suitable for carrying asynchronous data. Up to 1023 applications may be multiplexed in a packet mode transmission.

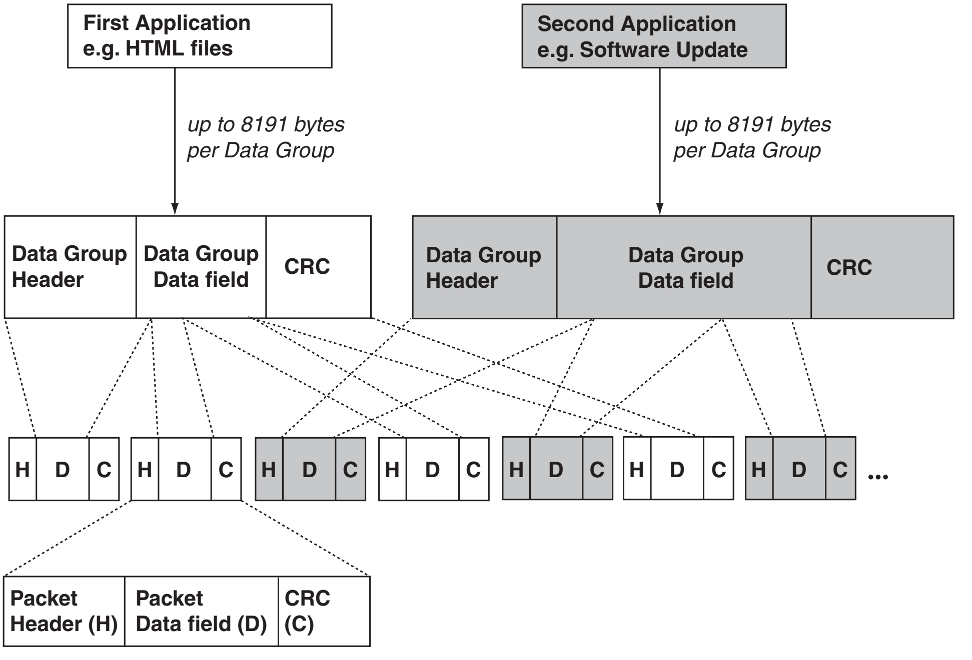

Figure 2.18 Example of a packet mode transmission multiplexing two applications

Figure 2.18 shows an example of a packet mode transmission. Two applications are carried in parallel. From each application, a suitable amount of data, for example one file, is selected to form a data group. The corresponding header is added to the data and the CRC is calculated. Each data group is mapped onto a number of packets, each of which may have a different length. The first, intermediate and last packets carrying data of one data group are marked accordingly in their headers. The packets of different applications use different packet addresses. Therefore, the sequence of the packets belonging to the first application may be interrupted by those of the second application. However, within each sequence of packets, the transmission order has to be maintained to allow the decoder to correctly reassemble the data group.

For more details of the packet mode, the data groups and packet headers the reader is referred to [TR 101496].

2.3.3.5 Enhanced Packet Mode (EPM)

For packet mode applications it is possible in addition to use the outer Reed-Solomon forward error correcting code (RS) to increase the performance of the transmission system. This is referred to as enhanced packet mode. The additional RS parity data bytes are transmitted in a way that also allows receivers not using this feature to decode the data packets. To achieve this, all packets of a packet mode channel are fed to the RS encoder irrespective of their packet address. The RS coding is applied to sequences of

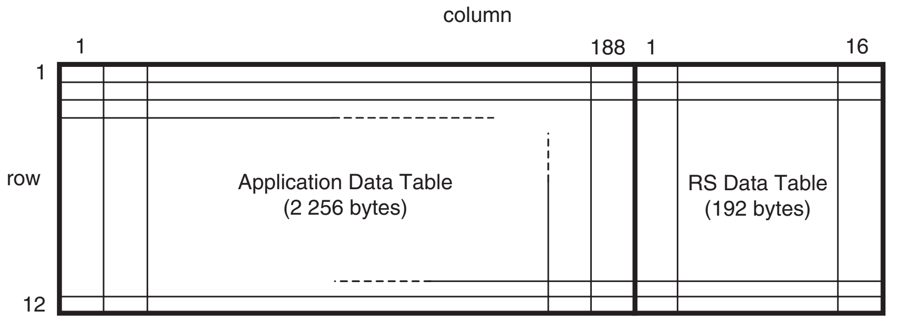

Figure 2.19 Generation of RS parity data in Enhanced Packet Mode

packets comprising 2256 = 12 * 188 bytes of application data, padding packets must be inserted if necessary (Fig. 2.19).

The Application Data Table is filled vertically column by column with the data of the packets. Then the Reed-Solomon parity bytes are calculated horizontally over the 188 bytes in the same row. The RS coding process generates 192 = 12 * 16 additional bytes of RS parity data, i.e., 12 shortened RS codewords are generated at a time (see section 2.2.4).

All application data columns are read out vertically as they are filled and are transmitted followed by the RS parity bytes which are also read out vertically (virtual time interleaving). These data are transported within nine consecutive 'FEC packets' immediately after the packets containing the application data (Fig. 2.20).

The enhanced packet mode was designed in a backwards-compatible way to the standard packet mode, i.e. legacy receivers will discard the FEC packets, but can still decode the application data packets if they are received error-free. For this purpose the FEC packets are different from ordinary data packets in two ways: they use a packet address that is not used for a Service Component, and their structure, including the position of the CRC, is different from that of the data

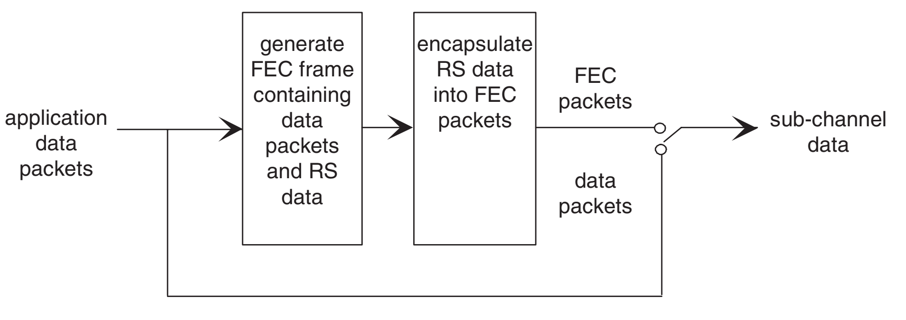

Figure 2.20 Insertion of FEC packets into a packet mode sub-channel

packets. This assures that the conventional CRC check will lead to discarding these packets in legacy devices.

For more details of the RS coding scheme the reader is referred to [EN 300401].

2.3.4 Fast Information Channel (FIC)

The FIC is used to signal the multiplex configuration of the DAB transmission and service information for the services contained. Therefore, it uses fixed symbols in the transmission frame which are known to the receivers. The receiver needs this information to be able to decode any of the sub-channels. Therefore, for instant data acquisition, the FIC data are not time interleaved (hence the name 'fast'). Instead, a high protection (code rate 1/3) and frequent repetition of the data is used to guarantee high availability.

The FIC consists of a number of Fast Information Blocks (FIBs) which carry the information (see Figure 2.21). Each FIB is made up from 32 bytes: two CRC bytes are used for error detection, the remaining 30 bytes are filled with Fast Information Groups (FIGs). The FIGs are distinguished by a type field. FIGs of type 0 are used for the MCI and Service Information (SI), FIGs of type 1 are used to send text labels, FIGs of type 5 are used to send general data in the FIC (Fast Information Data Channel, FIDC), FIGs of type 6 are used with access control systems and FIGs of type 7 are used for in-house data transmission by the broadcaster. Some FIG types use extensions to indicate different meanings of the data fields, e.g. FIG type 0 extension 1 (also referred to as FIG 0/1, etc.) is used to signal the sub-channel organisation while FIG type 0 extension 10 is used to send the date and time.

In special cases the data rate provided by the FIB symbols in the transmission frame (32 kbps) may not be sufficient to carry all of the FIC information with the desired repetition rate. In this case, part of this information can be sent in parallel in packet mode sub-channel #63 with packet address 1023. This is referred to as the Auxiliary Information Channel (AIC). The corresponding packet mode data group contains the

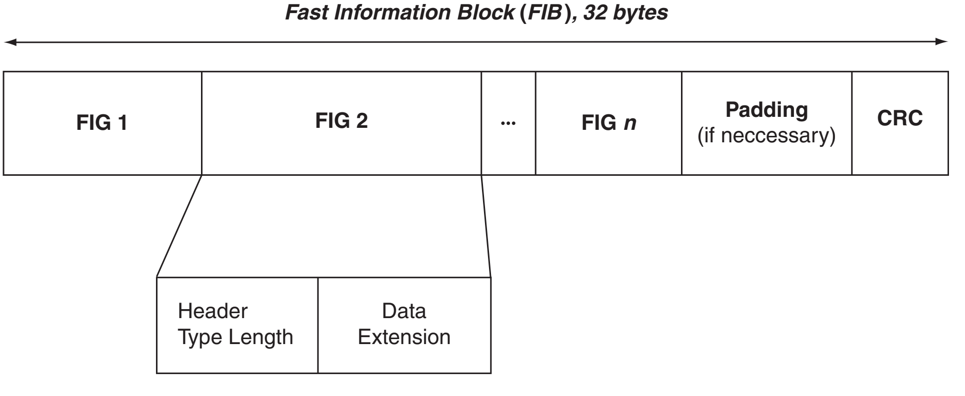

Figure 2.21 Structure of a Fast Information Block (FIB) and a Fast Information Group (FIG)

FIGs as defined for the FIC. Data concerning other DAB transmissions may be sent only in the AIC. For more details see section 2.5.2.4.

2.3.5 Transmission Frames

From FIBs and CIFs transmission frames are formed which are then mapped to OFDM symbols. Table 2.9 lists the duration of the frames and the number of FIBs and CIFs used in each transmission mode.

2.3.6 The Logical Structure of the DAB Multiplex

The different data streams carried in the DAB multiplex can be grouped together to form what is called a ‘service’. The services can be labelled (e.g. ‘Bayern 3’) and it is the services from which the listener makes his or her choice. All services taken together are referred to as an ensemble.

The different data streams (e.g. audio, data, etc.) which belong to one service are called its Service Components. Different services may share components, and the logical structure of services, service components and the position in the CIF where the data of the components are actually carried is signalled as part of the MCI in the FIC. Figure 2.22 shows an example of this structure. The ensemble labelled DAB 1 contains a number of services, among them Radio 11, Radio 12, Data 29 and Radio 3. Radio 3 is a service which consists of an audio component only, that is a ‘normal’ radio programme. Radio 11 and Radio 12 each consist of an audio component and share Radio 1 data as a common data component which carries information relevant to both programmes or programme-independent data such as traffic messages. At times, say during a news programme on the hour, Radio 11 and Radio 12 broadcast the same audio signal. Instead of transmitting the corresponding bits twice, it is possible to signal this at the level of service components. The capacity in the CIF usually used for the audio component of Radio 12 (i.e. sub-channel 7) can be reused during this time for another service or component, for example slow motion video. Data 29 is a data service without an audio component but with two separate data components. Figure 2.18 also shows how the different components are transported in the MSC. It can be seen that different

Table 2.9 The frames of the four DAB transmission modes

Mode	Duration of Transmission Frame, ms	Number of CIFs per Transmission Frame	Number of FIBs per Transmission Frame
TM I	96	4	12
TM II	24	1	4
TM III	24	1	3
TM IV	48	2	6

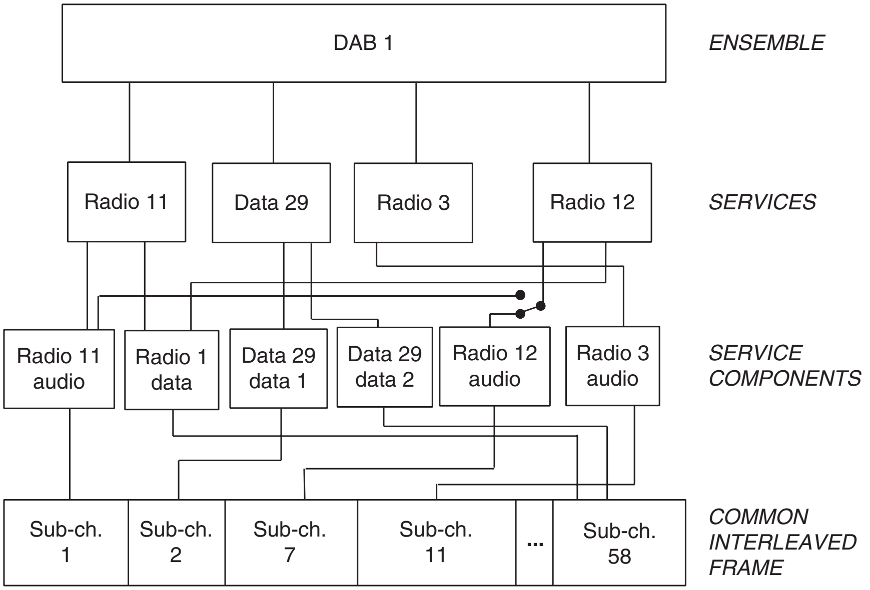

Figure 2.22 Example of the logical structure of the DAB multiplex

components (Radio 1 data and Data 29 data 2) may share a packet mode sub-channel while stream mode components each require an individual sub-channel.

It is also apparent in the figure which information has to be sent in the FIC: the sizes and positions of the sub-channels in the CIF and their respective code rates are signalled as 'Sub-channel organisation' in FIG type 0 extension 1, the services and their components are described by 'Basic service and service component definition' in FIG type 0 extension 2 and 3 (used for packet mode).

Of course there is a need to provide further information about each service, such as service label, programme type, programme number for recorder control, indication of announcements (e.g. for automatically switching to a service broadcasting traffic information), alternative frequencies and frequencies of other DAB transmissions. This kind of information may be displayed in order to assist the listener in operating the receiver or to enable the receiver to optimise the reception. The DAB system provides a number of mechanisms for this purpose, far beyond what has been realised by the Radio Data System (RDS) in FM broadcasting (see section 2.5).

2.3.7 Multiplex Reconfiguration

To achieve a high flexibility, the multiplex may be reconfigured from time to time during transmission. When the multiplex configuration is about to change, the new information, together with the timing of the change, is transported via the MCI in addition to the current one and indicates in advance what changes of the multiplex are going to take place. Because of the time interleaving, special precaution has to be taken when the size of a sub-channel changes. For details the reader is referred to Annex D of [EN 300401]. Due to the formation of super frames, some further rules of operation apply for DAB+, see section 8 of [TS 102563].

2.4 Conditional Access

New business models of modern broadcast systems demand protected distribution and controlled consumption of digital content. An example is subscription for pay-TV.

Whereas traditional broadcast systems distribute content to every member in the network, the integration of an access control system enables transmission to closed-user-groups.

In this context, a closed-user-group is a set of members that are authorized to consume an identical service. Services may be used for device internal processing, like update of navigation data for PNAs (Personal Navigation Assistant), or for direct human consumption, e.g. weather reports.

Access control systems protect content during distribution using modern cryptographic methods for encryption and authentication. The content can be consumed only by the respective group members that have the necessary entitlements, usage rights and cryptographic keys.

From a service provider's point of view, access control permits flexible business models in terms of:

- grouping content parts to services;
- offering pay-services, e.g. pay-TV;
- providing preview services;
- maintaining customer relations, e.g. by exclusively providing high value content;
- distribution of confidential data within a business-to-business relation;
- subscription per theme, i.e. either lifetime or time-limited; and
- protection of content owners rights.

Access control is an essential feature of digital broadcast systems and qualifies them for the future.

2.4.1 General Principles

Since no dedicated access control system is foreseen in [EN 300401] and [TS 102367], general aspects and concepts regarding CA Systems and DRM Systems are described in this section. They familiarize the reader with current methods of operation and point out special requirements for an application in DAB/DMB.

2.4.1.1 CA Systems and DRM Systems

Access control systems are known either as CA – Conditional Access systems or as DRM – Digital Rights Management systems.

CA systems have their origin in the TV-world and are applied, if the transmission should be protected and the consumption is subject to charge, e.g. pay-TV. In general, CA systems verify whether a potential user has the right to consume content or not. They do not distinguish between more detailed rights, as DRM systems do. Therefore the content is protected on its way through the broadcast channel ('transport-channel-protection') and is intended for singular consumption in case the potential user owns the relevant entitlements. Typically, CA terminal modules do not store the content for further use or distribution.

DRM systems meet the objectives of content owners, authors and publishers to protect their copyrighted music and videos from unauthorised use and sharing. They have been developed for controlled content download, e.g. via Internet and its repeated consumption and forwarding. Therefore content is not only protected on its way to the terminal devices but it remains protected for its future use.

Some DRM systems provide a large scale of different usage rights to their protected objects, including:

- Who is entitled to use the content?
- How many times can the content be consumed?
- How can the content be consumed, e.g. listened, printed out or viewed?
- What kind of device is the content entitled for?
- When will the rights expire?
- Is content forwarding allowed?
- Do geographical limitations exist?

Due to the fact that the distribution systems are widely combined and extended, today's meaning and the application of the two concepts, CA and DRM, have become more and more blurred. Hence DRM has become the catch-all term.

2.4.1.2 Access Control for Handheld Devices

Access control systems that are applied in a mobile environment need to cope with the following limitations:

- The bandwidth is always constrained and expensive.
- The devices are very limited in terms of processor, memory and battery power.
- Portable devices are not always turned-on.
- The devices lose signal and roam out of the coverage area from time to time.
- A permanent active backchannel is not always available.
- All kinds of one-to-one communication are expensive.
- Permanent change of own geographical position and therefore passing through different transmission zones (roaming) must be taken into account.

Hence existing and established CA and DRM systems have to be extended to meet the above mentioned requirements.

2.4.1.3 Broadcast Systems and Hybrid Solutions

A typical broadcast system does not provide a backchannel for rights acquisition and message acknowledgement. Considering the above mentioned restrictions, all kinds of management information has to be repeated permanently to ensure a high reception probability. To keep the management overhead reasonable, this requires efficiently coded and short messages. Efficient subscriber grouping and subscriber addressing algorithms are essential.

In addition to broadcast-only distribution systems, hybrid solutions can be realised. That is, content distribution together with essential additional information via the broadcast channel, but utilisation of alternative point-to-point systems, e.g. GSM for the directed transmission of usage rights and entitlements. Those concepts require a permanent back channel.

2.4.1.4 Proprietary Solutions and Open Concepts

The predominant intention of access control manufacturers is the provision of proprietary, 'closed' solutions, supplying all relevant components on server and client side from one source. They keep their system details secret. Examples for such solutions are 'Irdeto PIsys' and 'Windows Media DRM'.

In contrast, part of the industry continues to work for open solutions and standards, enabling interoperability, quality and technology spreading, e.g. OMA DRM 2.0.

Open solutions are based on Kerckhoffś principle that the security of a cryptosystem should rely on the secret of the used keys, even if the complete system design is public knowledge ('security through transparency').

2.4.1.5 Underlying Key Hierarchy

All access control systems are based on a multilevel key hierarchy. This is for the secure transmission of keys from the transmitter to the receiver side and for separation of different services and corresponding service providers.

In general a key hierarchy comprises three to four levels, as shown in Figure 2.23:

- Layer 4:
  The multimedia content is encrypted with a symmetric encryption algorithm using a content key (CW - control word).
- Layer 3:
  To descramble the content, the corresponding content key must exist in the terminal device. Hence the content key must be distributed within a corresponding management message (ECM – entitlement control message) in advance. This is done in a scrambled way, where the content key is scrambled with a service key (SK).
- Layer 2:
  The registered and entitled terminal devices have the service key stored and are therefore able to decrypt the current content key. The service key has been transmitted in advance (EMM – entitlement management message). On its way it is protected with a device key (DK).
- Layer 1:
  The device key is being distributed during device registration.

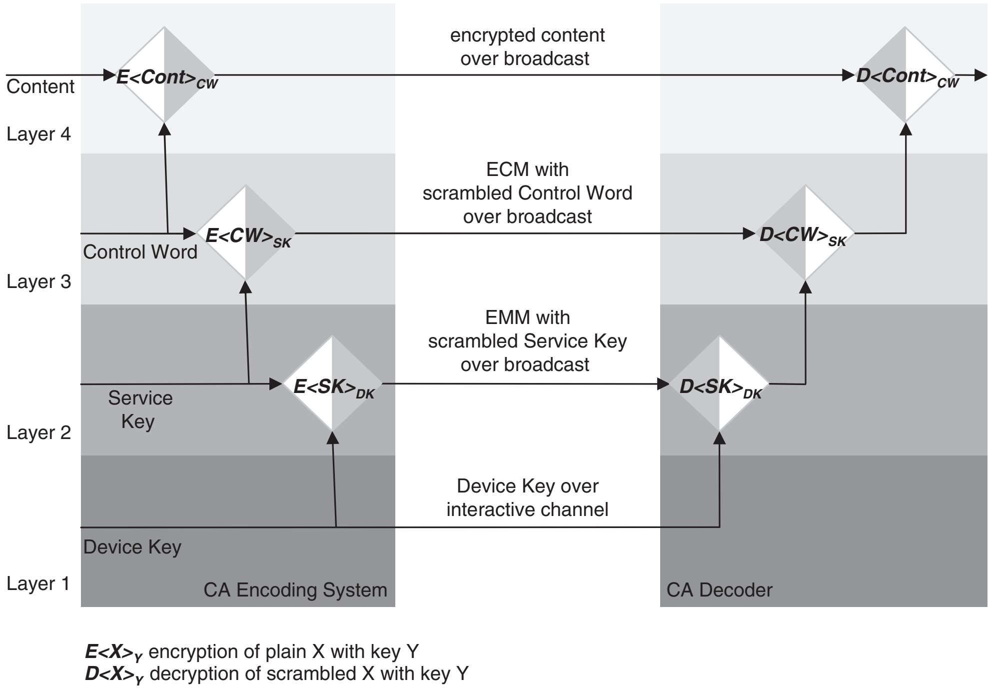

Figure 2.23 Multilevel key hierarchy of HECA – High Efficient Conditional Access system [www.HECA]

The device key could either be a symmetric key or a public-private key pair. All other keys and algorithms in use are symmetric ones. The naming of the keys and management messages described above might differ within the CA and DRM systems.

2.4.1.6 Key Change and Message Repetition

Access control systems are exposed to hacker attacks. Hence relevant keys are frequently changed.

Taking the particular characteristics of mobile handsets into account we cannot count on permanent terminal accessibility and active backchannel. Therefore the transmitted messages, containing keys and entitlements have to be repeated permanently.

Typical values for key change and message repetition are:

- content key change every 5–20 seconds;
- repeated transmission of management messages containing the current content key every 3–20 seconds (On start-up a terminal has to wait for the relevant management message before it can start content decryption.);
- service key change once a month;
- repeated transmission of management messages containing the current service key and entitlements every 10–20 minutes.

2.4.1.7 Applied Cryptographic Algorithms and Security Module

Access control systems are based on certified cryptographic algorithms, see [www.NIST].
They are applied for:

- content scrambling;
- key protection;
- authentication of messages;
- secure storage mechanisms of keys and rights in the terminal device.

Depending on the field of application, the requirements on cryptographic algorithms in terms of security, complexity and delay differ. Traditional radio content is supposed to always be up-to-date but has a low life-span. Therefore content scrambling has to be fast but only moderately secure. This could lead to an application of a symmetric algorithm with 128 bit key length. In contrast, for the protection of content keys and service keys longer keys will be used.

In general today the use of Advanced Encryption Standard (AES) encryption algorithms is favoured. Different AES modes, e.g. cipher feedback (CFB), cipher-based message authentication code (CMAC) and different key length, e.g. 128, 256 are applied for the above mentioned fields of application. AES implementations are widely available in software and hardware even for thin clients. They are secure, fast and easy to integrate. The Rivest-Shamir-Adleman (RSA) algorithm is in use for the asymmetric protection of device keys.

Dependent on the requirements in terms of security, dedicated hardware is applied on transmitter- as well as on receiver-side. Receiver might have integrated, soldered security chips or they use external, removable solutions in SIM card format or SD card format.

The tasks of a security module are:

- decoding and evaluation of management messages;
- decryption of service key and content key;
- secure storage of device key, service key and entitlements;
- delivery of content to descrambler;

In contrast to decryption of service keys and content keys that is performed in the security module itself, the content is descrambled in the receiver core for reasons of data throughput.

2.4.2 System Architecture of HECA (Fraunhofer-IIS)

With the High Efficient Conditional Access system HECA [www.HECA], a CA System has been designed that is especially tailored for digital broadcast systems with limited bandwidth, like DAB.

It meets the above mentioned requirements in terms of security, mobility and efficiency. It can be applied on different DAB transport layers, see section 2.4.3, as well as on other digital broadcast media and transmission protocols, e.g. DRM – Digital Radio Mondiale or TPEG – Transport Protocol Experts Group.

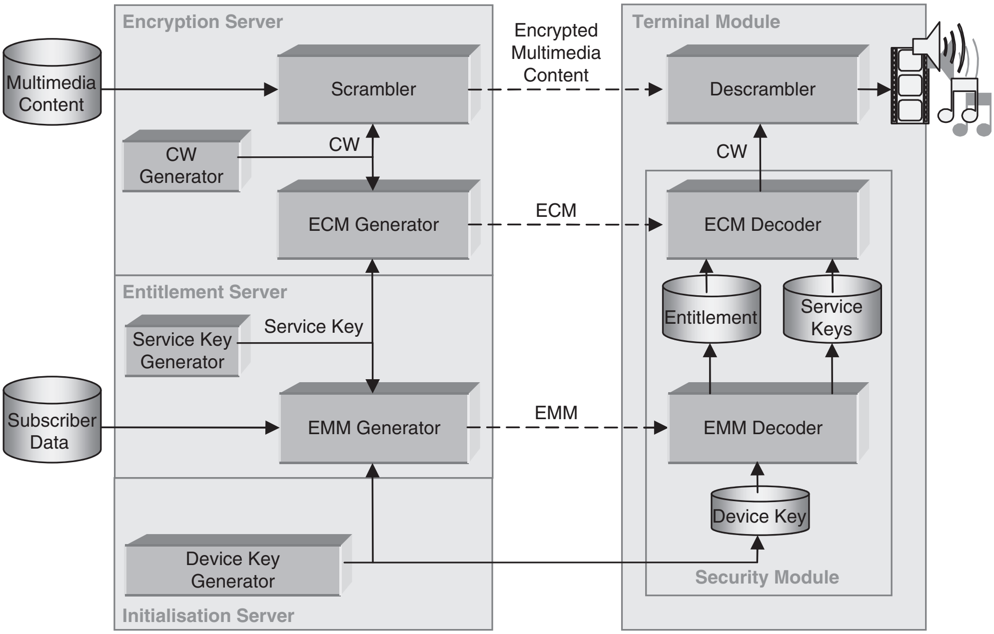

Figure 2.24 System Architecture of HECA – High Efficient Conditional Access system

Figure 2.24 shows a brief overview of its components and workflow.

The HECA Initialisation Server generates and delivers on demand from the device producer unique Device-IDs and device keys of high quality. The device producer implements them during the device manufacturing process.

At the request of a service provider, the HECA Entitlement Server configures services and stores the entitlements and subscriptions together with corresponding service keys in an internal or external database. It controls the service setup, service prolongation as well as token reload and it generates corresponding EMMs, which are interpreted and stored in the terminal device, respectively the security module.

The HECA Encryption Server scrambles service data with the corresponding control word, periodically generates ECMs and feeds both into the transmission chain.

The ECM is decoded in the receiver's security module. In the case of stored valid entitlement, the decrypted control word is provided for content descrambling.

2.4.3 Conditional Access Framework for DAB

Earlier versions of the DAB Standard [EN 300401] specified the scrambling and descrambling process and the coding of the related management messages. However, since the described concept was incomplete and not considered relevant, a more generic approach was chosen. Now a separate ETSI technical specification 'Digital

Audio Broadcasting (DAB); Conditional Access' [TS 102367] defines three possible levels where access control could be applied. It describes in detail how to signal scrambled content and where the related management messages have to be transported. It specifies neither a scrambling mechanism nor the coding of management messages.

Hence the technical specification is a framework that is open for the integration of several different CA or DRM systems and sets up corresponding tunnelling mechanisms.

Conditional access might be applied

- on DAB sub-channel level;
- on DAB data-groups level; or
- on MOT object level.

The underlying concept defining where related management messages have to be transported remains the same: They are not transported in separate additional channels that would need to be synchronised but on the same level, e.g. management messages for the encryption of a DAB sub-channel are found in the same sub-channel, management messages for DAB data-groups are transmitted within a data-group itself, and so on. This concept is essential for the accurately timed announcements of content key changes.

The three modes are pictured in the following sub clauses. In addition the technical specification allows the introduction of further proprietary modes.

2.4.3.1 DAB Sub-Channel Encryption

If access control addresses a complete DAB sub-channel, either an audio sub-channel possibly including PAD, an entire packet mode sub-channel or a stream mode sub-channel is protected. It is the most universal scrambling mode and represents the classical 'transport-channel-protection', shown in Figure 2.25.

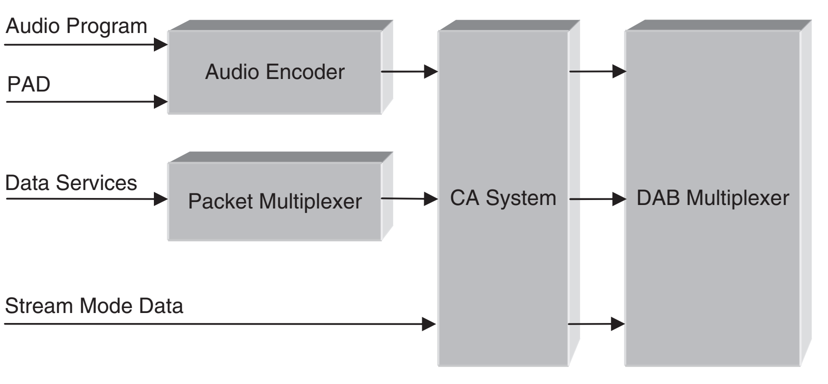

Figure 2.25 Location of access control system on DAB sub-channel level

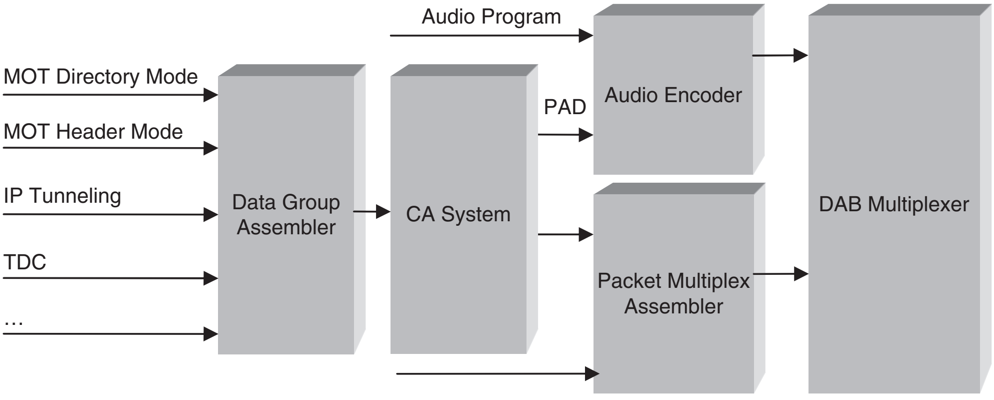

Figure 2.26 Location of access control system on DAB data-group level

2.4.3.2 DAB Data-Group Encryption

If protection of any data transfer protocol that uses MSC data-groups, e.g. MOT or TDC (see sections 4.3, 4.8.1), is desired, an access control system is placed directly after data-group-assembling as shown in Figure 2.26.

Data-group encryption protects MSC data-groups that are transported in PAD or in a packet mode sub-channel and might be applied to some data-groups or to all data-groups of a data application. A selective protection allows attractive service models, where the unscrambled part of an application can be used by any customer and receiver and the encrypted parts only by a subset. An example is a sports service, based on Journaline (see section 4.5.2): A table containing the results of the German Soccer League can be consumed by the public, whereas the information about the goal scorers etc. is encrypted and therefore addressed only to entitled users.

2.4.3.3 DAB MOT Encryption

Data applications that use the MOT directory mode, e.g. BWS, can be encrypted using an access control system on MOT level, see Figure 2.27. Within the same data application, scrambled objects can exist next to unscrambled ones. This leads to attractive business models. A weather service that uses MOT Broadcast Website (see section 4.4.2) is an example: Nationwide weather maps are presented by all receivers. Whereby regional weather forecast and additional information, e.g. wind strength can be decoded and presented only by entitled receivers that own the adequate usage rights and keys.

The MOT directory itself, which describes its structure, remains unscrambled.

In contrast to sub-channel encryption, scrambling on data-group level and on MOT level can be seen as an ‘object protection’ that extends the classical conditional access prospect as ‘transport-channel-protection’.

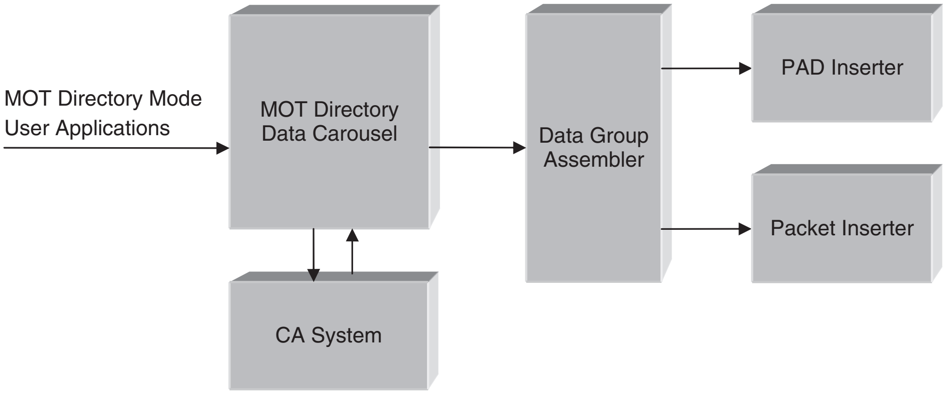

Figure 2.27 Location of access control system on DAB MOT level

2.5 Service Information

2.5.1 Introduction

Service Information (SI) provides additional information about the services carried in an ensemble and is intended to simplify service access and to provide attractive receiver features and functionality. Similar information carried in other ensembles and/or in FM (with or without RDS), DRM, AM (with or without AM Signalling System (AMSS)) services may also be signalled. Unlike the MCI (Multiplex Configuration Information), SI features are optional, that is broadcasters and receiver manufacturers may choose which to implement and when. An exception to this general rule is the user application information coded in FIG type 0 Extension 13. If data applications are broadcast, FIG type 0 Extension 13 must be used to signal type and parameters of the data application.

2.5.2 Coding of Labels

The Service Information uses various labels to textually identify ensemble, services, service components, X-PAD user applications or regions. All labels are encoded according to the same scheme.

Labels are encoded using FIG type 1 and optionally FIG type 2. In earlier revisions of the DAB standard, only FIG type 1 was used to signal labels of 16 bytes. Various Latin based characters sets were provided, but most receivers only supported the character set ‘Complete EBU Latin based repertoire’. This character set is suitable for many European countries, but not sufficient for international use like in Asian countries such as Korea or China. With the addition of the character set ISO 10646, many more characters are supported. ISO 10646 characters are encoded with UTF-8 (variable number of bytes per character) or UCS-2 (two bytes per character). Therefore, while a 16 byte label using character set ‘Complete EBU Latin based repertoire’ (one byte per character) permits labels comprising up to 16 characters, with character set ISO 10646 the length of the label was reduced to 5 (UTF-8 encoding) or 8 (UCS-2 encoding) Chinese characters. Many European receivers would also simply ignore the character set indicator and thus try to display Korean or Chinese labels with 'Complete EBU Latin based repertoire' characters, thus showing nonsensical labels.

In order to maintain backwards compatibility to existing receivers and increase the maximum label length for Asian characters, FIG type 2 was introduced. FIG type 2 supports labels of up to 16 characters. Since each character may comprise multiple bytes, the size of the label in bytes may be more than 16 bytes, e.g. up to 64 bytes in UTF-8 encoding. Therefore FIG type 2 supports segmentation of labels. A label can be segmented into up to 8 FIGs. The only permitted character set for FIG type 2 labels is ISO 10646. UTF-8 or UCS-2 encoding may be used. Both encodings use the same character set and only differ in the way a character number is encoded. UTF-8 uses a variable number of bytes per character, one byte to encode US-ASCII characters, two bytes for European special characters such as 'ä', 'à', 'ñ' or Greek, Cyrillic, Arabic, Hebrew, and three bytes for most other characters. With up to 6 bytes per character, UTF-8 is able to address all 2^{31} characters supported by ISO 10646. UCS-2 always uses two bytes per character. UCS-2 can therefore address the first 2^{16} (i.e., 65536) characters of ISO 10646, the so called Basic Multilingual Plane (BMP). UTF-8 is the more efficient encoding if most characters of the label are US-ASCII (i.e., they require a single byte per character) whereas UCS-2 is more efficient for most Asian scripts.

A receiver supporting FIG type 2 must support both UCS-2 and UTF-8 encoding. It is up to the broadcaster to choose the most efficient encoding individually for each label. If the broadcaster provides a label of FIG type 2, then he must also provide a FIG type 1 label with the same extension using the character set 'Complete EBU Latin based repertoire'.

ISO 10646 defines several thousand characters, but the receiver is not required to be able to render all characters. If a receiver is unable to display a FIG type 2 label, it must revert to the FIG type 1 label. The support of FIG type 1 labels and the 'Complete EBU Latin based repertoire' character set is therefore the minimum requirement for a receiver that presents labels. A more advanced receiver will also support ISO 10646 with both UCS-2 and UTF-8 encoding for FIG type 1 and FIG type 2 labels. The supported characters will depend on the target market.

FIG type 2 Extension 7 can be used to define characters using a 24×24 pixel grid. This feature is mainly used for Chinese characters not included in ISO 10646. Therefore a pixel grid is used that seems oversized for European characters, but is necessary for Chinese. FIG type 2 Extension 7 can be used to define characters that ISO 10646 does not provide (e.g., rarely used Chinese characters) or logo type images. The character number will be from the 'private use zone' (character codes 0xe000 to 0xf8ff) of the Basic Multilingual Plane (BMP) of ISO 10646. A character definition using FIG type 2 Extension 7 is valid for all labels signalled within the ensemble (including Dynamic Labels, see section 4.5.1).

Both FIG type 1 and FIG type 2 labels encode up to 16 characters and a character flag field of 16 bits (2 bytes). The 16 characters provide plain textual information. The character flag field is used to extract up to 8 characters from the provided 16 characters for displays that cannot present the full label. It contains a flag for each character in the label. When a bit is set to '1', the corresponding character of the label is included in an abbreviated label for display on receivers having a display of less than 16 characters. An example is shown in section 2.5.3.1. If the label comprises less than 16 characters, then the unused bits in the character flag field (having no corresponding character) must be set to ‘0’.

2.5.3 Basic Service Information

2.5.3.1 Ensemble Label

The ensemble label coded in FIG type 1 Extension 0 and FIG type 2 Extension 0 contains the Ensemble Identifier (EId) and a character field formed by a string which provides the listener with a clear textual description for the ensemble.

Example 2.1:

Ensemble label 16 characters:	__BBC_National__
Character flag field:	0011111110000100
Ensemble label 8 characters:	BBC_Natl

2.5.3.2 Date and Time

The date and time information in DAB is encoded in MJD (Modified Julian Date) and UTC (Co-ordinated Universal Time) fields. The MJD field is a continuous count of the number of days elapsed since 17 November 1858. The UTC field encodes the current time. As the name implies, MJD and UTC fields do not encode local time, but always UTC (Coordinated Universal Time), also referred to as Zulu time (Z). UTC is commonly used for communication systems, since it avoids confusion about time zones and daylight saving time.

The date and time feature is encoded in FIG type 0 Extension 10 which contains besides the *MJD* field and the *UTC* field also a Leap Second Indicator (LSI) and a confidence indicator. The ensemble provider should set the leap second indicator to ‘1’ throughout a UTC day that contains a leap second at the end of the same day. A leap second can theoretically occur at the end of any UTC month; but it is typically scheduled at the end of 30 June or 31 December. The leap second is introduced after 23:59:59 UTC. The confidence indicator is set to ‘1’ when the time information is within an agreed tolerance. The MJD is coded as a 17-bit binary number that increments daily at 00:00 (UTC) and extends over the range 0–99999. The following formula allows the MJD to be calculated from the year (Y), month (M) and date (D):

$$\mathrm{MJD} = 14956 + \mathrm{D} + \mathrm{int}\,(((\mathrm{Y} - 1900 - \mathrm{L}) \cdot 365.25) + \mathrm{int}\,((\mathrm{M} + 1 + \mathrm{L}\cdot 12) \cdot 30.6001).$$

For January and February (i.e. for M = 1 or M = 2) variable L = 1, otherwise L = 0.

Example 2.2:
28 October 2004 → $\mathrm{MJD} = 14956 + 28 + \mathrm{int}\,(((2004 - 1900 - 0) \cdot 365.25) + \mathrm{int}\,((10 + 1 + 0\cdot 12) \cdot 30.6001) = 53306$.

FIG 0/10 allows the time information to be provided at two different resolution levels. The standard resolution allows the time to be expressed in 1 minute intervals (short form of FIG 0/10) whilst the high-resolution level allows timing to be expressed in millisecond intervals (long form of FIG 0/10). Five bits are used to represent the hour (in the range 0–23) and 6 bits are used to represent the minutes. The long-form version contains additionally a 6-bit 'seconds' and a 10-bit 'milliseconds' sub-field.

Data applications such as the MOT Slideshow or the Electronic Programme Guide use the time provided by FIG 0/10 as the reference time. For applications that use FIG 0/10 as the reference time, an internal clock is required that is synchronised to the reference time sent by FIG 0/10. This internal clock must provide seconds accuracy no matter whether FIG 0/10 is broadcast in short form or long form.

If the **short form version of FIG 0/10** is used, then throughout the minute (seconds 00 to 59) the same FIG 0/10 will be broadcast. Therefore the receiver has to wait for a change in FIG 0/10 in order to determine when the (next) minute starts. This complicates synchronisation of the internal clock to the time sent by FIG 0/10 and provides poor accuracy. If the **long form version** is used, then after receiving a single FIG 0/10, the internal clock can be set very accurately. The time information carried in FIG 0/10 shall be taken to be the time of transmission of the start of the null symbol in the transmission frame carrying the time information in one of its FIBs.

It is recommended to use the long form version of FIG 0/10. If bit-rate for FIG 0/10 is limited, then it is recommended to reduce the repetition rate of FIG 0/10, but still use the long form version.

Example 2.3: 2004-10-28 14:45:25.050 (28 October 2004 at 14 h 45 min 25 sec 50 ms):

MJD	**Hours**	**Minutes**	**Seconds**	**Milliseconds**
1101000000111010	01110	101101	0110001	0000110010
(53306 = 28 Oct 2004)	(14)	(45)	(25)	(50)

2.5.3.3 Country, Local Time Offset, International Table

A number of basic, and mostly static, ensemble parameters are encoded in FIG type 0 Extension 9 (FIG 0/9):

The Ensemble ECC (Extended Country Code) specifies together with the country code of the EId a worldwide unique ensemble identifier. In an international ensemble it is possible to assign different ECCs to services. This is done in the extended field. The ECC is crucial for TMC (see section 4.6.1).

The Ensemble LTO defines the local time offset of the ensemble from the UTC transmitted in FIG type 0 Extension 10 (see section 2.5.3.2). It is expressed in multiples of half an hour (in the range −15.5 hours to +15.5 hours). It is possible that some services use a different LTO. This can be signalled in the extended field.

The Inter. Table Id indicates which international table is applicable in the ensemble. Two international tables are defined:

- Table 00000001 comprises the European table of RDS PTY codes (except for codes 30 and 31) as specified in [EN 50067] and the table of announcement types specified in [EN 300401].
- Table 00000010 comprises the RBDS list of PTY codes as specified in the RBDS standard and the table of announcement types specified in [EN 300401].

The coding allows the definition of future table identifiers, for example for use in Asian or African countries where there might be a need for another list of PTys and/ or another list of announcement types. The use of different international tables in one ensemble is not allowed. In Europe only Table 00000001 is applicable.

2.5.3.4 FIC Redirection

When there is a risk of overload in the FIC some FIGs may be transferred either as a whole or in part to the Auxiliary Information Channel (AIC). The AIC uses packet mode transport. FIC redirection is encoded in FIG type 0 Extension 31. The FIG type 0 flag field, the FIG type 1 flag field and the FIG type 2 flag field indicate which FIGs of type 0, type 1 or type 2 are carried in the AIC. All MCI FIGs should remain in the FIC. FIGs applying to other ensembles may be entirely transferred to the AIC. Other FIGs may be present partly in the FIC and partly in the AIC. For some of them the information is repeated more frequently in the FIC than in the AIC or vice-versa. This ensures that those receivers not equipped with a packet mode decoder will be able to access the basic data features though their performance may be inferior to other receivers. At the time of writing most receivers are not able to decode the AIC.

2.5.4 Service-related Information

2.5.4.1 Programme Service Label and Data Service Label

The service label feature provides the listener with a textual description of the service name. The character field is linked to a machine-readable Service Identifier (SId). In the case of a programme service label the SId comprises 16 bits and is similar in format to the RDS Programme Identification (PI) code: the 4 MSBs form the Country Id which distinguishes between 15 countries while the remaining 12 bits are the service reference. In the case of a data service label the SId comprises 32 bits: the 8 MSBs represent the service ECC, the next 4 bits the Country Id and the remaining 20 bits the service reference. The label of programme services is encoded in FIG type 1 Extension 1 and FIG type 2 Extension 1, the label of data services is encoded in FIG type 1 Extension 5 and FIG type 2 Extension 5.

Example 2.4: service labels

	Programme Service Label	Data Service Label
FIG	FIG 1/1 or FIG 2/1	FIG 1/5 or FIG 2/5
SId	16 bits	32 bits
	4 bits Country Id	8 bits service ECC
	12 bits service reference	4 bits Country Id
		20 bits service Reference
Example of service label: 16 characters	_ _BBC_R1_Digital_	_ _RADIO_1_TMC_ _ _
character flag field	0111111100000000	011 0 0 000111 1 1 100
Service label 8 characters	_BBC_R1_	_ R1_TMC_

2.5.4.2 Service Component Label

Service component labels are useful for the user to distinguish between service components of the same type. All service components are accessed at the service level, which means that service components can only be requested after their parent service has been selected. The receiver is required to support an additional layer of access: firstly to offer the services and present the primary service component. If the service also comprises secondary service components, then the receiver has to offer the components of the chosen service. The service component label is encoded in FIG type 1 Extension 4 and FIG type 2 Extension 4. The SCIdS is used to identify the service component within the service.

Service component labels may change frequently. It is recommended that the receiver follows any changes to these labels to avoid obsolete information remaining in the display.

Example 2.5: Service 'RADIO 1' with SId = F123 (hex) and service label 'RADIO 1' has in addition to the primary service component two secondary (audio) service components labelled as 'RADIO 1 EXTRA' and 'RADIO 1 PLUS':

Service	Service Label	SCIdS	Service Component Label
F123	RADIO 1	0001	RADIO_1_ EXTRA
		0010	RADIO_1_ PLUS

2.5.4.3 User Application Information

A DAB user application is a data application specified in a standard other than [EN 300401]. This implies that Dynamic Labels (see section 4.5.1) are not considered to be a user application. The most important information regarding data applications is the user application information provided in FIG 0/13. It tells the receiver which application software is needed to decode and present the broadcast data. The user application information is not part of the MCI and can thus be changed without needing a multiplex reconfiguration. For data service components (e.g., data applications transported in a packet mode sub-channel, a stream mode sub-channel or the FIDC), there will always be exactly one user application per service component. In case of data applications carried in the PAD channel of an audio service component, there can be up to six user applications per audio service component. All user applications within the PAD channel must be described by one single FIG 0/13 per audio service component.

For user applications carried in the PAD channel of an audio service component, FIG 0/13 also specifies DSCTy and other transport related parameters. Data service components signal the same parameters, but encoded in FIG 0/2 or FIG 0/3. The parameter X-PAD Application Type is used to separate the applications streams. The X-PAD Application Type (AppType) in a PAD channel thus has similar functionality as the packet address in a packet mode sub-channel.

Example 2.6:

Service 'RADIO 1' with SId = F123 (hex) and service label 'RADIO 1' has one primary programme service component with two X-PAD user applications; both X-PAD user applications are encoded in the same FIG 0/13:

Service	Service Label	SCIdS	User Application Type	X-PAD App Type	DSCTy
F123	RADIO 1	0001	0×002	12	60
			(Slideshow)		(MOT)
			0×44A	4	5
			(Journaline)		(TDC)

2.5.4.4 X-PAD User Application Label

X-PAD user application labels are useful for the user to distinguish between several user applications in the PAD channel of an audio service component. All X-PAD user applications are accessed at the audio service component level, which means that an X-PAD user application can only be requested after its parent service and then its (audio) service component has been selected. The X-PAD user application label is useful if the PAD channel carries multiple data applications that can not all be presented at the same time. For instance, if two Slideshows are carried in the same PAD channel, then the receiver might show the X-PAD user application labels to the user for selection. The X-PAD user application label is encoded in FIG type 1 Extension 6 and FIG type 2 Extension 6. The SCIdS identifies the service component within the service and the X-PAD AppType identify the X-PAD user application.

X-PAD user application labels may change frequently. It is recommended that the receiver follows any changes to these labels to avoid obsolete information remaining in the display.

Example 2.7: Service 'RADIO 1' with SId = F123 (hex), and service label 'RADIO 1' has one primary programme service component with two X-PAD data applications labelled 'WEATHER' and 'TRAFFIC':

Service	Service Label	SCIdS	X-PAD AppType	X-PAD user application label
F123	RADIO 1	0001	12	WEATHER
			4	TRAFFIC

2.5.4.4 Service Component Trigger

The beginning or the end of the transmission of a service component can be signalled by means of the Service Component Trigger (SCT) feature. It is primarily used with data services or multimedia services. In the case of receivers processing the FIC or the AIC in a continuous way, the SCT may be transmitted in the FIC or in the AIC at any time in order to signal the beginning or end of the transmission of a service component. The same SCT may be transmitted several times without any constraint on the transmission time. If a receiver is a member of a service user group and it satisfies the access conditions, it shall look for the service carried in the MSC in stream or packet mode or in the FIDC, at the time indicated by the SCT. Since there may be a delay between the real time of the reception of the service (component) and the time indicated in the SCT, the SCT should be transmitted by enough time in advance. For the start of the service, the actual beginning of the reception of the service (component) shall be later than the time indicated in the SCT. For the end of the service (component), when signalled, the end time indicated shall be later than the actual end time.

The SCT feature is encoded in FIG type 0 Extension 20 (FIG 0/20) which comprises the service component description in terms of the service component type (ASCTy or DSCTy). This FIG also signals where the component is located: in the FIDC or in the MSC in stream mode (short form of FIG 0/20) or in the MSC in packet mode (long form of FIG 0/20). FIG 0/20 further contains a number of flags. One of the flags signals whether the Logical Frame Number (LFN) applies to the beginning or end of service transmission. Another flag is used to signal whether the optional service user group information (defining specific groups of receivers) is present or not. Additional information may be provided to signal the time (UTC in 1 minute resolution) and conditional access parameters (CAId, SCCA).

If the time value is set to '0', or is absent, and if LFN = "1FFF" (hex), the transmission of the indicated service (component) is considered to have just started or ended.

2.5.4.5 Programme Number

The Programme Number (PNum) is a code that enables receivers and recorders designed to make use of this feature to respond to the particular programme item(s) that the user has pre-selected. In a receiver PNum may be implemented in a similar way as PTy in WATCH mode (see also section 2.5.5.2). However, whereas PTy WATCH is used to switch on or record a particular genre of programme, PNum allows a specific programme to be switched on or recorded.

PNum uses the scheduled programme time of the programme, to which the day of the month is added in order to avoid ambiguity. PNum together with the SId of the service that has scheduled the programme item form a unique identifier for this programme item. PNum is similar to the 16-bit PIN (Programme Item Number) in RDS. The first 5 MSBs (b15 to b11) represent the day of the month, in the range 01–31, the next 5 bits (b10 to b6) the hour, in the range 00–23, and the final 6 bits (b5 to b0) the minute, in the range 00–59. When the date part (b15 to b11) of PNum = 00000 special codes are allowed for the hours and minutes to signal a status code, a blank code or an interrupt code. PNum is encoded in FIG type 0 Extension 16 where it is associated with the SId to provide a unique reference for each programme. The Continuation flag indicates whether or not there will be a planned interruption while the Update flag indicates a re-direction to a different service (New SId) and time (New PNum). New SId thus indicates the SId of the target service of the redirection while New PNum specifies the new emission time or the time at which the programme will be resumed. It is coded in the same way as PNum.

The high data rate available in DAB makes it possible to develop more elaborate Electronic Programme Guide (EPG) systems for selection of services and programmes (see section 5.5).

2.5.5 Programme-related Features

2.5.5.1 Service Component Language

The Service Component Language (SC Lang) feature signals the language associated with the content of a service component. It is used for service (component) selection based on language rather than the SId, which is required for conventional service selection.

SC Lang serves two purposes:

1. In the case of audio service components, SC Lang is used to identify the spoken language associated with the audio content of a service component. SC Lang is required whenever a programme service has more than one secondary service component in a different language. If a service has no secondary service components (only a primary service component) then PTy language in FIG 0/17 (see

section 2.5.5.2) is used to signal the language of the programme. In this case SC Lang provides complementary redundant information that can be used for confirmation purposes.

2. In the case of data service components, SC Lang is used to identify the language associated with the content of the data. It allows the user to select data service components by dint of language.

SC Lang is encoded in FIG type 0 Extension 5 (FIG 0/5) where the language is associated with either the SubchId, the FIDCId or the SCId depending on whether the component is carried in the MSC in stream mode, in the FIC or in the MSC in packet mode. In the first case the short form of FIG 0/5 is used (L/S flag = '0'), in the latter case the long form is used (L/S flag = '1'). The MSC/FIC flag indicates whether, in the short form, the component is carried in the MSC in stream mode or in the FIC.

Example 2.8: The SC Lang for a service component carried in the MSC in stream mode is Swedish in sub-channel '3' and Finnish in sub-channel '4':

L/S	MSC/FIC flag	SubchId	Language
0	0 (MSC)	000011 (3)	00101000 (0x28) = Swedish
0	0 (MSC)	000100 (4)	00100111 (0x27) = Finnish

SC Language versus PTy Language

It is important to notice the subtle difference between SC language (FIG type 0 Extension 5) and PTy language (FIG type 0 Extension 17). SC Lang can be used to signal the language of any service component within an ensemble, being primary or secondary, audio or data. PTy language is intended to be used in combination with the programme types and it signals the language of the primary service component of a service. In addition, one language can be signalled for the secondary service component(s). Whereas SC language cannot be used to signal the language of service components in another ensemble, PTy language can be cross-referenced to another ensemble.

Whenever both SC language and PTy language are implemented their language fields should be consistent, that is, SC language should be equal to PTy language. If all services within an ensemble are in the same language, there is basically no need for signalling SC language. However, its use is strongly recommended because some receivers may have implemented a language filter based on SC language. In the case of static PTy language, the SC language should also be static. If the service provider has a service with programmes targeted at different language audiences then it should signal the language dynamically. If it has no means for dynamic signalling, static signalling is allowed on condition that the PTy language is set equal to '0'. SC language in this case is not transmitted.

2.5.5.2 Programme Type

The PTy feature provides another means of service access based on the categorisation of programme content by the service provider and the transmission of codes representing these categories to the receiver. The PTy classification in DAB has been extended beyond the 32 potential categories available with RDS. More than one code can be assigned at a time. This reduces the number of times that code '0' (no PTy) need be used and allows programmes to be more accurately described.

Static and Dynamic PTy Codes

Service providers may wish to implement the PTy feature in different ways. The first way minimises the expense of the studio infrastructure by keeping the codes relatively static. This approach serves to reflect the general service flavour, that is PTy codes (as well as the language, if signalled) are assigned to a service and remain unchanged for a long period and may be used to SEARCH for specific programme categories. It is obvious that in this case the PTy may not reflect the actual on-air programme but rather the format or genre of the service as a whole. Secondly, the PTy feature may be operated in a dynamic mode. In this case the code changes at programme item junctions, for example, from NEWS to SPORT or from NEWS in SWEDISH to NEWS in FINNISH. Dynamic PTy allows the receiver to operate in the WATCH mode as well as in SEARCH or SELECTION mode. Four PTy codes can be allocated to the primary service component of a programme service: two static PTy codes and two dynamic PTy codes.

The Int.code and Language Fields

PTy is encoded in FIG type 0 Extension 17 (FIG 0/17) where the SId of a service is linked to a language field, an Int. (International) PTy code and an optional complementary PTy code.

SId	S/D	Language	Int.code
A301	1	00001000	01110 (14)
		German	Classics
8217	1	00011101	00011 (03)
		Dutch	Information
F238	1	00001111	00100 (04)
		French	Sport
E265	1	00001010	01111 (15)
		Spanish	Other music

PTy of Primary and Secondary Service Components

The PTy categorisation applies to the primary audio service component of a service. Secondary service components, if available, cannot be individually categorised. They should be of the same 'type' as the primary component but they may or may not be given the same language. One language code can be assigned to a service, regardless of how many secondary components there are. The language therefore applies to the primary component, which is signalled by the *P/S flag* in FIG 0/17 being set to 'primary'. There is one situation where the language of a secondary component can be signalled. This is when the service comprises only two audio components, one primary and one secondary, and these two components have different languages.

Example 2.9: Suppose a Swedish service E456 has on-air a news programme in Swedish (as primary SC) and in Finnish (as secondary SC). This can be signalled by means of two FIGs 0/17:

SId	S/D flag	P/S flag	L flag	CC flag	Language	Int.code
E456	1 (dynamic)	0 (primary)	1	0	00101000	00001
					Swedish	News
E456	1 (dynamic)	1 (secondary)	1	0	00100111	00001
					Finnish	News

It is important to notice that when a service comprises secondary service components a SEARCH or WATCH will basically yield the primary service component.

Example 2.10: Service F123 with PTy code = SPORT has a primary service component transmitting a report of a TENNIS match and at the same time a secondary service component transmitting a report of a FOOTBALL match. A SEEK or a WATCH for SPORT will yield the report of the TENNIS match. If a service contains one (and not more than one) secondary service component in a different language a WATCH for the (common) PTy will yield the component in the desired language if the receiver has a language filter based on PTy language. In Example 2.9, of the Swedish service E456, a WATCH for NEWS in FINNISH will directly yield the Finnish news on the secondary service component of that service.

2.5.6 Announcements

The announcement feature is similar to the traffic announcement feature of RDS but on a larger scale, providing more categories such as news flashes or weather bulletins. Sixteen announcement categories can be coded of which 11 are currently defined.

2.5.6.1 The Interrupt Mechanism

Just like TA in RDS (see section 5.4.3) the announcement feature in DAB allows a listener to be directed from the current service/audio source to a programme service, which delivers a short audio message.

It is important to notice the subtle difference between the announcement feature and vectored dynamic PTys. 'News announcements' for example are basically short flashes, dynamic and unscheduled for short vectored interrupts. PTy 'News' on the other hand denotes a 'type of programme' which mostly lasts tens of minutes and which is scheduled in advance according to a programme number (see also section 2.5.4.5).

The DAB announcement interruption mechanism is dependent on the following filter criteria:

The type of announcement, i.e. is the type of announcement supported by the service and has the user selected it?

For every service which is allowed to be interrupted by announcements, static announcement support information is provided in FIG type 0 Extension 18 (FIG 0/18), which comprises 16 announcement support (Asu) flags. These indicate the announcement types (supported within the current ensemble) which can interrupt the service. This information may be used by the listener to select those announcement types for which he or she wants the audio source to be interrupted and to deselect those for which he or she doesn't want an interruption. The announcement switching information in FIG 0/19 comprises an announcement switching (Asw) flags field indicating the type of ongoing announcement; the SubChId to identify the sub-channel that generates the announcement, the cluster Id to identify the cluster to which the announcement is directed and optionally the region for which the announcement is applicable. The Asu flags are independent of the cluster Id. A special implementation of the announcement feature employs a dedicated announcement channel occupying its own sub-channel.

The cluster Id, i.e. does the service belong to the cluster to which the announcement is directed?

A cluster represents a group of services to which an announcement is directed with the intention of interrupting all the services participating in the cluster. A cluster Id can therefore be considered as 'a permission-to-interrupt' indicator. The announcement support information of FIG 0/18 includes a list of cluster Ids, which identify those announcement clusters a service is participating in. If an announcement is sent to a cluster that is not in the list, it will not interrupt the service. Cluster Id 0000 0000 specifies that an ongoing announcement is intended to interrupt the (audio) service components of all services having the signalled sub-channel identifier in their service organisation information. Cluster Id 1111 1111 is used for alarm announcements. The alarm announcement deviates from the other announcement types in that Cluster Id 11111111 causes the receiver to interrupt all services in the ensemble.

The Region Id (optional), i.e. does the service belong to the region to which the announcement is targeted and has the user selected this region?

When an announcement is targeted to a particular geographical region, the Region Id is appended to the announcement switching information of FIG 0/19. When the Region

Id is absent the announcement is targeted to the whole ensemble service area. In the receiver a region filter is considered to be a valuable tool that can allow listeners to select information about any region in the ensemble service area and not just in the area where the receiver is situated.

The New flag, i.e. is the announcement that is being broadcast a new message or not?

The New flag, signalled in the announcement switching information, is used by the service provider to distinguish between a message that is being broadcast for the first time ('new') and a message that is repeated ('old'). The New flag is suited for a cyclic announcement channel, that is a sub-channel reserved for repeated emission of announcements and possibly not belonging to a specific service. The detection of the flag by the receiver allows the listener (if desired) to avoid being interrupted by repeated messages announcements.

2.5.6.2 The Cluster Concept

Suppose a service provider is operating five services grouped in following three clusters:

Cluster Id 0000 0111 (7)	Cluster Id 0000 1000 (8)	Cluster Id 0000 1001 (9)
F301 (Radio 1)	F301 (Radio 1)	F302 (Radio 2)
F302 (Radio 2)	F303 (Radio 3)	F304 (Radio 4)
F303 (Radio 3)		

An example of possible announcement support information for these services by means of several FIGs 0/18 is shown below:

Example 2.11: Announcement support information

Sid	Asu Flags	No. of Clusters	Cluster Id 1	Cluster Id 2
F301	0000010000110010	00010	00000111 (7)	00001000 (8)
Radio 1	Traffic, News, Weather, Finance	(2)		
F302	0000000001000000	00010	00000111 (7)	00001001 (9)
Radio 2	Event	(2)		
F303	000001000001000	00010	00000111 (7)	00001000 (8)
Radio 3	Finance	(2)		
F304	0000001000000010	00001	00001001 (9)	
Radio 4	Sport, Traffic	(1)		

Assume that at a certain moment Radio 1 generates an area weather flash. Radio 1 is participating in two clusters: 7 and 8. If the service provider does not want to disturb the listeners of Radio 2 then it will direct, in the announcement switching information of FIG 0/19, the weather flash to cluster ‘8’ so that only Radio 3 listeners may be interrupted.

Cluster Id	Asw flags	New flag	SubchId	RegionId (lower part)
00001000	0000000000100000	1	011011 (27)	100010 (34)
(8)	Weather flash			Bretagne

This FIG 0/19 signals that the ongoing announcement is a new ‘weather’ announcement, generated in sub-channel ‘27’, directed to cluster ‘8’ and targeted to region ‘34’ (Bretagne). This weather flash will interrupt service F303 if the user has not deselected it and if either the receiver is located in Bretagne or the user has selected ‘Bretagne’ from the region menu.

It is clear that the cluster concept is a powerful tool for the service provider to control the interruption of services by the various announcements generated by the services in the multiplex. In multilingual ensembles, for example, the cluster concept can be used to prevent listeners from being interrupted by announcements in a language they don’t understand.

2.5.6.3 Other Ensemble Announcement Feature

Other ensemble (OE) announcement support information is encoded in FIG type 0 Extension 25 (FIG 0/25) and switching information in FIG type 0 Extension 26 (FIG 0/26). The latter is used to signal the identity of the ensemble providing the announcement, together with the Cluster Id Current Ensemble and the Cluster Id Other Ensemble.

In the same manner as for the basic announcement feature, the receiver can determine whether an interruption is permitted. The frequency information (see section 2.5.8.1) together with the EId Other Ensemble allows the receiver to determine the appropriate centre frequency of the other ensemble. The receiver can then retune to that frequency and decode the basic announcement switching information (FIG 0/19) in the other ensemble.

Consider several other ensembles with overlapping service areas and covering an area within the service area of the ensemble to which the receiver is tuned. When an announcement is ready to be broadcast in one of the other ensembles, the announcement switching information (FIG 0/19) for that ensemble is relayed to the provider of the tuned ensemble so that the appropriate OE service announcement switching information (FIG 0/26) can be generated.

The Region Id specified within the switching information allows the ensemble provider to derive and supply one or more appropriate region identifiers from knowledge of the service area of the OE service. The Region Ids must be chosen from those defined for use within the tuned ensemble. When Region Ids are not allocated for the current ensemble, Region Id current Ensemble = 000000 must be used. When Region Ids are not allocated for the other ensemble the region flag must be set to '0' or Region Id Other ensemble '000000' must be used.

Example 2.12: OE announcement support (FIG 0/25):

SId	Asu flags	No. of EIds	EId 1	EId2
F123	0000000000010000 Area weather flash	0010 (2)	F444	F555

This FIG 0/25 signals that in two other ensembles (F444 and F555) 'area weather flash' announcements are supported which can possibly interrupt service F123. It is important to notice that only one announcement support flag can be set at a time. For each other supported announcement type a separate FIG 0/25 must be transmitted.

Example 2.13: OE announcement switching (FIG 0/26):

Cluster Id CE	Asw flags	Region Id **CE**	EId OE	Cluster Id **OE**	Region Id **OE**
00000010 (2)	0000000001000000 Sport report	100010 (34) Bretagne	F555	00000111 (7)	001110 (14) Rennes

This FIG 0/26 signals that a 'sport report' announcement is now ongoing in the other ensemble F555 where it is directed to cluster '7'. This cluster corresponds to cluster '2' in the current ensemble, so that the announcement can possibly interrupt service F123. In ensemble F555 the event announcement is targeted to region '14' (Rennes) which maps to region '34' (Bretagne) in the current ensemble.

2.5.6.4 FM Announcements Feature

FM announcement support is encoded in FIG type 0 Extension 27 (FIG 0/27). FM announcement switching is encoded in FIG type 0 Extension 28 (FIG 0/28).

Consider an RDS service covering an area within the DAB ensemble coverage area. When a traffic announcement (TA) is ready to be broadcast on the RDS service, the 'TA' flag should be relayed to the DAB ensemble provider so that the appropriate FM service announcement switching information (FIG 0/28) can be generated. Great care has to be taken to synchronise the information that is transmitted at the start of the message. Therefore a raised TA flag must be present in the RDS service as long as the announcement is going on. The ensemble provider can supply one or more appropriate Region Ids from knowledge of the service area of the RDS service. The Region Ids must be chosen from those defined for use within the tuned ensemble. The selected region allows the receiver to filter out only those RDS traffic announcements relevant to that region. However, RDS services may not be available over the whole DAB ensemble coverage area. The receiver must therefore check as a background task, using the frequency information feature (see section 2.5.8.1), which RDS services are currently available at the receiver's location. Instead of the listener choosing a particular region(s), the relevant 'local' region may be established automatically by using an accurate receiver location-finding mechanism based on the TII feature (see sections 2.5.8.3 and 7.3.7) and regional identification features (see section 2.5.8.4).

Example 2.14: FM announcement support (FIG 0/27):

SId	Nr of PI codes	PI1	PI2	PI3	PI4
F123	0100 (4)	FA11	FB22	FC33	FC44

This FIG 0/27 signals that four RDS stations, FA11, FB22, FC33 and FD44, are TP stations on which traffic announcements can be generated which can possibly interrupt DAB service F123 in the current ensemble.

Example 2.15: FM announcement switching (FIG 0/28):

ClusterId CE	RegionId CE	PI
00000001 (1)	001110 (14) Rennes	FC33

This FIG 0/28 signals that an ongoing traffic announcement on RDS station FC33 (Radio Rennes) is relayed by the DAB ensemble provider to the current ensemble and directed to cluster '1' so that it can possibly interrupt DAB service F123. As the traffic announcement from Radio Rennes is rather of local importance, the DAB ensemble provider has decided to target it to region 'Rennes' rather than region 'Bretagne'.

2.5.7 Other Ensemble Services

The Other Ensemble Services (OE services) feature indicates which other ensembles are carrying a particular service. The OE services feature can be used for both programme services and data services. The feature is encoded in FIG type 0 Extension 24 (FIG 0/24) where a link between the SId of a service and EIds of other ensembles carrying that service is provided.

When the OE flag in the header of FIG 0/24 header is set to '0', FIG 0/24 provides a link between a service carried in the current ensemble and other ensembles carrying that service. When the OE flag is set to '1', FIG 0/24 provides a link between a service carried in another ensemble and other ensembles carrying that service.

Example 2.16 below illustrates the case where a programme service F123 (16 bits) is an unscrambled service which is carried in three other ensembles F444, F555, F666.

Sid	Rfa	CAId	Number of EIds	EId1	EId2	EId3
F123	0	000	0011	F444	F555	F666

Although the OE services feature is optional it becomes essential if the frequency information associated with a particular service needs to be signalled. However, a receiver, using the OE information alone in order to present the listener with a list of possible services from which to make a selection would produce a long list for the whole country, the majority of which may not be available to the listener. Therefore the Transmitter Identification Information (TII) feature (see section 2.5.8.3) is important because receivers using TII codes in order to identify the transmitter(s) they are tuned to estimate their own location. OE information should be kept up to date as far as possible. After a multiplex reconfiguration in another ensemble there may be some obsolete information sent for some time. This period should be minimised.

2.5.8 Tuning Aids

2.5.8.1 Frequency Information

The Frequency Information (FI) feature is used to signal the radio frequencies for other DAB ensembles and other broadcast services, such as FM (with or without RDS), DRM, AM (with or without AMSS).

When used in combination with the region definition feature (see section 2.5.8.4) a geographical area filter may be provided to allow the receiver to determine which of the (many) frequencies listed are worth checking in the area where it is situated.

The FI feature allows mobile receivers leaving the ensemble coverage area to retune to an alternative frequency (service following). The alternative frequency may apply to an identical DAB ensemble (same EId), another ensemble carrying the equivalent primary service component or an FM (with or without RDS), DRM, AM (with or without AMSS) service. The services can be identical (SId = PI or dummy code) or hard-linked (see section 2.5.8.2). If there is no alternative source for the same component the FI feature can help receivers to find a primary service component, which is soft-linked to the currently selected one.

The FI feature also helps receivers to get faster access to other DAB ensembles that are available in certain regions. An ensemble scan may be speeded up if the receiver knows the centre frequencies of all other DAB ensembles.

In combination with the OE/FM announcement feature (see sections 2.5.6.3, 2.5.6.4) the FI feature can establish a link between services and frequencies needed for announcement switching. In conjunction with other ensembles or the FM PTy WATCH function FI allows fast access to services starting to broadcast a programme item of the desired type.

FI Lists per Region Id

The FI feature is encoded in FIG type 0 Extension 21 (FIG 0/21) where, per Region Id, one or more FI lists are given. An FI list can contain up to two DAB ensemble frequencies and up to eight RDS frequencies so that a maximum of four DAB ensemble frequencies or 17 RDS frequencies can be signalled per region in one FIG 0/21.

Thanks to the Region Id the receiver can check only a restricted and yet directly relevant list of alternative frequencies but it requires the receiver to know where it is located. The Region Id is the full 11-bit version so that the region can, but need not, be a labelled region that is known to the user.

Region Id = '0' means that no area is specified. In this case the FI is valid for the whole coverage area of the signalled ensemble. The R&M field allows differentiating between DAB ensembles and FM services (with or without RDS), DRM and AM services (with or without AMSS; MW in steps of 9 kHz or 5 kHz).

Continuity Flag

The continuity flag is an indication whether 'continuous output' can be expected or not, for example, when switching to another DAB ensemble or when switching to equivalent and hard-linked services. The continuity flag may be set to '1' if the conditions listed in Table 2.10 are fulfilled.

The frequency of an ensemble is denoted as geographically adjacent if this ensemble is receivable somewhere in the region specified in the Region Id field. This is important in the case of a multiple frequency network (MFN) comprising several SFNs at different frequencies. Not-geographically-adjacent-area means that it is not guaranteed that the ensemble for which the centre frequency is signalled is receivable somewhere in the region specified in the Region Id field.

Importance of the OE Flag

Particular attention should be given to the OE flag in the sub-header of FIG 0/21.

The OE flag is set to '0' if the FI applies to the whole tuned ensemble or to an FM (with or without RDS), DRM or AM (with or without AMSS) service carrying a primary service component from the tuned ensemble.

The OE flag is set to '1' if the FI applies to frequencies of ensembles other than the tuned ensemble or to FM (with or without RDS), DRM or AM (with or without AMSS) services which are not identical to a primary service component from the tuned ensemble.

Table 2.10 *Conditions for putting continuity flag = "1'*

In case of frequencies of DAB ensembles (covering adjacent areas)	In the control field the transmission mode must be signalled. If the frequency of an ensemble is signalled, this ensemble must be synchronised in time and frequency with the current (tuned) ensemble. This means that the null symbols of corresponding transmission frames must coincide (tolerance = guard interval duration). The centre frequency of the signalled ensemble must not deviate by more than 10 Hz from the nominal CEPT frequency. Services present in the tuned and referenced ensemble should have identical CU start address, sub-channel size and error protection profile for their primary service component. For the primary service component the CA conditions need to be unique and the CIF counter values must be equal. *These continuity requirements apply to all frequencies in the "Freq. list" field.*
In case of frequencies of FM/AM services	The time delay between the audio signal of the DAB service and the analogue service must not exceed 100 ms.

Examples of FIG 0/21

Example 2.17: The following FIG 0/21 illustrates the signalling of the centre frequency of a DAB ensemble F888 in the region of Marseille. The OE flag is set to '1'. The ensemble operates in transmission mode II and is receivable somewhere in the Marseille region so that the control field is coded as 00100, meaning 'geographically adjacent, transmission mode II'.

OE flag	RegionId	Id field	R&M	Cont. flag	Length of freq. list	Control field	Freq 1
1	0x028	F888	0000	0	011 (3	00100	0x1C53C
	Marseille	(EId)	(DAB)		bytes)	geog. adj. mode II	(1463.232 MHz)

Example 2.18: The following FIG 0/21 illustrates the signalling of three RDS frequencies of France Inter for the region of Marseille.

RegionId	Id field	R & M	Cont. flag	Length of freq. list	Freq1	Freq2	Freq3
0x028	F201	1000	0	011	0x26	0x2A	0x4F
Marseille	(PI) Fr. Inter	(RDS)		(3 bytes)	91.3 MHz	91.7 MHz	95.4 MHz

2.5.8.2 Service Linking

When different services carry the same audio programme, service linking information is used so signal whether or not the services may be treated as identical; in this case the services are said to be 'hard-linked' together. The second use allows services to be generically related. This means that they do not necessarily carry the same audio but the kind of programmes supported is similar in some way, for example different local services from the same service provider. In this case the services are said to be 'soft-linked' together.

When the listener is moving outside the reception area of an ensemble, the service linking feature helps the mobile receiver to find alternatives if the originally tuned service is no longer available. The listener may in this way 'follow' and retain the same programme. In the case of dual DAB/RDS receivers the feature allows service following between the same service on DAB and on RDS. In the case of simulcast programmes in different ensembles or simulcasting on DAB and RDS common identifiers may be used instead of service linking because in this case frequency information provides all the information needed for service following.

Short and Long Form of FIG 0/6

Service linking is encoded in FIG type 0 Extension 6 (FIG 0/6) which has two versions: a short form and a long form.

The short form is used for rapid delinking or to signal a Change Event (CEI) and contains the Linkage Set Number (LSN) together with a number of flags such as the LA flag (Linkage Actuator), the S/H flag (Soft/Hard) and the ILS (International Linkage Set Indicator).

The long form has an Id list of service identifiers appended (PI codes or dummy codes referring to the linked services).

The LSN is a 12-bit code, unique to each set of services that are actually or potentially carrying a common programme.

The LA flag indicates if a service is actually (LA = 1) or potentially (LA = 0) a member of the set of programme services described by the LSN.

The ILS flag indicates that the link applies to only one country (national) or several countries (international).

The combination of the LSN, ILS and S/H parameters identifies a particular set of services and constitutes the "key" for the service linking feature. This key must be unique for the area where it can be received and also for any adjacent area. There is no relation between sets with a common LSN, but with different settings of the ILS. When a link ceases the LSN may be reused again for another set of services.

The IdLQ (Identifier List Qualifier) in the Id list usage field is used to indicate whether the Id in the Id list is a DAB SId, a DRM Sid, an AMSS Sid, an RDS PI code or a dummy code.

The Shd flag is a shorthand indicator used to indicate that the identifiers in the Id list, having the second nibble in the range '4' to 'F', each represents a list of up to 12 services each sharing the same country Id and the same 8 LSBs of the service reference.

Reference Identifier

If one of the services of a linkage set belongs to the current ensemble the OE flag in the FIG 0/6 header is set to '0'. The identifier of this service is a 'reference identifier' and must be included at the beginning of the Id list regardless of the IdLQ.

Example 2.19: Assume a set of 16 DAB services with SId = F211, F212, . . . , F227 and a set of 16 RDS services with PI = F301, F302, . . . , F316.

If the current ensemble contains service F211 then this is a reference service and is put first in the Id list. As there are in total 32 services to be put in the Id list, the list has to be split into four parts, carried in four different FIGs 0/6:

FIG index	OE	SIV	IdlQ	Id list
a	0	0 (start list)	01 (RDS PI)	F211 (reference), F301, . . . , F311
b	0	1 (cont. list)	00 (DAB SId)	F212, F213, . . . , F224
c	0	1 (cont. list)	00 (DAB SId)	F225, F226, F227
d	0	1 (cont. list)	01 (RDS PI)	F312, . . . , F316

If none of the services belongs to the current ensemble, the OE flag is set to '1'. In this case no reference identifier is defined so that the order of identifiers within the Id list is arbitrary.

Example 2.20: Same assumptions as above except that none of the services in the linkage set belong to the current ensemble:

FIG index	OE	SIV	IdlQ	Id list
a	1	0 (start list)	01 (RDS PI)	F301, F302, . . . , F312
b	1	1 (cont. list)	00 (DAB SId)	F211, F212, . . . , F223
c	1	1	01 (RDS PI)	F313, F314, F315, F316
d	1	1	00 (DAB SId)	F224, F225, F226, F227

2.5.8.3 Transmitter Identification Information Database

The Transmitter Identification Information (TII) database feature helps the receiver to determine its geographical location. TII codes can be inserted into the synchronisation channel of the transmitted DAB signal. The 'pattern' and 'comb' of the TII signals correspond to the MainId (Main identifier) and SubId (Sub-identifier) of the transmitter.

The TII database feature provides the cross-reference between these transmitter identifiers and the geographic location (in terms of coarse and fine longitude and latitude) of the transmitters. This can be useful in cases where a DAB receiver needs to make a decision to retune to another frequency and the decision criterion is marginal using the TII list alone. The TII database feature provides a much better specified area definition than is possible with only the geographical description in the region definition. If signals can be received from three or more transmitters, the receiver can use this information to perform a triangulation and pinpoint its position (see section 7.3.7).

The TII database feature is encoded in FIG type 0 Extension 22 (FIG 0/22). The M/S flag allows differentiation between signalling the reference location of the MainId (M/S flag = 0) and signalling the location of transmitters relative to that reference location (M/S flag = 1). The location reference does not necessarily need to coincide with a transmitter location, but if it does then the latitude offset and longitude offset are set to '0' for that transmitter.

For terrestrial frequency planning it may be necessary to delay somewhat the various transmitter signals constituting an SFN in order to optimise the combined signal at the receiver. The TII database feature therefore comprises a TD (Time Delay) field, which signals the constant time delay (in microseconds) of the transmitter signal in the range 0 to 2047 μs.

In the receiver the latitude of the received transmitter is calculated by converting the two's complement value given in the combined latitude coarse and fine fields (20 bits) to its decimal equivalent and then multiplying this value by $90°/2^{19}$. The longitude of the transmitter is calculated in a similar way but the decimal equivalent has to be multiplied by $180°/2^{19}$. The resolution of latitude is 0.6″ which is about 19 metres, while the resolution of longitude is 1.2″ which corresponds to about 38 metres at the equator (or 19 metres at 60° latitude).

When M/S flag = 1 the location of the transmitters relative to the reference location is given in terms of the latitude and longitude offset of the transmitter from the reference associated with the same MainId. The maximum offset from the reference location that can be signalled is 5.625° of latitude and 11.25° of longitude.

Example 2.21 illustrates the case where the reference position corresponds to a transmitter located at 48°07'29″ North and 001°37'33″ West and which has in its null symbol a TII code with pattern number = '3' and comb number = '1', which means that Main Id = 0000011 and Sub Id = 00001. The latitude is +48°07'29″ with 07' = (07/60) ° = 0.11666° and 29'' = (29/3600) ° = 0.008055° so that +48°07'29″ = 48.1247222°.

In FIG 0/22, for M/S = '0', the latitude is coded as a 20-bit number comprising a 16-bit latitude coarse field and a 4-bit latitude fine field. A two's complement number has to be calculated such that when the decimal equivalent 'L' of this number is multipliedby $90°/2^{19}$ the result is +48.1247222°. $L \cdot 90°/2^{19} = +48.1247222°$ so that L = +280346.8262 = 0100 0100 0111 0001 1010 or 0x4471A.

Example 2.22: Calculation of the latitude/longitude coarse and fine fields in FIG 0/22:

	Latitude Coarse	Longitude Coarse
Geographical co-ordinates	48°07'29" (North)	001°37'33" (West)
In decimal degrees	+48.1247222°	−1.6258333°
Formula	$L \cdot (90°/2^{19}) = +48.124722$	$L \cdot (180°/2^{19}) = -1.625833$
Decimal value of L	+280346	−4735
Binary value of L	0100 0100 0111 0001 1010	0000 0001 0010 0111 1111
One's complement of L		1111 1110 1101 1000 0000
Two's complement of L		1111 1110 1101 1000 0001
Value to be put in coarse field	0100 0100 0111 0001 (0x4471)	1111 1110 1101 1000 (0xFED8)
Value to be put in fine field	1010	0001

Since L is a positive number the two's complement of L corresponds to the binary value. The 16 MSBs of this number are put in the latitude coarse field and the 4 LSBs are put in the latitude fine field. The values for the longitude coarse and fine fields can be derived in a similar way. Since the longitude is West of Greenwich, value L is negative. The examples show how the values to be put in the longitude coarse and fine fields are derived from the two's complement number.

This results in the following coding of FIG 0/22 for M/S = 0 (Main Identifier):

M/S	MainId	Latitude coarse	Longitude coarse	Latitude fine	Longitude fine
0	0000011 (3)	0100 0100 0111 0001 (0x4471)	1111 1110 1101 1000 (0xFED8)	1000 (8)	0101 (5)

Since transmitter 3/1 is the reference position the latitude offset and longitude offset are equal to zero so that FIG 0/22 for M/S = 1 for transmitter 3/1 would look like:

M/S	MainId	SubID	Time delay	Latitude offset	Longitude fine
0	0000011 (3)	00001 (1)	00000000000 (0 µs)	0000 0000 0000 0000 (0x0000)	0000 0000 0000 0000 (0x0000)

Suppose that around this reference position in the SFN there are three transmitters (3/2, 3/3, 3/4) with MainId = 0000011 (3) and with SubId = 00010 (2), 00011 (3) and 00100 (4). Suppose that the latitude offset for transmitter 3/2 is 00°13'14" North and the longitude offset is –000°14'15" West (see Example 2.23). The time delay for this transmitter is 14 µs.

Example 2.23: Calculation of the latitude/longitude offset fields in FIG 0/22

	Latitude Offset	Longitude Offset
Geographical co-ordinates	00°13'14'' North	000°14'15'' West
Decimal degrees	+0.220555°	−30.2375°
Formula	$L \cdot (90°/2^{19}) = +0.220555°$	$L \cdot (180°/2^{19}) = -0.2375°$
Decimal value of L	+1285	–692
Binary value of [L]	0000 0101 0000 0101	0000 0010 1011 0100
One's complement of L	1111 1101 0100 1011	
Two's complement of L	1111 1101 0100 1100	
Hexadecimal value of L	0x0505	0xFD4C

FIG 0/22 for M/S = 1 (Sub Identifier) for transmitter 3/2 could look like:

M/S	MainId (7 bits)	SubId (5 bits)	Time delay (11 bits)	Latitude offset (16 bits)	Longitude offset (16 bits)
1	0000011	00010	00000001110	0000 0101 0000 0101	1111 1101 0100 1100
	(3dec)	(2dec)	(14 µs)	(0x505)	(0xFD4C)

The coordinates of transmitter '3/2' are +48°7'29'' +00°13'14'' = +48°20'43'' latitude North and –001°37'33'' –000°14'15'' = –001°51'48'' longitude West.

2.5.8.4 Region Definition and Region Label

Geographical regions within an ensemble coverage area can be identified by means of the region definition feature of FIG type 0 Extension 11 (FIG 0/11) and the region label feature of FIG type 1 Extension 3 (FIG 1/3). The region definition feature, which is based on lists of transmitter identification codes, allows a receiver to determine roughly where it is located. This knowledge allows a receiver to filter out information which is relevant only to that region. For example, a much shorter list of alternative frequencies can be checked for service following and alarm announcements can be confined to the region concerned.

The region label feature of FIG 1/3 provides a textual description of a region (e.g. Bretagne, Oberbayern, Kent, etc.). Region labels help a listener to select a region in conjunction, for example, with the announcement feature. Traffic information concerning the destination and the route can be requested as well as that for the starting point of a journey. The user can select, say, 'Kent' from a region menu if he or she is interested in getting information from this region or for filtering (traffic or weather) announcements.

A region is uniquely identified by means of an 11-bit Region Id, which is composed of two parts: a 5-bit upper part and a 6-bit lower part. This allows up to 2047 regions to be identified within an ensemble coverage area. The region label uses only the 6-bit lower part (the 5-bit upper part is set to all zeros) so that up to 64 regions per ensemble can be labelled.

Description of a Region

With FIG 0/11 there are two ways to describe a region:

1. By means of a list of transmitter identifiers (TII list). In this case a Region Id is associated to a group of transmitters with one common Main Id and different Sub Ids. By means of this list the receiver can determine in which region it is located. This is useful for receiver processing in the case of frequency information to select a suitable alternative frequency.

Example 2.24:

Region Id	Main Id	SubId1	SubId2	SubId3	SubId4	SubId5
00000 10110	0001011	00001	00010	00011	00100	00101
(0x002E)	(0x0B)	Crystal	Alexandra	Guildford	Bluebell	Reigate
Greater		Palace	Palace		Hill	
London						

2. By means of a spherical rectangle. In this case a Region Id is associated to a geographical area in terms of a spherical rectangle from which the coordinates of the South-west corner are given as latitude/longitude coarse coded as a 16-bit two's complement number. The size of the spherical rectangle is defined by the extent of latitude/longitude coded as a 12-bit unsigned binary number. Region identification based on the spherical rectangle allows navigation systems based on GPS to be used for automatic receiver location.

Example 2.25:

RegionId	Latitude coarse	Longitude coarse	Extent of latitude	Extent of longitude
00000000001	1101001000111010	1111100111010010	000111111111	000010100000
(0x001)	(0xD23A)	(0xF9D2)	(0x1FF)	(0x0A0)
	–32°11'12" (South)	–008°41'26" (West)	(01°24'12")	(000°52'44")
			±156 km	±100 km at
				equator

Importance of the G/E Flag

The G/E flag in FIG 0/11 determines the validity area of the Region Id, that is whether the Region Id applies to the ensemble (E) coverage area only (e.g. the Bretagne ensemble), or to a 'global' (G) coverage area, that is the area defined by the country Id and ensemble ECC (e.g. the whole of France).

Each ensemble provider in a country may use none, some or all of the globally defined Region Ids that it finds appropriate for its purposes. All other Region Ids may be defined for use ensemble-wide, even if they were globally defined through the above-mentioned agreement but not used as global ones in this ensemble.

Suppose that in France there is an agreement to use code 00000001110 (0x000E or 14 dec) as a globally defined Region Id to denote 'Bretagne'. An ensemble provider in Paris may use this code in its Paris ensemble to denote 'Versailles' on condition that the provider puts in FIG 0/11 the G/E flag = 0 (E) for Region Id 00000001110 and refrains from using it as the global code for 'Bretagne'.

For announcements in other ensembles or in FM the selected region allows the receiver to filter out only those announcements relevant to that region. However, RDS or OE services may not be available over the whole DAB coverage area. The receiver must therefore check as a background task, using the FI feature, which RDS or OE services are currently available at the receiver's location in order to prevent the receiver from retuning to an RDS or OE service and finding that it cannot be received.

3

Audio Services and Applications

3.1 General

The DAB system is designed to provide radio services and additional data services. This chapter focuses on the main audio services as the central application. The audio services use MPEG Audio Layer II to provide mono, stereo and multichannel programmes to the listener. Although the first broadcast applications and receivers will only supply stereo programmes, the migration to multichannel services is nevertheless already included in the system.

Also, data services will become more and more important and one can currently see many new ideas on their use being developed and tested. Most advanced is the Internet, but it is easy to envisage that the data capacity of the DAB system can be used for very similar services, for more details also see Chapter 4.

With the introduction of the compact disc and its 16-bit PCM format, digital audio became popular, although its bit rate of 706 kbps per monophonic channel is rather high. In audio production resolutions of up to 24-bit PCM are in use. Lower bit-rates are mandatory if audio signals are to be transmitted over channels of limited capacity or are to be stored in storage media of limited capacity. Earlier proposals to reduce the PCM rates have followed those for speech coding. However, differences between audio and speech signals are manifold since audio coding implies higher values of sampling rate, amplitude resolution and dynamic range, larger variations in power density spectra, differences in human perception, and higher listener expectations of quality. Unlike speech, we also have to deal with stereo and multichannel audio signal presentations. New coding techniques for high-quality audio signals use the properties of human sound perception by exploiting the spectral and temporal masking effects of the ear. The quality of the reproduced sound must be as good as that obtained by 16-bit PCM with 44.1 or 48 kHz sampling rate. If for a minimum bit rate and reasonable complexity of the codec no perceptible difference between the original sound and the reproduction of the decoded audio signal exists, the optimum has been achieved. Such a source

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

coding system, standardised by ISO/IEC as MPEG-1, MPEG-2 Audio or newer standards, allows a bit-rate reduction from 768 kbps down to about 100 kbps or less per monophonic channel, while preserving the subjective quality of the digital studio signal for any critical signal. This high gain in coding is possible because the noise is adapted to the masking thresholds and only those details of the signal are transmitted which will be perceived by the listener.

3.2 Audio Coding and Decoding

3.2.1 Perceptual Coding

3.2.1.1 General

Two different mechanisms can be used to reduce the bit rate of audio signals. One mechanism is determined mainly by removing the redundancy of the audio signal using statistical correlation. Another one reduces the irrelevancy of the audio signal by considering psycho-acoustical phenomena, like spectral and temporal masking. Only with both of these techniques, making use of the statistical correlation and the masking effects of the human ear, a significant reduction of the bit rate down to 200 kbps per stereophonic signal and below could be obtained. Such coding schemes are called perceptual coding, in contrast to other technologies, such as wave form coding, source oriented coding (like speech coding), or lossless coding. Most of recent perceptual coding systems were developed by the MPEG (Moving Picture Experts Group) and are standardised in ISO/IEC International Standards etc. as described below.

3.2.1.2 Masking of Audio Components

In order to reduce the bit rate of an audio signal whilst retaining the bandwidth, dynamic structure and harmonic balance, an audio codec has to reduce the quantisation step size of the signal. A CD signal is typically quantised with 16 bits per sample, which for a 0 dBFS sine wave results in a signal-to-quantisation-noise ratio of 96 dB. Reducing the quantisation step size will increase the quantisation noise relative to the audio signal. If the human auditory system were susceptible to this increase under all conditions, we could not reduce the bit rate.

Fortunately, the human ear is a very imperfect organ. When we are exposed to a tone at one frequency, this excites the inner ear and the sensory cells in the cochlea in such a way that signals close (in frequency) to this signal may be inaudible, if their level is sufficiently low. This effect, which is called masking, decreases with the distance of the so-called masker and the masked signal [Zwicker, 1967].

A second imperfection is the fact that we do not hear audio signals below a certain level. This forms an absolute hearing threshold.

Both effects vary for human beings and have been measured in the past. They form the basis of what is called perceptual coding. The way to exploit these effects in a codec is to split the signal into small frequency regions (bands) and increase the quantisation noise in each frequency region such that it is as large as possible while still being masked by the signals in that frequency region.

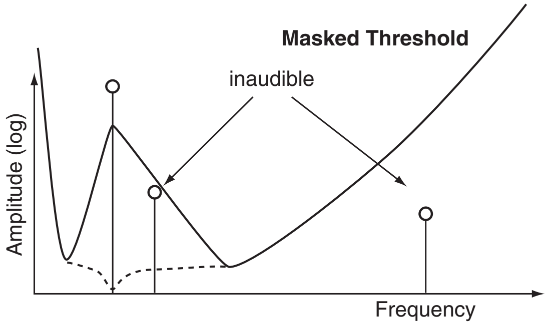

Figure 3.1 Masking of individual sine tone and absolute hearing threshold

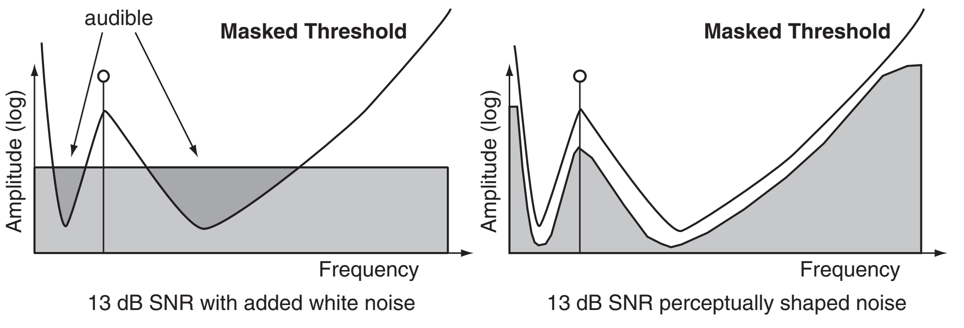

Figure 3.2 Added quantisation noise, without (left) and with (right) shaping

The effect of masking of a single sine tone and its combination with the absolute hearing threshold is depicted in Figure 3.1. Two sine tones, which are masked, are shown in the figure as well.

In Figure 3.2, the same amount of noise has been added to a sine tone. In the left part of the figure the noise has equal amplitude across all frequencies and is therefore audible. In the right half of the figure the noise has the same energy but has been shaped according to the masking curve. In this case the noise is inaudible. In the MPEG Audio codecs the shaping of the noise is achieved by splitting the signal into 32 sub-bands before the reduction of the quantisation step size.

The masking properties have been measured for single sine tones and for narrow-band noise signals [Zwicker, 1967]. The amount of masking depends on the nature of a signal and furthermore on the perceived tonality of a signal. The more tonal a signal is, the stronger the masking. It is therefore necessary to measure the tonality of a component to decide its masking properties. A ready assumption that is made in perceptual coding is that masking also occurs for complex signals, such as music and speech.

3.2.2 MPEG-1/2 Audio Layer II

3.2.2.1 Basic Principles

The block diagram of the MPEG-1 Audio Layer II encoder, called 'MUSICAM' [Dehery, 1991] during its time of development and before standardisation in ISO/IEC, is shown in Figure 3.3.

Consequently, Audio Layer II is identical to the MUSICAM coding scheme, whereas Audio Layer I has to be understood as a simplified version of the MUSICAM system. The basic structure of the coding technique which is more or less common to both, Layer I and Layer II, is characterised by the fact that MPEG Audio is based on perceptual audio coding.

One of the basic functions of the encoder is the mapping of the 20 kHz wide PCM input signal from the time domain into sub-sampled spectral components. For both layers a polyphase filterbank which consists of 32 equally spaced sub-bands is used to provide this functionality.

The output of a Fourier transform which is applied to the broadband PCM audio signal in parallel to the filter process is used to calculate an estimate of the actual, time-dependent masked threshold. For this purpose, a psycho-acoustic model, based on rules from psycho-acoustics, is used as an additional function block in the encoder. This block simulates spectral and, to a certain extent, temporal masking too. The sub-band samples are quantised and coded with the intention to keep the noise, which is introduced by quantising, below the masked threshold. Layers I and II use a block compounding technique with a scale factor consisting of 6 bits, valid for a dynamic range of about 120 dB and a block length of 12 sub-band samples. With this kind of scaling technique,

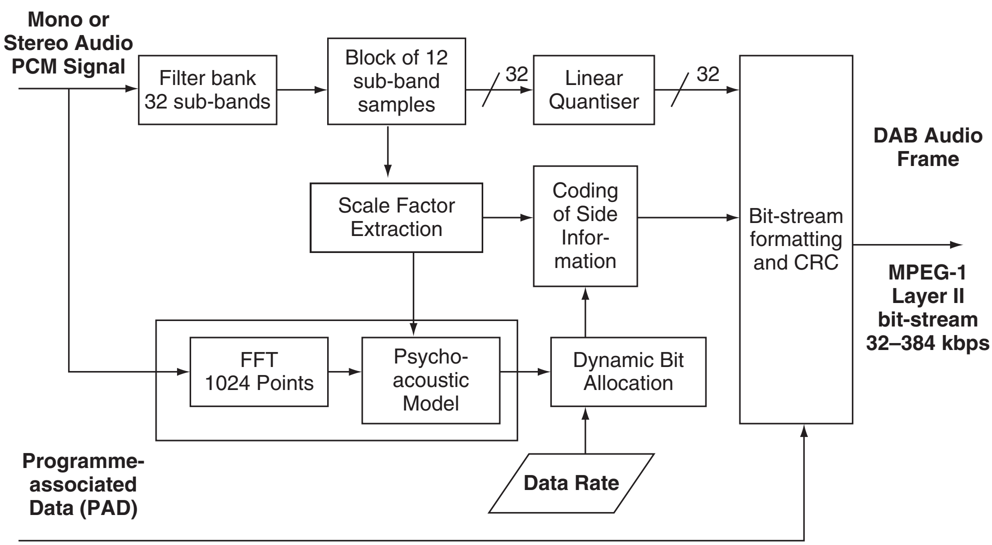

Figure 3.3 Block diagram of the MPEG-1 Audio Layer II encoder, as used in the Eureka147 DAB system

Audio Layer I and Audio Layer II can deal with a much higher dynamic range than CD or DAT, that is conventional 16-bit PCM.

In the case of stereo signals, the coding option 'Joint Stereo' can be used as an additional feature. It exploits the redundancy and irrelevancy of typical stereophonic programme material, and can be used to increase the audio quality at low bit rates and/or reduce the bit rate for stereophonic signals [Waal, 1991]. The increase of encoder complexity is small, and negligible additional decoder complexity is required. It is important to mention that joint stereo coding does not enlarge the overall coding delay.

After encoding of the audio signal an assembly block is used to frame the MPEG Audio bit-stream which consists of consecutive audio frames.

The frame-based nature of MPEG Audio Layer II leads to a delay through the encode/decode chain which is of the order of two frames (48 ms) when the processing and the tie for transmission are not taken into account. Typical delays through real transmission chains are of the order of 80 ms, see also section 2.2.5.

3.2.2.2 Psychoacoustic Model

The psycho-acoustic model calculates the minimum masked threshold, which is necessary to determine the just-noticeable noise level for each band in the filterbank. The minimum masked threshold is the sum of all individual masking curves as described in the previous section. The difference between the maximum signal level and the minimum masked threshold is used in the bit or noise allocation to determine the actual quantiser level in each sub-band for each block. Two basic psycho-acoustic models are given in the informative part of the MPEG-1 standard [IS 11172]. While they can both be applied to any layer of the MPEG Audio algorithm, in practice most encoders use model 1 for Layers I and II. In both psycho-acoustic models, the final output is a signal-to-mask ratio for each sub-band of Audio Layer II. A psycho-acoustic model is necessary only in the encoder. This allows decoders of significantly less complexity. It is therefore possible to improve the performance of the encoder even later, by relating the ratio of bit rate and subjective quality. For some applications, which do not demand a very low bit rate, it is even possible to use a very simple encoder without any psychoacoustic model.

The fundamental basis for calculating the masked threshold in the encoder is given by the results of masked threshold measurements for narrow-band signals considering a tone masking noise and vice versa. Concerning the distance in frequency and the difference in sound pressure level, very limited and artificial masker/test-tone relations are described in the literature. The worst case results regarding the upper and lower slopes of the masking curves have been considered, assuming that the same masked thresholds can be used for both simple audio and complex audio situations.

The output of the FFT is used to determine the relevant tonal (i.e. sinusoidal) and nontonal (i.e. noise) maskers of the actual audio signal. It is well known from psychoacoustic research that the tonality of a masking component has an influence on the masked threshold. For this reason, it is worthwhile to discriminate between tonal and nontonal components.

It is also known that the perception of the tonality of a component is time dependent. A pure tone is not perceived as tonal initially and builds up over time. This time dependency is part of more sophisticated psycho-acoustic models.

The individual masked thresholds for each masker above the absolute masked threshold are calculated depending on frequency position, loudness level and tonality. All the individual masked thresholds, including the absolute threshold, are added to the so-called global masked threshold. For each sub-band, the minimum value of this masking curve is determined. Finally, the difference between the maximum signal level calculated by both the scale factors and the power density spectrum of the FFT and the minimum masked threshold is calculated for each sub-band and each block.

The block length for Audio Layer II is determined by 36 sub-band samples, corresponding to 1152 input audio PCM samples. This difference of maximum signal level and minimum masked threshold is called the signal-to-mask ratio (SMR) and is the relevant input function for the bit allocation.

3.2.2.3 The Filterbank

A high-frequency resolution is small sub-bands in the lower frequency region, whereas a lower resolution in the higher frequency region with wide sub-bands should be the basis for an adequate calculation of the masked thresholds in the frequency domain. This would lead to a tree structure of the filterbank. The polyphase filter network used for the sub-band filtering has a parallel structure, which does not provide sub-bands of different widths. Nevertheless, one major advantage of the filterbank is given by adapting the audio blocks optimally to the requirements of the temporal masking effects and inaudible pre-echoes. A second major advantage is the small delay and complexity. To compensate for the lack of accuracy of the spectrum analysis of the filterbank, a 1024-point FFT for Audio Layer II is used in parallel with the process of filtering the audio signal into 32 sub-bands.

The prototype QMF filter is of order 511, optimised in terms of spectral resolution and rejection of side lobes, which is better than 96 dB. This rejection is necessary for a sufficient cancellation of aliasing distortions. This filterbank provides a reasonable trade-off between temporal behaviour on one side and spectral accuracy on the other side. A time/frequency mapping providing a high number of sub-bands facilitates the bit rate reduction, due to the fact that the human ear perceives the audio information in the spectral domain with a resolution corresponding to the critical bands of the ear, or even lower. These critical bands have a width of about 100 Hz in the low-frequency region, which is below 500 Hz, and a width of about 20% of the centre frequency at higher frequencies.

The requirement of having a good spectral resolution is unfortunately contradictory to the necessity of keeping the transient impulse response, the so-called pre- and post-echo, within certain limits in terms of temporal position and amplitude compared to the attack of a percussive sound. Knowledge of the temporal masking behaviour [Fastl, 1977] gives an indication of the necessary temporal position and amplitude of the pre-echo generated by a time/frequency mapping in such a way that this pre-echo, which normally is much more critical compared to the post-echo, is masked by the original attack. In association with the dual synthesis filterbank located in the decoder, this filter technique provides a global transfer function optimised in terms of perfect impulse response perception.

3.2.2.4 Scale Factors

The sub-band samples are represented by a combination of scale factor and actual sample value. The scale factor is a coarse representation of the amplitude of either one individual sample or a group of samples. This approach allows for a higher reduction of the bit rate than to code each individual sample.

The calculation of the scale factor for each sub-band is performed for a block of 12 sub-band samples. The maximum of the absolute value of these 12 samples is determined and quantised with a word length of 6 bits, covering an overall dynamic range of 120 dB per sub-band with a resolution of 2 dB per scale factor class. In Audio Layer I, a scale factor is transmitted for each block and each sub-band, which has no zero-bit allocation.

Audio Layer II uses additional coding to reduce the transmission rate for the scale factors. Owing to the fact that in Layer II a frame corresponds to 36 sub-band samples, that is three times the length of a Layer I frame, three scale factors have to be transmitted in principle. To reduce the bit rate for the scale factors, a coding strategy which exploits the temporal masking effects of the ear has been studied. Three successive scale factors of each sub-band of one frame are considered together and classified into certain scale factor patterns. Depending on the pattern, one, two or three scale factors are transmitted together with additional scale factor select information consisting of 2 bits per sub-band. If there are only small deviations from one scale factor to the next, only the bigger one has to be transmitted. This occurs relatively often for stationary tonal sounds. If attacks of percussive sounds have to be coded, two or all three scale factors have to be transmitted, depending on the rising and falling edge of the attack.

This additional coding technique allows on average a factor of 2 in the reduction of the bit rate for the scale factors compared with Layer I.

3.2.2.5 Bit Allocation

Before the adjustment to a fixed bit rate, the number of bits that are available for coding the samples must be determined. This number depends on the number of bits required for scale factors, scale factor select information, bit allocation information, and ancillary data.

The bit allocation procedure is determined by minimising the total noise-to-mask ratio over every sub-band and the whole frame. This procedure is an iterative process where, in each iteration step, the number of quantising levels of the sub-band that has the greatest benefit is increased with the constraint that the number of bits used does not exceed the number of bits available for that frame. Layer II uses only 4 bits for coding of the bit allocation information for the lowest, and only 2 bits for the highest, sub-bands per audio frame.

3.2.2.6 Sub-band Samples

First, each of the 12 sub-band samples of one block is normalised by dividing its value by the scale factor. The result is quantised according to the number of bits spent by the bit allocation block. Only odd numbers of quantisation levels are possible, allowing an exact representation of a digital zero. Layer I uses 14 different quantisation classes, containing $2n - 1$ steps, with $2 <= n <= 15$ different quantisation levels. This is the same for all sub-bands. Additionally, no quantisation at all can be used, if no bits are allocated to a sub-band.

In Layer II, the number of different quantisation levels depends on the sub-band number, but the range of the quantisation levels always covers a range of 3 to 65535 with the additional possibility of no quantisation at all. Samples of sub-bands in the low-frequency region can be quantised with 15, in the mid frequency range with seven, and in the high-frequency range only with three different quantisation levels. The classes may contain 3, 5, 7, 9, 15, 63, ..., 65535 quantisation levels. Since 3, 5 and 9 quantisation levels do not allow an efficient use of a codeword, consisting only of 2, 3 or 4 bits, three successive sub-band samples are grouped together into a 'granule'. Then the granule is coded with one codeword. The coding gain by using the grouping is up to 37.5%. Since many sub-bands, especially in the high-frequency region, are typically quantised with only 3, 5, 7 and 9 quantisation levels, the reduction factor of the length of the code words is considerable.

3.2.2.7 Audio Layer II Bit-stream Structure

The bit-stream of Audio Layer II was constructed in such a way that a decoder of both low complexity and low decoding delay can be used, and that the encoded audio signal contains many entry points with short and constant time intervals. The encoded digital representation of an efficient coding algorithm specially suited for storage application must allow multiple entry points in the encoded data stream to record, play and edit short audio sequences and to define the editing positions precisely. For broadcasting this is important to allow fast switching between different audio programmes.

To enable a simple implementation of the decoder, the frame between those entry points must contain all the information that is necessary for decoding the bit-stream. Owing to the different applications, such a frame has to carry in addition all the information necessary for allowing a large coding range with many different parameters. In broadcasting, frequent entry points in the bit-stream are needed to allow for an easy block concealment of consecutive erroneous samples impaired by burst errors.

The format of the encoded audio bit-stream for Layer II is shown in Figure 3.4. Short, autonomous audio frames corresponding to 1152 PCM samples characterise the structure of the bit-stream. Each audio frame starts with a header, followed by the bit allocation information, scale factor and the quantised and coded sub-band samples. At the end of each audio frame is the so-called ancillary data field of variable length that can be specified for certain applications. Each frame can be accessed and decoded on its own. With 48 kHz sampling frequency, the frame duration is 24 ms for Layer II.

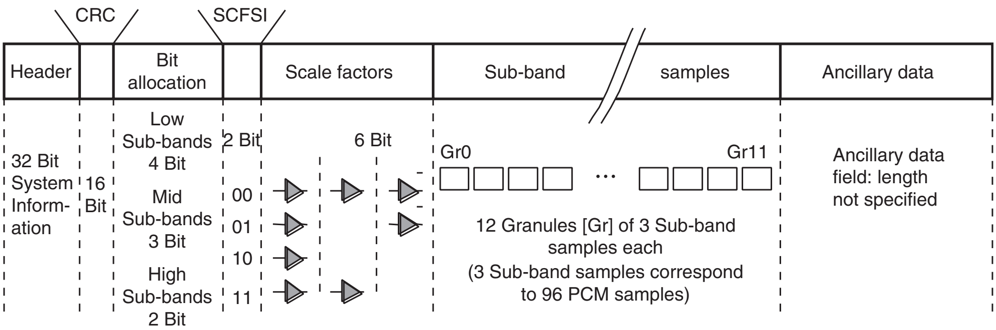

Figure 3.4 Audio frame structure of MPEG-1 Audio Layer II encoded bit stream

3.2.3 MPEG-2/4 HE AAC

The original DAB system based around MPEG-1Audio Layer II coding had limitations with regards bit-rate and quality, as would any audio coding scheme. In the years since the original Eureka 147 DAB system was designed, both audio coding algorithms have improved and processing power available to chip-set manufacturers has increased. The advanced broadcasting system DAB+ (see also section 3.4.2) aims to exploit this by implementing a newer audio coding scheme. The audio coder chosen was a particular version of High Efficiency Advanced Audio Coding version 2 (or HE AAC v2 for short). DRM (Digital Radio Mondale) already uses HE AAC v2 for its coding scheme, so it was a proven technology in a bandwidth-limited broadcast application.

The AAC (Advanced Audio Coding) algorithm comes in many varieties depending upon its particular application. A summary of some of these different versions is listed below:

- MPEG-2 AAC – the original standard coder;
- MPEG-4 AAC-LC – low complexity version;
- MPEG-4 HE AAC – high efficiency version of AAC-LC with SBR (Spectral Band Replication);
- MPEG-4 HE AAC v2 – as HE AAC but with PS (Parametric Stereo coding) included.

MPEG-4 HE AAC v2 is a superset of the AAC-LC core codec. This superset structure permits to use plain AAC for high bit rates, AAC and SBR (HE AAC) for medium bit rates or AAC, SBR and PS (HE AAC v2) for low bit rates.

DAB+ uses HE AAC v2 with the following key properties:

1. Spectral band replication (SBR).
2. Parametric stereo (PS).
3. 960-point transform.

These properties will be described later, but they are all aimed to provide a high audio quality at a lower bit-rate than was possible with MPEG-1 Audio Layer II.

3.2.3.1 Overall View

HE AAC v2 consists of three main blocks: the core AAC coder, the SBR block and the PS block. The core AAC coder performs the conventional transform audio coding, generating a set of parameters for each audio frame based on the psycho-acoustically weighted properties of the audio signal. However as the bit-rates used are relatively low, the bandwidth of the audio signal analysed is restricted to ensure artefact-free reproduction. To overcome this bandwidth restriction, the SBR part comes into play. SBR synthesizes the missing higher frequencies by analysing both the full-band original audio signal and the band-limited coded signal. The amount of data required to synthesize these high frequencies is much less than for the core AAC coder. Therefore SBR allows full bandwidth audio to be coded, but at a lower overall bit-rate. Further bit-rate saving can be made by exploiting redundancy in the stereo aspect of the audio. Instead of coding the two channels (left and right) explicitly, the audio can be coded as a mono signal, but with extra stereo information to recreate a close approximation to the original stereo image. The PS part provides this function, giving yet further savings of bit-rate.

3.2.3.2 AAC Core Encoder

To give an impression of the complexity of the AAC algorithm, the standards document ISO/IEC [13818-7] is about 150 pages long with numerous tables, diagrams and pseudo-code. Therefore to describe every single aspect of the AAC algorithm would not be possible. There are other publications available [Bosi, 1997] [Wolters, 2003], which are recommended as they provide a more detailed guide for this issue. The emphasis of the following sections is to explain the key points of the algorithm in a clear and simple way.

Figure 3.5 is giving an overall picture of the various building blocks for the AAC encoder.

3.2.3.3 The Filterbank

AAC is essentially a transform coder, where the time-domain audio signal is transformed into a different domain that is more conducive to data reduction. Whereas MPEG-1 Audio Layer II used a relatively coarse 32-band filterbank, AAC improves on this by using a MDCT (modified discrete cosine transform) which produces a time-frequency representation of the audio at a far finer frequency resolution.

The size of the MDCT is not fixed, but can be one of two sizes. In the standard AAC coder these two sizes are 2048 and 256, but for the DAB+ implementation with a shorter window size, this has been changed to 1920 and 240. The reason for having two different window sizes is to give the option of either good frequency resolution (large transform) or good time resolution (small transform). The choice of which transform is dependent upon the nature of the audio signal. Where the audio is in a reasonably steady-state (e.g. sustained musical notes) then the large transform is preferred. When the audio is changing drastically with transitions (e.g. hand-clap) then a short transform is more suitable.

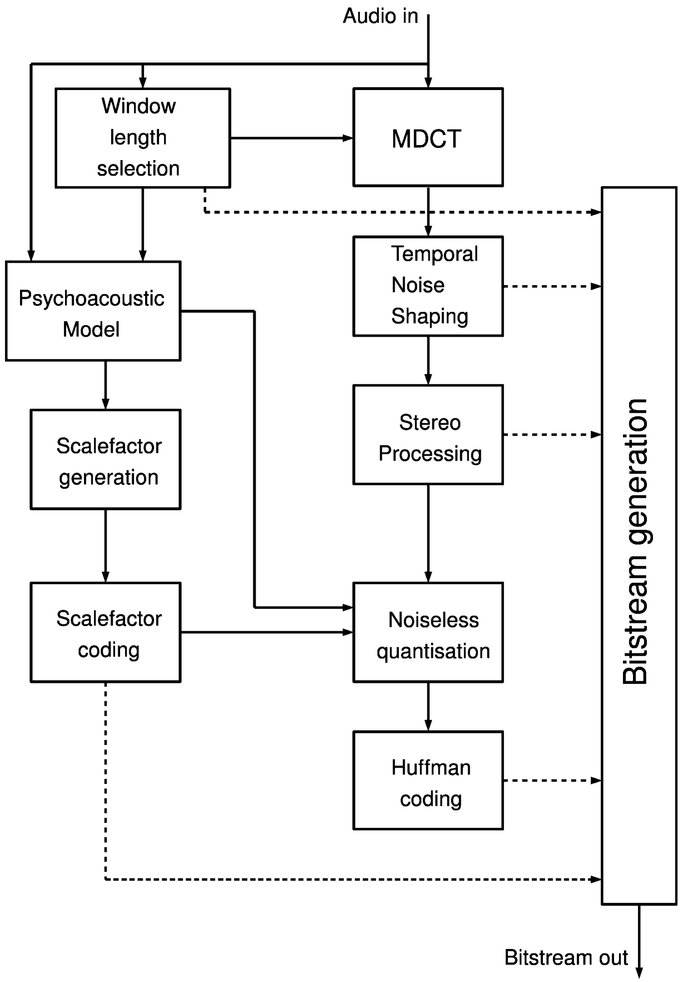

Figure 3.5 AAC core encoder block diagram

3.2.3.4 Windowing

Before the audio samples are transformed by the MDCT they must be windowed. Windowing can affect the frequency selectivity and rejection of the transformed signal. In the same sense that the nature of the audio can decide whether to use a short or long transform, it can also determine the best shape of window to use. Two shapes of window are used: the sine window and the Kaiser-Bessel derived (KBD) window. The KBD window is better at separating spectral components when they are more than 220 Hz apart and gives good rejection beyond this. The sine window gives more poor rejection, but is better separating spectral components closer than 140 Hz to each other.

With two shapes of windows and two sizes of MDCT it follows that four different windows are possible. However windows in successive frames (or in the case of the short windows, sub-frames) must overlap-add to produce a constant unity value. If a short sine window is followed by a long KDB window, then the overlap of the two windows would not result in unity. Therefore windows are broken into three chunks: start, middle and stop chunk. The start chunk of the current window must overlap and add to give unity with the stop chunk of the previous window. The middle chunk must just sit at unity. Therefore a long window may start off with a short sine window start chunk, then have the unity middle chunk and then have a long KBD stop chunk. Short windows have no middle chunks, nor do long windows that have long windows preceding and following them.

3.2.3.5 Temporal Noise Shaping

The temporal noise shaping (TNS) filter is an optional part of the encoder. If it is used, it sets a flag in the bit-stream so the decoder is ready to use it.

A typical artefact of audio coders is pre-echo. This is where an impulse or transition gets smeared out, often over a frame or sub-frame, so that energy is introduced before the actual sound starts. With spectral psychoacoustic masking, the magnitude spectrum of a frame containing a transient will effectively remove the information of the transient's occurrence. Therefore a masking curve when translated back into the time domain will allow quantisation noise that is audible before the transient occurs.

TNS filtering is a technique that gets around this problem, and significantly improves the audio coding quality. The MDCT coefficients are passed through a predictor, and the residual is output and then passed onto the subsequent blocks of the encoder (this is why the TNS must be flagged as the MDCT coefficients are no longer absolute values for quantisation). Open-loop or forward predictive coding applied to spectral values over frequency results in the quantisation noise being adapted to the temporal (i.e. time domain) shape of the audio signal. This effectively temporally masks the noise.

TNS filtering can be applied to parts of the spectrum, not just the whole spectrum, and the order of prediction filter can be varied up to a maximum of 20. The residual, filter order, frequency range and number of filters used are all encoded and transmitted. The decoder essentially performs the inverse of this process to generate absolute MDCT values from the residual given all the information about the TNS filtering.

3.2.3.6 Stereo Processing

Coding a stereo signal offers more opportunities for coding gain than just by treating it as two independent channels. Two methods of so called joint stereo coding are used in AAC which can be combined. The first method is mid/side (M/S) stereo coding (also known as sum-difference) and the second method is intensity stereo coding. For each frame of audio it is possible to apply either method to different frequency bands, or even just code bands as independent channels.

M/S stereo coding transforms the stereo signal (L, R) into a sum and difference pair of signals. This technique is useful in controlling the coding noise ensuring it is imaged in a manner that keeps it masked. While there may be redundancy in the correlation of the left and right channels, it is possible that stereo signals can require larger masking thresholds due to unmasking effects of phase shifted or uncorrelated sounds between the channels. M/S coding is designed to counter these issues.

Intensity stereo coding exploits, particularly at high frequencies, that spectral values can be shared across both channels, and just a magnitude weighting can be used. So it is like turning frequency bands into mono signals that are then panned towards the channel that is appropriate.

These stereo coding techniques can help to reduce the bit-rate of stereo material, and can also be used in multichannel coding where the channels are handled in their symmetrical pairs.

3.2.3.7 Scale Factors

It is very unlikely that all the MDCT coefficients will be values close to the maximum possible for normal audio signals. Usually they will be considerably lower, and often the higher frequency coefficients will be lowest of all. When small values are quantised they will suffer from more quantisation noise than larger values; therefore by boosting the smaller values, quantisation noise can be reduced.

The MDCT coefficients are broken down into either 49 frequency bands for the large transform or 14 bands for the small transform, which are based on the human ear's critical bands. At lower frequencies these bands are narrower and widen at higher frequencies. The diagrams given in Figures 3.6 and 3.7 show how these bands are arranged. The X scale represents the MDCT coefficients and the height of each scale

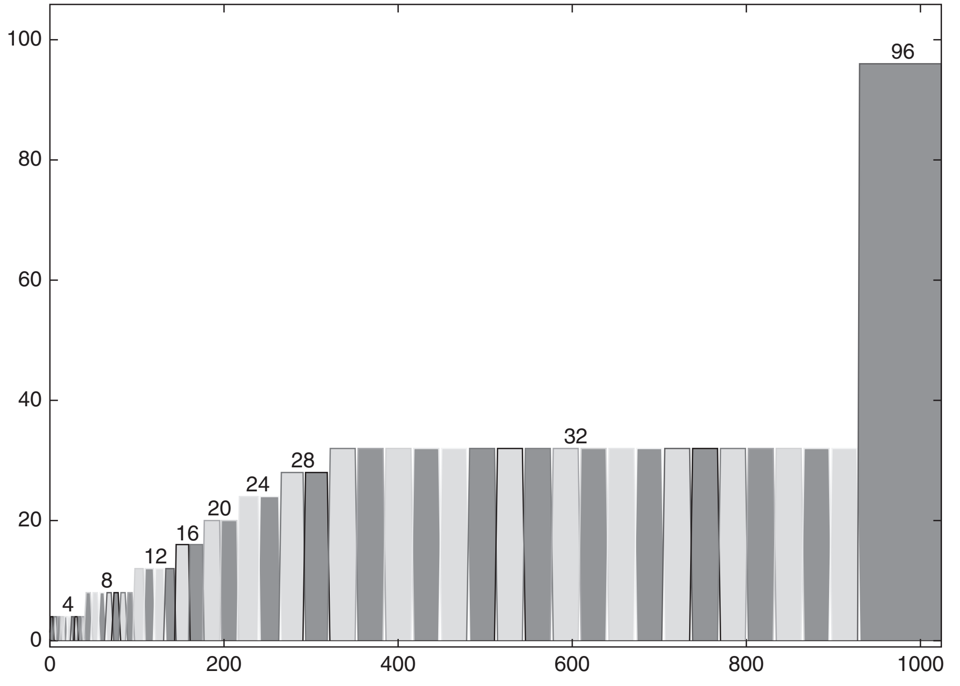

Figure 3.6 Scale factor bands for the large transform

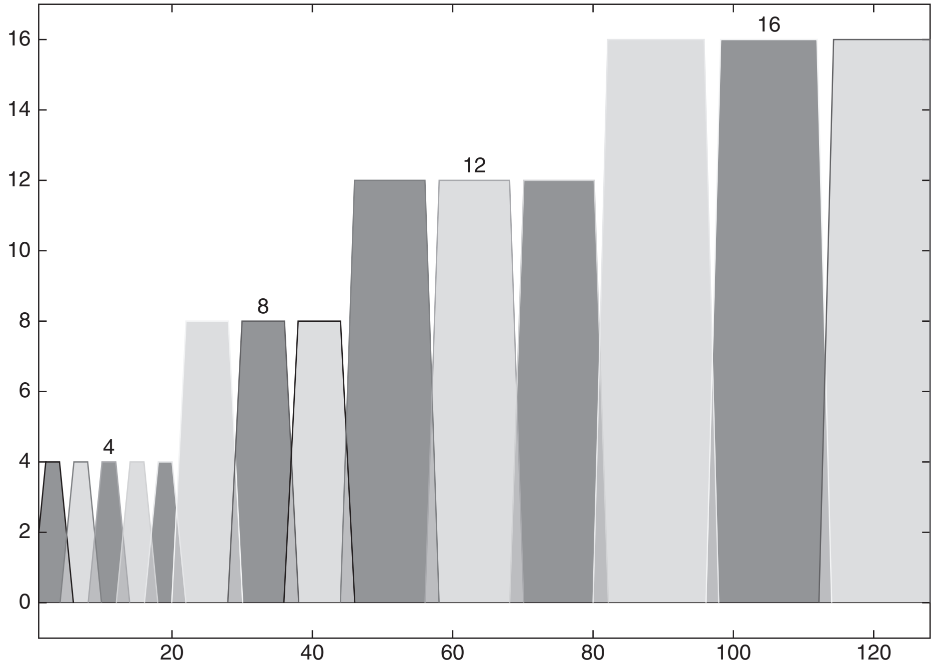

Figure 3.7 Scale factor bands for the small transform

factor band represents its width. For each of these 49 or 14 bands a scale factor value is calculated, and this amplifies all the coefficients in that particular band. A single value known as the global gain is also generated, which represents the maximum scale factor of the whole frame, and is used to offset the scale factor values, which are encoded as differential values from the previous scale factors.

The method for calculating the scale factors is very much down to the implementation and is not specified by any standards. Usually they are calculated by using information from the psychoacoustic model, the bit rate used and the likely bit requirement for the quantised coefficients and the scale factors themselves.

For short windows, where there are 8 groups of 128 coefficients, the values are grouped and interleaved to improve the coding efficiency of the scale factors and coefficients.

3.2.3.8 Quantisation and Huffman Coding

The quantisation stage is where the serious data reduction occurs. Here the spectral data is quantised in a manner that aims to minimize the perceptual coding artefacts. The quantiser is driven by the psychoacoustic model that provides information to which spectral bands require more bits than others. As with any audio coder, there is a limit to how many bits are available for encoding this spectral data (i.e. the bit-rate setting on the encoder). Therefore there is a likelihood, which increases as the bit-rate drops, that insufficient bits are available to satisfy the psychoacoustic model’s demands. Hence the quantising process must compromise in some way to do the best it can with the available bits and minimise the quantisation noise, or some other metric.

The quantiser used is nonuniform, which has the advantage of in-built noise shaping. As the human ear responds to levels in a nonlinear way, it follows that a quantiser should do so too. The relationship between MDCT coefficients and their quantised indices is:

$$xq = sign(x) * nint\left(\left[\frac{|x|}{(2^{s/4})}\right]^{\frac{3}{4}} - p\right)$$

where x is a MDCT coefficient,
xq is the quantised coefficient,
s is the quantiser step-size,
p is an offset value,
$sign$ is the sign of of a value (i.e -1 or 1) and $nint$ is the nearest integer value.

The decision of what step-size to use for quantisation is based on minimising the quantisation noise and the available bits. As there are up to 49 bands to deal with, often an iterative process is used to do this. Each stage of iteration will calculate the overall quantisation noise that will result for a given set of scale factors and step-sizes; then adjust them accordingly to minimise this. There are many different approaches to tackling this, and certain assumptions and starting points can be made from the psychoacoustic model and nature of the signal.

The quantised coefficients are then coded using Huffman coding. The coefficients are grouped into either two or four values, and depending upon the maximum absolute value of the coefficients one of 12 codebooks is used.

Huffman coding is an efficient method of data reduction for numerical data. The coding gain is achieved by the knowledge of the likelihood of each data value. For example, the original data is 6 bits in length and the value zero ('000000') occurs 40% of the time, then representing 000000 (6 bits) as 0 (a 1 bit word) would make a significant saving. A rarely occurring value (let's say 33, or '100001' for this example) might only occur 0.2% of the time, therefore it could be represented by a 10 bit word. While this value might be longer than its uncoded version, it is so unlikely to occur it does not significantly add to the overall bit-rate. Therefore Huffman coding can losslessly reduce the data rate for the quantised values. It also follows that the design of the Huffman codes must be tailored to the probability distribution of the quantiser values. As these distributions vary with different window sizes and quantiser step-sizes, then different Huffman codebooks are required; hence the need for 12 different codebooks in the encoder.

3.2.4 Spectral Band Replication (SBR)

Traditionally perceptual waveform codecs limit the bandwidth of the audio spectrum to maximise the quality of the perceptually more important lower frequencies. At lower bit-rates this cut-off frequency can result in dull sound audio, albeit one without significant coding artefacts. If these lost higher frequencies can be recreated in some way, then a brighter and more realistic sound may be possible.

Spectral band replication (SBR) for the ISO/IEC specification [TS 14496] (see also [Meltzer, 2002] and [Meltzer, 2006]) is a technique to synthesise the higher frequencies of an audio signal. For example, if the core AAC coder limits the bandwidth of a 16 kHz bandwidth audio signal to 0 to 9 kHz, then SBR can be used to generate the frequencies in the 9 to 16 kHz band. SBR only requires a relatively small number of bits to describe these higher frequencies as it utilises the high correlation of the signal in the lower frequencies with the higher frequencies.

In simple terms, the technique transposes the lower frequencies that would be coded by the core codec into the higher frequency area, and then spectrally shapes these transposed frequencies to match the original spectral envelope as shown in Figure 3.8. The information required to represent the spectral envelope, along with some other guiding parameters can be more compactly coded than if it was treated as a conventional waveform coding.

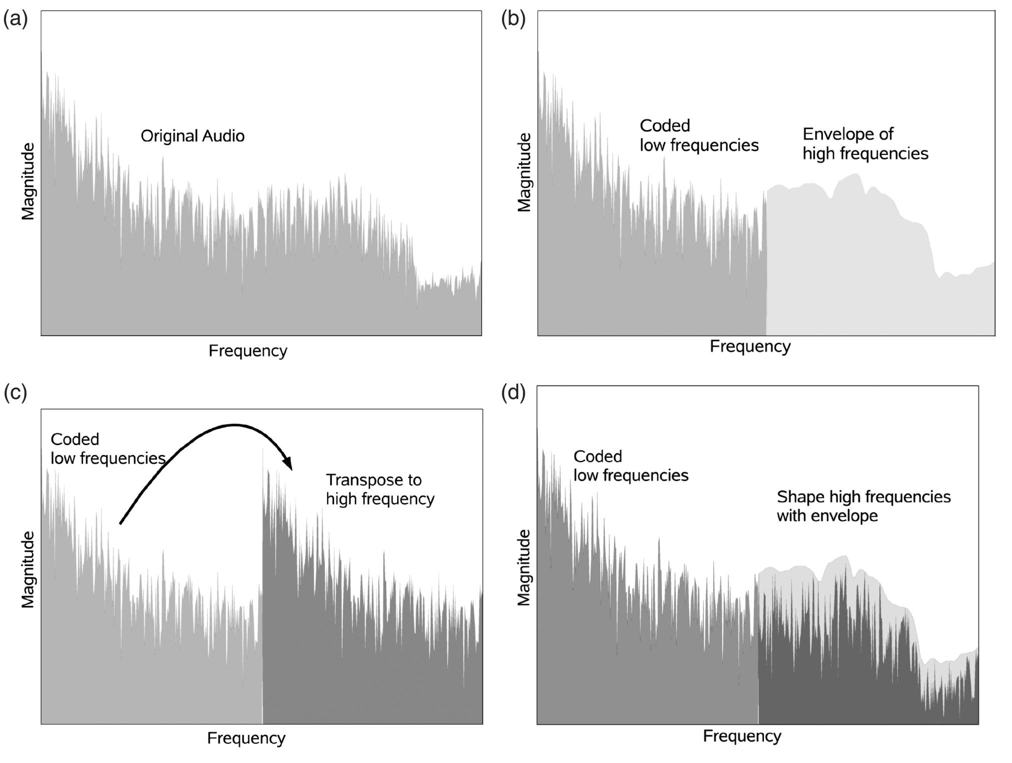

Figure 3.8 SBR audio spectra
(a) Original audio spectrum
(b) Low frequencies conventionally coded and the envelope of high frequencies
(c) Low frequencies transposed to the high frequency range
(d) High frequencies shaped with original envelope

There is more to the technique than simple transposition and envelope shaping, as many factors need to be dealt with. Such issues as transients, uncorrelated high and low frequencies, noise-like signals and isolated high frequency sounds all require extra analysis and guiding information. It is also important to maintain both good time and frequency resolution to ensure the SBR signal sounds realistic and seamless with the core audio signal.

The crossover frequency chosen for the SBR depends upon the bit-rate used. The lower the bit-rate the lower the crossover frequency required. For example, at 20 kbps (AAC, stereo) the crossover maybe set at 4.5 kHz, with the SBR dealing with up to 15.4 kHz. Whereas at 48 kbps, the crossover could be set at 8.3 kHz, with the SBR making the rest up to 16.9 kHz. If you were to use an AAC codec without SBR at 20 kbps, you may set the cut-off at 8 kHz. With SBR at 20 kbps, the SBR part may take as little as 1 kbps, then you are coding the 0 to 4.5 kHz range with 19 kbps with the AAC core; therefore the quality of the 0 to 4.5 kHz signal would be greater than the 0 to 8 kHz coded signal (i.e. more bits per kHz). The SBR part will then generate a higher bandwidth signal, producing a less dull sound than the nonSBR coded audio.

This is why SBR can produce higher quality audio at lower bit-rates than the core codec alone. However, at higher bit-rates the advantages of SBR wear off, and it is preferable to use the core codec to deal with the full audio bandwidth. This is because SBR cannot reproduce audio signals as accurately as the core codec if the core codec is given sufficient bits. For AAC with stereo audio, the bit-rate where SBR becomes unnecessary is around the 96 to 128 kbps mark. A good implementation of the HE AAC (i.e. AAC with SBR) algorithm should automatically deal with this crossover.

Another feature of HE AAC is that nonSBR versions of AAC decoders can still decode the audio, with the downside that the higher frequencies dealt with by SBR will be missing. Dull but clear sounding audio is better than no audio at all!

As SBR removes the need of the AAC core codec to deal with higher frequencies, it becomes possible for the core codec to operate at a lower sampling frequency. This can allow the core codec to reproduce better quality audio as improved frequency resolution is available to it. For example if the audio source is sampled at 48 kHz, then the core AAC encoder can operate at 24 kHz sampling while, SBR runs at the original 48 kHz. A simple 2:1 down sampler is used in the encoder, and a 2:1 up sampler in the decoder, along with a mixer to generate a decoded 48 kHz sampled signal.

The SBR encoder does not encode the audio on the raw time domain signal, but transforms it using a quadrature mirror filter (QMF) into 64 frequency bands before carrying out the SBR analysis as shown in Figure 3.9. It follows that in the decoder, the AAC decoder's output is fed to a 32-band QMF analysis filter to be combined with a SBR 32-band signal, before passing through a QMF synthesis filter as shown in Figure 3.10.

An AAC coder with SBR is also known as HE AAC v1.

3.2.5 Parametric Stereo Coding (PS)

Coding the left and right channels of a stereo signal separately produces the best imaging, but is only feasible at high bit-rates if the audio quality is not to be compromised. Many

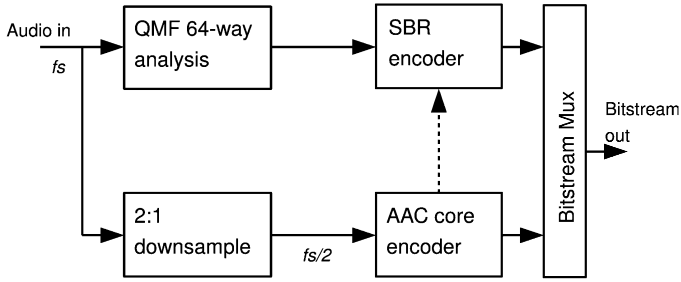

Figure 3.9 AAC with SBR encoder block diagram

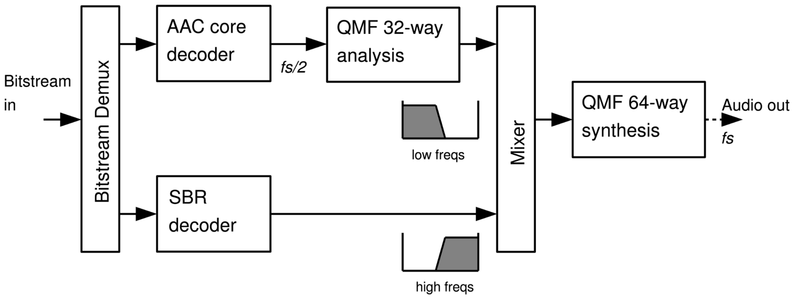

Figure 3.10 AAC with SBR decoder block diagram

audio codecs look at both channels to code them together with various techniques, which rely on the similarity between the channels to reduce the bit-rate requirement. Both MPEG 1/2-Audio Layer II and AAC coding has a stereo mode known as intensity stereo, which is an efficient method of coding stereo signals to maximise the quality of the core signal at the expense of the stereo image, without dropping down to a mono mix.

As most stereo audio signals have a high correlation between the left and right channels, there is a large amount of redundancy to exploit. By mixing left and right channels into a mono signal and transmitting a helper signal to allow this mono signal to be spread back out into stereo at the decoder, we have a very efficient method of encoding a stereo signal. Parametric stereo (PS) [Meltzer, 2006] is a technique that generates a helper signal from the original stereo audio, and transmits that alongside the mixed-down mono audio. The bit-rate requirement of the PS data is very small compared to that of the audio data itself, giving significant bit-rate reductions over separately coded left and right channels.

The parameters used by PS that can be derived from a stereo signal are:

- Inter-channel Intensity Difference (IID) – this is the intensity or magnitude difference between the channels. A great difference in intensity represents a hard panning of the signal towards one channel.

- Inter-channel Phase Difference (IPD) – this is the phase difference between the channels. With many recordings sounds reach one microphone before another (much like our ears), therefore a phase or time difference will occur. Recreating this difference maintains the localisation of the sound.
- Inter-channel Cross Correlation (ICC) – this is the correlation between the channels. A high correlation means the channels are similar, therefore IID or IPD are likely to work well. If this value is low then the channels are likely to be dissimilar and IID and IPD will be less effective.

As with SBR, PS is a helper to a core coder. Therefore if a decoder cannot decode PS data, the decoded audio will simply be mono. Also, the PS algorithm (which also includes a stereo to mono downmixer) uses the same QMF domain signal as SBR, so can be placed before the SBR encoder. In the decoder the PS decoder generates a stereo signal in the QMF domain that can be turned back into an audio signal with a 64-band QMF synthesis filterbank.

The diagrams in Figures 3.11 and 3.12 show how an encoder and decoder with both SBR and PS are arranged. An AAC coder with SBR and PS is also known as HE AAC v2.

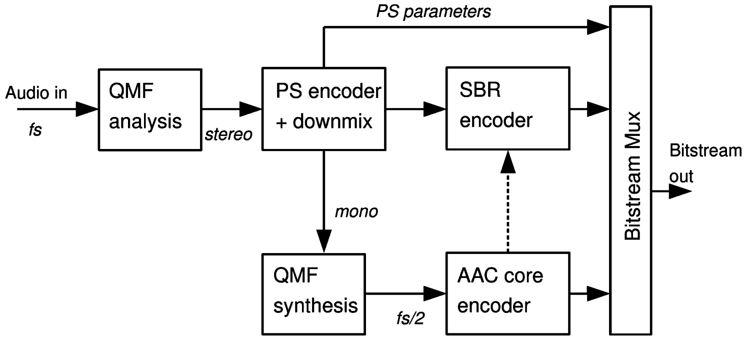

Figure 3.11 HE AAC v2 encoder block diagram

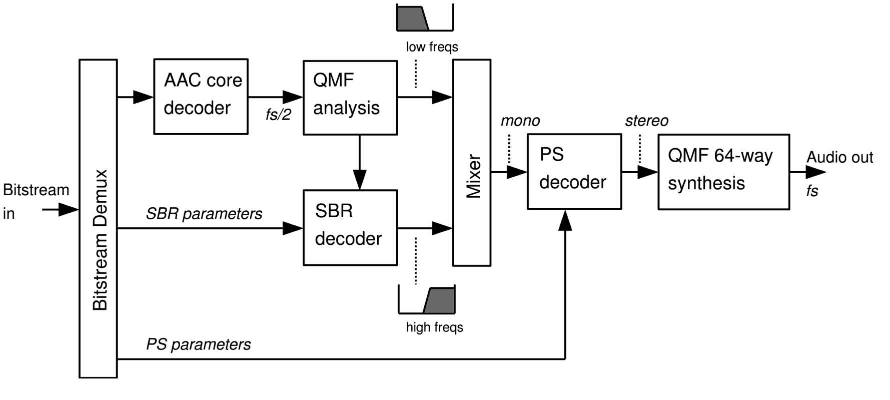

Figure 3.12 HE AAC v2 decoder block diagram

3.2.6 Audio Decoding

3.2.6.1 Audio Layer II Decoding

The block diagram of the decoder is shown in Figure 3.13. First of all, the header information, CRC check, the side information (i.e. the bit allocation information with scale factors), and 12 successive samples of each sub-band signal are extracted from the ISO/MPEG Audio Layer II bit stream.

The reconstruction process to obtain PCM audio is again characterised by filling up the data format of the sub-band samples regarding the scale factor and bit allocation for each sub-band and frame. The synthesis filterbank reconstructs the complete broadband audio signal with a bandwidth of up to 24 kHz. The decoding process needs significantly less computational power than the encoding process. The relation for Layer II is about 1/3. Because of the low computational power needed and the straightforward structure of the algorithm, Audio Layer II can be easily implemented in special VLSI chips.

3.2.6.2 HE AAC v2 Decoding

The decoding process for HE AAC v2 consists of the AAC core decoder, SBR decoder and PS decoder. The block diagram for the three components was shown in Figure 3.12. The SBR and PS decoders are described in sections 3.2.4 and 3.2.5 respectively. The AAC core decoder is essentially the reverse of the encoder.

Figure 3.14 shows the block diagram of the AAC core decoder. The incoming bitstream is converted into the data for each of the parameters, so the quantised coefficients, scale factors and block control data are all recovered. The scale factors and quantised coefficients are recombined to produce coefficients ready for the IMDCT. The IMDCT then converts the frequency domain signal into the time domain signal, which is overlap-added according to the windowing sizes chosen at that instant.

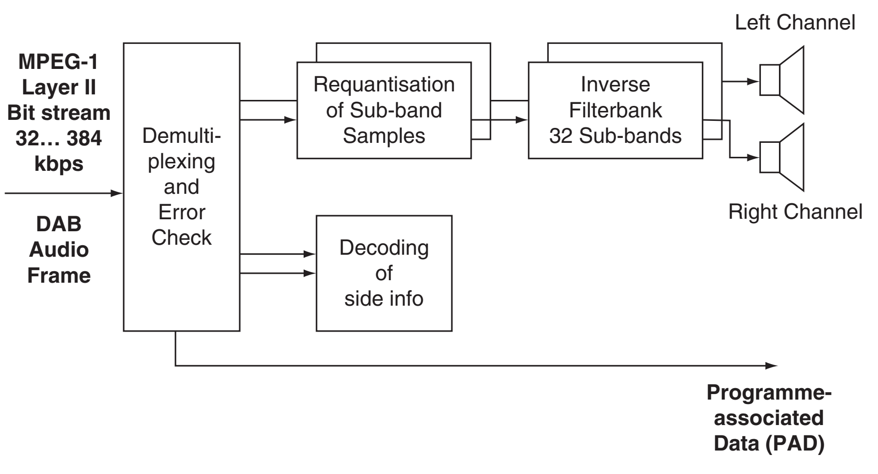

Figure 3.13 Block diagram of the MPEG-1 Audio Layer II decode

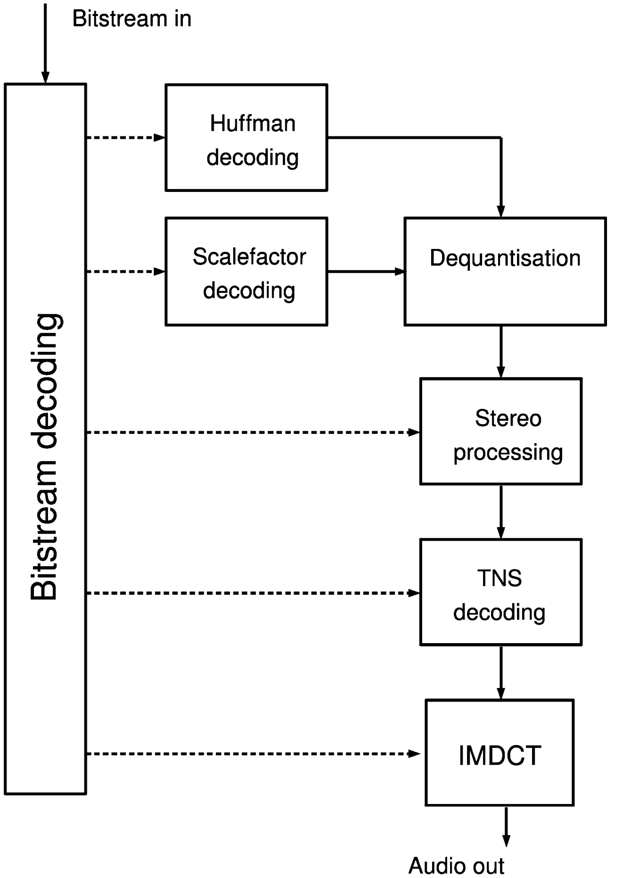

Figure 3.14 AAC core decoder block diagram

As with most audio codecs the decoding process is far less complex than the encoding process. It is also standardised completely, so all implementations should produce identical outputs, unlike the encoder.

3.3 Characteristics of DAB Audio Coding

The EU147 DAB source coding system permits a digital audio broadcast receiver to use a standard MPEG-1 and MPEG-2 Audio Layer II decoder. Besides the emission of the digital audio broadcast signal, the MPEG Audio Layer II coding technique and its encoded audio bit-stream can be used in a number of other applications, including contribution between broadcast studios, primary and secondary distribution, and news/sports reporting links. These different applications require a flexible coding scheme offering a wide range of parameters, in particular concerning the bit rates, audio modes (i.e. mono, stereo and multichannel representation), protection level of the coded bit-stream and the possibility to carry Programme-associated Data (PAD), which enable completely new applications.

More detailed information can be found in [Stoll, 1995] or [AES, 1996].

3.3.1 Audio Modes

The audio coding system, used in Eureka 147 DAB, supports the following modes:

- Mono (one-channel) mode.
- Stereo (two-channel) mode.
- Dual-channel mode. In this mode, the two audio channels can be either bilingual, or two mono channels, but with only one header. At the decoder a choice is made on which of the two programmes should be decoded.
- Joint stereo mode. In the Joint Stereo mode, the encoder exploits the redundancy and irrelevancy of stereo signals for further data reduction. The method used for Joint Stereo in Audio Layer II is 'intensity stereo coding'. This technique still preserves the spectral envelope of the left and right channel of a stereo signal, but transmits only the sum signal of the sub-band samples of the left and right channel in the high audio frequency region [Chambers, 1992].
- Low sampling frequency coding with f_S 1/4 24 kHz.
- Provisions are made in the Eureka 147 DAB standard [EN 300401] for the inclusion of MPEG-2 Audio Layer II backwards-compatible Surround Sound coding.

3.3.2 Sampling Rate

The audio coding algorithm of DAB allows two sampling rates: 48 kHz and 24 kHz. The higher sampling rate can be chosen to have full audio bandwidth of 20 kHz for the transmitted signal and to allow for a direct broadcasting of studio signals without the need for sampling-rate conversion. The audio quality of a PCM signal improves with increasing resolution of the input signal. Thus, the MPEG Audio Layer II standard can handle a resolution of the input signal up to 22 bits/sample.

The lower bit rate can be chosen to deliver a high quality, in particular for speech signals at very low bit rates, that is at bit rate of 64 kbps per channel. However, this does not mean that a new sampling frequency will be introduced outside of the DAB system. Instead a down-sampling filter from 48 kHz to 24 kHz in the audio encoder and an up-sampling filter from 24 kHz to 48 kHz in the decoder are used.

With DAB+, the sampling rate options are greater, with 48, 32, 24 and 16 kHz all possible with the HE AAC v2 coding algorithm. The AAC core encoder runs at either 48/32 kHz (no SBR used) or at 24/16 kHz (with SBR). The 24 kHz and 16 kHz sampling rates are the half-sample rates for the AAC core, but unlike the 24 kHz sampling rate with Layer II that is band-limited to 12 kHz, the SBR part of HE AAC provides the band-limit of the full sample rate at these half sampling rates.

The 32 kHz and 16 kHz sampling rates may be better choices at very low bit-rates where the programme material is not demanding for encoding, such as speech. More half sampling rate features are described in section 3.7.2.

Note: In contrast to other broadcasting systems (e.g. ADR) a sampling frequency of 44.1 kHz is not applicable, which means a source taken from a CD has to be sample-rate converted.

3.3.3 Frame Structure

The DAB audio frame is based on the MPEG Audio Layer II frame which includes all the main and side information which is necessary for the decoder to produce an audio signal at its output again. Additionally, each DAB audio frame contains a number of bytes that may be used to carry the PAD, that is the information which is intimately related to the audio programme. This structure is depicted in Figure 3.15. The PAD field contains 2 bytes of Fixed PAD (F-PAD), and an optional extension called the extended PAD (X-PAD). Functions available for the PAD include Dynamic Range Control (DRC), Music/speech indication (M/S), programme-related text and additional error protection. The PAD features are explained in section 3.5.

3.3.4 Bit Rates

Depending on the type of programme, the number of programmes within the DAB multiplex and the protection level, different bit rates for Audio Layer II lie in the range of 32 to 192 kbps in single channel mode can be selected at the audio encoder. The choice of the bit rate is not fully independent of the audio mode. Depending on the bit rate, the audio modes given in Table 3.3 can be used. Any combination of these options may be used within the DAB multiplex [Chambers, 1992]. For the half-sampling-rate coding of 24 kHz, 14 different bit rates in the range of 8 to 160 kbps can be chosen, irrespective of the selected audio mode. For the multichannel extension, Table 3.3 is valid for the base

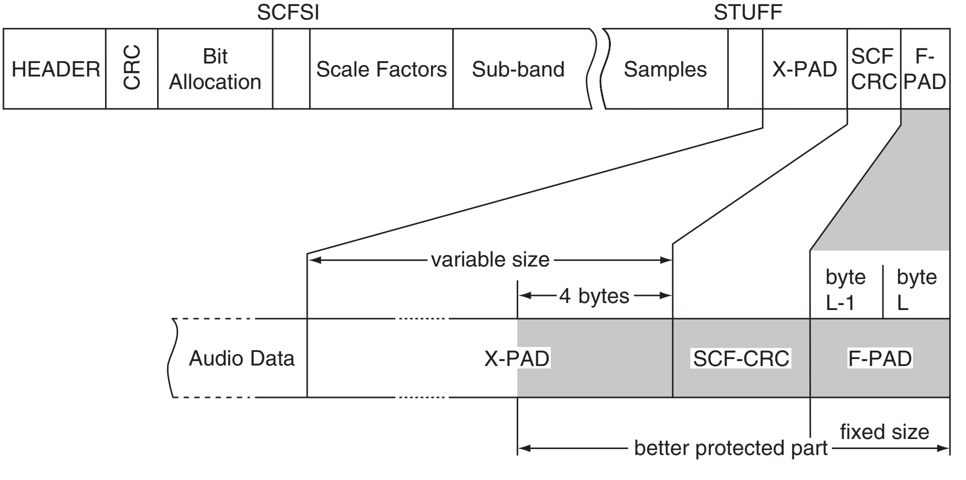

Figure 3.15 Structure of the DAB audio frame including Fixed and Extended Programme-associated Data (F-PAD and X-PAD) and Scale Factor CRC (SCF-CRC)

bit-stream only. The details of the lower sample rates and the multichannel extension are explained below.

The wide range of bit rates allows for applications that require a low bit rate and high audio quality; for example, if only one coding process has to be considered and cascading can be avoided. It also allows for applications where higher data rates up to about 180 kbps per channel could be desirable if either cascading or post-processing has to be taken into account. Experiments carried out by the ITU-R [AES, 1996] have shown that a coding process can be repeated nine times with MPEG-1 Layer II without any serious subjective degradation, if the bit rate is high enough, that is 180 kbps per channel. If the bit rate is only 120 kbps, however, no more than three coding processes should occur.

The bit rates available for both DAB and DAB+ MPEG audio Layer II coding are very extensive and beyond what is sensible for use as tolerable quality and economy. A sub-channel will have a size assigned to it, but not only does it have to accommodate the bits for the audio coding, but it must also carry the PAD bits. Therefore the bit rate for audio in the sub-channel will be less than the indicated value. For example, at 80 kbps the actual bit rate available for audio may only be around 72 kbps (the exact figure depends upon PAD size and sampling rate). Therefore the quality of the audio is marginally lower than would be expected for the indicated bit-rate. This can become significant at bit-rates that are at the bottom end of the quality range of the audio coding algorithm, where the quality begins to fall away sharply.

3.4 DAB+ Coding Schemes

3.4.1 Is Audio Layer II Still Up-to-date for DAB?

The MPEG-1 Audio Layer II coding system was developed and intensively tested in the early 1990s, and standardised for use with the DAB system in 1993. Even the last revision of the DAB standard [EN 300401] is based on MPEG-2 Layer II, which now includes lower sampling rates.

In the time after, several different audio coding systems were exploited, especially for use with very low bit-rates (between 8 kbps and 64 kbps). They are in particular aimed at Internet-streaming applications and related services. For example, this is the worldwide MP3 coding scheme (based on MPEG-1 Audio Layer III), and the MPEG-2/4 AAC (Advanced Audio Coding) system [IS 13818-7]. In principle, at lower bit-rates the AAC codec can provide audio quality better than MPEG-1 Layer II, but it is not compatible to Layer II at all. Results from different tests [BPN 029] indicate, that the quality of Layer II at 192 kbps can be reached with AAC already at 128 kbps, sometimes also at 96 kbps. That would mean a significantly higher spectrum efficiency if used for DAB.

Therefore, the question was raised, if Layer II is still the right choice for use with the DAB system. Investigations were made (for instance, see [Zieglmeier, 2000]) to compare those coding systems with critical programme material, considering the requirement of cascading of at least two codecs (which is typical for a broadcast chain).

Some of the last results of audio coding research were the so called Spectral Band Replication (SBR) technology [Meltzer, 2002], and the so called Parametric Stereo (PS) coding, see sections 3.2.4. and 3.2.5 respectively. That techniques can enhance the efficiency of existing perceptual audio codecs.

MPEG-4 AAC completed with SBR is called HE AAC. Apart from the new components of the DAB system family as DAB+ (see below) and DMB (see Chapter 9), HE AAC is also used for audio coding with the digital broadcasting system at frequencies below 30 MHz, this is DRM (Digital Radio Mondiale, see section 1.6).

3.4.2 DAB+ Audio Coding

Considering the mentioned constraints an update to the original DAB system was standardised by WorldDMB, the so called DAB+ system.

DAB+ is based on the original DAB standard but uses a more efficient audio codec. Whereas DAB uses MPEG-1 Audio Layer II, DAB+ uses HE-AAC v2 (also known as AAC+). This allows equivalent or better subjective audio quality to be broadcast at lower bit rates. The increased efficiency offers benefits for Governments and Regulators (even better spectrum efficiency), broadcasters (lower costs per station) and consumers (a wider choice of stations).

DAB+ is designed to provide the same functionality as the original DAB radio services including the following (e.g. to the same service on another DAB ensemble or its FM simulcast), traffic announcements and PAD multimedia data (e.g. dynamic labels such as title artist information or news headlines; complementary graphics and images etc.).

More benefits of DAB+ compared with the basic DAB system are listed in section 1.3.2.

The HE-AAC v2 audio codec is at the time the most efficient audio compression scheme available and allows for up to three times as many services per multiplex as the original DAB: A 40 kbps subchannel with HE-AAC v2 provides a similar audio quality (even slightly better in most cases) as MPEG Audio Layer II at 128 kbps.

3.4.2.1 Coding Technology of DAB+

HE-AAC v2 combines three technologies:

- The core AAC audio codec.
- A bandwidth extension tool SBR (Spectral Band Replication), which enhances efficiency by using most of the available bit rate for the lower frequencies (low band) of the audio signal. The decoder generates the higher frequencies (high band) by analysing the low band and side information provided by the encoder. This side information needs considerably less bit rate than would be required to encode the high band with the core audio codec.
- Parametric stereo (PS): a mono down-mix and side information is encoded as opposed to a conventional stereo signal. The decoder reconstructs the stereo signal from the mono signal using the side information. HE-AAC v2 is a superset of the AAC core codec. This superset structure permits plain AAC to be used for high bit rates, AAC and SBR (HE-AAC) for medium bit rates or AAC, SBR and PS (HE-AAC v2) for low bit rates. Therefore HE-AAC v2 provides the highest level of flexibility for the broadcaster. A detailed description of HE-AAC v2 is available in section 3.2.3.

HE-AAC v2 provides the same perceived audio quality at about one third of the subchannel bit rate needed by MPEG Audio Layer II. The same audio coding is also used in DRM (see section 1.6.) and DMB (see Chapter 9) e.g. for television audio. Devices, which also include DMB or DRM capabilities can benefit from the fact that the audio coding for this range of technologies is essentially the same.

3.4.2.2 Bit Rates with DAB+

Whereas DAB with Audio Layer II offered two sampling rates, 24 and 48 kHz; with DAB+ and HE AAC v2, there are now four possible sampling rates: 16, 24, 32 and 48 kHz. The half sample rates for DAB+ are therefore 16 and 24 kHz.

The provided sampling rates are used by the AAC core coder. If SBR is not used, then the AAC core uses 32 or 48 kHz. If SBR is used, then the AAC core runs at half these sampling rates (i.e., at 16/24 kHz), and the SBR algorithm deals with the full sampled version of the audio. For example a 48 kHz sampled stream of audio can be encoded by DAB+ using SBR at 40 kbps. In this case the actual sampling rate used by the AAC core will be 24 kHz (half of 48 kHz as SBR is used). So the core codec will code 0 to 12 kHz of the audio and SBR will deal with the 12 to 24 kHz region. The DAB+ receiver will use the 24 kHz sampling rate for the AAC core, but as it has an SBR decoder, the decoded audio will be sampled at 48 kHz with full audio bandwidth.

At higher bit-rates where SBR is not required, the full sampling rates (32 and 48 kHz) would be used. The 32 kHz sample rate (and 16 kHz half sample rate) allows more flexibility with using lower bandwidth audio where quality can be optimised at lower bit-rates, or where the audio material is more suited to it (speech programmes would benefit at these sampling rates). It is worth remembering the audio in FM radio rolls off at 15 kHz, which would be within a 32 kHz sampling rate.

The bit rates for DAB+ are shown in Table 3.1, where the actual bit rates available at each of the possible sampling rates are listed [TS 102563].

Unlike Audio Layer II in DAB, DAB+ with HE AAC v2 allows the full range of sub-channel bit-rates from 8 to 192 kbps at all the sampling rates. Naturally, some of these combinations would be completely irrelevant in terms of quality and coding efficiency.

It should also be noted, that there would be a significant difference in the processing time delay between Audio Layer II (which needs approximately 70 ms) and AAC (takes about 300 ms).

3.4.2.3 DAB+ Features

All the functionalities available for DAB with MPEG Audio Layer II services are also valid for DAB+:

- service following (e.g. to FM or other DAB ensembles);
- traffic announcements;
- PAD multimedia (dynamic labels such as title artist information or news headlines;
- PAD dynamic control issues (DRC, M/S control, see below);
- still images (such as weather charts and other multimedia content);
- service language and programme type information (e.g. Classical music, Rock music, Sports) etc.

Table 3.1 Bit-rates for DAB+

Sub-channel index, s	Sub-channel size [kbps]	Bit-rate available for audio [kbps]			
		AAC core sampling rate			
		16 kHz	24 kHz	32 kHz	48 kHz
1	8	6,733	6,533	6,267	5,800
2	16	14,067	13,867	13,600	13,133
3	24	21,400	21,200	20,933	20,467
4	32	28,733	28,533	28,267	27,800
5	40	36,067	35,867	35,600	35,133
6	48	43,400	43,200	42,933	42,467
7	56	50,733	50,533	50,267	49,800
8	64	58,067	57,867	57,600	57,133
9	72	65,400	65,200	64,933	64,467
10	80	72,733	72,533	72,267	71,800
11	88	80,067	79,867	79,600	79,133
12	96	87,400	87,200	86,933	86,467
13	104	94,733	94,533	94,267	93,800
14	112	102,067	101,867	101,600	101,133
15	120	109,400	109,200	108,933	108,467
16	128	116,733	116,533	116,267	115,800
17	136	124,067	123,867	123,600	123,133
18	144	131,400	131,200	130,933	130,467
19	152	138,733	138,533	138,267	137,800
20	160	146,067	145,867	145,600	145,133
21	168	153,400	153,200	152,933	152,467
22	176	160,733	160,533	160,267	159,800
23	184	168,067	167,867	167,600	167,133
24	192	175,400	175,200	174,933	174,467

MPEG Audio Layer II and HE-AAC v2 radio services can coexist in the same ensemble. However, legacy receivers might list HE-AAC v2 radio services even though they will not be able to decode them.

Short 'Zapping' Delay

An important design criterion for DAB+ was to provide a short 'zapping' delay. Both the time it takes to switch from one radio service to another station on the same DAB ensemble as well as the time it takes to tune to a radio service on another DAB ensemble was minimized.

Surround Sound

Currently all DAB radio services are mono or stereo. However, DAB+ also provides the means to broadcast surround sound in a backwards compatible way. Using MPEG Surround (see section 3.6.4) it is possible to broadcast a stereo signal together with surround side information (e.g. 5 kbps for side information). Standard stereo radios will ignore this side information and only decode the stereo signal. MPEG Surround receivers will evaluate the side information and reproduce surround sound (e.g. in 5.1 or other multichannel formats). So at a comparatively low additional bit rate, the broadcaster can increase the audio experience on surround sound receivers, and still provide high quality sound to all other radios.

Performance of DAB+ in Field Tests

During the standardisation process, field tests were conducted in the UK and in Australia. They gave a number of interesting results.

They showed that the geographical coverage area of radio services using HE-AAC v2 is slightly larger than that for radio services using MPEG Audio Layer II. The multimedia information carried in PAD of an HE-AAC v2 radio service is much better protected against transmission errors than PAD data of a radio service using MPEG Audio Layer II.

The error behaviour of MPEG Audio Layer II is different to that of HE-AAC v2. Audio services using HE-AAC v2 performed about 2 to 3 dB better at the threshold of audibility. With MPEG Audio Layer II, the weaker the DAB signal gets, the more audible artefacts can be heard. HE-AAC v2 produces no audible artefacts, but when the signal gets too weak, an increased number of audio frames will be lost and this causes short periods of silence (fade-out and fade-in). Compared to radio services using MPEG Audio Layer II, radio services using HE-AAC v2 will fail later (they can cope with a slightly lower DAB signal quality), but the margin from error free reception to loss of reception is smaller.

3.5 Programme-associated Data

The DAB system not only provides a very high audio quality for the listener, but also includes provision for many additional data services [BPN 007], [Riley, 1994]. Most of these data services are independent of the audio programme, but some of them are closely related to the audio signal. The latter form is the so-called Programme-associated Data (PAD). A fixed and optionally a flexible number of bytes are provided to carry the PAD in the DAB audio bit-stream, see Figure 3.15 in section 3.3.3. Additional capacity is provided elsewhere in the DAB multiplex (or ‘ensemble multiplex’) to carry more independent data or additional information, such as text, still or moving pictures, see Chapters 4 and 5.

Examples of Fixed Programme-associated Data (F-PAD) include:

- Dynamic Range Control (DRC) information, which may be used in the receiver to compress the dynamic range of the audio.
- Speech/music indication to allow for different audio processing of music and speech (e.g. independent volume control).

As these features are directly related to the audio programme signal, they are described in detail in the following. For more details on other programme-associated information carried within the F-PAD or X-PAD (Extended Programme-associated Data), see Chapters 2, 4 and 5.

In reproduction of audio broadcast programmes there will often be the need to apply dynamic range compression to some types of programme material because of listeners' requirements. Furthermore the satisfactory balancing of the loudness of different types of programme, particularly music and speech, depends principally upon the listeners' requirements. In conventional broadcasting it has never been possible to satisfy different individual habits of listening.

3.5.1 Dynamic Range Control (DRC)

The DAB DRC feature enables broadcasters to transmit programmes with a relatively wide dynamic range, accompanied by a DRC signal which the listener may use to effect unobtrusive compression of the programme dynamics, if required [Hoeg, 1994].

The dynamic range of an audio programme signal (sometimes termed the programme dynamic) is the range between the highest and the lowest useful programme signal level. The problems associated with programmes having a wide dynamic range, and with achieving a satisfactory loudness balance between different parts of the radio programme (such as speech or music), are well known from experience with VHF/FM broadcasting [Müller, 1970]. In many cases the source programme dynamic or the dynamic range commonly used by the broadcasters (approx. 40 dB) may be much larger than the usable dynamic range (the so-called reproduction dynamic) in noisy environments such as in a moving car. The reduction required in the dynamic range of the programme may be 10 to 30 dB (or more).

Taking into account the listening conditions and the requirements of the listener, these problems can only be solved at the receiver. For this to be feasible, additional information must be provided by the broadcaster concerning the gain adjustments, which may be needed to reduce the dynamic range. Therefore, at the broadcaster's premises a DRC signal is generated at the studio side, which describes the audio gain to be applied in the receiver, as a succession of values. This DRC signal is transmitted in a coded form (DRC data = DRCD) together with the audio.

It is a requirement of the DAB specification [EN 300401] that the audio signal is transmitted with its original programme dynamic, without any pre-compression. The DRC data is incorporated in the DAB bit-stream as PAD. In the receiver, the regenerated DRC signal may be used optionally to control the audio gain in order to match the dynamic range of the received audio programme to the requirements of the listener, or to improve audibility in difficult conditions.

The user can choose any appropriate compression ratio between the anchors:

- *No compression*: the audio programme is reproduced with the dynamic range as delivered by the broadcaster;
- *Nominal compression*: the audio programme is reproduced with an appropriate degree of dynamic compression as adjusted by the broadcaster (normally the dynamic range is compressed by about 1.3:1);

- *Maximum compression*: the audio programme can be reproduced with extremely strong compression of the dynamic range (e.g. > 2.0:1). This makes sense for poor reproduction conditions (e.g. in a car) and for special programme types only, because the sound quality is likely to be affected.

3.5.2 Music/Speech Control (M/S)

It is well known from any kind of radio programme transmission that there may be very different requirements in the loudness balance between music and speech programmes, depending on the interests of the listener [Ilmonen, 1971]. A Music/Speech (M/S) control signal, which is also transmitted in the DAB bit-stream, now enables the listener to balance the loudness of different types of programme according to taste.

There is the option to signal four states of the so-called M/S flag within the PAD:

00 programme content = Music

01 programme content = Speech

10 programme content = not signaled

11 reserved for future use (e.g. for a programme containing music and speech with artistically well-balanced levels, for instance a drama or musical).

This M/S flag information can be generated at the studio side automatically (e.g. derived from the corresponding channel fader at the mixing console). As an item of programme-related data, the information is transported within the PAD, and may be used at the receiver to control the output signal level (volume) in a predetermined manner, probably in a level range of about 0 to –15 dB, to enhance either the music, or the speech.

3.5.3 Transport Mechanism of DRC and M/S Data

Data traveling in the PAD channel is intimately related to the audio programme. It is assembled together with the coded audio in the same DAB frame, so the time relation established at the source is maintained throughout the complete DAB transmission chain. Audio and PAD must not be subject to different transmission delays. Figure 3.16 shows a simplified block diagram of the DRC system for DAB, which is also valid in principle for transmission of the M/S control flag.

The DRC signal is generated by a characteristic which is linear over most of the dynamic range, with an appropriate compression ratio. An overall range of gain variation of about 15 dB is linearly coded in a 6-bit digital word, for each 24 ms frame of the broadcast audio signal. The intention is to provide long-term compression of the dynamic range whilst preserving the impact of dramatic changes in the programme dynamics (e.g. that caused by a *crescendo* or *subito fortissimo*). Therefore the generating algorithms use specific sets of audio programme controlled time characteristics, which are not the subject of any specification.

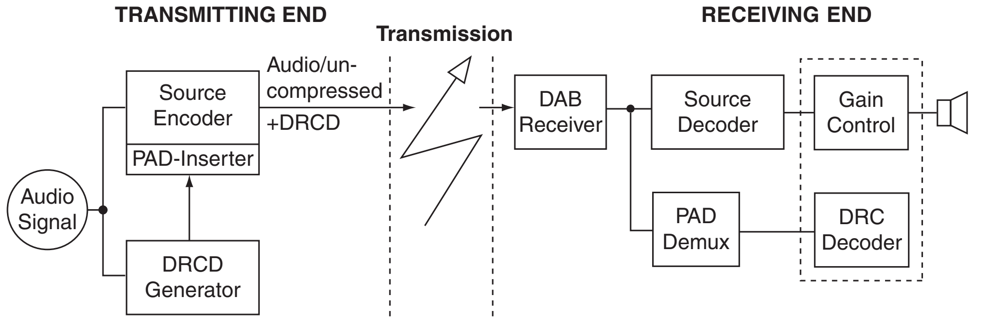

Figure 3.16 Block diagram of a DRC system for DAB

The DAB audio programme signal itself is transmitted without any dynamic compression applied after the production/post-production stage. During the production of some types of programmes, compression may be applied for 'artistic' reasons. However, programme signals compressed at this stage are treated in the same way as uncompressed signals by the DRC system. Because the DRC data is transmitted discontinuously in bursts at time intervals of one DAB frame (24 ms) it is necessary to have a look-ahead time of one complete DAB frame in which to analyse the level of the signal, to avoid overshoots (i.e. brief periods in which the reproduced signal is excessively loud). This means that a delay of at least 24 ms must be introduced into the signal path at the DRCD generator so that the incoming audio signal may be examined prior to generation of the DRC data. The DRC data for each DAB audio frame is conveyed in the L-byte of the PAD of the preceding DAB audio frame of the broadcast signal (see [EN 300401]). Thus the DRC data arrives at the receiver immediately before the audio with which it is associated, avoiding the need for further delays to be introduced by the receiver.

Some dynamic range control algorithms may require longer delays, in excess of 24 ms and up to several seconds. These would be associated with the DRCD generator at the transmitting end, and would not require any different implementation or treatment in the receiver.

DRC is also defined for an AAC coding bit-stream and can so be used with DAB+. In contrast to MPEG Audio Layer II, here the Dynamic Range Control (DRC) data shall not be carried in F-PAD. If Dynamic Range Control is used, the DRC data shall be encoded utilising AAC specific means: DRC data is stored in the data field dynamic_range_info, which is contained in the so called extension payload – a syntactic element of AAC [TS 102563]. The reason is, not to use extra data capacity in the PAD field, but to transmit the DRC data directly within the audio stream.

3.5.4 The Receiving End

A common and simple receiver concept realises the necessary DRC functions. These functions do not depend upon the type of algorithm in the DRC signal generator used at the transmitting end. The same options for listening to the uncompressed programme, the programme with the nominal degree of compression and the programme with a greater degree of compression are always available.

The DRC data is demultiplexed from the PAD area, and a linear interpolation regenerates a continuous signal to control the gain of the audio channel, in order to match the dynamic range of the received programme to the individual listener's needs. The maximum gain change in the compression mode will be about 15 dB. If the error protection applied to the PAD fails, specifically defined limiting characteristics for the rate of gain change will protect the compressed audio signal against severe impairments.

The conventional way for controlling the dynamic range is to use a simple multiplication in the digital domain, or a VCA (or DCA) at the analogue end of the receiver to realise all necessary gain adjustments (DRC, manual or automatic volume control and other gain-related features such as the balancing of music and speech loudness).

Solutions fulfilling the DAB requirements for DRC data have been developed by several parties, see also [Hoeg, 1994]. These systems normally need a look-ahead time, typically from 24 ms up to 3 s. There are several configurations of DAB receiver equipment (see Chapter 8) which support DRC and M/S functionality.

Besides the DRC system with control data being transmitted as described in the DAB standard [EN 300401], self-contained compressors for use in receivers which are operating without the need for transmitted control data have also been proposed. One of these [Theile, 1993] is based on scale factor weighting in the MPEG-1 Audio Layer II decoder.

3.6 Multichannel Audio with DAB

The use of discrete multichannel audio (3/2 or 5.1 format respectively) in the home is becoming adopted with the introduction of the new storage medium DVD and the advent of digital terrestrial television (DVB-T, see section 1.6).

One of the basic features of the MPEG-2 Audio standard [IS 13818] is the backward compatibility to MPEG-1 [IS 11172] coded mono, stereo or dual-channel audio programmes. This means that an MPEG-1 audio decoder is able to decode correctly the basic stereo information of a multichannel programme. The basic stereo information is kept in the left and right channels, which constitute an appropriate downmix of the audio information in all channels. This downmix is produced in the encoder automatically. The backward compatibility to two-channel stereo is a strong requirement for many service providers who may wish to provide high-quality digital surround sound in the future as there is already a wide range of MPEG-1 Audio Layer I and Audio Layer II decoder chips which support mono and stereo sound.

3.6.1 Characteristics of the MPEG-2 Multichannel Audio Coding System

A generic digital multichannel sound system applicable to television and sound broadcasting and storage, as well as to other non-broadcasting applications, should meet several basic requirements and provide a number of technical/operational features. Owing to the fact that during the next few years the normal stereo representation will still play a dominant role for most of the consumer applications, two-channel compatibility is one of the basic requirements. Other important requirements are interoperability between different media, downward compatibility with sound formats consisting of a smaller number of audio channels and therefore providing a reduced surround sound performance. The intention is to provide for as wide a range of applications and facilities as possible, including multilingual services, clean dialogue and dynamic range compression.

MPEG-2 audio allows for a wide range of bit rates from 32 kbps up to 1066 kbps. This wide range can be realised by splitting the MPEG-2 audio frame into two parts:

1. The primary bit-stream which carries the MPEG-1 compatible stereo information of up to 384 kbps.
2. The extension bit-stream which carries either the whole or a part of the MPEG-2 specific information, that is the multichannel and multilingual information, which is not relevant to an MPEG-1 audio decoder.

Figure 3.17 shows the ISO/IEC 13818-3 MPEG-2 Audio Layer II multichannel audio frame consisting of the MPEG-1 compatible part and the so called extension part.

The 'variable length' of the ancillary data field in the MPEG-1 compatible audio frame, or the MPEG-1 compatible part of the MPEG-2 frame, gives the possibility of carrying some multichannel extension information. However, for high-quality Audio Layer II coding the bit rate for the multichannel audio signal may exceed 384 kbps, requiring an extension part to be added to the MPEG-1 compatible part.

All the information about the compatible stereo signal has to be kept in the MPEG-1 compatible part. In this case, the MPEG-2 audio frame consists of the MPEG-1 compatible and the (non-compatible) extension parts. The MPEG- 1 standard provides for a maximum of 448 kbps for Audio Layer I and 384 kbps for Layer II. The extension bit-stream must therefore provide such additional bit rate as may be necessary. If, in the case of Layer II, a total of 384 kbps is selected, the extension bit-stream can be omitted.

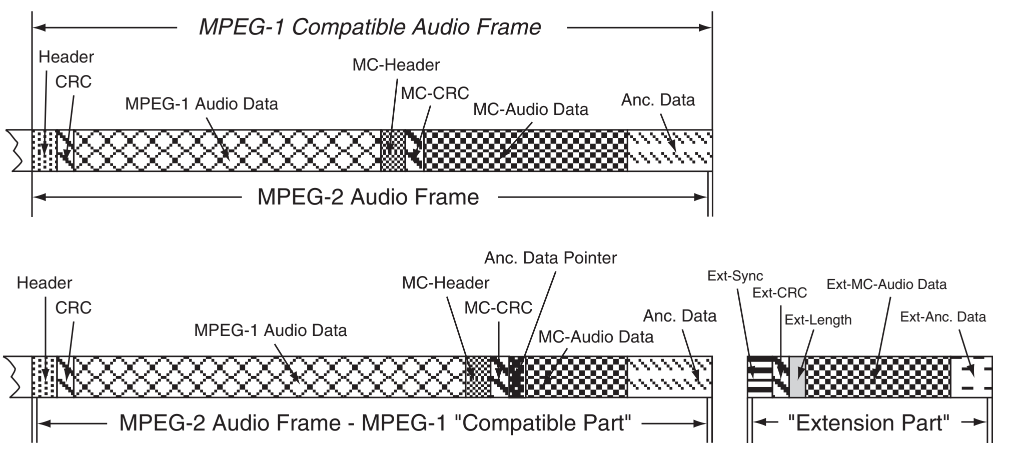

Figure 3.17 MPEG-2 Audio Layer II multichannel audio frame, consisting of the MPEG-1 compatible part and the extension part

The MPEG-2 standard does not require the bit rate to be fixed, and in applications such as ATM transmission, or storage applications such as DVD (Digital Video Disk) a variable bit rate may be used to advantage.

Serious quality evaluations [BPN 019], [Wüstenhagen, 1998] have shown that a bit rate of about 512 to 640 kbps will be sufficient to provide a reasonable audio quality for multichannel audio transmission using MPEG-2 Audio Layer II coding (non-coded transmission of five signals would need a net rate of $5 \times 768 = 3.840$ kbps).

3.6.2 3/2-Multichannel Presentation Performance

3.6.2.1 The ITU-R Multichannel Audio Format

The universal multichannel audio format named 3/2 or 5.1 is based on the generic international standard given in ITU-R Recommendation [BS.775] and other related EBU documents [R96], [BPN021] and SMPTE [RP173] as well.

These formats can be used by DAB or other digital TV services such as DVB-T. The DVD audio recording disc format additionally supports the 5.1 multichannel audio standard.

ITU-R BS.775-1 defines the following source and reproduction signals/channels as Format 3/2 which means three signals for the left, right, centre channels to the front, two signals for left surround and right surround channels, and an optional low frequency effect (enhancement) signal, as designed below:

Code	Signal/channel
L	Left Front
R	Right Front
C	Centre
LS	Left Surround
RS	Right Surround
LFE	Low Frequency Effect (optional)

As an option, because it is common in the film industry, the reference 3/2 format can be supplemented by an additional low-frequency effect (LFE) channel for signals intended for low-frequency effect speaker(s). This channel is used for special effects in motion pictures etc. It has a restricted bandwidth of 20 to 120 Hz and up to 10 dB higher reproduction level.

An important requirement for the new multichannel system is to provide backward compatibility to existing audio formats using a lower number of channels/signals. This leads to a hierarchy of compatible sound systems (see Table 3.2), from mono (1/0) via two-channel stereo (2/0) up to the new universal multichannel audio system (3/2).

Table 3.2 Hierarchy of compatible multichannel sound systems for broadcasting and recording (according to ITU-R [BS.775])

System	Channels	Code	Application
Mono	M	1/0	Radio/TV
2-channel stereo	L/R	2/0	Radio/TV, CD/MC recording
2-channel + 1 surround	L/R/MS	2/1	(practically not used)
2-channel + 2 surround	L/R/LS/RS	2/2	(some special recordings on CD)
3-channel stereo	L/C/R	3/0	(practically not used)
3-channel + 1 surround	L/C/R/MS	3/1	Matrixed surround (Dolby, etc.)
3-channel + 2 surround	L/C/R/LS/RS	3/2	Universal multichannel system for film, TV/radio, recording (DVD)

In this hierarchy, up- and down-mixing of multichannel sound originated for broadcasting, cinema or recording is possible as well as programme exchange or broadcasting at different format levels. That means a receiver can reproduce a programme transmitted or recorded in the 3/2 format also in 2/0 at an appropriate quality, depending on technical and other conditions on the reception side.

The increasing penetration of discrete multichannel audio systems into the consumer market worldwide, means that broadcasters should not be left behind.

Multichannel sound reproduction was primarily developed for film sound and is mostly used for this medium and for DVD recordings now.

Otherwise, pure multichannel audio programmes can be transmitted by radio, as the revised coding scheme for DAB (Digital Audio Broadcasting) now allows the use of MPEG-2 Layer 2 multichannel audio. Although there are some limitations for the maximum bandwidth (bit-rate) in a DAB ensemble multiplex (1.5 Mbps per block). Recent tests made by the EBU [BPN 019] have shown that a bit-rate between 512 and 640 kbps is sufficient to provide a reasonable quality for a 5.1 multichannel audio programme.

The provision of multichannel audio by radio or TV can be an argument for the user to replace their conventional reception devices with new receivers for DAB and/or DVB.

3.6.2.2 Low-frequency Effect Channel (LFE)

According to [BS.775], the 3/2-stereo sound format should provide an optional Low-frequency Effect (LFE) channel (sometimes also called Low-frequency Enhancement), in addition to the full range main channels. The LFE is capable of carrying signals in the frequency range 20 Hz to 120 Hz. The purpose of this channel is to enable listeners, who choose to, to extend the low-frequency content of the audio programme in terms of both low frequencies and their level. From the producer's perspective this may allow for smaller headroom settings in the main audio channels. In practice, the LFE channel is mainly used with film sound tracks. The mentioned 3/2 format is commonly called '5.1 format', where '.1' means the LFE channel.

3.6.3 Compatibility Aspects

3.6.3.1 Backward/forward Compatibility with MPEG-1

The multichannel audio decoder has to be backward/forward compatible with the existing two-channel or monophonic sound format.

Backward compatibility means that the existing two-channel (low-price) decoder should decode the basic 2/0-stereo information from the multichannel bit-stream (see Figure 3.18) correctly. This implies the provision of compatibility matrices [ten Kate, 1992] using adequate downmix coefficients to create the compatible stereo signals L_0 and R_0, shown in Figure 3.19. The inverse matrix to recover the five separate audio channels in the MPEG-2 decoder is also shown in the same figure.

The basic matrix equations used in the encoder to convert the five input signals L, R, C, LS and RS into the five transport channels T_0, T_1, T_2, T_3 and T_4 are:

$$T_0 = L_0 = (\alpha \times L) + \beta \times (\alpha \times C) + \gamma \times (\alpha \times LS)$$
$$T_1 = R_0 = (\alpha \times R) + \beta \times (\alpha \times C) + \gamma \times (\alpha \times RS)$$
$$T_2 = C^W = \alpha \times \beta \times C$$
$$T_3 = LS^W = \alpha \times \gamma \times LS$$
$$T_4 = RS^W = \alpha \times \gamma \times RS$$

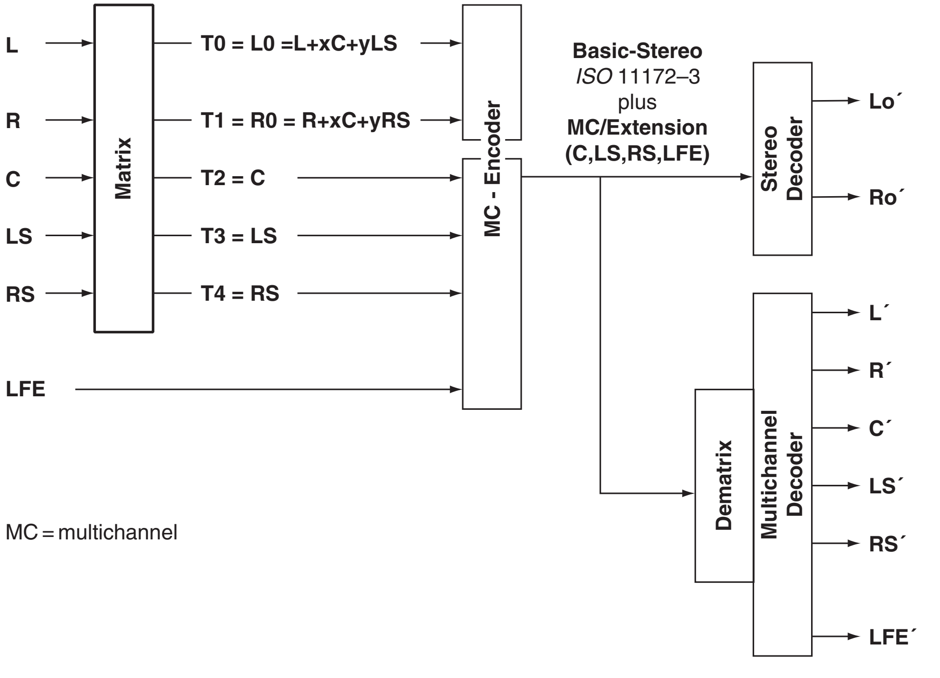

Figure 3.18 Backward compatibility of MPEG-2 Audio with MPEG-1

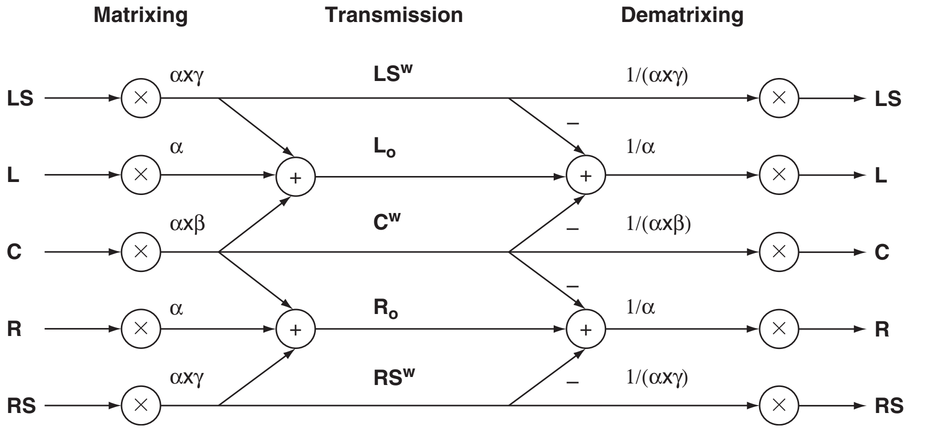

Figure 3.19 Compatibility matrix (encoder) to create the compatible basic stereo signal, and the inverse matrix (decoder) to re-establish the discrete five audio channels

Four varying matrix procedures with different coefficients α, β and γ have been defined and can be chosen in the MPEG-2 multichannel encoder, dependent from the individual character of the programme. Three of these procedures add the centre signal with 3 dB attenuation to the L and R signals. The surround signals LS and RS are added to the L, respectively R, signals with either 3 dB or 6 dB attenuation. The possibility of an overload of the compatible stereo signals L_0 and R_0 is avoided by the attenuation factor which is used on the individual signals L, R, C, LS and RS prior to matrixing. One of these procedures provides compatibility with Dolby Surround decoding. Being a two-channel format, compatibility can already be realised in MPEG-1. MPEG-2 allows such transmissions to be extended to a full, discrete five-channel format.

The fourth procedure means that no downmix is included in the bit-stream, which actually constitutes a Non-backwards Compatible (NBC) mode for the MPEG-2 multichannel codec. An MPEG-1 decoder will produce the L and R signals of the multichannel mix. In certain recording conditions this ‘matrix’ will provide the optimal stereo mix.

Forward compatibility means that a future multichannel decoder should be able to decode properly the basic 2/0-stereo bit-stream.

The compatibility is realised by exploiting the ancillary data field of the ISO/IEC 11172-3 audio frame for the provision of additional channels (see Figure 3.17).

In its first realisation the DAB system [EN 300401] will not provide multichannel sound. Therefore the extension to digital surround sound has to be backward/ forward compatible with an MPEG-1 Audio decoder.

3.6.3.2 Compatibility with Matrix Surround

Matrix surround systems, such as Dolby ProLogic, have found wide acceptance in consumers’ homes. Many movies and drama programmes are produced not only in discrete multichannel but also in matrix surround for delivery on video tape and analogue broadcast.

To broadcasters this legacy of matrix surround systems means that they should be able to continue to broadcast matrix surround material with the digital system. The audio coding system should be compatible with the matrix surround system.

The BBC conducted a study on the compatibility of MPEG Audio Layer II and Dolby ProLogic I (then the dominant matrix surround system) and found that, provided some precautions are taken, the surround information is completely preserved [Meares, 1998].

3.6.4 MPEG Surround for HE AAC

In section 3.2.5 the technique of Parametric Stereo (PS) was described, where a downmixed mono signal can be turned into a stereo signal with the use of a parametric description of the stereo features and a suitable synthesizer. MPEG Surround [Breebart 2005] is a system which takes this philosophy and extends it to surround sound, or multichannel audio.

As with Parametric Stereo (PS), an MPEG Surround encoder will have the original multichannel signals to analyse, and will aim to reproduce them as realistically as possible. Instead of transmitting six channels of audio (which would be the case for 5.1 surround sound), a downmixed version of the original can be coded and transmitted, which may either be stereo or even mono. The audio decoder will then decode a downmixed version of the audio, and a MPEG Surround decoder then can turn this downmixed audio into a six-channel version. Therefore a non-MPEG Surround enabled audio decoder will still be able to reproduce a useful downmixed output. The spatial information in the original six-channel audio is encoded by the MPEG Surround encoder and transmitted alongside the audio coding data, but usually at a far lower bit-rate. This side information can then be decoded to recreate the six-channel sound.

It therefore becomes possible with an HE AAC base codec combined with MPEG Surround functionality to encode 5.1 surround sound as low as 64 kbps with a quality than many listeners would be more than content to listen to. The bit-rate of the spatial information in this case could be as low as 3 kbps.

Apart from the standard format 5.1, MPEG Surround is able to transmit various multichannel audio formats if necessary, such as 6.1, 7.1 up to 22.1 formats.

3.6.4.1 General Approach

The spatial encoding technique consists of several different components and analysis techniques. Some of them are very similar to those used in Parametric Stereo described earlier, such as level differences and cross channel correlation. The diagram in Figure 3.20 shows how a typical encoder and decoder combination would work. The N channel input signals ($x(1 \ldots N)$) are downmixed to (though not exclusively) a stereo signal ($d(1,2)$), or can use a hand-downmixed stereo signal $(x(L,R))$. Spatial parameters are calculated from the multichannel and downmixed signals and are transmitted to the decoder along with the coded downmixed audio signals. The decoder stage will effectively upmix the stereo signal according to the spatial parameters the recreate an N channel output.

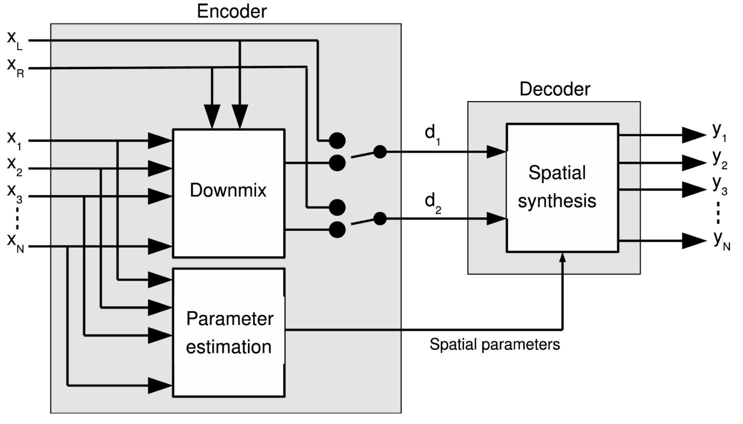

Figure 3.20 Spatial encoding and decoding

3.6.4.2 Building Blocks of the Decoder

Yet again there are similarities with Parametric Stereo in how MPEG Surround synthesizes the audio signals. The initial stage is to transform the downmixed input into the frequency domain using a hybrid QMF analysis filterbank. The hybrid aspect of this filterbank gives it a non-uniform frequency resolution which more closely matches the response of human hearing.

The second stage is to generate a multichannel version of the frequency domain signal. The signal is upmixed using the upmixing matrixes. Then the spatial parameters control the processing of the upmixed version of the signal, including the use of decorrelators to generate a spatially realistic sound.

The final stage is a hybrid QMF synthesis filterbank to turn the signal back into the time domain, but with the multiple channels.

3.6.4.3 Downmixing and Upmixing

The downmixing parts of the encoder and upmixing parts of the decoder each consist of two different elements, from which any combination of down/upmixing can be achieved (e.g. 5.1 to 2 channel, 7.1 to 5.1). The first element is a One-To-Two (OTT), which when in a decoder upmixes a single channel signal to a two-channel signal. Likewise, in an encoder its equivalent would downmix from two to one channel. The second element is a Two-To-Three (TTT) which, as the name suggests, converts two channels to three channels in the decoder, and vice-versa in the encoder version.

An OTT encoder element will mix the two channels together and will generate two spatial parameters, and a residual signal along with the downmixed mono signal. The decoder version of the OTT will take in the mono signal and use the spatial parameters to drive the generation of the two channel output. If the residual signal is available to the decoder, it can be add this to reproduce the original two channel signal.

The spatial parameters used in an OTT element are Channel-Level Difference (CLD) and Inter-channel coherence (ICC). These are very similar to IID and ICC as used in Parametric Stereo (see section 3.2.5).

The TTT encoder element will generate a third channel from the two input channels assuming a linear combination of the three. This may typically be a left-right to left-centre-right transformation. As with OTT elements it generates spatial parameters, in this case known as Channel Prediction Coefficients (CPC). It also generates a residual signal which can be used to improve the quality that the parameters cannot model.

The diagrams in Figure 3.21(a) and (b) show how a 5.1 to stereo encoder and a stereo to 5.1 decoder can be constructed with OTT and TTT elements. Note how the channels are grouped with each element so that the most likely similar sounding channels are together.

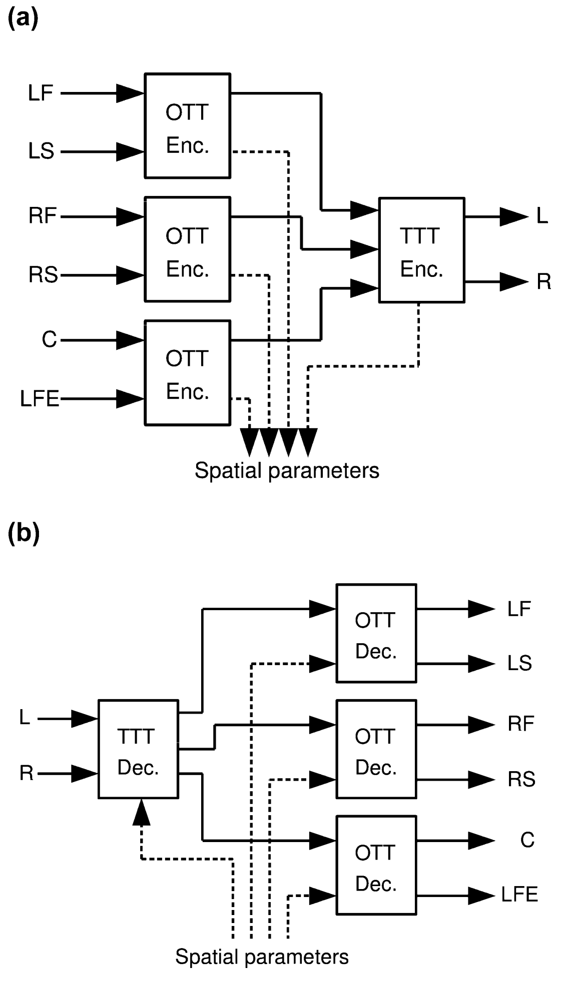

Figure 3.21 Downmixing/upmixing decoders with OTT and TTT elements **(a)** converting 5.1 to 2 channels, **(b)** converting 2 channels back to 5.1

Decorrelation, Temporal Shaping and Smoothing

The OTT and TTT encoder elements generate spatial parameters, including the ICC values that represent how spatial a sound is. To synthesise this diffuseness, the channels must be made to be more dissimilar to each other than just in levels. By using decorrelators in the frequency domain, this can be achieved. These decorrelators are implemented with lattice all-pass filters. To balance the amount of diffuseness in the sound, the decorrelated signals are mixed with the unfiltered versions. A side effect of this is temporal smearing, an artefact well known to audio coding experts.

To counteract temporal smearing (also known as pre-echo) some processing is carried out in the time domain. Each of the QMF bands can be processed separately and transformed in the time domain, and only the higher frequency bands are usually dealt with. The signals are modified by a time-domain envelope that has been calculated to minimise the smearing artefacts.

If the spatial parameters change values too rapidly from frame to frame, which is a particular problem at low bit-rates where the quantisers are very coarse, then unpleasant spatial artefacts can occur (e.g. image jumping around). The counteract this problem, the decoded spatial parameters can be smoothed over time to remove step-changes to the image.

Residual Coding

The OTT and TTT elements, as described earlier, produce two output streams. One stream contains the spatial parameters that are used in the main bit-rate reduction scheme. The other, less commonly used stream, is the residual signal. This is essentially the difference between the decoded spatial audio and the downmixed signal. By adding the residual to the synthesised multichannel signal the original multichannel signal can be recovered. Naturally, it would not be practical to send this pure residual signal as it would require a high bit-rate. However, it can also be encoded to reduce its data requirements, so can be transmitted along with the spatial parameters data to improve quality. Normally the residual signal would be discarded as the spatial parameters alone would provide sufficient quality for many applications; but at higher bit-rates the residual signal could help raise the quality still further.

Mono or Stereo Downmixing

MPEG Surround is capable of working with either stereo or mono downmixed versions of the multichannel original. While using a stereo downmix would produce a better quality in terms of imaging, and also better downwards compatibility with stereo material; the option of using a mono downmix is a useful one for lower bit-rate applications. Where the bit-rate is at a premium coding the audio with a mono signal would produce better audio quality than a stereo signal (if you compare Audio Layer II at 96 kbps mono and 96 kbps joint-stereo this becomes blatantly obvious!). Therefore a compromise with spatial quality becomes very worthwhile when the fundamental quality of the audio can be improved. The non MPEG Surround capable decoder can evaluate parametric (PS) data to generate a two-channel stereo output signal.

3.7 Other Advanced Audio Applications

The introduction of MPEG-2 and MPEG-4 coding features into the DAB standard provides additional possibilities which distinguish once more the DAB system from conventional radio systems. These features are taken further in DAB+ with the HE AAC v2 coding scheme, using the MPEG Surround functionalities.

3.7.1 Multilingual Extension

A good case for providing a multilingual service in combination with surround sound is given when the spoken contribution is not part of the acoustic environment that is being portrayed. Surround sound sports effects can be combined with multiple language mono commentary channels. In contrast, surround sound with drama would require a new five-channel mix for each additional language.

An important issue is certainly the 'final mix in the decoder'; that means the reproduction of one selected commentary/dialogue (e.g. via a centre loudspeaker) together with the common music/effect stereo downmix (examples are documentary film, sport reportage). If backward compatibility is required, the basic signals have to contain the information of the primary commentary/dialogue signal, which has to be subtracted in the multichannel decoder when an alternative commentary/dialogue is selected.

3.7.1.1 MPEG-2 Audio Layer II

MPEG-2 Audio provides alternative sound channel configurations in the multichannel sound system, for example the application of the 'second stereo programme' might be a bilingual 2/0-stereo programme or the transmission of an additional binaural signal. Other configurations might consist of one 3/2 surround sound plus accompanying services, e.g. clean dialogue for the hard-of-hearing, commentary for the visually impaired, multilingual commentary, etc. For such services, either the multilingual extension or the ancillary data field, both provided by the MPEG-2 bit-stream, can be used. This provides excellent speech quality at a bit rate of 64 kbps and even below, making very little demand on the available capacity of the transmission channel [BPN 021].

3.7.1.2 HE AAC v2

Providing additional audio channels for alternative audio services with HE AAC with parametric surround or stereo is not something that can be achieved easily. Parametric surround is based on the audio channels having a level of correlation and the assumption that it is single programme, rather than several independent pieces of audio. However as HE AAC can produce good quality at lower bit-rates, the practical route to providing additional channels would be to use another coded stream. For example, the main stereo programme could be coded with HE AAC v2 at 64 kbps and an additional clean dialogue channel could be provided with HE AAC at 24 kbps.

At higher bit-rates where surround sound programmes are used and encoded with discrete channels (i.e. not using parametric surround) then encoding an additional channel as part of the main coded stream would be possible. For example a 5.1 programme with an additional stereo commentary could be encoded using a single eight-channel codec rather than two separate codecs.

In addition to these services, broadcasters should also be considering services for hearing-impaired consumers (for more details see section 3.7.3). In this case a clean dialogue channel (i.e. no sound effects) would be most advantageous.

All those pointed features can be solved advantageously by using the new MPEG-4 coding functionality of object oriented coding; this is the Spatial Audio Object Coding (SAOC). Similar to regular SAC (Spatial Audio Coding) and based on MPEG Surround (MPS), see section 3.6.4, the object-oriented SAOC encoder receives the individual input signals (i.e. audio objects) and produces one or more downmix signals plus a stream of side information. On the receiving side, the SAOC decoder produces a set of object outputs that are passed into a rendering stage generating an output for a desired number of output channels and speaker setup. The parameters of the renderer can be varied according to user interaction and thus enable real-time interactive audio composition [Herre, 2007]

3.7.2 Half Sampling Rate Coding

MPEG-1 audio was extended through the MPEG-2 extension to lower sampling frequencies in order to improve the audio quality for mono and conventional stereo signals for bit rates at or below 64 kbps per channel, in particular for commentary applications. This goal has been achieved by reducing the sampling rate to 16, 22.05 or 24 kHz, providing a bandwidth up to 7.5, 10.5 or 11.5 kHz. The only difference compared with MPEG-1 is a change in the encoder and decoder tables of bit rates and bit allocation. The encoding and decoding principles of the MPEG-1 Audio layers are fully maintained.

Table 3.3 shows audio modes and bit rates for the 48 kHz and 24 kHz sampling rate. For a comparison of audio quality at half-sample rate and full rate.

Table 3.3 Audio modes and bit rates for 48 kHz and 24 kHz sampling rate

Sampling Rate	48 kHz	48 kHz	48 kHz	48 kHz	24 kHz
Mode	Single Ch.	Dual Ch.	Stereo	Int. Stereo	All
Bit Rate [kbps]					Bit Rate [kbps]
32	X				8
48	X				16
56	X				24
64	X	x	X	x	32
80	X				40
96	X	x	X	x	48
112	X	x	X	x	56
128	X	x	X	x	64
160	X	x	X	x	80
192	X	x	X	x	96
224		x	X	x	112
256		x	X	x	128
320		x	X	x	144
384		x	X	x	160

3.7.3 Audio Broadcasting for the Hearing Impaired

It is well known that the percentage of people with impaired hearing is increasing dramatically, both within the younger and the older generations. For many years the ITU-R [CCIR, 1991] has been looking for a means of broadcasting audio with higher intelligibility for people with impaired hearing. The target is to provide better audibility in general and higher speech intelligibility in particular for hearing-impaired customers of audio and TV broadcasting services. Ideally, an additional speech programme would be provided with higher intelligibility for interested people. This would be pre-processed in an appropriate manner, and could be further individually adjusted in the receiver according to the listener's need or preference.

New digital broadcasting systems such as DAB and Digital Video Broadcasting (DVB) provide possibilities for incorporating additional (associated) services for the hearing impaired, using the multi-programme features of MPEG-2 coded audio bit-streams, and probably other features of MPEG-4. Some initial discussions between a number of broadcasters, network providers and medical institutes led to some basic proposals [Hoeg, 1997], but no further progress has been reported, yet.

3.7.3.1 Signal Processing at the Studio Side

Some technological measures at the studio side can be made during the production process of broadcast programmes in order to improve the audibility of the programme, in particular the speech part.

Signal-to-noise ratio. Firstly, it is necessary to avoid introducing into the studio mix of the audio programme anything which is at a sufficiently high level that it significantly interferes with the audibility or intelligibility of the programme element which is of the greatest importance to the listener. For example, if the dialogue in a film or drama production is accompanied by intentional background noise (such as music or effects) it is important to ensure good intelligibility of the dialogue, if necessary by setting the level of the music or effects at a relatively low level. If we apply the term 'signal to noise ratio' (SNR) here to mean the ratio between the levels of the most important programme element and the background noise (i.e. the ratio between the dialogue and the effects) one can say that for people with normal hearing a SNR of about 6 dB would be satisfactory. For the hearing-impaired listener a SNR of more than 10 dB is necessary in order to ensure the intelligibility of speech [Mathers, 1991].

Separation of the speech programme. Ideally, a separate 'clean' speech channel should be produced and transmitted for the benefit of listeners with impaired hearing. The hearing impaired listener could then adjust the relative levels of speech and effects in an individual final mix, or even dispense altogether with the effects if required.

If this is not possible, the so-called 'Music/Speech-Flag' (see section 3.5.2) could be used to activate appropriate measures to improve the intelligibility of speech in the receiver.

Filtering. Typically, the frequency range of the hearing impaired is limited to the medium frequency bands. Therefore, the separate speech signal could be filtered at the studio side in order to avoid the transmission of extreme high or low frequencies, which can adversely affect intelligibility. A further consequence of such filtering would be to reduce the data rate requirements of MPEG audio coding systems carrying the bandwidth-reduced signal.

At the receiving side, more specific filter characteristics could be used, depending upon the individual hearing problems of the user.

3.7.3.2 Features of MPEG Audio Coding

MPEG audio coding, in particular its specification for DAB [EN 300401], provides a number of additional features which can be profitably used for the transmission and processing of programme signals for the hearing impaired. A general benefit of MPEG-4 audio coding [IS 14496] is the very low bit-rate which can be used for transmission of the additional speech channel. It can be coded with a bit-rate much less than 64 kbps (probably using the SBR Spectral band replication enhanced AAC coding schemes [Meltzer, 2002], [Stoll, 2002], see also section 3.2.4).

MPEG-4 audio coding provides a number of additional features such as Pitch control and Speed control which can be profitably used for the transmission and processing of programme signals for the hearing impaired. An MPEG-4 coded bit stream can be additionally transmitted easily within an MPEG-2 Audio bit stream, for instance with the so-called Ancillary data.

Dynamic Range Control. As already explained in section 3.5.1 in detail, in many cases the dynamic range commonly used by the broadcasters (approx. 40 dB) may be much larger than the wanted dynamic range (the so-called reproduction dynamic). This is particularly important for listeners with many types of hearing impairment, including most forms of age-related and noise-induced deafness.

As previously described, a DRC signal can be generated at the studio side, which describes the audio gain to be applied in the receiver. This DRC signal is transmitted together with the audio signal within the MPEG bit stream (MPEG-2 uses the Programme associated data, PAD; MPEG-4 can transmit those data within the audio bit-stream). In the receiver, the regenerated DRC signal may be used optionally to control the audio gain in order to match the dynamic range of the received audio programme to the requirements of the listener, in order to reduce discomfort and improve audibility (in particular the intelligibility of speech).

Multilingual channels. An MPEG-2 audio bit-stream can transmit up to seven separate channels additionally to the main service in two-channel or 5.1.channel format. MPEG-4 HE AAC can transmit an unlimited number of extra channels/signals. Those additional multilingual channels can be used not only for extra commentary channels, but also for a separate clean speech signal for the hearing impaired. All of these channels can be decoded simultaneously by the same decoder chip in a receiver.

Pitch control. This special feature of MPEG-4 audio coding means that the pitch of an audio signal can be changed at the receiving end without changing the duration of presentation. This could be used to align the frequency range of the audio signal to the individual needs of the hearing impaired customer.

Speed control. Another interesting MPEG-4 feature is to change (in particular, to reduce) the presentation speed of the speech programme in real time in the decoder (without changing the pitch of the signal), in order to improve the intelligibility of the speech content.

A similar method of speech rate conversion in order to improve the audibility for elderly people has been tested successfully in Japanese TV, see [Miyasaka, 1996].

3.7.3.4 Proposal of a Possible Scenario

The measures mentioned above could be combined in a system solution for DAB (or DVB) in order to improve the audibility of audio programmes for the hearing impaired, as proposed in [Hoeg, 1997]. All these components are already contained in the corresponding DAB or DVB system standards and require little additional bit-rate. For a block schematic diagram of a possible arrangement see Figure 3.22.

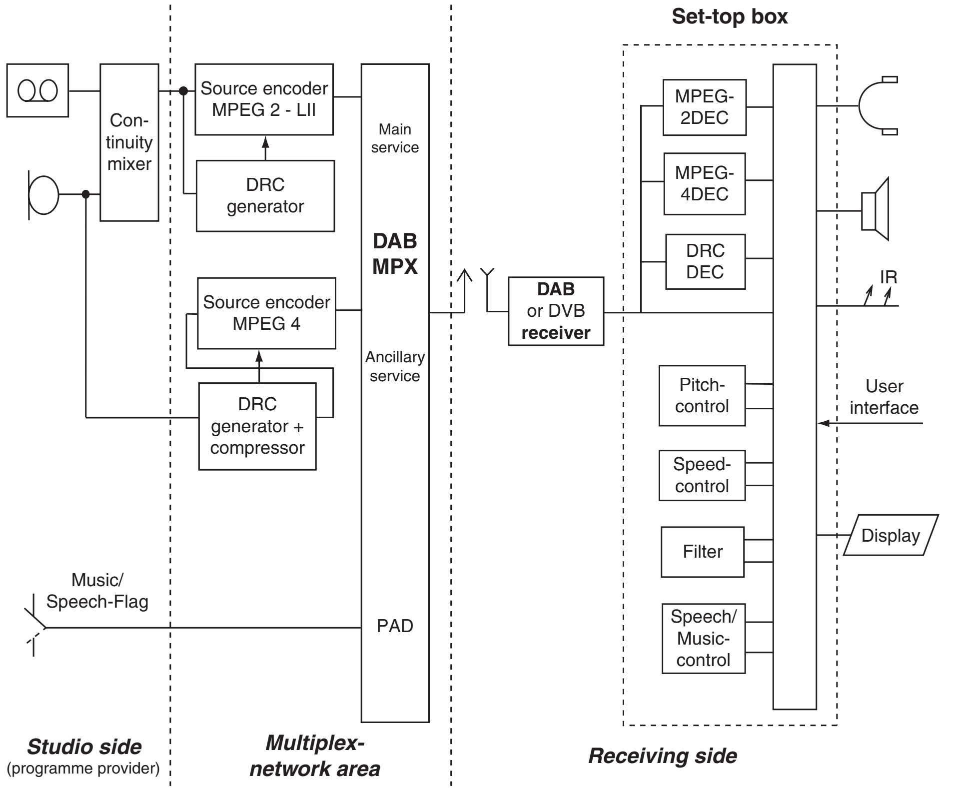

Figure 3.22 A possible scenario for programme transmission incorporating facilities for the hearing-impaired [Hoeg, 1997]

The idea is to produce (to extract) at the studio side a separate speech (commentary) programme signal, to pre-process it in an appropriate manner and to transmit it in parallel to the (non-processed) main programme signal, which may contain common audio information such as speech, music, environmental information etc. as usual. The pre-processing of the separate speech signal should be focused on those parameters which are relevant to the most frequently occurring problems of hearing, such as restricted frequency range, and discomfort with a wide dynamic range.

In particular, a setting for Dynamic Range Control (DRC) can be applied individually to the two programme components (speech and main programme). The use of the Music/Speech-flag can help to choose the right parameter settings at the receiver, depending upon the character of the programme.

The main programme will be encoded in an MPEG-1/2 audio bit-stream according to the DAB standard, which also carries the DRC and M/S information in the PAD. The separate speech channel can be encoded either in MPEG-2 (half-sample rate coding, see section 3.7.2.) or in MPEG-4, probably using one of the new SBR enhanced coding schemes. In the first case, the speech signal can be transmitted with a very low additional bit-rate within the multilingual channels of the MPEG-2 bit-stream. In the other case, the MPEG-4 bit-stream can be inserted into the ancillary data field of the MPEG-2 bit-stream.

After transmission the completed programmes are received with a conventional DAB or DVB receiver. In a specially designed set-top box (or a specially equipped receiver) the programme can be adjusted to the individual needs of the hearing-impaired customer. This includes:

- decoding of the transmitted MPEG-2 and/or MPEG-4 bit-streams, the DRC data and the M/S control signal;
- application of the special features such as filtering, Dynamic range control, Pitch control and/or Speed control;
- production of a final output signal by taking the special signal for the hearing-impaired listener, applying such processing as may be required, and possibly mixing this with the main programme, to create audio with improved audibility and intelligibility.

3.8 Quality of Service

The quality of service in digital transmission systems is typically much more constant than with analogue transmissions, such as FM or AM. In a digital transmission system, two layers of coding are used. The channel coding provides a very robust, transparent data channel. Error protection and correction measures are tailored to the specific transmission environment, such as a moving, mobile receiver in the presence of large echoes. The source coding is adapted to the nature of the data; in the case of DAB this is the audio signal. Its purpose is to reduce the necessary channel capacity (bandwidth) of the signal, without major impairments of audio quality.

The audio quality in a DAB system is therefore mainly determined by the source coding algorithm, implementation and the operational parameters, such as bit rate, sample rate and coding mode.

3.8.1 Analogue vs Digital Transmission

In an analogue transmission, any change of the transmission channel and any deviation from the optimum channel/directly influence the audio quality. Typical effects are interference between stations, fading due to reflections from buildings or cars and noise injection from nearby engines, ignition systems and others. So common are these impairments that we often fail to notice them at all. Little or no remedy is available.

In a digital transmission system a clear separation between the channel coding and the source coding enables these impairments to be minimised or eliminated. The error protection of the channel coding is so strong that it can provide a transparent channel for the source data under almost all conditions. Time varying changes of the transmission path do not affect the source data, unless the channel coding breaks down, which happens rarely.

The resulting audio quality is now mainly determined by the source coding scheme and its operational parameters, such as sample frequency, bit rate or coding mode etc. and is largely independent from the characteristics of the actual transmission path. This also means that even with a perfect transmission path, the audio quality can be limited, if the operational parameters are not set appropriately. In the event of transmission errors, the audio quality can drop dramatically, depending on the error protection and concealment that is available in the channel or source decoders.

In the age of digital transmission, the audio quality is no longer defined by objective measures like the signal-to-noise ratio, linearity, distortion and bandwidth. With the advent of perceptual audio coding, a signal with 13 dB SNR (Signal-to-Noise Ratio) can be a truthful representation of a 16-bit signal. Audio quality needs to be defined in relation to the uncompressed original signal and the perceived differences between this reference and the signal that is delivered to the listener. Traditional quality measures provide only limited information about the actual audio quality and can be misleading. It is necessary to develop new methods, as described in subsequent sections.

The quality of the reference is a moving target itself. Most consumers regard CD quality as state of the art, whereas years ago vinyl discs used to enjoy this reputation. With the introduction of higher quality formats, such as Super Audio CD (SACD) and DVD-Audio, this changes again. At the same time many people find the quality provided by highly compressed audio data (for instance produced by MPEG-2 Layer3 encoding, usually designed as MP3) perfectly well suited for casual listening, although this provides lower perceived quality than a standard CD. Against this background it is clear that any quality measure must be based on the difference between the uncompressed original signal and the coded version.

The impairments introduced by a coding system include linear distortion, quantisation noise, pre-echoes, bandwidth limitations, alias components, change of stereo image and timbre. Table 3.4 shows the most relevant categories of artifacts that may occur with digital coding or transmission techniques [BS.1284].

The AES have published an excellent educational/tutorial multi-media CD-ROM with many specific audio examples and explanations of audio coding effects [AES, 2002].

Table 3.4 Categories of artefacts that may occur with digital coding or transmission techniques (similar to [BS.1284])

Artefact Category	Explanation
Quantisation defect	Defects associated with insufficient resolution, e.g. granular distortion, changes in noise level
Distortion of frequency characteristic	Lack of high or low frequencies, excess of high frequencies, formant effects, comb-filter effects
Distortion of gain characteristics	Change in level (gain) or dynamic range of source signals, level jumps (steps)
Periodic modulation effect	Periodic variations of signal amplitude, such as warbling, pumping or twitter
Non-periodic modulation effect	Effects associated with transients, e.g. deformation of transient processes
Temporal distortion	Pre- and post-echoes, "smearing" (loss of time-transparency)
Extra sounds (noise)	Spurious sounds not related to the source material, such as clicks, noise, tonal components
Missing sound	Loss of sound components of the source material, e.g. caused by incorrect decisions of the masking model exacerbated by a general shortage of bits
Correlations effect (crosstalk)	Linear or non-linear crosstalk between channels, leakage or inter-channel correlation
Distortion of spatial image quality	All aspects including spreading, movement, localisation stability or accuracy of phantom sources, balance, change of spaciousness

3.8.2 Audio Quality vs Bit Rate

The perceived audio quality of any perceptual coding scheme depends on the bit rate in a nonlinear fashion. At the higher bit rates differences between the original and coded-decoded signals can be imperceptible. As the bit rate is reduced, a point is reached where the quality begins to deteriorate. At this point the coding scheme simply cannot provide satisfactory results and a number of different artefacts (such as quantisation noise, band-width limitation and change of the stereo image) can be perceived.

The MPEG-1 standard (as it is typically used at the time in DAB) specifies a sample rate of 48 kHz. MPEG-2 has defined an extension to 24 kHz. Both are allowed in DAB.

The extension to lower sample rates provides a better frequency resolution for signals which can be band limited, such as speech programmes. These constitute a significant part of broadcasts.

The data presented in Figure 3.23 are taken from [EBU, 1999]. It is a summary of multiple different listening tests, performed by several international organisations. The listening tests were executed according to ITU recommendation [BS.1116] and the scale used (see Table 3.5 below) is that of absolute impairment when compared to the original. The three points for each bit rate give the mean score for the least critical item and the

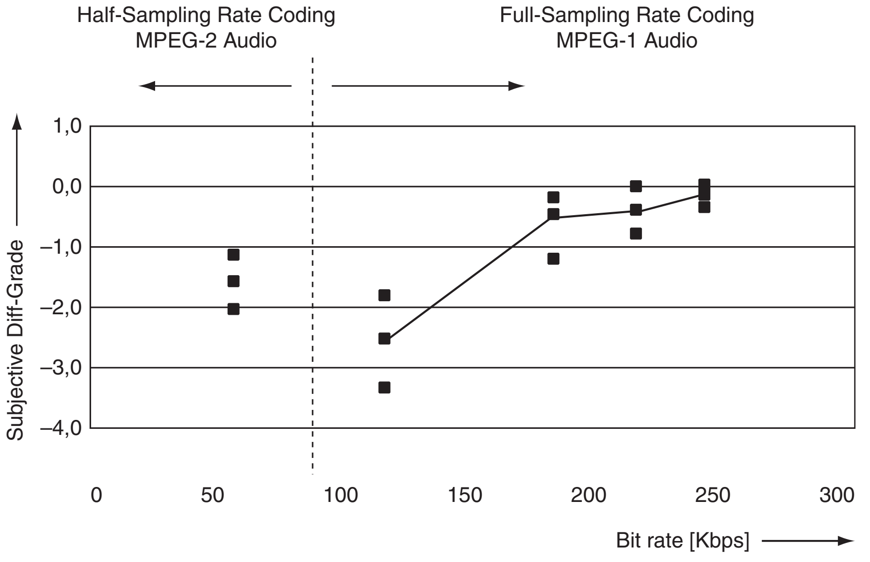

Figure 3.23 MPEG Audio Layer II, stereo: Quality versus bit rate and sampling rate

most critical item above and below the mean score over all items in the middle. The audio quality for a certain bit rate and setting depends on the actual codec implementation, because the psycho-acoustic model and the bit allocation strategy are not part of the standard and are subject to continuing research.

The figure shows that, as the bit rate of a 48 kHz-sampled signal is reduced, a point can be reached where the perceived quality is improved by reducing the sample rate to 24 kHz. The results given above can only be an indication of the general trend, but should not be used as the only rule to set all the coding parameters in the daily practice.

For further information on the relationship between quality and bit rate see [AES, 1996].

3.8.3 Subjective Quality Assessment

The ITU has developed recommendations for subjective listening tests for small impairments, after realising that the development of new and better coding schemes and their widespread use will require a sensitive method for assessing the quality of coded material and that test results need to be comparable and reproducible. Since the impairments depend to a large extent on the coding system and the test items, considerable effort has been put into procedures for the preparation of test material and the selection of test items. Finally the method is not defined in terms of known artifacts, but recognises that each new coding scheme is likely to introduce new types of impairments. The resulting method can be utilised, independent of the quality of the reference and the types of artifacts introduced. The only limitation is posed by the fact that the impairments should be small. In the presence of large impairments (such as those typically introduced by a speech codec) the method is not suited to resolve small differences between codec implementations.

Finally, it should be noted that the MPEG-Audio Layer 2 coding system, which is used with DAB, is one of the best tested coding schemes worldwide (see for instance [AES, 1996], [EBU, 1999]).

3.8.3.1 Test Methodology

Since broadcasters try to provide the best quality possible, a measure for small impairments has been created by the ITU [BS.1116], [BS.1283], [BS.1284].

The semi-continuous five-point impairment scale, given in Table 3.5 is used to measure differences between the original and the coded and decoded version. This relative scale allows for arbitrary absolute quality of the reference and normalises any impairments.

The highest grade (5.0) is given when no difference is perceptible between the reference and the signal under test. The lowest grade (1.0) is given when the differences are very annoying. All grades must be within this range. The subjective difference grade (SDG) is defined as the difference between the grade given to the hidden reference signal and the signal under test in a listening test. It allows comparisons between different tests and to normalise the results between different tests. This is described in detail in the next section.

Since each listener has different physiology and expertise, the SDG given by one listener may be different from that given by another listener. In a listening test, this is taken into account by presenting the SDG with confidence intervals that represent the spread between all expert listeners involved.

An expert listener has been trained to notice and identify these artifacts when comparing the coded version with the original, and he can often even notice them without explicit knowledge of the original signal. Untrained listeners can usually acquire this expertise in a matter of days. Repeated exposure to artifacts can make them obvious even to a casual listener in an imperfect environment. It is for these reasons that the method described in the next section strives to be as sensitive as possible, even if most listening is done in imperfect environments (e.g. noisy cars) by average listeners.

Small impairments may not be obvious to all listeners, under all listening conditions and their presence depends on the actual test signal. Since the methods discussed in this chapter strive to detect even the smallest impairments, they require expert listeners, a

Table 3.5 ITU Five Point Impairment Scale [BS. 1284]

Grade	Description of Difference with Reference
5.0	Imperceptible
4.0	Perceptible, but not annoying
3.0	Slightly annoying
2.0	Annoying
1.0	Very annoying

controlled and optimal listening environment and the most critical items for a given source coding scheme.

The core of the test is a listening session where an expert listener compares two versions of the material with the original (here called 'reference'). One of the two versions being evaluated is the reference, the other is the coded and decoded signal. The listener is not told which of the two versions being evaluated is the reference, therefore it is known as the 'hidden' reference (to distinguish it from the signal known to be the original). The listener has two tasks: first to identify the hidden reference (which must be given a grade of 5.0), then to grade the version that he identified as the coded signal. Since impairments are supposed to be very small or not noticeable at all, the listener may err in the first part of the task. In this case he may grade the hidden reference with something less than 5.0. When the results are then converted to subjective difference grades, these can take on negative values and the resulting confidence intervals from all listeners may extend beyond the zero line. This typically happens when a codec is so good that it does not introduce any noticeable artifacts (or only very small ones).

The test is said to be 'double blind' since neither the listener nor the test operator knows which of the signals is the hidden reference and which is the coded and decoded version. This set-up prevents any bias on the side of the listener or interference from the test operator. It is well known that a person who is asked to grade a signal that he knows to be coded will give it a lower grade than in the blind assessment.

The ITU-R recommendation [BS.1116] describes the necessary listening environment and requires full disclosure of acoustic and electro-acoustic measurements.

Prior to the actual grading sessions, listeners are trained in small groups. The listeners learn the test procedure during the training and can familiarise themselves with the test material, the artifacts and the equipment used.

Another test method called MUSHRA (Double-Blind MUlti Stimulus test with Hidden Reference and Anchor) was developed by the EBU and specified after by the ITU-R [BS.1342], especially designed for tests of systems or signals with lower quality levels, the so called Immediate quality level.

3.8.3.2 Test Preparation

The outcome of any subjective test depends strongly on the items that are being graded. Perceptual codecs often provide good quality across a wide range of material, but for each codec a specific set of items reveals its weaknesses. Since subjective tests are often used to determine which codec is to be used in an internationally standardised system (for digital radio, digital TV and storage based systems), any test must strive to provide a fair and balanced selection between test items.

Items should not include artificial signals, whose only purpose is to break a codec, but must be what is called 'typical broadcast material'. The reasoning is that a test should predict the behaviour in a practical set-up, instead of being solely a scientific investigation of the potential weaknesses of a coding scheme. An independent selection panel makes a selection from many different items, before picking some 10 to 15 for the final test.

The ITU described method has been crafted to allow comparison between tests performed at different times by different organisations. So-called anchors can be included in a test. These are coded versions of well-known items that have been included in many tests with the exact same encoding applied. If test results are comparable, then the average grades of these anchors should be the same across different tests. If they are, results from different tests can be compared directly. If they are not, this may be an indication that something during the test was very different from other tests and that caution should be applied when interpreting the results or comparing them with the outcome of other tests.

3.8.3.3 Analysis and Presentation of Results

The ITU recommendation [BS.1116] requires a full disclosure of the listening test procedure in a report. Any deviation from the procedure, any anomalies and reasons for rejecting listeners needs to be explained. Without a thorough description of all steps involved, a test report probably conceals more than it reveals.

A statistical analysis follows the grading phase once all listeners have concluded the sessions. A simple t-test is performed on the grades to identify and reject those listeners who were not reliably capable of identifying the hidden reference. The anchors used during the test usually lie at the lower end of the scale and an expert listener must be able to spot these. People may fail to do so because they misunderstood the instructions for the test execution, became bored during the grading session or are simply not expert listeners. Whatever the reason, these listeners' results need to be removed, before any conclusion can be drawn.

The absolute grades for each test session are converted to difference grades (grade of actual signal under test minus grade of reference). A proper identification of the hidden reference results in a negative difference grade, a wrong identification results in a positive difference grade. The data from all listeners for each item and codec is presented with confidence intervals and mean values. The interpretation is that an expert listener will give a difference grade somewhere inside of the confidence interval with a probability of 95%.

When the confidence intervals of two codecs overlap, for a particular item of test material, there is no statistically significant difference between the two, even if a diagram may suggest so. Only when the confidence intervals do not overlap at all, can one draw the conclusion that one codec might sound better than the other for this particular item.

Although tempting, it may not be very meaningful to average the grades given to all items for one codec. It is recommended that a codec must perform up to a certain level for any and all items, not on average across all items. The presentation of the results per item allows a better judgement of worst-case performance, average performance and performance to be expected in the field.

3.8.4 Subjective Tests for AAC Codec Family

While the AAC codec has been around for over a decade now, the SBR and PS extensions that give us the HE AAC v1 and v2 codecs, and MPEG Surround are far more recent developments, and therefore have not been so widely tested. At the time of writing there has yet to be a set of tests covering the full range (of sensible bit-rates at least) of AAC bit-rates and combinations for DAB+. However, an EBU group known as D/DABA is currently working on subjective tests for DAB+, including performance in error-prone (i.e. realistic radio reception) channels.

To help give an impression of the performance of AAC and its various offspring, some subjective tests covering various versions over the past few years are summarised here.

3.8.4.1 AAC vs Audio Layer II

In 1998, the Communications Research Centre (CRC) carried out some of the earliest listening tests on AAC [Soulodre, 1998]. Among the codecs tested were MPEG AAC (the core codec in those early days) and MPEG Audio Layer II, which are of most interest to us for the purposes of DAB and DAB+.

The listening tests were performed using the BS.1116 technique, which gives an impairment scale of 0 (no perceptible difference) down to -5 (very annoying). This method is aimed at higher quality coded audio, and cannot be directly compared with an absolute quality scale that would be used in a MUSHRA test.

There were numerous codec and bit-rate combinations covered in the tests, ranging from 64 kbps to 192 kbps; with MPEG Audio Layer II (hardware and software versions), Layer III, MPEG AAC, Dolby AC-3 and PAC all tested.

To highlight the combinations that are of interest to DAB, four out of these combinations have been selected for comparison. These are:

1. MPEG Audio Layer II at 128 kbps at 32 kHz sampling frequency.
2. MPEG Audio Layer II at 192 kbps at 48 kHz sampling frequency.
3. AAC at 96 kbps at 32 kHz sampling frequency.
4. AAC at 128 kbps at 48 kHz sampling frequency.

While 128 kbps is the most common bit-rate for DAB with Layer II, it is worth noting these tests used a 32 kHz sampling frequency, which is different to the 48 kHz used in DAB, so the results may differ somewhat. For these tests eight different test items (critical musical excerpts) were chosen.

In Figure 3.24, the results for each codec are shown for each test item and the average over all test items.

It can be seen that AAC at 128 kbps out-performs Layer II at 192 kbps, and AAC at 96 kbps is of similar performance to Layer II at 192 kbps. Some test items really highlight the difference in performance between codecs. For example, the test item *Pitch pipe* ('pitchp') is very tough on Layer II at 128 kbps, but AAC at 128 kbps deals with it very well.

It is worth remembering the tests were carried out in 1998, and the encoder algorithms have improved since then; but it does give an indication of the performance of the AAC core codec in comparison with Audio Layer II. Naturally, it is worth reading the actual publication to see the complete set of results and methodology used in the tests.

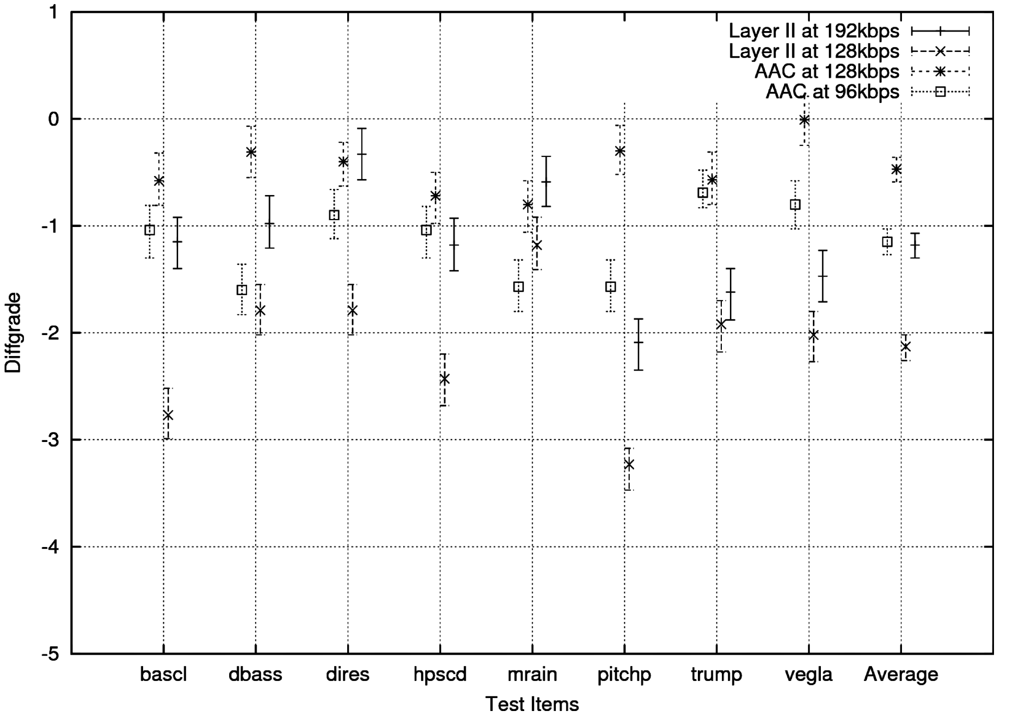

Figure 3.24 Test results for MPEG AAC and Audio Layer II codecs

bascl = *Bass clarinet arpeggio;* dbass = *Bowed double bass;* dires = *Dire Straits;* hpscd = *Harpsichord arpeggio;* mrain = *Music and rain;* pitchp = *Pitch pipe;* trump = *Muted trumpet;* vegla = *Susan Vega with glass*

3.8.4.2 HE AAC at 24 kbps Mono

In 2002 the BBC performed some listening tests on the AAC codec comparing its various versions with or without SBR at a particular bit-rate. The results were reported in [Meltzer, 2002], which summarises how SBR can be combined with AAC.

The codec combinations, all at 24 kbps mono unless stated, used in the MUSHRA tests were:

1. AAC Pure – Standard AAC.
2. AAC SBR – HE AAC.
3. AAC SBR Core – The AAC core codec only in HE AAC.
4. AAC Wideband – Standard AAC (this was set to 32 kbps, not 24 kbps).
5. 3.5 kHz anchor.
6. 7 kHz anchor.
7. Hidden Reference – the original signal, not bandwidth limited.

The results in Figure 3.25 show how SBR (HE AAC) improves the scoring over standard AAC by nearly 30 points at this bit-rate. It also shows that HE AAC at 24 kbps is better than standard AAC at 32 kbps.

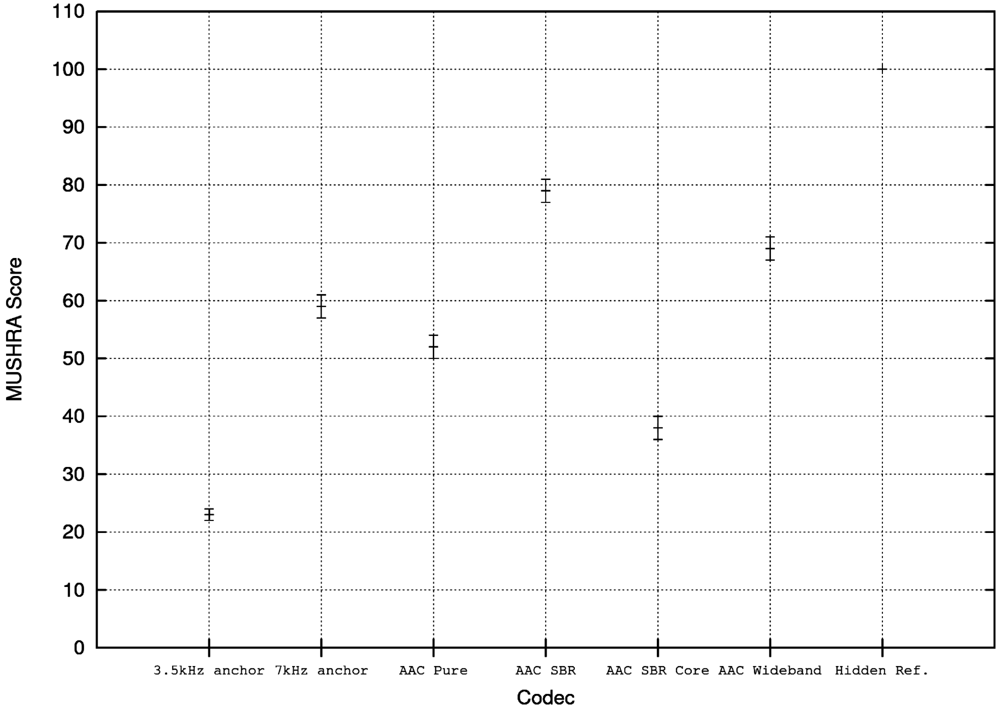

Figure 3.25 Test results for 24 kbps mono AAC versions

3.8.4.3 HE AAC with Surround Sound

In 2006 the EBU performed some audio quality tests on multichannel audio codecs. The codecs included were Dolby Digital, Dolby Digital+, Windows Media Audio (two versions), MPEG Layers II & III, DTS, AAC and HE AAC. A variety of bit-rates ranging from 1500 kbps down to 64 kbps were tested all with 5.1 channels. Of particular interest to DAB+ are the results of the HE AAC codecs that were tested at five different bit-rates. Some of the codecs were still under development at the time of the tests, so improvements have been made since the tests. In fact, the tests provided valuable information to the codec designers to help with these improvements. The full results can be found in EBU doc [Tech3324].

The codec and bit-rate combinations of interest to DAB+ are:

- HE AAC + MPEG Surround at 64 kbps and 96 kbps;
- HE AAC 5.1-channel at 128 kbps, 160 kbps and 192 kbps;
- AAC 5.1-channel at 256 kbps and 320 kbps.

Ten test items were used, carefully selected to be both critical and represent a wide range of material.

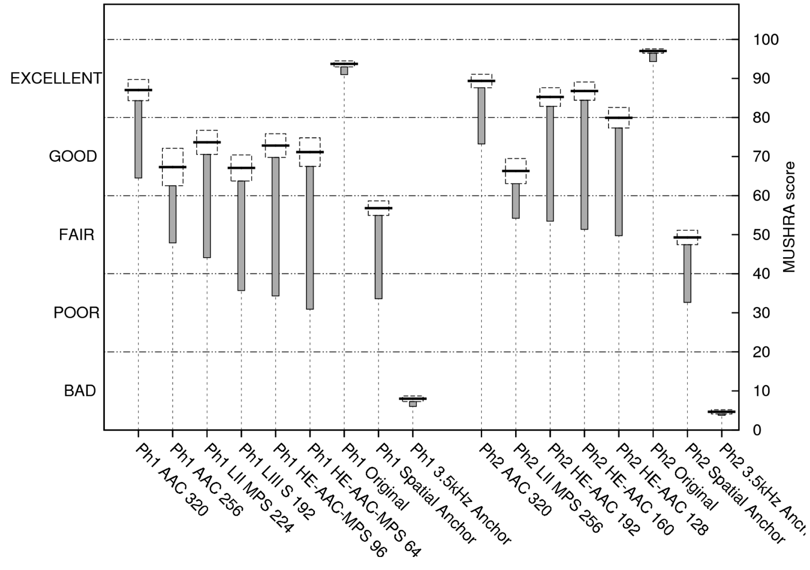

Figure 3.26 Test results for multichannel codecs

The MUSHRA methodology [BS.1342] was used in these tests, despite the reasonably high quality expectation. Due to the large size of the test, it was felt that [BS.1116] would be too time consuming for the listeners. One aspect of the results that was felt important to include was the score of the worst scoring item. Listeners will always notice bad artefacts even if the overall audio quality is generally high, and by just looking at an overall average, this problem can be overlooked; hence the need to check for troublesome test items.

The results of interest are shown in Figure 3.26. The scores for the hidden reference, spatial anchor (an image reduced version of the original), and a 3.5 kHz low-pass filtered anchor are shown for reference. It is worth remembering that the two phases used different test items, so the average scores are likely to differ somewhat, hence the use of anchors to help highlight these differences. The horizontal mark for each codec is the averages score over all test items, with the box around it the 95% confidence interval. The bottom of the thick line descending from each box represents where the average (over all listeners) for the worse case item lies.

The results show that HE AAC at 128 to 192 kbps performs very well with the average in the good or excellent region. However, the worst-case test item scores in the fair region, which is more of a concern. In this case the item is ‘applause’, which is a commonly occurring sound in broadcasting. Partly due to the results (though the applause issue was quite well known before the tests), the codec developers knew where to concentrate their efforts on improving the algorithm. Since these tests there have been improved versions of the HE AAC codecs developed, where the applause item would be expected to score significantly higher.

3.8.5 Error Protection and Concealment

Only very limited error protection for the audio bit-stream is provided in the MPEG standard. Error protection has to take into account the characteristics of the source data and the transmission channel. The MPEG Audio standards have been written for a wide range of applications with very different transmission channels, ranging from nearly completely transparent (i.e. error-free transmission channels, like storage on computer hard disks) to very hostile transmissions paths, like mobile reception with DAB [EN 300401].

3.8.5.1 Error Protection

In the presence of only a very few errors, say with a bit error ratio of about 10^{-5} to 10^{-6} and lower, the optional CRC check, provided in the ISO standard, will in general be an efficient tool to avoid severe impairments of the reconstructed audio signal. Errors in the most sensitive information, that is header information, bit allocation (BAL) and scale factor select information (SCFSI), can be detected. The chance of bit errors in this part of the audio frame are small. If, however, one single bit error occurs in these fields, the result will be the loss of a complete audio frame. In this case, the result of a single bit error is the same as if a complete audio frame is lost by a burst error or cell loss.

To protect a listener of ISO/MPEG coded audio signals from annoying distortions due to bit errors, channel coding has on the one hand the task of correcting as many bit errors as possible, and on the other detecting any bit errors remaining after correction. In the Eureka 147 DAB system, the data representing each of the programme services being broadcast is subjected to energy dispersal scrambling, convolutional coding and time interleaving. The convolutional encoding process involves adding redundancy to the service data using a code with a constraint length of 7. In the case of an audio programme, stronger protection is given to some bits than others, following a pre-selected pattern known as the Unequal Error Protection (UEP) profile, shown in Figure 3.27. The average code rate, defined as the ratio between the number of source-encoded bits and the number of encoded bits after convolutional encoding, may take a value from 0.33 (the highest protection level) to 0.75 (the lowest protection level). Different average code rates can be applied to different audio sources. For example, the protection level of audio services carried by cable networks may be lower than that of services transmitted in radio frequency channels.

Even with the best error correction scheme residual bit errors cannot be completely avoided, and have to be detected, especially for information which is very sensitive to bit errors.

This includes the header and control information, which has been taken into account by the ISO CRC, but also for scale factors, which has been only partially considered by ISO/MPEG.

The optimisation criterion for protecting encoded audio signals against bit errors is not the minimisation of the bit error ratio as the most important issue, but to minimise the perception of audio signal distortions in the case of bit errors. The subjective annoyance of bit errors depends strongly on the kind of disturbed data for the ISO/MPEG Audio coded signal.

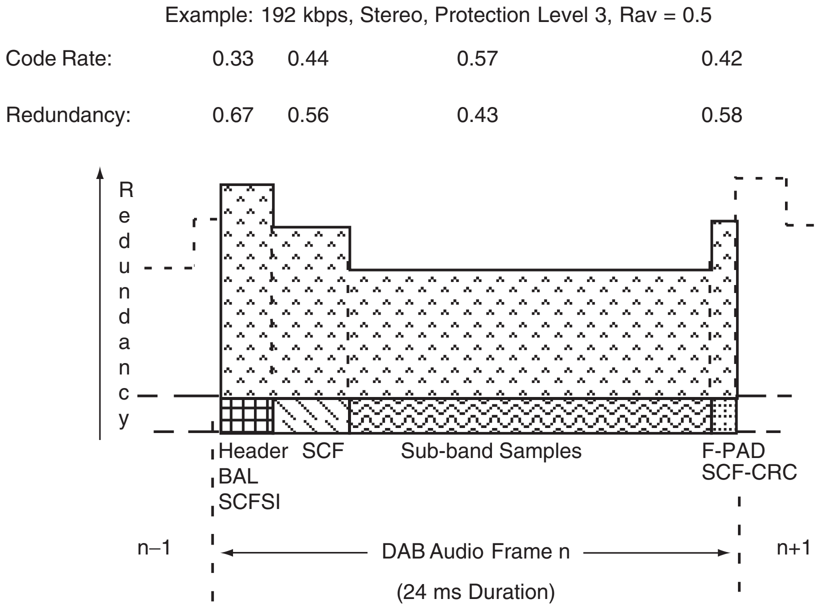

Figure 3.27 Unequal error protection scheme used in the DAB system for an MPEG Audio Layer II frame

Whereas a bit error ratio of 10^{-4} for sub-band samples results in barely audible degradation, any single bit error in the control information (header, BAL or SCFSI) causes a total frame error of 24 to 36 ms. The individual bit error ratios have to be balanced with respect to a certain degradation with the goal of minimising the overall distortion. Besides the economical aspect due to minimal additional redundancy, a UEP scheme allows for 'graceful degradation' which may be very convenient for the listener in the case of an increase of the bit error rate.

In the case of the Eureka 147 DAB system, the following measures for residual error detection strategies can be used:

Frame CRC (First Cyclic Redundancy Check), specified by ISO/IEC [IS 11172]. A 16-bit parity check word can be used for the detection of errors in the main audio information within the encoded bit-stream, that is ISO header, BAL and SCFSI. If the CRC is violated a concealment based on the replacement of non-reliable frames is requested.

Scale factor CRC (Second Cyclic Redundancy Check), specified by Eureka 147/ ETSI [EN 300401]. Up to four 8-bit parity check words can be used for the detection of errors in single scale factors within sub-band groups. If the CRC is violated a concealment based on the replacement of non-reliable scale factors is requested.

Reliability information If conventional or punctured convolution codes are used for the error protection, additional reliability information could be derived by a Viterbi channel decoder. Reliability information derived by the channel decoder will give more information on the actually violated data and would support the adaptation of concealment to the error situation.

Failure characteristics of the DAB system have also been tested by subjective assessment, see for instance [Lever, 1997] or [BPN 011].

3.8.5.2 Concealment Measures

Error correction and error detection strategies can be optimised concerning both the encoded audio signal and the listener's perception. Although some measures for error correction may have been applied, the channel coding technique has to provide for the detection of residual bit errors, because every channel code can be overloaded. Several error concealment strategies in combination with low bit rate coding, like muting, substitution, repetition or estimation of the disturbed parts of the received encoded audio data, have already been proposed and described in [Dehéry, 1991]. Those techniques improve the audio quality in case of transmission errors, but some of them, for example left/right-substitution, are not compatible with the audio bit-stream, and it cannot be expected that those techniques will be applied.

Applying error concealment techniques should result in the following improvements for the listener:

- Improved sound quality at the edge of the coverage area (single and burst errors).
- Improved sound quality during difficult mobile reception conditions (single and burst errors).
- Improved sound quality during cell loss in ATM distribution networks (burst errors).
- Performance and costs trade-offs between simple and cheap receivers and sophisticated and expensive receivers.

None of these benefits are available to the user in the old analogue system.

3.8.5.3 Frame-oriented Error Concealment

A complete frame or even some successive frames have to be processed with frame-oriented error concealment if the frame CRC indicates an error. Depending on the receiver's capabilities several concealment techniques can be applied.

Muting. The simplest method of frame-related error concealment is to mute frames which are indicated as non-reliable or even non-decodable. Compared with a very annoying reception the subjective annoyance caused by occasional muting of these frames might be assessed as lower.

Repetition. Instead of muting, non-decodable frames can be replaced by previously correct decoded frames. The audio quality of a repetition depends strongly on the signal itself; usually it is much better compared to a simple muting. Compared with muting, the repetition of frames needs memory of at least one frame (768 bytes at a bit rate of 256 kbps). A longer break of some frames can be bridged by either multiple repetition of one frame or, depending on the memory provided in a receiver, by a single repetition of the necessary number of previously correctly decoded frames.

3.8.5.4 Error Concealment of Scale Factors

Scale factors represent the maximum absolute value of a sequence of samples in each sub-band. Errors in scale factors do not lead to a distortion of one complete frame, but can be compared to a change produced by an equaliser where one or more sub-bands are amplified or attenuated by the disturbed scale factor. If MSBs of scale factors are not decoded correctly, 'beeping'-type impairments are produced and perceived as 'very annoying'.

The probability density function of differences of successive scale factors shows that for 90% of the time there are only small changes in the range of plus or minus one scale factor class. It is very difficult to detect such an error, which is limited to one sub-band and to an amplification or attenuation of about 2 dB. Based on these statistical data of scale factors, very simple error concealment can be applied. The disturbed scale factors are replaced by previously correct decoded scale factors. This can be considered as a concealment technique with no or only small impairments of quality.

3.8.5.5 Error Concealment by Plausibility Checks

The statistical properties of different information within one audio frame consisting of a DAB header, BAL, SCFSI, scale factors and sub-band data can be exploited for error detection.

An analysis of the successive scale factors of each sub-band from the point of view of statistical properties of audio signals can give additional and more detailed information on the reliability of the received scale factors. During an attack of the audio signal (e.g. a piano chord), scale factors may change greatly within a short time and may be followed by scale factor values which decrease slowly. During the decay of an audio signal continuously decreasing values of scale factors can be expected. This means that a high increase followed by a high decrease is not plausible. Thus, the scale factor has to be indicated as non-reliable and has to be concealed by, for example, repetition or interpolation. Compared to other possibilities of error protection and detection codes for the scale factors, this type of plausibility analysis does not need additional channel capacity and thus may lead to a similar evaluation of the error situation and performance.

In addition, the received scale factor values can be compared to the range of values expected by the SCFSI. If the SCFSI indicates the transmission of three scale factors within one sub-band, a certain difference (depending on the encoder algorithm) between the successive scale factors can be assumed. A more detailed investigation of these interdependencies might lead to a basis for a predictor model comparing received audio data with predicted data and classifying the received audio data as more or less reliable.

3.8.5.6 Subjective Assessment of the Error Performance of the DAB System

In 1994 and 1996, the Eureka 147 DAB project conducted tests to study the audible impairment of the DAB system at carrier-to-noise ratios approaching the failure point. Conventional subjective tests of the kind normally performed for this type of study would have presented some problems, however. Such tests require many presentations of different conditions to each listener, for sufficient results to be obtained to provide meaningful graphical data for the various conditions to be tested. Furthermore, with Rayleigh fading, the normal, short test sequences would not be appropriate at the higher carrier-to-noise ratios, because under these conditions the signal impairment results from short bursts of errors or muting distributed over relatively long periods of time. A new approach to this kind of testing was therefore devised, in which the carrier-to-noise ratio was adjusted at the input to the receiver to determine the point at which the effects of bit-errors first become noticeable (the 'OoI, onset of impairment') and the point at which the disturbance or muting becomes so frequent that the programme is no longer of interest to the listener, or where speech becomes unintelligible (the 'PoF, point of failure'). The data obtained from such tests is not suitable for graphical presentations (e.g. to show the subjectively-assessed quality as a function of the carrier-to-noise ratio) but it does convey the most useful indications of the error performance.

Test Conditions

A laboratory DAB chain was used commencing at the MPEG Audio Layer II encoder and ending at the output of a receiver, with a radio-frequency (RF) path simulation capable of presenting an RF signal with a wide range of carrier-to-noise ratios at the receiver input. Details may be found in [Gilchrist, 1994], [Lever, 1996], [Lever, 1997].

In the 1996 tests, a second DAB receiver was used to measure the bit-error ratio (BER) on a separate data channel in parallel with the receiver being used for the listening tests. This data channel had the same or a higher bit rate than the audio channel, and was protected with an equal error protection (EEP) profile at an equivalent code rate. This measurement enabled to verify that the receivers were performing in accordance with expectations at different values of carrier-to-noise ratio (C/N) in Gaussian and Rayleigh channels. Figure 3.28 shows the BER as a function of C/N for different code rates.

The audio test material selected for the tests comprised recordings of critical musical items, and of English speech items. The source of the programme items was the EBU SQAM Compact Disc [EBU, 1988]. In each test, the items were repeated until the listeners had reached a decision. Significantly longer periods of listening were needed for the tests involving a Rayleigh channel than for those involving a Gaussian channel.

The first tests were conducted with only a small range of bit rates and code rates, but with stereo, joint stereo and mono coding. In the subsequent tests, a wide range of code rates was tested at one of the bit rates, and a range of bit rates at one code rate. The audio bit rates and average code rates selected for these later tests are shown in Table 3.6.

In the listening tests, the test subjects asked for the carrier-to-noise ratio to be adjusted whilst they searched for the OoI and the PoF of the received DAB signal. The test conditions were chosen to ensure that audio services with sampling frequencies of 48 kHz and 24 kHz were tested over a reasonable range of bit rates and, in the case of the 24 kHz-sampled service, with both equal and unequal error protection (EEP and UEP). Onset of impairment (OoI) as the point where three or four error-related events could be heard in a period of about 30 seconds. The failure point PoF was defined as the point where the error-related events occurred virtually continuously, and signal muting took place two or three times in a period of about 30 seconds. The carrier-to-noise ratios at the OoI and at the PoF were found in the manner described, with a Gaussian channel (i.e. just with Gaussian noise added to the received signal) and with Rayleigh fading of the

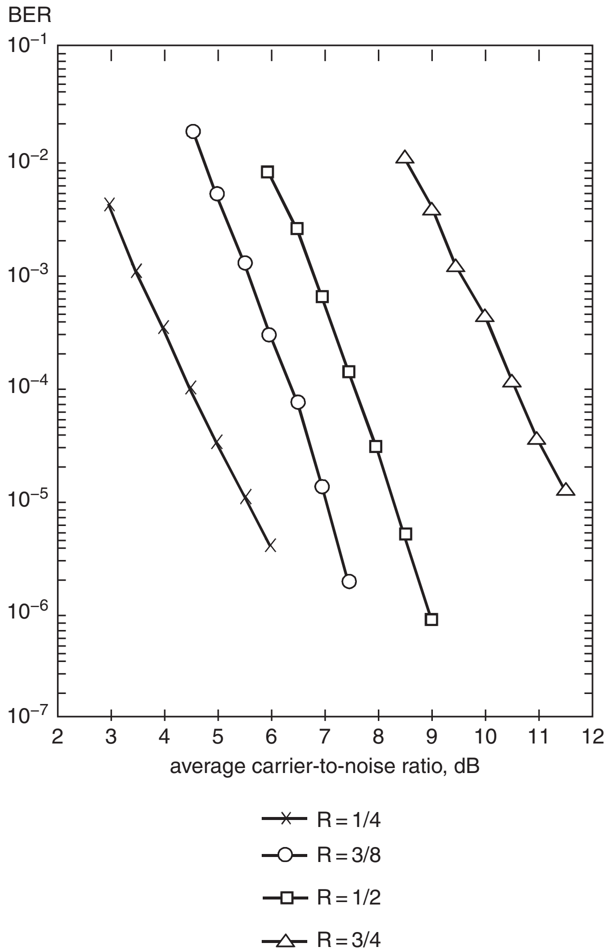

Figure 3.28 Bit-error ratio measured in a data channel of the fourth generation DAB receiver, as a function of the carrier-to-noise ratio

received signal, simulated using the Grundig 'FADICS' (FADICS stands for **FADI**ng **C**hannel **S**imulator equipment [Zumkeller, 1994]). The tests with the Gaussian channel represented the conditions in which a static receiver operates, and the FADICS was used to simulate two mobile reception conditions. One was reception in a vehicle moving at 15 km/h (approximately 9.3 miles per hour) in an urban environment, and the other was reception in a vehicle moving at 100 km/h (approximately 62 miles per hour) in a rural environment.

Tests were conducted with simulated Rayleigh fading in the urban and rural conditions described above in Band III for transmission mode I, and in L-Band for mode II. In some of the tests with Rayleigh fading, it was found impossible to obtain error-free reception, even at high carrier-to-noise ratios. Under these circumstances, the onset of impairment

Table 3.6 Conditions used in the 1996 error-performance tests

	Bit rate [kbps]	Sampling Frequency [kHz]	Mode	UEP/EEP	Code Rate	Protection Level
1	256	48	Stereo	UEP	0.50	3
2	224	48	Joint stereo	UEP	0.35	1
3	224	48	Joint stereo	UEP	0.40	2
4	224	48	Joint stereo	UEP	0.50	3
5	224	48	Joint stereo	UEP	0.60	4
6	224	48	Joint stereo	UEP	0.72	5
7	192	48	Joint stereo	UEP	0.51	3
8	160	48	Joint stereo	UEP	0.51	3
9	64	48	Mono	UEP	0.50	3
10	64	24	Mono	UEP	0.50	3
11	64	24	Joint stereo	UEP	0.50	3
12	56	24	Mono	UEP	0.50	3
13	56	24	Mono	UEP	0.60	4
14	40	24	Mono	EEP	0.25	1
15	40	24	Mono	EEP	0.38	2
16	40	24	Mono	EEP	0.50	3
17	40	24	Mono	EEP	0.75	4
18	32	24	Mono	UEP	0.50	3
19	24	24	Mono	EEP	0.38	2
20	16	24	Mono	EEP	0.38	2

could not be found, but where possible an approximate indication was obtained of the point at which the rate of occurrence of error-related events ceased to fall, as the carrier-to-noise ratio was increased from the failure point. This point was termed the ‘transition point’ because a transition occurred from a carrier-to-noise ratio dependent impairment condition to a static condition where a steady residual (or ‘background’) error rate was evident. This was used in place of the onset of impairment.

Test Results and Conclusions

The listeners in both series of tests had similar experiences concerning the precision with which the onset of impairment (OoI) and the point of failure (PoF) could be determined subjectively. There is typically an uncertainty of about 0.5 dB in determining the failure point. When determining the onset of impairment (or the transition point, where there was a residual error rate even at high carrier-to-noise ratios) the less critical programme items (speech) tended to mask the error-related impairments to some degree, typically introducing an uncertainty of about 0.5 dB. This latter phenomenon appears to be particularly noticeable under rural Rayleigh fading conditions with mode II transmissions at L-Band. However, there was an uncertainty of about +/–1.0 dB in determining the mean carrier-to-noise ratio for the tests with simulated Rayleigh fading.

Comparing the results for the DAB signal with added Gaussian noise, in mode I and mode II at the bit rates and code rates common to both sets of tests showed that any

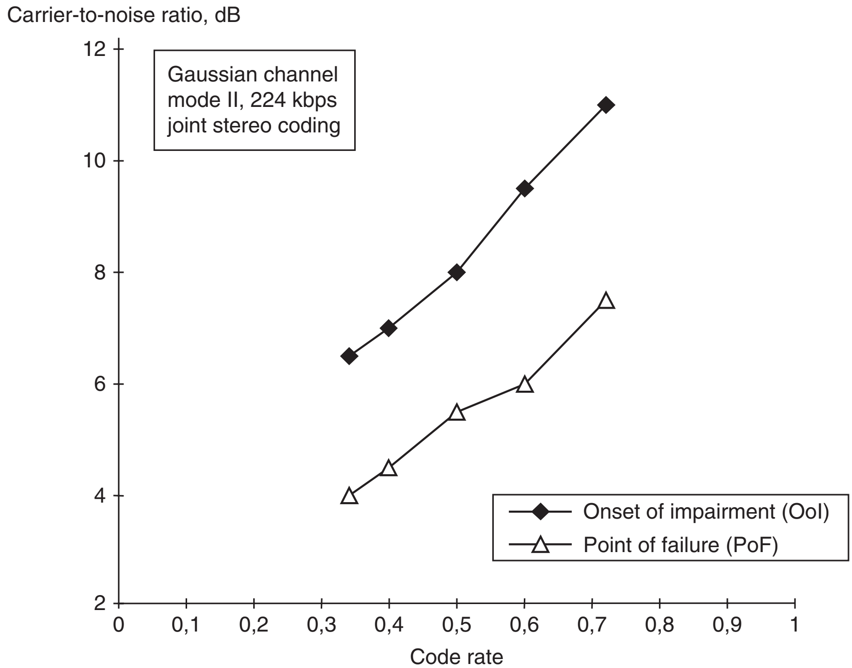

Figure 3.29 Carrier-to-noise ratio at the onset of impairment (OoI) and at the point of failure (PoF), as a function of the code rate

differences were very small, usually only 0.5 dB. As the differences were comparable with the expected error in determining the carrier-to-noise ratio at the OoI and PoF, they are probably not significant.

As would be expected, the error performance improved as the code rate was reduced. Figure 3.29 shows the relationship between the carrier-to-noise ratio at the OoI and the PoF as a function of the code rate for a Gaussian channel using transmission mode II. When subject to Rayleigh fading, the DAB receiver needed a consistently higher mean carrier-to-noise ratio. Rural Rayleigh fading conditions (simulating the reception in a faster-moving vehicle) required higher carrier-to-noise ratios than urban conditions.

The results showed that the bit rate and whether the audio is coded as mono, stereo or joint stereo have little, if any, effect upon the performance of the receiver at low carrier-to-noise ratios. They showed, too, that reducing the audio sampling frequency from 48 kHz to 24 kHz has no significant effect upon performance.

The use of equal or unequal error protection (EEP or UEP) is determined by the bit rate of the service. The results of the tests showed that UEP gives an improvement in receiver performance at the point of failure equivalent to between 1.0 and 3.5 dB in the carrier-to-noise ratio, at a code rate of 0.5. The greatest benefit is obtained under Rayleigh fading conditions. No significant effect was observed upon the onset of impairment when the type of error protection was changed. Thus one effect of changing from UEP to EEP is to bring the failure point closer to the onset of impairment.

No significant differences were found in performance when comparing mode I Band III transmission and mode II L-band transmission.

3.8.6 Objective Quality Assessment

Formal listening tests are very time consuming and expensive. The preparation of test material, the training, listening and statistical analysis require a great deal of effort and experience. Conducting such a test requires expertise and resources that are often beyond those available in an organisation that needs to monitor quality during ongoing operations or verify codecs for specific applications. Test results obtained from one implementation of a codec cannot be applied safely to another implementation. This motivated the ITU to develop an objective test methodology called Perceptual Evaluation of Audio Quality (PEAQ), see ITU-R recommendation [BS.1387]. The goal was to formalise and automate the lengthy subjective procedures into a tool and methodology that encompasses the experience and knowledge from numerous listening tests and today's understanding of the human auditory system into a toolset that can be applied quickly.

The method is not aimed at replacing formal listening tests, but rather to approximate the results of these. Its output is an estimate for the basic audio quality, as it would have been determined by a subjective listening test. The toolset provides an excellent means to monitor codecs in an operational environment and to perform comparisons between different implementations of a coding scheme. Developers can use it to assess modifications of a coding algorithm quickly.

An expert listener is trained to identify various artifacts of a codec. Since each new codec is likely to produce new artifacts, it is essential to realise that a tool that has not been trained for these particular artifacts may not be able to recognise and score them appropriately.

3.8.6.1 Test Methodology

The test method requires the reference (i.e. the original, unprocessed signal) and the coded signal as inputs. It evaluates all differences between the two and scores them according to psycho-acoustic criteria. The individual differences are combined to form an estimate of the subjective difference grade, called the objective difference grade (ODG). The data flow through the analysis system is presented in Figure 3.30.

The first step in this process is to transform the input signals to representations of the excitation pattern along the basilar membrane, by means of an FFT with subsequent morphing to a pitch scale and in parallel by means of a filterbank with level dependent filters. The excitation patterns are then corrected for the absolute hearing threshold and a spreading function is applied that represents some of the masking effects. The resulting excitation patterns are then used to derive several intermediate measures, such as the noise to mask ratio, linear distortions, differences in bandwidth, differences in modulation and the actual masking curves. Standard values for the just noticeable differences are then used to determine a detection probability of each individual difference found between the reference and coded signal.

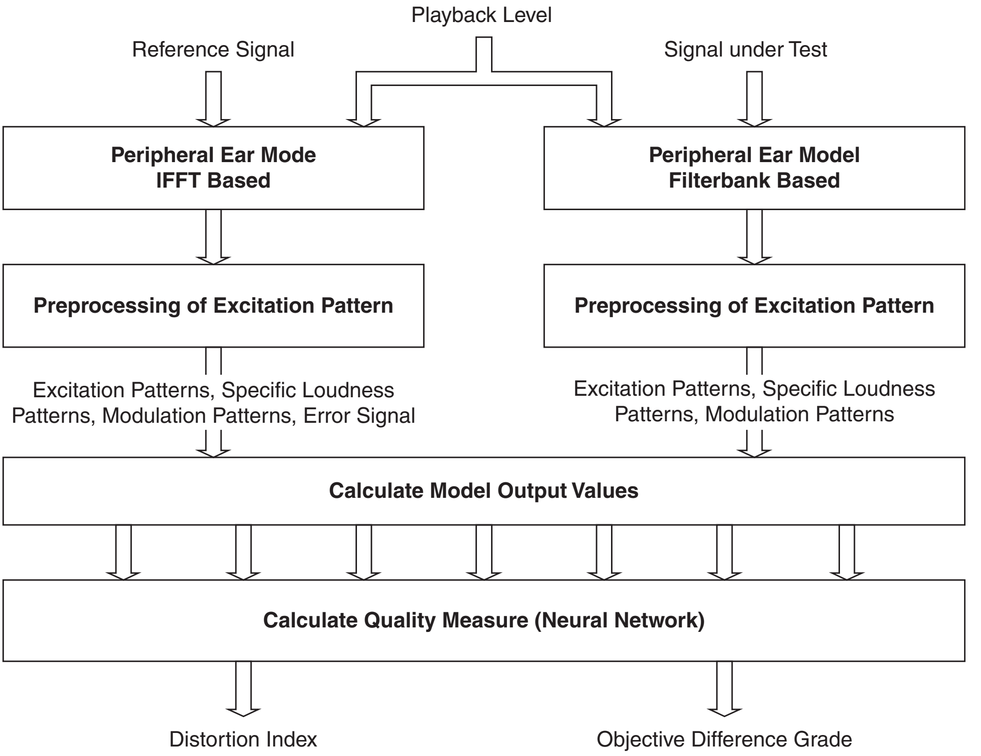

Figure 3.30 Data Flow in PEAQ Analysis [BS.1387]

The different measures are combined into an estimate of the subjective difference grade by a neural network, since their relationship is very complicated and not easily modeled. This network has been trained and verified with the results of several listening tests, some of which were conducted prior to the development of this ITU recommendation, some during the development. The network approximates the judgement of the individual differences as an average expert listener would.

The PEAQ methodology does not include a statistical component. The output is a single number for the mean difference grade. No attempt is made to determine the size and potential overlap of confidence intervals. As we have seen during the discussion of the subjective listening test, the confidence intervals often say as much about a comparison between codecs as do the mean values. Here this information is lost and one can only hope that the method provides a monotonous relationship between perceived quality and ODG. In the following we will see that this is not necessarily so.

3.8.6.2 Comparison between Objective and Subjective Assessment

The training of the neural network for PEAQ was performed with the utmost care. Of necessity a balance had to be struck between the number and severity of outliers (in this framework of statistical analysis, outlier means a measured value which lies significantly

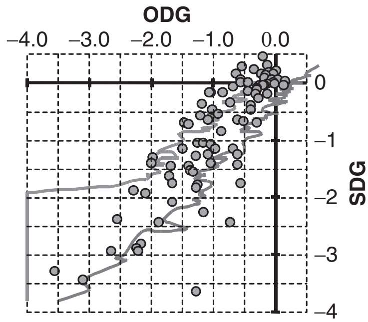

Figure 3.31 PEAQ: Advanced model output versus listening tests [Thiede, 2000]

outside of a given tolerance field), the accuracy of the individual listening test result in terms of the size of the confidence interval and the bias towards overestimating the severity of impairments. Two versions of the system were developed, with different complexity. The basic version has a large number of output variables, so the use of the advanced version is recommended for all applications.

Figure 3.31 depicts the outcome of the verification tests of the advanced model, comparing the subjective (SDG) and objective difference grades (ODG) of different programme items. Each of the plotted dots stands for an individual test item. The solid (grey) lines depict the tolerance scheme that was used during the model training process. The relative differences between the ODG and the SDG partly are fairly large. One can easily see that an individual item can be misjudged by as much as two difference grades – a so called outlier.

Studies have shown that a much better correlation can be achieved if ODGs are averaged across a large number of items. In this case the absolute error is always smaller than 0.5 difference grades. The drawback of averaging over several items is that the insight into worst case and best case performance of the codec is lost. A codec developer has to be very critical of the outcome of each individual item. For online monitoring and assessment of the validity of an installation, the averaging method may be very well suited.

The ITU recommendation [BS.1387] has been developed to facilitate the rapid evaluation of systems that include perceptual audio codecs and as an aid to developers and parties performing listening tests. It fully achieves these goals. It should not, however, be used as the sole method to determine which of two codecs is better suited for a particular application or standard. The final word in quality assessment remains with a subjective listening test, conducted according to the ITU recommendation [BS.1116].

3.9 Audio Levels

3.9.1 Audio Signal Level Alignment

Careful control of the audio level is very important in both the analogue and the digital environment. In the analogue environment, noise and distortion limit the capability of the system to accommodate signals with a wide dynamic range. In the digital environment, the system can be designed to accommodate signals with a wide dynamic range, but it is essential to avoid the hard 'clipping' of programme signal peaks which occurs when the signal level is too high. However, there can be problems other than those directly concerned with dynamic range. For example, different approaches to controlling levels may cause problems with the alignment of connections during the exchange of programmes between broadcasters.

The capability of handling signals with a wider dynamic range does not necessarily solve all the problems of programme level adjustment, see also [Gilchrist, 1998].

3.9.1.1 Programme Signal Metering and Control

In the past, broadcasters have used either the so-called Volume Unit (VU) meter or the Peak Programme Meter (PPM) to measure and set programme levels [IEC 60268], [BS.645]. The VU meter is a rectifier voltmeter with a rather long integration time (200 ms) which indicates approximately the average value of the voltage. It does not respond to short peaks in the programme. The PPM is a quasi-peak meter, with a rapid response to increasing level (the integration time is typically 10 or 5 ms respectively, depending upon the particular model) but the peak readings decay slowly. It gives a much better indication of the programme peaks than the VU meter, and the slow decay characteristic makes it easier for operators to observe the programme peaks.

There are a number of versions of the PPM, all having substantially the same ballistic characteristics, but with differences in the markings on the scale. Figure 3.32 displays the indications produced by various types of programme meters with the ITU-R recommended test signals.

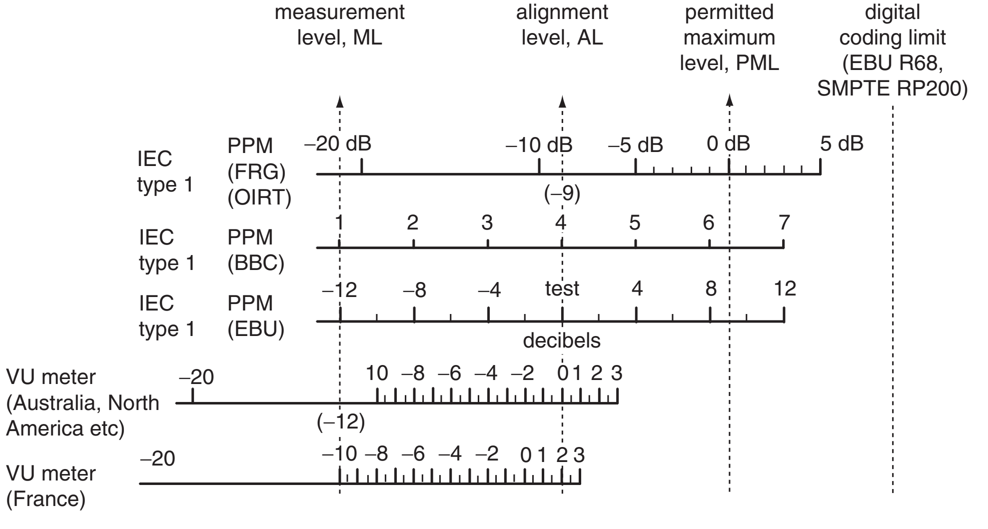

Figure 3.32 Programme level metering, according to [BS.645]

Most analogue systems do not introduce really 'hard' clipping when overloaded, and the onset of distortion is usually gradual. This means that the occasional gentle clipping of short-duration signal peaks normally does not produce unacceptable distortion. Nevertheless, broadcasters will generally arrange for some 'headroom' between the highest level indicated by the meter and the overload point of the transmitter, or provide a protective limiter, or similar device, to 'catch' the highest signal peaks and reduce their level.

Electronic level meters (so-called true peak programme meters with a very short integration time < 0.1 ms) have been available for a number of years. They have the capability of capturing and displaying accurately the level of even the briefest of signal peaks.

Any modification of programme peaks by a limiter may introduce distortion, or 'pumping', and the degree of annoyance depends upon the extent to which the peaks are modified. Some broadcasters, particularly those operating popular music stations, intentionally drive the limiters sufficiently hard to invoke the compression of all but the lowest signal peaks. This type of processing is used in order to produce a programme which sounds loud to the listener, possibly with the intention of sounding as loud as, or louder than, rival radio stations which may be competing for the attention of the listener.

3.9.1.2 Level Profile

The level profile of an audio transmission channel shows how the different levels of programme and test signals are located in the range between the upper system limit level (i.e. the clipping level in a digital domain, designed with 0 dBFS = 0 dB 'Full Scale'), and the noise signal level.

Figure 3.33 shows the principal level profile of a transmission channel. For broadcasting systems (recording, distribution, emission) a set of audio test levels is defined by the Recommendations ITU-R [BS.645] and EBU [R68].

The EBU recommends that the 'Permitted Maximum Signal level' L_{PMS} is set to 9 dB below the digital clipping level (Full scale). The intention is that, with 9 dB headroom, programme signals controlled using quasi-peak meters with an integration time of 10 ms will not be clipped, even if they contain transients with a very short duration. If true peak reading meters are used, they will give higher peak readings than the PPM on some programme material, whereas VU meters will typically under-read relative to the PPM as they have a long integration time.

In broadcasting and some studio recording operations, where programme interchange compatibility is of primary importance, it is normal to work to international standard guidelines that define an 'Alignment Signal level', L_{AS}, for alignment of the recording equipment or the transmission lines, respectively. ITU and EBU recommendations, among others, specify a digital alignment signal level of –18 dBFS, whereas SMPTE recommendations [RP200] specify a level of –20 dBFS. Both are likely to be encountered in operational practice, and it is therefore important to indicate clearly which alignment level is adopted, in order to avoid subsequent confusion and possible mis-alignment of levels.

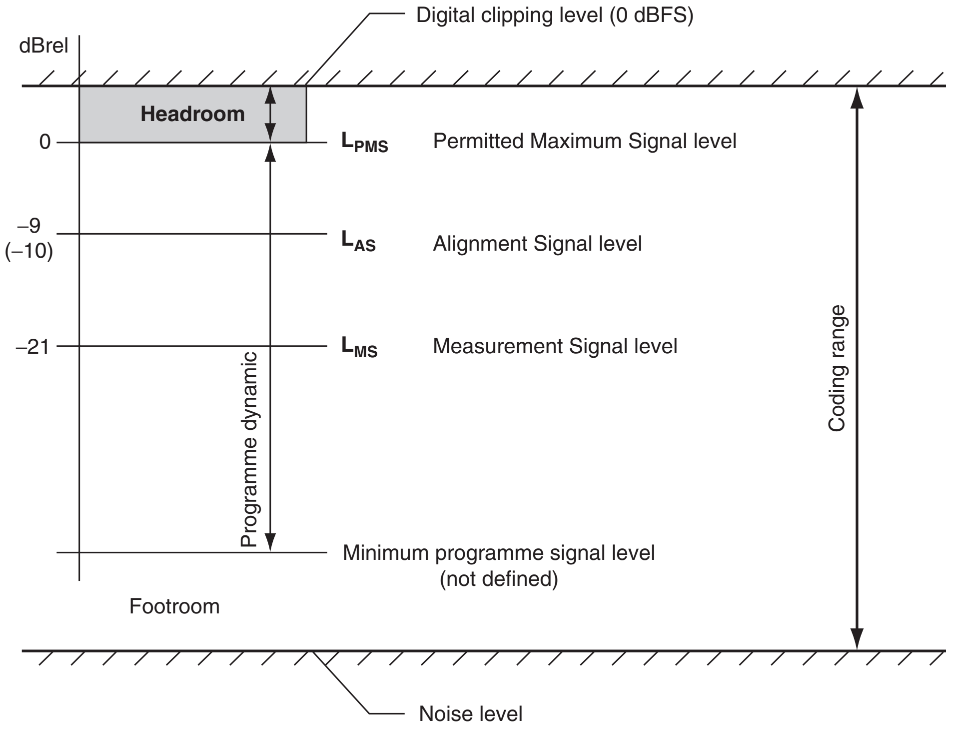

Figure 3.33 Principal level profile of a digital audio broadcast transmission channel

In situations where the alignment signal recommended by the ITU-R [BS.645] and EBU [R68] at −18 dBFS is used, the indications given by the various meters are as shown in Figure 3.32. In situations where the SMPTE reference level [RP 200] at −20 dBFS is used for alignment, the various types of PPM indicate a level 2 dB lower than with the alignment signal recommended by the ITU-R and the EBU, corresponding to '-11 dB' (FRG/OIRT) '$3^1/_2$' (BBC) and '−2 dB' (EBU). The VU meter indications would be 2dB lower, too, being either '−2' (Australia, North America, etc.) or '0' (France) but the recommendations of the SMPTE imply a change from the situation shown in Figure 3.32.

In countries such as North America, where the VU meter is ubiquitous, the use of the SMPTE Recommended Practice [RP200] to align and control signal levels prior to digital coding corresponds to the practice shown as 'VU meter (France)' in Figure 3.32, with the SMPTE reference signal causing the meter to read '0 VU'. Thus, when programmes are exchanged in digital form between organisations using PPM and VU metering in accordance with the SMPTE Recommended Practice, the relative responses of the meters shown in Figure 3.32 should remain valid with the exception of the upper VU meter scale (i.e. the scale labelled 'VU meter (Australia, North America, etc.)'). Under these circumstances this particular scale is no longer valid, or necessary, and should be disregarded. The remaining VU meter scale in Figure 3.32 (labeled 'VU meter (France)' in the diagram) applies to all VU meters during the digital exchange of programmes according to the SMPTE Recommended Practice.

A lower signal level, the Measurement Signal level (L_{MS}) is defined at –21 dB relative to L_{PMS}, for measurement purposes only (for instance measurement of the frequency response). This level has no significance for operational purposes.

3.9.1.3 Programme Levels and Level Alignment in Digital Broadcasting

Digital techniques offer the broadcaster the ability to handle audio signals with a considerably wider dynamic range than that accommodated by analogue techniques. Digital audio systems are characterised by noise and distortion levels which are determined principally by the design of the system (i.e. the resolution of the analogue-to-digital converters (ADCs) and any bit rate reduction techniques which may be employed), and by 'hard' clipping when the amplitude of the analogue audio signal is sufficient to drive the ADC beyond the limit of its coding range.

With a sufficiently high digital resolution (i. e. 16 bits/sample for CD recording, but up to 24 bits/sample for applications using an AES/EBU interface) connections for contribution, distribution and exchange can be implemented which provide a high signal-to-noise ratio and the capability to handle programme material with a wide dynamic range. The same is true for broadcast emission.

The application of bit rate reduction inevitably affects the noise levels, but practical digital radio services depend upon this for their existence, and today's advanced bit rate reduction techniques such as MPEG-1/2 Audio Layer II used for DAB have little or no effect upon the perceived audio quality provided the audio signal is afforded sufficient bit rate [BS.1115].

The metering and control of audio signals should therefore conform to the recommendations of the EBU [R68] and ITU-R [BS.645], or SMPTE [RP 200], as appropriate, in particular with regard to the provision of sufficient headroom for transient signal peaks which exceed the permitted maximum signal level.

3.9.2 Programme Loudness

Broadcasters have to handle a wide variety of audio programme material for both radio and television. Some of it has a very wide dynamic range (e.g. orchestral and choral music, film sound tracks) and some has a relatively narrow dynamic range (e.g. talks, interviews and music which has been subjected to a high degree of dynamic compression). The normal practice for analogue broadcasting has been for many years to control the audio signal so that the highest peaks just approach 100 per cent modulation. Trained operators use programme level meters to align levels in the broadcasting chain and control signal peaks, insofar as their own and the meters' performance enables them to do this. Usually, limiters are provided to prevent over-modulation of the transmitters. The intention is to maximise the signal-to-noise ratio and thereby the coverage of the transmitters.

One result of using this approach to audio level control is to set the mean programme level, and consequently the loudness, of programmes with a relatively narrow dynamic range significantly higher than that of orchestral and choral music. Thus broadcasts of operas, and choral and orchestral concerts, tend to sound much quieter than 'pop' music and chat shows, for example.

As already mentioned, some broadcasters drive their limiters hard in order to make their programmes sound as loud as possible, in the belief that listeners are attracted to the radio stations which sound the loudest.

However, listeners are becoming increasingly aware of audio quality issues, and in consequence less tolerant of distortion and changes in the perceived loudness of audio programme material. Loudness changes can cause discomfort to the listener, and most frequently occur when the listener switches from one broadcaster's channel to another, or when listening to a sequence of different types of programme from one broadcaster. Advertising material is frequently blamed, with some justification, for being presented at an uncomfortably loud level.

In a digital broadcasting system, no benefit to the coverage results from consistently high modulation levels. Furthermore, the system should be capable of handling material with a wide dynamic range. This means that dynamic range compression is unnecessary, unless the broadcaster requires it to produce a particular type of sound or the listener requires it (e.g. for listening in a noisy environment). In the former case, the broadcaster should apply the compression, but ensure that the mean programme level is not raised in consequence. In the latter case, the listener should use the dynamic range control signal provided in the PAD (programme associated data) in conjunction with the facilities in the receiver, see section 3.5.

A number of meters have been developed specifically for the purpose of monitoring the loudness of audio programme material, such as [Hoeg, 1996], [Emmett, 1994]. However, the removal of annoying loudness variations will depend upon changes to some broadcasters' current (at the time of writing) practices and their adoption of new standards. Progress towards this end has been made in the ITU-R with the publication of Recommendations for algorithms [BS.1770] and meters [BS.1771] for indicating loudness and true-peak audio level.

ITU-R [BS.1771] defines requirements for a meter that can perform measurement of loudness according to [BS.1770] by weighting with a modified L_{eq}-scale (Long term rms-equivalent measure), and optionally indicate true digital peaks, too. The tested weighting L_{eq}(RLB) uses a high-pass frequency weighting curve referred to as the revised low-frequency B-curve (RLB or RL2B). A new unit is introduced, known as LU (Loudness Unit) which is related to a dB-scale in such a way that a fall in level of 10 dB results in a reduction of 10 on the LU-scale.

The EBU has formed a Project group (P/LOUD) to draw up guidelines for the use of the ITU-R recommended metering either in combination with, or in place of, peak programme meters.

4

Data Services and Applications

4.1 General

While DAB early development mainly focused on *audio* broadcast, data capabilities have been designed into the system right from the beginning. After successful presentations of the new digital audio broadcast system at the beginning of the 1990s, these data capabilities gained more and more interest and prototype data applications were demonstrated by different project partners of Eureka 147 at related events (e.g. at NAB in Las Vegas, Spring 1992). In order to harmonise these prototype data applications and to reach compatibility between the equipment and services, data interfaces, protocols and applications were defined to match the needs of broadcasters, service providers and end users. In the meantime it is generally accepted that the usage of some DAB channel capacity, especially for multimedia data applications, is crucial both for user acceptance and for financing of the networks of this new digital broadcast medium. All the necessary hooks are available to follow this route.

4.2 Data Application Signalling and Access

A DAB multiplex can carry audio and multimedia data. In order to present multimedia data to the user, a special software is needed on the receiver side. Such software is usually called application software. The data application software is therefore the special piece of software needed on receiver side to present a specific type of data carried via DAB. Standards define how data is structured, transported and how it should be presented on receiver side. For example, the Dynamic Label specification within [EN 300401] indicates how short textual information related to the audio can be carried. If a data application is not defined in [EN 300401], then it is called a user application.

If we say that a specific data application is broadcast, we mean that data according to a specific data application specification is broadcast and that it can be presented by an application software complying to the specific data application specification.

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

The DAB multiplex is organised into services and service components (see section 2.3.6). Services and service components are logical components the user can select. If a service component is presented to the user, then data from a specific data channel is fed to a specific decoder. The DAB signalling indicates where the data is transported (this depends on the type of service component) and how the data is decoded, i.e., to which data application specification the decoder must comply. Data applications can be carried as a data service component in a packet mode sub-channel, in a stream mode sub-channel or in the FIDC. In addition, data applications can also be carried in the PAD channel of an audio service component (see 2.3.3.3). Similar to a packet mode sub-channel, a PAD channel may carry data for multiple applications in parallel. But whereas a single packet mode sub-channel may contain several data service components (one per data application), in DAB terminology the entire audio service component including everything contained in its PAD channel is still considered one single audio service component. There are no data service components inside a PAD channel.

DAB services consist of at least one service component. The type of the first service component, the so called primary service component, determines the type of the service. If the primary service component is an audio service component, then the service is called an audio service. If the primary service component is a data service component, then the service is called a data service. Both audio and data services may have additional (so called secondary) audio and/or data service components.

A DAB service forms an integrated whole of one or multiple service components. If the user selects a service, then all its service components should be presented to the user. Presentation starts with the primary service component. If the service has additional service components or data applications, then they should automatically be presented as well, provided they do not compete for the same output device. Data applications in the PAD channel of an audio service component must only be presented if the audio service component is also presented.

For example: an audio service comprises one primary audio service component, one secondary audio service component and one secondary data service component. Both audio service components carry Dynamic Labels and an MOT Slideshow. The data service component is an MOT Broadcast Web Site.

If the user selects this service on a portable device with a colour screen, then the device first determines the primary service component. The primary audio service component is then decoded and put out via the loudspeakers. Since the audio service component also carries Dynamic Labels and an MOT Slideshow, both will be automatically presented on the screen of the device. Neither the secondary audio service component nor the secondary data service component can be presented in addition, because they compete for loudspeakers/screen which are already in use by the primary service component and its data applications. A special button may permit the user to switch to the secondary audio service component. In this case its data applications will also replace those of the primary audio service component. The user will also be able to replace the Dynamic Labels/MOT Slideshow by the MOT Broadcast Web Site of the secondary data service component. However, the user will not be able to select the data applications of the secondary audio service component while listening to the primary audio service component or vice versa.

The example shows that the closest relation of an audio service component and a data application is among the audio service component and all data applications traveling in its PAD channel. If the audio service component is selected, then its data applications are implicitly selected as well. Additional data service components of the same service will only be automatically selected, if they do not compete for the same output device.

The labels provided by FIG type 1 and FIG type 2 support the user when selecting services, service components and/or data applications traveling in a PAD channel.

Any receiver that supports a certain user application must be able to extract it from any data channel that supports this user application. For instance a service provider might choose to carry a Slideshow in the PAD channel of the primary audio service component or as a secondary data service component in a packet mode sub-channel. The latter is useful if this service component is shared by several services.

It should be noted that while DAB permits very complex combinations of services and service components, at the moment most real-life services consist of one single (primary) service component. In case of audio service components, usually at least Dynamic Labels are provided in the PAD channel. In the future, more data applications are expected both inside the PAD channel as well as in secondary data service components.

The basic configuration information about data service components is part of the Multiplex Configuration Information (MCI, see section 2.3.6) sent in the Fast Information Channel (FIC). In contrast to audio service components, which are always sent in stream mode and coded according to the DAB audio coding schemes based on MPEG Audio Layer II or as DAB+, there are many possible coding formats and transport mechanisms for data service components. To enable the receiver to decode the service components, it is necessary to signal precisely which transport protocol is used and what presentation engine is required. 'User application' standards (such as the MOT Slideshow, Journaline or the Electronic Programme Guide standard) define the software needed to decode and present the data carried by the data service component. The list of all DAB user applications is provided in [TS 101756]. Several FIGs are used to identify the broadcast user applications and their data channels.

FIG 0/2 signals the Service Identifier (SId) of the service (see section 2.5.3) and lists all service components of the service. For audio and data service components carried in a stream mode sub-channel, FIG 0/2 indicates the sub-channel; for data service components carried in the FIDC (Fast Information Data Channel), the FIDCId (Fast Information Data Channel Identifier) is provided. In both cases this information is sufficient to extract the application data from the DAB data channel. In case of a data service component carried in a packet mode sub-channel, it is necessary to specify sub-channel and packet address. Since FIG 0/2 is unable to carry these two parameters, an additional FIG 0/3 is used per data service component to provide them. Each data service component has an associated Data Service Component Type (DSCTy). This parameter is signalled in FIG 0/3 for service components carried in a packet mode sub-channel, and in FIG 0/2 for service components carried in a stream mode sub-channel or the FIDC. In earlier revisions of the DAB standard, this parameter identified the type of the data service component, e.g., Traffic Massage Channel or emergency warning system. Today this parameter is usually used to define the transport protocol used by the service component, e.g., MPEG Transport Stream, Transparent Data Channel (TDC) or embedded IP packets. One entry in this table refers to the Multimedia Object Transfer Protocol (MOT) used by many DAB data applications.

The most important information regarding data applications is the user application information provided in FIG type 0 Extension 13. This mandatory parameter tells the receiver which application software is needed to decode and present the broadcast data. In case of user applications carried in the PAD channel of an audio service component, FIG 0/13 also specifies DSCTy and other transport related parameters; data service components use the same parameters, but encoded in FIG 0/2 or FIG 0/3. The PAD channel permits to carry multiple applications in parallel and therefore a parameter 'X-PAD AppType' is used to separate the applications streams. The 'X-PAD AppType' in a PAD channel thus has similar functionality as the packet address in a packet mode sub-channel.

Legacy PAD decoders will assume that an MOT Slideshow or MOT Broadcast Web Site always uses AppTypes 12 and 13 (and in addition AppTypes 14/15 for scrambled segments and conditional access (CA) messages). Therefore AppTypes 12 to 15 must not be used for other user applications.

4.3 The Multimedia Object Transfer Protocol

4.3.1 General Principles

The Multimedia Object Transfer (MOT) Protocol [EN 301234] is a transport protocol for the transmission of multimedia content in DAB data channels to various receiver types with multimedia capabilities. It is the common transport mechanism for transmitting information via different DAB data channels (PAD, packet mode) to unify access to multimedia content within the DAB system. MOT ensures interoperability between different data applications, different receiver device types as well as for equipment from different manufacturers. A basic support for multimedia presentations is also provided.

4.3.2 Structure of MOT

All information is conveyed from the information source to the destination as 'MOT objects' of finite length. The maximum object length that can be signalled is about 255 Mbytes. No constraints are applied on the content that can be transmitted. An example of such a content is an MP3 file or an image file.

Each object consists of 'header information' and 'body'. The object body contains the payload data to be transported, for example an HTML file. The header information carries all meta data of the object. The meta data consists of two types of parameters. MOT parameters are used to identify the object and to support reassembly, decompression/descrambling, and cache management in the MOT decoder. User application specific parameters provide additional information to the user application decoder on receiver side.

Transport of the header information of an MOT object depends on the MOT mode. MOT provides two modes:

- In 'MOT header mode', a sequence of MOT objects is broadcast. The next object can only be transmitted after the transmission of the current object is completed. The MOT decoder forwards every received MOT objects to the user application decoder.
- In 'MOT directory mode' a set of MOT objects is broadcast. The MOT directory describes all objects that belong to the current set of objects. MOT directory and bodies are broadcast in parallel. The MOT decoder manages the received set of files and handles cache management and object expiration.

In MOT header mode, the header information describing the current object is carried as the MOT entity 'MOT header'.

In MOT directory mode, the header information of all objects within the current set of objects is combined. Together with parameters within the MOT directory (MOT directory parameters) that apply to the whole set of objects, all header information of all objects is carried as the MOT entity 'MOT directory'.

There are five different types of MOT entities:

- MOT header (MOT header mode only);
- MOT body (unscrambled);
- MOT body (scrambled on MOT level);
- MOT directory (MOT directory mode only); and
- compressed MOT directory (MOT directory mode only).

During transport an MOT entity is split into segments. The type of MOT entity is signalled in the data group type field of the MSC data groups carrying MOT segments.

The user application defines which MOT mode is used and which MOT parameters and user application specific parameters are carried in the header information of an object. If MOT directory mode is used, then additional parameters referring to the whole set of objects may be used within the MOT directory.

It should be noted that the MOT header mode does not restrict the user application to one single object. In MOT header mode, the MOT decoder only manages one single object at any time. But it is possible to have a user application where several objects provided one after the other by the MOT decoder are managed by the user application decoder. An example is data applications that collect data received via MOT header mode, for instance a sequence of podcasts. The more podcasts have been received, the bigger the database gets on receiver side. In this case a user application specific management of the received objects is applied.

4.3.3 Transmission of MOT and Overhead Considerations

For transmission the MOT object is split into several segments and then mapped to packets (for transport in a packet mode sub-channel) or X-PAD data sub-fields (for transport in a PAD channel). Details are provided in [EN 301234].

When choosing the segment size for an entity, several factors have to be taken into account:

- The smaller the segment size, the less data is lost in case of a reception error. Note that this only makes a difference if entities are repeated. If an entity is broadcast without any repetitions, then large segments may be used. Since any segment lost due to a reception error will cause the loss of the entire MOT object anyway, it does not matter how big the lost segments are.
- If the enhanced packet mode with additional error protection (see section 2.3.3.5) is used, then the error probability is strongly reduced. In this case, larger segments may be used.
- The smaller the segment size, the higher the segmentation overhead. Each segment has an overhead of usually 11 bytes; 2 bytes for the segmentation header, 9 bytes for MSC data group header and MSC data group CRC.
- Additional overhead is caused by the lower layers of the protocol stack. If MOT is used in a packet mode sub-channel, then an MSC data group carrying the MOT segment will be mapped to a sequence of packets. Unused (padding) bytes in the last packet increase the overhead. This overhead can be reduced by using large segments and/or by choosing segment sizes that require no padding on packet level.
- Since the number of segments per MOT entity is limited to 32 768, the segment size might have to be increased for very large entities.

To minimise the total transport overhead, it is necessary to analyse the total overhead. Figure 4.1 shows the overhead on various protocol levels in a packet mode sub-channel. Overhead is for instance caused by the mapping of MOT segments to MSC data groups. This segmentation overhead depends on the size of the segment. Since the segmentation overhead is a fixed number of bytes, it can simply be reduced by increasing the segment size. For a segment size of around 1 kbytes, the relative segmentation overhead is around 1%.

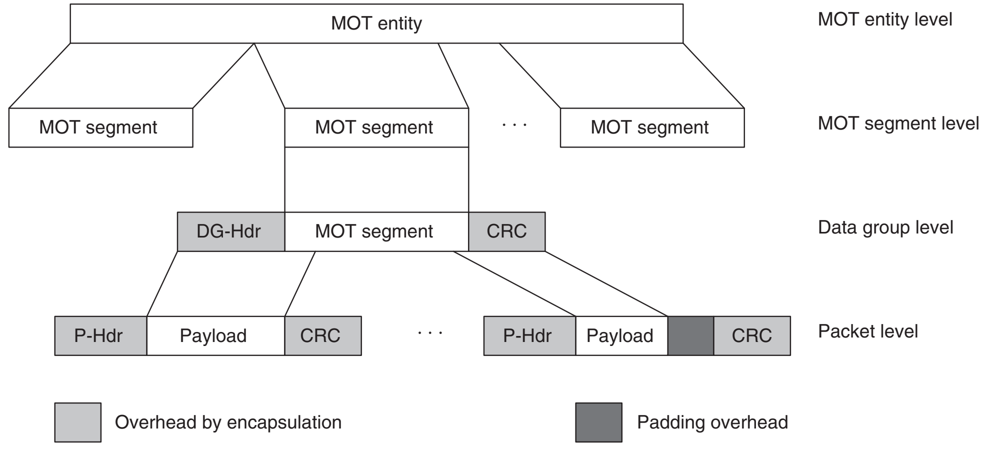

Figure 4.1 Transport overhead of an MOT entity carried in a packet mode sub-channel

If MOT is used inside a packet mode sub-channel, then the overhead caused by the mapping of MSC data groups (containing the MOT segments) to packets is much more significant. There are two types of overhead. Overhead caused by packet headers/packet CRCs and overhead caused by padding. DAB provides four different packet sizes (24, 48, 72, 96 bytes) and requires an integral number of packets per logical frame (i.e., each 24ms). This means that, depending on the sub-channel bit-rate, it is not always possible to use packets of 96 packets that would cause the lowest possible overhead. Each packet has a fixed overhead of 5 bytes for packet header and packet CRC.

The following examples illustrate this:

Example 4.2: A packet mode sub-channel of 8 kbps permits only packets of 24 bytes each. The relative overhead is 5 bytes per 19 bytes net, i.e., 26%.

Example 4.3: A packet mode sub-channel of 56 kbps requires at least two packets per logical frame, one 96 byte packet and one 72 byte packets. In this case the relative overhead is 10 bytes per 158 bytes net, i.e., 6.3%. All other possible combinations of packet sizes have a higher overhead.

Example 4.4: A packet mode sub-channel of 64 kbps requires at least two 96 byte packets per logical frame. In this case the relative overhead is 10 bytes per 182 bytes net, i.e., 5.5%. All other possible combinations of packet sizes have a higher overhead.

The examples show that the relative overhead on packet level can be quite high. The overhead is highest for small sub-channel bit-rates and lowest for sub-channels whose bit-rate is a multiple of 32 kbps. It can also be seen that it is bitrate-efficient to combine packet-mode sub-channels with low bit-rate to one larger sub-channel. In example 4.2, the relative overhead is more than 4 times higher than in example 4.4. The lowest overhead shows example 4.4 whose bit-rate is the sum of the bit-rates in example 4.2 and 4.3. While technically the combination of packet-mode sub-channels is beneficial, there might be political reasons against it or the resulting bit-rate might get too high for some types of receivers. If the bit-rate of the packet mode sub-channel is a multiple of 32 kbps and each MSC data group is transported without any interleaving on packet mode level, then most of the packets that carry an MOT segment will have the lowest possible overhead. At the beginning and at the end of the MSC data group, smaller packets might have to be used, thus slightly increasing the total overhead. One reason for such smaller packets than the optimum (maximum) packet size is padding. Padding is needed, if the gross size of the MOT segment (i.e., including MSC data group overhead and segmentation overhead) does not completely fill the last packet used to carry the MOT segment. Therefore the encoder should try to use segment sizes that require little or no padding. If, for instance, MOT segments of 1024 bytes are used and the last packet is a 24 byte packet that is only half full, then the relative overhead caused by this padding is 1%.

If MOT is used inside a PAD channel of an audio service component, then the overhead caused by the mapping of MSC data groups (carrying the MOT segments) to X-PAD data sub-fields is much lower than the overhead caused by a packet mode sub-channel. On the other hand, the error protection for PAD in Audio Layer II is usually worse.

Looking at the overhead caused by various protocol levels, segment sizes of around 1 kbyte seem useful. If MOT is used inside a packet mode sub-channel, then it is recommended to use a sub-channel bit-rate that is a multiple of 32 kbps and a segment size that requires no padding on packet level. The enhanced packet mode permits to use segment sizes of several kbytes. However, given the overhead on packet level and the additional overhead caused by the forward error correction, even much larger segments will not significantly reduce the total overhead.

4.3.4 MOT Header Parameters

Depending on the type and characteristics of the user application, the 'MOT header extension' may contain different parameters which the application needs for data management and presentation. The parameters may appear in any sequence. This section describes some of the most important 'header extension' parameters for MOT objects; a list of all general MOT parameters can be found in [EN 301234], application specific parameters are defined in the application specification (e.g. the MOT Broadcast Website specification [TS 101498]).

MOT basic transport parameters. These parameters identify the object, indicate its type and signal whether the object is compressed during transport.

MOT caching support parameters. These parameters are used in MOT directory mode only. They support cache management. Using these parameters, the MOT decoder can easily determine if the objects in its cache are still valid. So on start-up or if the user changes between different DAB ensembles, the MOT decoder only has to wait for the next reception of the MOT directory and then it can tell which objects are still valid and whether outdated objects may be presented until the current version of a specific object has been received. Two other important issues are addressed by these parameters.

The service provider can control object expiration (MOT parameter 'Expiration') and thus ensure that an MOT decoder that no longer receives any data (e.g., because the user is inside the sub-way) will not present data whose validity can no longer be verified. The service provider can specify a time interval and in case of a loss of reception longer than the specified time interval, the MOT decoder will refrain from presenting objects whose validity it can no longer verify.

Another parameter controls the MOT decoder behaviour in case of an object update (MOT parameter 'PermitOutdatedVersions'). User applications such as the MOT Broadcast Web Site have web pages that may contain inline objects such as images. If the web page is updated, it can be signalled to the receiver whether the old web page can still be presented until the new version of the web page has been received. If consistency reasons require that the new web page be presented with the new versions of the images only, then this can be signalled to the receiver. However, in this case a user will notice that this web page is available in the MOT decoder until it receives the MOT directory containing the meta information for the new web page and the new images. Thereupon the MOT decoder will remove the old web page and the outdated images and wait for the new version of web page and images. Receiving these objects can take a significant amount of time. Normally, the user will prefer to see the outdated version of a web page until the update is received. In order to avoid inconsistencies (mismatch between inline images and the web page itself), the service provider must ensure that updated inline images having a logically different content also use different object identifiers (MOT parameter 'ContentName').

4.3.5 Advanced Features of the MOT Directory Mode

The latest revision of the MOT standard [EN 301234] includes features to reduce the receiver footprint and to increase the receiver performance:

- It is recommended that the list of header information inside the MOT directory is sorted in ascending order of the MOT parameter 'ContentName' (mandatory parameter that identifies each MOT object). The MOT parameter 'Sorted-HeaderInformation' then tells the receiver that when it compares two versions of the MOT directory, it can efficiently compare two sorted lists (old and new MOT directory). Comparing an unsorted list requires many more resources.
- Earlier MOT directory mode implementations first reassembled the MOT directory. Processing and storage of MOT body segments started after the MOT directory was completely reassembled. These simple implementations unnecessarily discarded MOT body segments (those received before the MOT directory) and thus slowed down the start-up procedure. Since the start-up behaviour is critical for user acceptance, the MOT standard now indicates how MOT body segments can be collected and processed as soon as the MOT directory is reassembled. Such implementations can utilise all MOT body segments and they provide a much better performance on start-up and after changes to the MOT directory.

4.4 Standardised MOT User Applications

Several user applications based on the MOT Protocol have been defined and standardised. This allows data applications to run on data terminals from different manufacturers.

4.4.1 MOT Slideshow

An MOT Slideshow [TS 101499] consists of a sequence of slides (images) which should be presented to the user one after the other. The MOT Slideshow (like the Dynamic Labels) is a broadcaster controlled data application. Once this user application is started on the receiver, the broadcaster determines when which slide will be presented; user interaction is neither needed nor possible.

The MOT parameter TriggerTime within the header information of all slide objects determines when a slide will be shown on the display. It is possible to indicate an absolute time (e.g., 2008-02-17 12:07:40 UTC) or 'Now'. 'Now' indicates that the slide should be shown on the display as soon as the slide is completely received and decoded. An absolute trigger time must only be used if the broadcaster is able to ensure that the receiver has enough time to receive and decode the image before it has to be presented. It is also possible to send a slide without any trigger time and trigger it later on with an MOT header update that provides an absolute trigger time or 'Now'.

The current MOT Slideshow standard uses one single object buffer on receiver side. This reduces the receiver footprint, but implies that for instance only the last received slide can be triggered by an MOT header update. Image formats JPEG and PNG are supported and the image size is limited to 50 kbytes. The recommended screen resolution is 320x240 pixel.

The MOT Slideshow supports two image formats. JPEG compression is best suited for photographs. However, especially at a high compression ratio, JPEG is not well suited for embedded text or diagrams. PNG is a loss-less compression scheme and best suited for diagrams such as station logos or embedded text. In order to get the best quality at the smallest image size in bytes, it is beneficial to choose the image format depending on the content of the image.

The MOT Slideshow is currently revised and the Slideshow decoders compliant to the new revision will provide 450 kbytes of memory to store several slides for later triggering. It will then for instance be possible to upload several large slides ahead of a news block and trigger the slides within several seconds matching the current news. In the current MOT Slideshow specification, the time between the presentation of two slides can not be shorter than the transmission time of the slides. The new revision will also support animated images (APNG format) in a backward compatible way.

The feasibility and application of such an MOT Slideshow have been demonstrated at exhibitions [Hallier, 1994b] and in some pilot projects. Mobile DAB devices with colour screens now appear in the market, therefore the MOT Slideshow gets used more and more.

4.4.2 MOT Broadcast Web Site

The user application 'MOT Broadcast Web Site' [TS 101498] offers the user a web based data application with local interactivity. Such a service consists of linked objects (HTML files, images, animations, audio files, etc.) which are organised typically in a tree-like structure below a start page. The objects of such a service are broadcast cyclically in a 'data carousel'. To minimise the start-up time for a user who turns on his or her terminal or selects the 'Broadcast Web Site' application, the data objects of the application should be prioritised and broadcast accordingly. It is advisable to broadcast the start page very frequently, say every 15 . . . 30 seconds. The next level of objects – normally the objects which are directly referenced by the start page – should also be broadcast rather frequently, the next levels less frequently and so on. Also the priority of larger JPEG pictures could be handled independently of the HTML pages; this can further improve the response time. This priority management has to be done at the data multiplexer and is for example implemented in the data server of one supplier *(Fraunhofer IIS)*. Using such a priority management, the user may start to navigate through the upper level of information already received while the lower levels are received in the background.

It is possible to update single objects or parts of the application simply by transmitting the updated objects.

Using the MOT directory, the object management is quite easy: having received the MOT directory, the terminal can decide for example how to organise the storage, which objects should be received or discarded, etc. The updating and deletion of objects is explicitly controlled by the service provider broadcasting an update of the MOT directory. In addition, the integrity of the data set contained within the data carousel will be ensured by using the MOT directory. It is recommended to support the MOT caching functionality, especially if the receiver also has persistent memory. Start-up of the BWS on receiver side can be greatly improved by supporting the collection of MOT segments even before the MOT directory is completely received, see [EN 301234] for details.

Please note that even if an MOT Broadcast Web Site is in many aspects similar to web sites on the Internet, it is usually not possible to simply broadcast an existing Internet web site. The reason is that most Internet web sites are usually too large to be transported via a bandwidth-limited DAB channel in reasonable time. Mobile terminals will also usually have smaller screens and limited scrolling capabilities, therefore the content has to be specifically tailored towards the capability of the mobile device. Also the content has to be selected depending on the needs of a mobile user.

4.4.3 Electronic Programme Guide (EPG)

This user application also uses the MOT protocol. The EPG is explained in detail in section 5.5.

4.5 Text Based Services

4.5.1 Dynamic Label

The DAB Dynamic Label application enables the presentation of short text messages, the so called *dynamic labels*, on the receiver display. The basic functionality resembles that of 'RDS Radiotext' known from analogue FM broadcasts. The update of information is triggered by the broadcaster.

The provided content mainly refers to the currently running audio programme: song title and artist, name of the show and the announcer, phone numbers, short news flashes, etc. Due to the close link of this service to the audio programme they are carried within the audio stream's PAD channel. Almost every DAB (and DRM) programme today is accompanied by these short text strings, and basically all available digital radio receivers with a display present this kind of information.

The maximum length of a label string is 128 bytes. During transport, each label is divided into up to eight segments. Each segment carries up to 16 bytes of the label string.

The segments are transported in X-PAD data groups. These data groups carrying Dynamic Label segments are signalled with the X-PAD application types '2' (start of data group) and '3' (continuation of data group). The basic structure of an X-PAD data group carrying a Dynamic Label segment is shown in Figure 4.2.

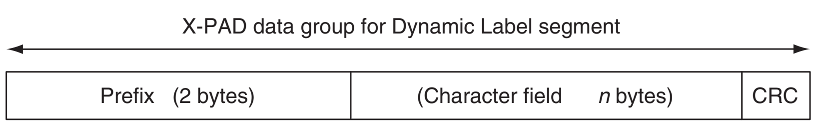

Figure 4.2 X-PAD data group carrying Dynamic Label

Each X-PAD data group carrying a Dynamic Label segment consists of a 16 bit prefix with control information, followed by a character field of up to 16 bytes as part of the complete label string. At its end the data group carries 2 bytes (16 bits) of CRC information for error detection.

The segments of a single Dynamic Label string are identified by the segment number carried in the prefix field. They can be transmitted in any order. Repetitions of identical segments can be used to increase the reception probability in case of high bit error rate in the transmission channel, but it is not allowed to interleave segments of different label strings.

Dynamic Labels use the character set 'Complete EBU Latin based repertoire' or ISO 10646. Similar to labels encoded in FIG type 1, support of the character set 'Complete EBU Latin based repertoire' is mandatory for the receiver. In addition, the character set ISO 10646 should be supported with both UTF-8 and UCS-2 encoding. The (sub-)set of characters that can actually be presented on the display will depend on the target market of the receiver.

For markets such as China it might be useful for a receiver to support the definition of characters via a pixel grid, to allow the presentation of very specific and broadcaster-defined characters. This is supported by FIG type 2 extension 7, see also section 2.5.2.

In order to allow for some minimum formatting capabilities for the labels and to optimise the label presentation for smaller displays, some special control characters are defined. Using these control characters, preferred word and line breaks as well as headline sections may be defined. In addition, a command to clear the display is provided. The additional control codes count as normal characters; therefore the maximum number of 128 displayable characters within one Dynamic Label string will be reduced by the number of control characters used within the label string. For more information regarding the coding and formatting of Dynamic Label strings, see section 5.4.4.

In 2007 the DAB Dynamic Label application was extended by two specifications: 'Dynamic Label Plus' and 'Intellitext'. Both definitions optionally enhance the existing DAB Dynamic Label application with specific functionality in a backward compatible way. The general limitations of the Dynamic Label service are respected. This refers to the maximum possible text length of 128 characters per label, and the minimum delay between label updates to give the user sufficient time for reading a label shown on screen.

'**Dynamic Label Plus**' [TS 102980] makes it possible to mark up parts of a message, and to indicate the content type of these parts, like song title and artist name. This enables a receiver for example to keep the current song title permanently on screen. The extension provides basically the same functionality as the 'RadioText Plus (RT+)' specification defined by the RDS Forum.

As an example, the DAB Dynamic Label string 'You are listening to "Yayayeyo" by The Splendids'. could be tagged to carry the title of the current song from index position 22 with a length of 8 characters, and the artist name starting at index 36 with a length of 13 characters.

Several types of content can be defined. The pre-defined type codes comprise music and audio description (like ID3 tags known from MP3 files), information items (like weather, lottery), programme description (like now, next, host), interactivity indicators (like phone or sms numbers), and special items. The list of defined type codes is part of the ETSI standard.

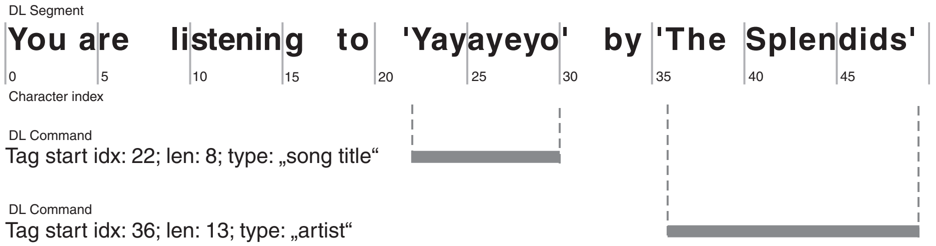

Figure 4.3 Example of Dynamic Label Tags

The tagging information is carried in a DL Plus Command message of type ‘DL Plus Tags’, immediately following the associated DAB Dynamic Label message. An Item Toggle Bit allows various pieces of information to be correctly assigned (spanning multiple DL Plus Command messages) per programme item (e.g. a song or show). The Item Running Bit can be used for example to indicate the temporary interruption of an item (e.g. a tagged show being briefly interrupted by news).

The ‘**Intellitext**’ extension to the DAB Dynamic Label service was finalized in 2007 [TS 102652]. It defines a specific text string format for dynamic label messages, which enables a receiver to make earlier received messages available through a multi-level selection structure.

Other than the Dynamic Label Plus application that requires new Dynamic Label commands (i.e., an extension of the transport protocol), Intellitext purely relies on a specific format of the visible text of a label string to enable this functionality. The general structure of an Intellitext compatible Dynamic Label message is ‘<menu> <submenu_index> - <submenu> <data_index> : <data> <time_to_live>’. Some of these elements are optional. For example, the Dynamic Label string ‘Football - Prem Lge Table[1]: 1. Chelsea 27 pts; 2. Spurs 18 pts; 3. Charlton 16 pts’ would be available to the listener through the Intellitext menu structure as three list items ‘1. Chelsea 27 pts’, ‘2. Spurs 18 pts’, and ‘3. Charlton 16 pts’ in the menu ‘Football’, submenu ‘Prem Lge Table’. The index value ‘[1]’ after the submenu name allows for a later update of specific items within the main menu. Using the <time to live> element at the end of an Intellitext compatible Dynamic Label string, the broadcaster can optionally influence the timeout value for any Intellitext message using ‘.’ (dot) characters: One ‘.’ character represents a timeout value of 24 hours, two ‘..’ indicate 12 hours, and three ‘. . .’ characters finally demand a maximum of 1 hour.

All Dynamic Label messages are visible to those DAB receivers not equipped with Intellitext. A receiver aware of Intellitext may still choose to show all Dynamic Labels in real-time on screen, while only those strings conforming to the Intellitext format would be stored in the menu structure for later access by the user.

4.5.2 Journaline

4.5.2.1 Introduction

The benefit of the straightforward DAB Dynamic Label user application (see 4.6.1) is its simplicity and subsequently its support in almost any modern radio receiver. However, the application is mainly intended to provide highly programme-related content for quick lookup on the display, like the current song title and artist. The content and its presentation order are pre-defined by the broadcaster. There is no possibility for the users to interactively and immediately access detailed information that is currently relevant to them, let alone access to information that may not even be programme related.

For that kind of interactive and on-demand access to information, another data application had originally been designed – the MOT Broadcast Website (see 4.4.2). This solution is very flexible. But at the same time it requires a relatively high bit rate to transport the content. The decoding, presentation and user interaction is demanding in terms of cache memory requirements, HTML parsing and rendering, required PC like graphical user interfaces (ideally with a pointing device), and so on. Early field trials in Germany showed that one of the most disturbing aspects for broadcasters and content generators was the fact that the web standards based application promised a predictability of rendering results that could not have been fulfilled– any radio receiver would present the carefully designed layout of the HTML pages in a different way.

Therefore a new data service, Journaline, was developed as part of the publicly funded project RadioMondo [Kilian, 2005]. That application should combine the benefits of simple text messages and interactively available information, and at the same time take into account the tough bandwidth restrictions of the new DRM broadcast standard.

Journaline provides the radio user with interactively accessible textual information. The typical use case is comparable to the teletext service on a TV set or an electronic magazine. The Journaline specification – originally developed with the tight bitrate restrictions of DRM in mind – was picked up by the WorldDMB forum as the new text based information service, and standardised in September 2007 through ETSI [TS 102979] for use in both DAB/DMB and DRM transmissions.

The data service is designed with simplicity and efficiency as primary goals. This involves all parts of the broadcast chain: Broadcasters can easily reuse existing data sources like RSS feeds and XML data. The content is encoded in a binary form and compressed to minimise the required transmission bandwidth. On receiver side both the decoder footprint in terms of CPU and memory, as well as the minimum required user interface functionality are very small.

This last aspect enables the integration of a Journaline decoder into the full range of digital radio receiver classes: from price sensitive bedside and kitchen radios to high-end multimedia devices with graphical displays. The application even supports special environments like car radios by providing speech hinting information for high quality text-to-speech playback.

4.5.2.2 Functionality

The concept of Journaline is based on the following requirements:

- provision of pure text information with worldwide applicability;
- interactive, immediate access by the user to all provided information;
- easy integration of existing information sources by the broadcaster (RSS, XML standards);

- efficient information encoding and transport to support any radio broadcast standard, including the narrow bandwidth DRM system;
- smallest possible decoder, memory, and user interface footprint in the receiver;
- optional support for text-to-speech playback of text information in receivers, interactivity (back channel establishment and linking to web URLs, e-mail, SMS, other Journaline pages, other DAB services, etc.), and geo tagging of information to enable location based services;
- easy future extensibility, maintaining full backward compatibility to existing decoders.

Figure 4.4 demonstrates the basic idea and functionality of the resulting Journaline application: Textual information is selected through a menu structure.

The immediate access to all available information is enabled by organising all service elements in a hierarchical tree structure (as shown in Figure 4.5). Menus allow for a quick selection of currently relevant topics and information.

Every menu and every text page is a self-describing and self-contained object. At no moment does the receiver need to build up or evaluate any internal object hierarchies or directories – one building block for low minimum memory requirements.

Hence, the decoder could even work without any cache memory: by waiting for the reception of the main menu and presenting it to the user, by then waiting for the selected target object and presenting it upon reception, and so on. Of course the availability of some cache memory greatly enhances the service responsiveness to the user.

Every object (menu or text) is efficiently encoded in JML (Journaline Markup Language), a binary representation of XML formatted content. Finally a compression scheme is applied to the plain textual information.

In addition, all aspects of the service are restricted to sensible maximum values, which should hardly ever be reached in a real-world Journaline service. This applies to object size, maximum object count per service, maximum number of selection targets per menu,

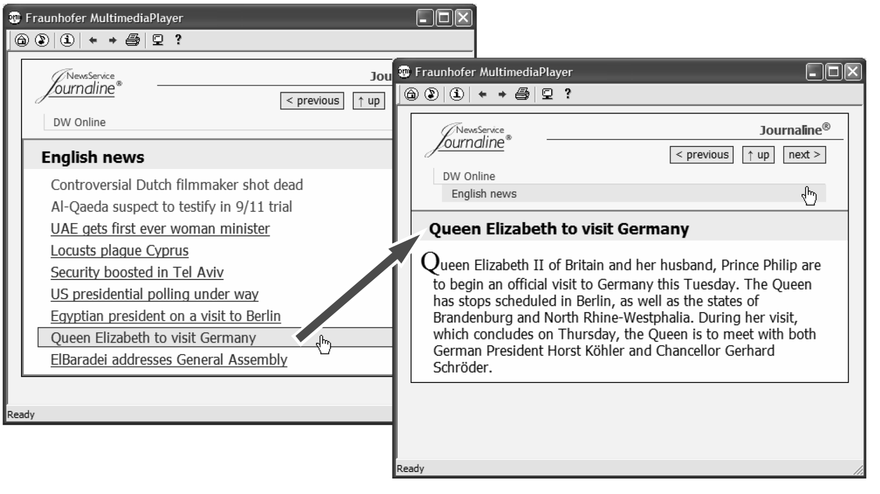

Figure 4.4 Navigating through Journaline content (Windows based decoder)

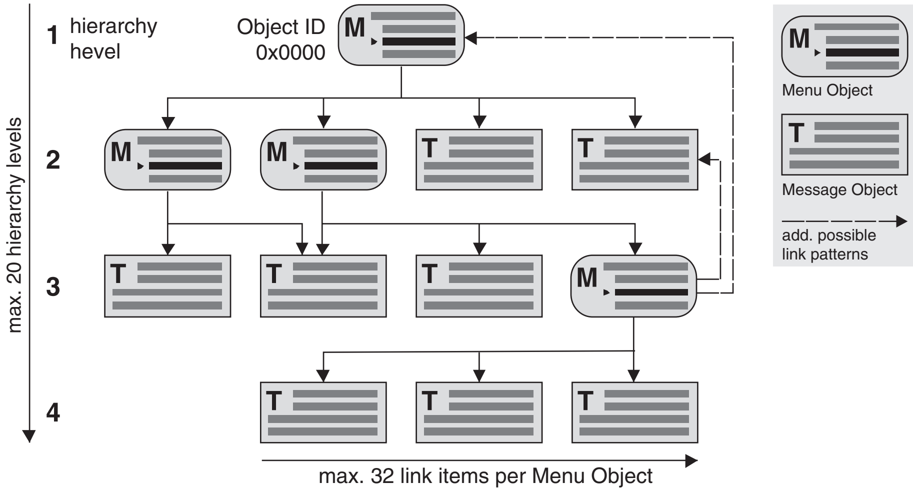

Figure 4.5 Hierarchical tree structure of menus and text pages

maximum number of menu hierarchy levels, and so on. These restrictions enable the efficient implementation of a Journaline decoder even on embedded and restricted microcontroller platforms, and thereby serve the goal of supporting all classes of digital radio receivers.

4.5.2.3 Types of Information Pages

Every information page (i.e. object) represents one menu or one text page. It is encoded and transmitted independently from any other object, and distinguished by a unique ID within the Journaline service.

Currently four types of information pages are defined. They will briefly be introduced below along with their individual features and behaviour rules. In general, the minimum requirement for a Journaline compliant receiver is to make all textual content accessible to the user. However, the Journaline specification does not define any strict or precise rendering requirements beyond the general rules stated below, to allow for a flexible optimisation of the text presentation for any receiver type.

'**Menus**' (see Figure 4.6) enable the user to select other (sub-) menus and text pages. They are composed of a title and a range of link targets, while marking the current

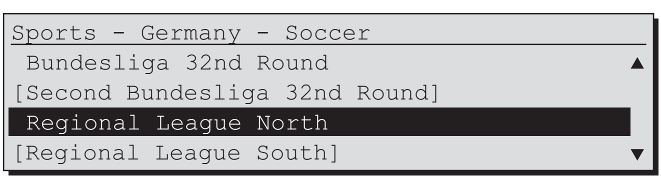

Figure 4.6 Example of an information page 'Menu'

selection position of the user. Optionally the cache status of a link target may be visually hinted – if the linked information page is already in the cache, the user can access it immediately without waiting for its next reception.

Upon reception of a new version of the currently presented ‘Menu’, it shall be updated immediately on the screen while maintaining the user’s current selection position.

A ‘**Plain Text Message**’ (see Figure 4.7) is composed of a title and the body text. The user can vertically scroll through the full text. If an updated version of the text is received, this may be indicated to the user, but no automatic update of the screen content should be performed.

Information pages of type ‘**List/Table**’ (see Figure 4.8) are especially helpful when it comes to presenting sports results or financial information like stock values. If an updated version of such a page is received while it is shown on screen, the display should be redrawn immediately while keeping the user’s current scroll position. So if the user has scrolled down to have their favourite sports club visible on screen during a match, they will just notice the scores go up over time.

A ‘**Ticker Message**’ (see Figure 4.9) is composed of a single title element without any body text. If possible, it should be completely visible on screen without any user

```
Germany tops EU service report (17:13)  ▲
Research by the European Union's
statistical arm Eurostat has placed
Germany top of the EU productivity
league in a range of services. The      ▼
```

Figure 4.7 Example of an information page ‘Plain Text Message’

```
Soccer - Bundesliga 32nd Round (16:15)
TSV 1860 - Cottbus      3:0            ▲
Dortmund - Nürnberg     4:1
Hertha - Bayern         3:6
Stuttgart - Bremen      0:1            ▼
```

Figure 4.8 Example of an information page ‘List/Table’

```
                                        ▲
The housing boom continued in the first
quarter - Home prices nationwide spiked
7 percent.
                                        ▼
```

Figure 4.9 Example of an information page ‘Ticker Message’ interaction. Other than plain text messages, ticker messages are automatically updated on screen as soon as a new version of the object is received. This functionality resembles that of DAB Dynamic Labels and DRM TextMessages – however, as part of the Journaline menu structure each ticker message page covers one specific topic. This feature enables services like news headline tickers (every ticker message optionally linked to a detailed information page), dynamic programme accompanying content (like song lyrics for Karaoke or closed captioning for editorial pieces), or video subtitles in various languages.

4.5.2.4 Formatting of Information Pages

All information pages are restricted to a maximum size of 4 kbytes (incl. 4 bytes MSC Data Group Header and CRC), independent of their type.

The Unicode representation format UTF-8 is used for all text character encoding. This enables a Journaline service to support any international script and character set like Chinese, Arabic right-to-left, Cyrillic, etc.

Figure 4.10 describes the format of each Journaline Object. The Standard Header carries the 16-bit Object ID, which uniquely identifies a Journaline Object within a Journaline service. The Object Type distinguishes Menus, Plain Text Messages, etc. The Compression Flag indicates whether the Content Section is compressed using a deflate mechanism. The Revision Index finally provides the Journaline decoder with a convenient way to quickly identify whether the content of a newly received Journaline Object was changed compared to the version currently kept in cache.

The Static Flag supports the bookmark/favourite feature of Journaline, which allows the user to store the link to a specific Journaline page for quick later access (e.g. by pressing the radio's station key). Many Journaline pages within a Journaline service (defined by their respective Object ID) carry random information. Therefore the

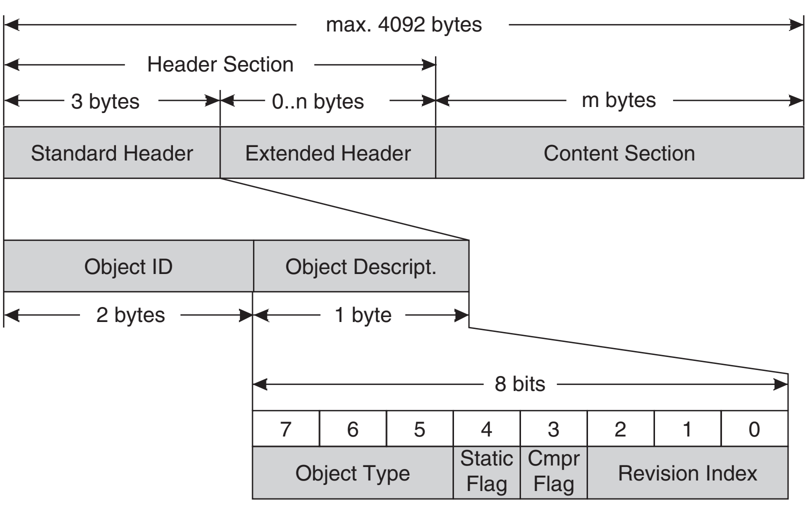

Figure 4.10 Journaline Object Format

bookmark feature should only be enabled for those pages that will also in future carry the same type of content (e.g. the local weather forecast), statically assigned to one specific Object ID. This is signalled to the receiver by enabling the Static Flag.

The length of the Extended Header section is signalled as part of the service component description. It is currently not used but will allow for a later extension of the Journaline specification. A Journaline decoder must at least be able to ignore this Extended Header section in case it is present.

4.5.2.5 Transmission of Journaline in DAB

Journaline supports two ways of organising the transmission of objects, and both ways can easily be mixed within one single Journaline service: carousel mode and real-time transmission.

Regular content is broadcast in the form of a data carousel. As soon as a receiver starts decoding a Journaline service, it collects all received information objects and stores them in the cache for immediate future access by the user – until the objects are updated or removed from the carousel. An updated object can be retransmitted immediately at any time, as there is no previous adjustment to any global directory structure required. Using individual priority classes for different parts of the Journaline service, regularly updated content can be broadcast more often than static content. And menus like the main menu page can be broadcast more often than other information pages to make the initial user access to the service as quick as possible.

Information pages in the form of ticker messages are updated regularly with changing content. In the case of an update, these ticker messages are broadcast immediately and thereby briefly ‘interrupt’ the running data carousel(s).

Each Journaline object is transmitted as one MSC Data Group, either in a packet mode sub-channel or within the X-PAD fields of a DAB/DMB audio/video sub-channel. It can be presented to the user as a stand-alone data service, or as a PAD data application accompanying an audio/video programme.

If the same Journaline service component is linked to multiple services, each link may indicate its specific page of entry. This way a broadcaster carrying multiple programmes in a DAB Multiplex could signal a Journaline service component carried in packet mode along with all its programmes as PAD. However, depending on which programme the user is currently listening to when accessing Journaline, they will be presented with a programme specific entry page. Nevertheless, the whole Journaline content is accessible for interactive exploration.

4.5.2.6 Current Status

Journaline has been available on-air as part of DRM transmissions for several years. Various international radio stations are providing Journaline information along with their radio programme as a service to their listeners. Some of these Journaline services consume as little as 200 to 600 bits per second. DAB/DMB transmissions of Journaline services started in 2007 with the availability of professional grade broadcast encoder equipment.

This platform-spanning availability of Journaline is appreciated by receiver implementers, including car manufacturers, as it eliminates the need for individual platform specific implementations of data decoders and user interfaces.

Professional Journaline encoder solutions are available for various broadcast platforms. Journaline encoder modules are part of several commercially available broadcast server systems, including those products based on the Fraunhofer ContentServer line.

Journaline decoder modules for integration into receivers are available for both open-source projects under GPL as well as for commercial grade applications. Figures 4.11 and 4.12 visualise some examples of Journaline decoder implementations for various platforms. Car manufacturers have presented their own DAB data service developments based on the Journaline user application.

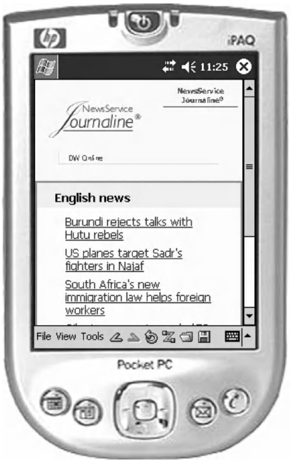

Figure 4.11 Journaline decoder on Windows CE

Figure 4.12 Journaline decoder on a microcontroller platform

4.6 Traffic Information Services and Navigation Aids

In this section two approaches to traffic and travel information (TTI) services and support for differential GPS based on DAB are described. For a broader view on travel information broadcasting see [Kopitz, 1999].

4.6.1 Traffic Message Channel (TMC)

Traffic Message Channel (TMC) is a basic data application consisting of digitally coded traffic messages, originally defined for RDS (Radio Data System, see [EN 50067]) services in FM broadcast. All messages are coded according to the 'Alert C' Protocol and carry the following information:

- characteristics of the event: detailed information describing the traffic problem or weather situation;
- location: area where the problem occurred;
- severity of the event;
- duration: estimated duration of the event;
- alternative route.

All this information is given in numeric code values which refer to the various tables (event, location, direction, duration, etc.) of the Alert C Protocol; these code values are assembled in 'groups'. Each group consists of up to 37 bits of information.

Additional information can be given with special messages, e.g. a more precise definition of the event's location or a description of an exceptional event not covered by the standard message table.

The handling of the messages (validation, time-out processing, priority management, etc.) as well as the presentation to the user could be done by an external decoder. The presentation can be done either by an alphanumeric display or by a speech synthesiser which generates spoken messages.

In DAB, the TMC messages are carried in the FIDC in FIG 5/1. Several messages may be assembled within one FIG 5/1 data field. A detailed specification how the TMC user messages, system messages, tuning information and encryption administration data is mapped from RDS groups to FIG 5/1 and how the TMC service should be signalled in FIG 0/2 and FIG 0/13 was developed within the DIAMOND project and is now standardised in [TS 102 368].

A traffic information service based on TMC carried on DAB was first demonstrated in 1995 when an external TMC decoder box which was originally designed for FM-RDS TMC was connected to the serial port of a DAB data decoder. Since the infrastructure for collecting the traffic information data was rather poor at that time, the service was for demonstration only. In addition, the real-time insertion of the TMC messages into the FIC was rather complicated. During the last few years this infrastructure has improved significantly and therefore DAB-TMC is now used in several countries for setting up a useful traffic information service. Also the capabilities of the multiplexers and the data inserters have been improved with the introduction of the 'Service Transport Interface' (STI) which allows easy insertion of TMC messages assembled in FIG 5/1 (see section 5.3.3).

4.6.2 TPEG

TPEG stands for the Transport Protocol Experts Group of the EBU which developed an end-user-oriented application for the delivery of traffic and travel information (TTI) [Marks, 2000]. At the end of 2007, the TISA organisation (Traveller Information Services Association) took over the activities undertaken by the TMC Forum, the TPEG Forum and the German Mobile.Info project.

The TPEG specification defines the core protocol and the service layer. Service providers may operate services which use one or more delivery technologies (e.g. Internet, DAB) from one message generation process. The application layer is scalable and allows a range of receiver types to be used simultaneously – from simple receivers with only a small alphanumeric display up to integrated navigation systems. One of the key characteristics of TPEG is the fact that there is no need for the end user to have a location database before using the service. The location database is derived from digital maps at the service provider side and the location information will be transported to the end user embedded in the TPEG application. Another important characteristic is language independence. Messages are not carried as free text, but as numerical values the receiver translates to a message in a specific language. Therefore, for instance, a French car driver can get the traffic information in French, even when travelling through Germany.

Several TPEG applications exist:

- RTM (Road Traffic Messages): road based travel disruption information;
- PTI (Public Transport Information): comprehensive public transport information;
- PKI (Parking Information): parking space availability and pricing;
- CTT (Congestion and Travel Time): information about congestions and the time it takes to travel on a network;
- TEC (Traffic Event Compact): describes events and their effect on the road network. It is optimised for use by navigation systems;
- WEA (Weather information): information about weather.

At this time, only the first two TPEG applications are accepted standards, the others are awaiting final approval.

TPEG is organised as a self-contained stream of data which contains all messages as well as provisions for synchronisation and error detection. A TPEG stream should be broadcast over DAB using the 'Transparent Data Channel' (TDC) specification [TR 101759], see section 4.7. The Mobile.Info project (now part of TISA) requires the use of TDC with data groups, each TPEG frame is mapped to one MSC data group.

First implementations of TPEG use an external application decoder connected to a DAB data receiver and have been available as prototypes since the beginning of 2001.

4.6.3 Differential GPS

For navigation purposes the satellite-based 'Global Positioning System' (GPS) is widely used. A miniaturised satellite receiver receives information from five or more satellites and calculates from the data sent by the satellites and the receiving delay its current position.

GPS was originally designed for military applications and the accuracy for civil uses was artificially deteriorated. The artificial noise added to the GPS signal was removed several years ago. Since then GPS is accurate enough for most applications.

In order to increase the positioning accuracy for high precision applications, Differential GPS (DGPS) can be used. The basic principle is to use a stationary reference receiver with a well-known fixed location. This reference receiver compares its known location with the location calculated from the satellite data. The difference between these two locations is transmitted in the standardised 'RTCM 2.1' format via a broadcast channel to mobile GPS receivers. These receivers may then correct the position calculated from the satellite data by the additional information and so reduce the positioning error.

DGPS data is broadcast using the Transparent Data Channel (TDC, see section 4.7) feature of DAB. The use of TDC with data groups is recommended.

4.7 Other Data Transmission Mechanisms

4.7.1 Transparent Data Channel

The Transparent Data Channel (TDC) [TR 101759] provides DAB with the ability to transport any type of data using the following DAB data channels:

- packet mode data service component;
- stream mode data service component;
- PAD channel inside an audio service component.

TDC data inserted into a stream mode data service component must be synchronously provided at exactly the rate of the stream mode sub-channel. Since all DAB sub-channels are a multiple of 8 kbps, the application using the TDC must either have a bit-rate that is a multiple of 8 kbps, or it must use padding to fill up the TDC bit-rate to the next multiple of 8 kbps.

TDC in a packet mode data service component or the PAD channel of an audio service component are asynchronous. Data can be provided at any bit-rate up to a maximum bit-rate. It is possible to use the TDC with or without MSC data groups. Communication systems use blocks of bytes at their lower protocol layers. To map these protocols to DAB, usually a block of data (e.g., a so called 'frame') is mapped to an MSC data group. Therefore many non-DAB protocols can be carried via DAB by using MSC data groups. For completeness, it is also possible to carry a sequence of bytes (TDC without data groups). This is similar to a pipe where bytes inserted at the transmission side are output again at the receiver side. However, reception errors may cause bytes to be corrupted or lost. Therefore the receiver must have synchronisation methods to re-synchronise to such streams, individually for each proprietary framing. This adds unnecessary complexity to a receiver compared to TDC with data groups. For all applications that provide data in blocks of bytes ('frames'), the use of TDC with data groups is recommended.

Typical applications using the TDC protocol are Journaline (see section 4.5.2), DGPS or TPEG.

4.7.2 IP-Tunnelling

4.7.2.1 General Principles

There are two different ways to distribute data across DAB. The MOT protocol, developed for DAB applications, enables the transport of multimedia objects as described in section 4.3. However, the Internet Protocol (IP) is the basic transport mechanism that most computer networks are using. Clients and servers use IP addresses to communicate with each other. The Transmission Control Protocol (TCP), [RFC 793] and the User Datagram Protocol (UDP), [RFC 761] are subsets of the widely used IP protocol. It is also possible to use other protocols than UDP or TCP on the Internet transport layer. The multimedia objects are split into IP packets on the transmission side. Depending on the client application, incoming packets are reassembled or handed over directly to the application (e.g. Media Player). The advantage of this technology is that it is possible to transmit the multimedia content via different distribution mediums without having to convert between different transmission protocols.

An important feature that IP supports are streaming data applications, in contrast to the MOT protocol which does not support streaming data applications since it was designed for the transmission of whole data objects. The MOT protocol transmits whole data objects at once whereas the IP datacasting protocol is defined for streaming variable sized data chunks.

To enable the transport of Internet Protocol (IP) packets [RFC 791] in a DAB data channel, an appropriate mapping of IP packets on the protocol structure of DAB packet mode was developed. This is called IP datacasting. Essentially, IP packets are 'tunnelled' through the DAB protocol stack. The distribution of IP packets in a DAB data channel is unaffected by their contents.

The available IP data rate depends on the DAB multiplex configuration as well as the error protection profile used. The maximum data rate for an IP datacasting service is limited by the data rate of an entire DAB ensemble, e.g. 1152 kbps, with an equal error protection of protection level 3 (see Chapter 2).

The IP tunnelling standard [ES 201735] describes a transport mechanism for the distribution of IP services via DAB. The encapsulated Internet Protocol is used as a common network layer. Given DAB broadcast properties, IP tunnelling is limited to unidirectional communication. If bi-directional communication is necessary, TCP via UMTS, GSM or GPRS is used as a return channel. The corresponding standards are [TS 101736], [TS 101737].

4.7.2.2 IP-Tunnelling-Standard

The ETSI Standard [ES 201735], named 'Internet Protocol datagram tunnelling', describes the transport of IP packets in a DAB data channel. The tunnelling of IP packets via DAB is shown in Figure 4.13. The IP packets are tunnelled through a DAB packet mode service component. This is done by encapsulating the IP packets in a Main Service Channel (MSC) data group on packet mode transport level (see section 2.3.3.3).

The payload of one MSC data group defines exactly one IP packet. According to the MSC data group data field specification [EN 300401] the maximum size cannot exceed

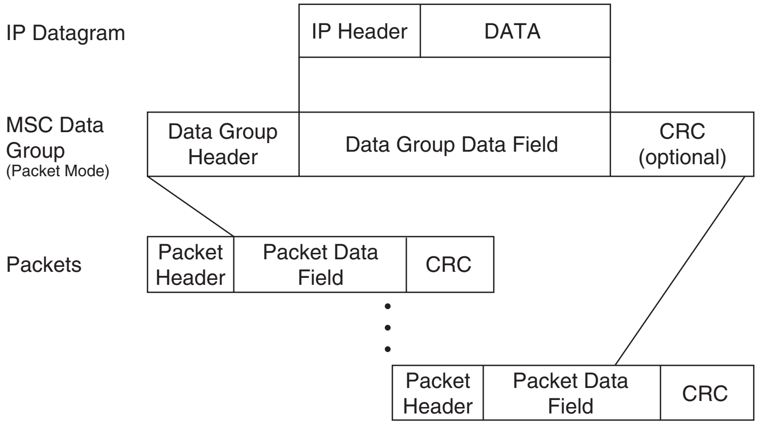

Figure 4.13 Encapsulation of IP packets for DAB transport

8191 bytes. If an IP packet exceeds this maximum value, the data payload will be repackaged into smaller datagrams. [RFC 791] specifies the way a packet is fragmented. If fragmentation occurs the payload remains unchanged.

The signalling of encapsulated IP packets as well as the identification of IP packet services is defined in the Fast Information Channel, see section 4.2. The corresponding Data Service Component Type (DSCTy), which indicates the type of the data, is called 'Embedded IP packets'.

The DAB broadcast channel provides a guaranteed quality of service and bandwidth to all services carried within the multiplex. For data applications different error protection levels may be used. The reception reliability can be increased by repeating data groups on the DAB packet mode transport level. Most IP implementations do not work correctly receiving multiple copies of the same IP datagram and therefore the duplications of identical IP datagrams must be eliminated by the DAB receiver/data decoder before forwarding the IP datagrams to the application.

The IP as well as the MAC Address is configured during the installation by the local network administrator. The configured IP and MAC Address has no influence on received content. The secure addressing of different user groups can be realised with a conditional access system within the receiver or on the IP application level.

Unlike the DAB native MOT protocol the encapsulation of IP packets into the DAB protocol requires additional overhead due to its header information. IP packets carry different additional information into the header, e.g. destination address, type of service, time to live, etc. which are needed for distribution into networks.

From the viewpoint of the information provider the distribution of data via IP packets simplifies the broadcast architecture by avoiding the need to convert content into a DAB specific protocol. Unlike MOT the IP encapsulation enables real time streaming like digital TV as well as other applications. For further details of IP based multimedia services see section 9.3.

5

Provision of Services

5.1 The DAB Service Landscape

The introduction of DAB took place during a period of great change in the radio world, in general. Following the digitalisation of studios and consumer products, now the means of transmission to living rooms and cars are going to be digital. Furthermore, in addition to speech and music, until now the only forms of presentation to audiences, new kinds of multimedia applications will be introduced with the Internet, digital broadcasting and new generations of mobile communication systems.

The migration from the existing to necessary new infrastructures for the provision of multimedia services is a major challenge for content and service providers. Depending on the intended amounts of investments, the appearance of service providers in DAB will be very variable. Most of the radio stations are starting with a simulcast of existing analogue audio services. Some stations are also providing data services for a better general service for the audience. However, few providers are launching new audio services. In all these cases, the accessibility of listeners plays a decisive role. As a result of the normally slow appearance of new technologies, broadcasters have to exploit synergetic aspects of other media during the generation of a service. One time generating – several times distributing.

The intention of this chapter is to describe the service generation, the adaptation of the DAB system to interfaces of other media and the gradual integration of the necessary infrastructures in broadcasting houses. Most of the examples described are taken from the experiences of German broadcasters.

5.1.1 Structure of DAB Service Organisation

In the past, only a simple structure of responsibility was needed in a broadcast system: the programme provider (editor, supervisor or similar) was responsible for the production of the final content and form of a radio programme and for the studio output into the broadcast chain for distribution and transmission to the customer, without any further changes in content or quality. This was true for AM and FM/ VHF radio, and partly also for the first digital radio services (DSR, ADR, etc.).

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

The very complex structure of the contents of DAB (i.e. audio programmes with different formats and quality levels, programme associated data, independent data services, programme service information) and its grades of freedom to change dynamically several parameters of the multiplex requires a more diverse responsibility for managing the final content, form and quality of a DAB service by

- the programme provider;
- the service provider;
- the ensemble provider.

Figure 5.1 shows a simplified structure of the processes of service provision for a DAB service ensemble.

5.1.1.1 Programme Provider

For audio programmes, the programme provider may have the same tasks concerning content as in a conventional audio broadcast service, but in addition will also provide some other programme-related information (Programme-associated Data PAD) such as DRC (Dynamic Range Control) data, M/S (Music/Speech) flag (see section 3.5), programme SI (Service Information) such as dynamic label, language assignment, programme type assignment, announcement type (see section 2.5.), and so on. All these data are of a dynamic type, which means that they will change in a close time relationship to the current programme content.

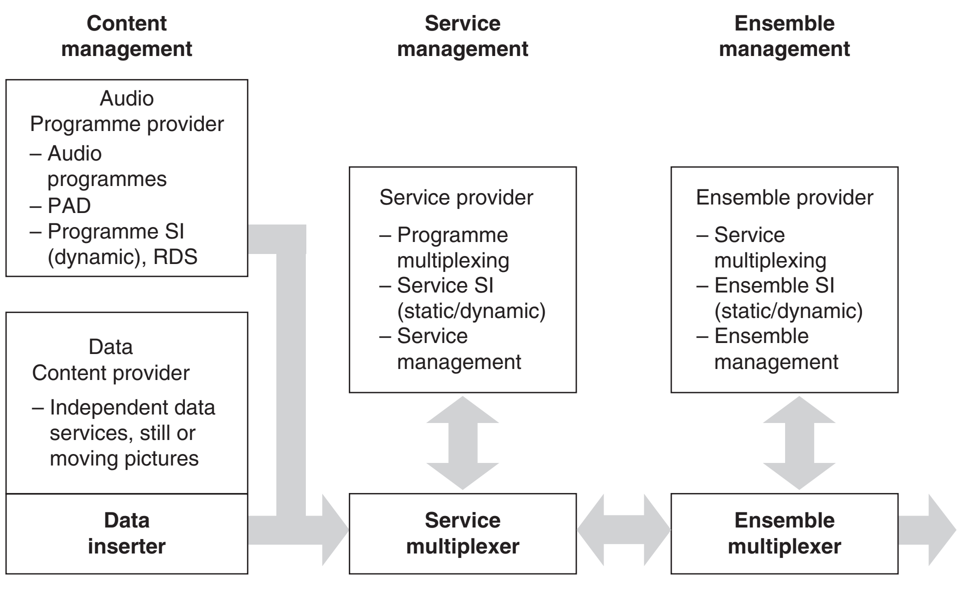

Figure 5.1 Structure of the management process of DAB services

In a similar manner the programme (or content) provider for independent data services will manage the contribution to a certain DAB service. This can be located apart from the broadcast studio. The audio coding process itself can be allocated either in the broadcasting operating system or in the service multiplexer, which is the central component of the service management process.

5.1.1.2 Service Provider

The programme output is passed to a service provider, and a series of programmes is then assembled into a programme service which may also contain other components in addition to the audio programmes, for instance SI similar to RDS for FM/VHF broadcasting, or new, independent data services.

The SI can be of a static type (such as programme service label, programme type, regional identification, or of a dynamic type, such as programme type preview, programme number (see section 2.5.). Also new data services (e.g. traffic information or still pictures for different purposes) which are completely independent of the main (audio) service can be assembled in a service multiplex. The service provider's task is to define the organisation of the service (i.e. the description of each service component and how they link together), see [BPN 007].

Another task of the service provider will be the management of the service multiplex, such as reconfiguration requests or changes in audio parameters, such as data rate, mono/stereo and so on. Each digital audio bit stream has an individual data rate and audio type (mono, stereo, Joint Stereo, etc.).

5.1.1.3 Ensemble Provider

Finally, a number of programme services (about four to seven), including the other components of the Main Service Channel (MSC) and the Fast Information Channel (FIC), are multiplexed by the ensemble provider into one DAB service ensemble and finally fed to the DAB transmitter(s). The ensemble provider is also responsible for particular static and dynamic SI as shown in Figure 5.1. In general, this task will be managed by a separate organisation apart from that providing programme content or services.

More details of selected aspects of the provision of services and the necessary infrastructure of the production processes for DAB are given in the following sections. As these technologies are still not very common at the time of writing, the descriptions may be understood as an overview by means of examples or proposals for future implementation. See also Chapters 2 (system concept), 3 (audio services and applications), and 4 (data services) for details of the service components mentioned. Chapter 6 (collection and distribution networks) details the functionality of the service and ensemble management processes and the interfaces between them.

5.1.2 DAB Main Services

This section explains the structure of DAB services in some detail, for a better understanding of the specific abstracts and coherence used later in the text.

Audio services, data services and the necessary configuration information are assembled as a so-called DAB multiplex. This multiplex is configurable and provides a data rate of approx. 1.2 Mbps. It is designed to transport an FIC and an MSC including the individual sub-channels which transport the audio and data services.

The DAB MSC provides mainly four service transport mechanisms:

- Service Information (SI) and TMC (within FIDC);
- Sub-channel for audio services;
- Programme-associated Data (PAD) carried within audio services;
- Packet mode data carried in individual sub-channels for data services.

An example of a multiplex configuration with multiple audio services used in practice is shown in Table 5.1.

5.1.2.1 Example of a Typical DAB Ensemble

The DAB system is based on a multiplex that can assign different bit rates to each individual programme and that also allows dynamic reconfiguration. The latter could be used to split one high-quality programme into two of lower quality or vice versa.

The actual protection ratio and bit-rate for each individual programme depend on the codec used, the material for that programme and the intended audience. Although it would be desirable from a quality point of view to always use high bit rates, this is not economical. More programmes at acceptable quality are a better deal than fewer programmes at guaranteed CD quality.

Table 3.1 (see section 3.3.4) provides possible values taken from [BPN 007]. Table 5.1(a) lists the type of programme with recommended bit rates and channel configurations for certain quality levels. We can see a fairly wide range of bit rates for all different quality levels.

Table 5.1 Examples of typical DAB ensembles

(a) Sets of suitable parameters for different materials and quality levels of audio services

Programme Type	Format*	Quality Level	Sample Rate kHz	Protection Level	Bit Rate kbps
Music/Speech	1/0	Broadcast quality	48	UEP_2 or 3	112 to 160
Music/Speech	2/0	Broadcast quality	48	UEP_2 or 3	128 to 224
Music/Speech	3/2	Broadcast quality	48	UEP_2 or 3	384 to 640
Speech only	1/0	Acceptable quality	24 or 48	UEP_3	64 to 112
Messages	1/0	Intelligible message	24 or 48	UEP_4	32 or 64
Data	None			UEP_4	32 to 64

*Format codes: 1/0 mono
2/0 two-channel stereo
3/2 multichannel

Table 5.1 *(continued)*

(b) DAB multiplex configuration for Ensemble Bavaria, Germany (Channel 12D)

No.	Service	Service ID (Hex)	Audio Mode	Bit rate (kbps)	SubCh ID (Hex)	SubCh Size (Cu)	Error Prot.
1	Bayern 2 plus	D312	Joint Stereo	128	0×14	96	UEP3
2	Bayern 4 Klassik	D314	Stereo	192	0×11	140	UEP3
3	Bayern Plus	D316	Joint Stereo	128	$0 \times 0F$	96	UEP3
4	BR Verkehr	D31E	Mono	64	0×13	48	UEP3
5	On3Radio	D317	Joint Stereo	128	0×10	96	UEP3
6	B5 plus	D315	Mono	96	0×12	70	UEP3
7	Rock Antenne	D319	Joint Stereo	192	0×07	140	UEP3
8	Radio Galaxy	D31B	Joint Stereo	160	0×06	116	UEP3
9	Mobil Dat Bayern	E0D01008	none		$0 \times 0B$		EEP3A
10	EPG Bayern	E0D41008	none	total	$0 \times 0B$	total	EEP3A
11	Mobile Info	E0D01007	none	64	$0 \times 0B$	48	EEP3A
12	TPEG-BR-Test	E0D21008	none		$0 \times 0B$		EEP3A

Note: UEP = Unequal Error Protection level, see section 2.3.3.

The coding mode (stereo or Joint Stereo) is not included in this table, but experience shows that the use of Joint Stereo coding can reduce the required bit rate. Again the existing codec implementations differ significantly, and the actual audio quality therefore does so too.

A typical ensemble, as once broadcast in Bavaria, Germany, is shown in Table 5.1(b). The bit rates reflect the development that has taken place over the years in the encoders used. They are slightly lower than given in [BPN 007]. Also joint coding is employed for all stereo programmes.

5.1.2.2 Sub-channel for Audio Services

The sub-channels allow audio transmission according to the DAB family audio standards: the basic DAB standards MPEG-1 [TS 11172] or MPEG-2 [IS 13838] and the related new standards DAB+ [TS 102563] and DMB [TS 102428]. The transport capacities are ordered from content providers at the network providers in the form of Capacity Units (CU). The number of CUs defines the audio bit rate or packet mode bit rate at the chosen error protection level (i.e. 140 CU; protection level 3 = 192 kbps). For considerations of service variety and audio quality the transport of approx. six to seven audio services per multiplex is possible (see also Table 5.1(b) above).

5.1.2.3 Programme-associated Data (PAD)

Parts of the audio sub-channel capacities can be used for the transmission of ancillary data. The information is transported synchronously within the MPEG-1 audio bit stream. The PAD comprises a Fixed PAD (F-PAD) control channel with a data rate of 0.7 kbps and an optional Extended PAD (X-PAD) transport channel with capacities up to 64 kbps. Most of the actually offered audio encoders have a restriction of 16 kbps.

The following information can for instance be transported in PAD:

- F-PAD:
  - Size of the extended data transport channel (X-PAD);
  - Control information for different listener situations DRC, M/S flag, see Chapter 3;
  - Ordering information (European Article Number, EAN, International Standard Recording Code, ISRC);
  - In-house Data.
- X-PAD:
  - Dynamic label (Radio text), Dynamic label plus;
  - Multimedia Object Transfer (MOT) protocol, for details see Chapter 4;
  - TPEG (within Transparent Data Channel, TDC);

Owing to the necessary time synchronisation of audio signals and the associated data, the PAD insertion should be operated under the responsibility of the content providers.

As a result of the synchronous transport of audio and PAD in a common bit stream, the following problem arises. During the audio encoding process the available bit rate is determined from the selected audio data, determining the PAD data rate. In cases of low audio bit rates, for instance 96 kbps, and a maximum PAD data rate of 16 kbps, a reduction in audio quality may occur. The remedy is a sensible configuration of the capacities for audio and PAD data.

5.1.2.4 Packet Mode Data

Besides the audio services, additional sub-channels can be configured in the MSC for the transport of packet mode data services. Under the primary aim of covering the existing analogue radio market in DAB during introduction of the system, capacities of approx. 64 kbps for packet mode data services were realistic. With additional DAB frequencies in the future, higher data rates will be feasible for multimedia or telematic services. The packet mode data are to be inserted either at the locations of the service providers (into the service multiplexer) or at the network providers (into the ensemble multiplexer).

5.1.3 Data Services

5.1.3.1 Dynamic Label/Dynamic Label plus

The Dynamic Label format transports transparent text information and control characters with a length up to 128 characters within the PAD channel. The service can be easily presented with alphanumeric text displays and thereby readily realised with cheap DAB receivers. The receiver supports the presentation of text according to the implemented display type, that is 32 characters per two lines or 64 characters per four lines. The first receivers with possibilities for incremental or scrolling functions are now available on the market. The broadcasters are responsible for a sensible service duration of single Dynamic Labels for text presentations in moving cars, see also section 5.4.4. The Dynamic label plus feature is backwards compatible to Dynamic label.

5.1.3.2 Multimedia Object Transport (MOT) Protocol

The MOT protocol allows the standardised transport of audio visual information, such as still pictures and HTML content. The use of MOT is similarly possible in the PAD and packet mode. Three applications of the MOT protocol are used in DAB: the services Broadcast Website, BWS [TS 101498], Slideshow, SLS [TS 101499], and Electronic Programme Guide, EPG. For details see sections 4.4.2, 4.4.1 and 5.5, respectively.

5.1.3.3 Broadcast Web Site (BWS)

The BWS is a local interactive service, where the user selects the information already received with a browser. This form of a 'radio web' is based on the mark-up language HTML. Besides the application, profile types are also fixed in the standard. The profile types rule the technical requirements of the presentation platform. Two profiles are associated with the BWS service. One is for services at integrated DAB data receivers, that is for car PCs or navigation systems, with a display resolution of 1/4 VGA (320×240 pixels), HTML ver.3.2 and a storage capacity at least of 256 Kbytes. The second profile allows a non-restricted service presentation on PC platforms. Supporting the first profile has resulted in a larger accessibility for users, because the 1/4 VGA profile can be received at PC platforms as well as at integrated receivers. For more details see also section 4.4.

5.1.3.4 Slide Show (SLS)

This second application describes, on the basis of JPEG or PNG (Portable Network Graphics) files, sequences of still pictures. The order and presentation time of this service are generated by the provider. This service provides no local interactivity to the user. The transmission time depends primarily on the file sizes of the pictures and the PAD data rate (see Table 5.2).

Table 5.2 Transmission time of MOT services (examples)

Content	Number of Files	File Size	Transmission Time (Data rate = 16 kbps)	
			PAD	Packet Mode
CD-Cover (JPEG) Resolution 320×240	1	14 kbytes	7 s	7 s
CD-Cover (JPEG) Resolution 640×480	1	42 kbytes	22 s	22 s
HTML file Text only	1	1 kbyte	0.5 s	0.5 s
HTML files	37	129 kbytes	1.10 min	1.05 min

5.2 Use of Existing Infrastructures

The radio world is going to be digital. This statement is valid for radio waves as well as for audio signal recording and transmission and the work flows in the broadcasting houses. The introduction of computer-aided radio has lead to more efficient and simplified work flows. Computer modules connected over digital networks satisfy the requirements at a higher rate during information processing, easy format checks and faster programme production. The usage of computer-aided radio technologies provides a ready source for new services.

5.2.1 Broadcasting Operation Systems

Broadcasting operation systems are the technical core of the audio production of every modern radio station. The systems include function modules for capturing, archiving, scheduling and broadcasting on-air. The minimisation of the complexity of broadcasting operation systems allows broadcasters to launch new programme formats with less effort.

Audio takes are captured data reduced in the MPEG-1 audio format. In order to avoid reductions of audio quality, in cases of encoder cascading, the archived data rate should be chosen with the highest value (i.e. 384 kbps). During the archiving process, additional text information, like title, track and artist, is also captured. This information can easily be used as ancillary data for PAD information. The broadcasting operation systems work in two different modes: full automatic or with moderation in a semi-assistant mode.

Editors arrange the stored audio information in daily play lists. During the runtime of the play lists, the stored audio takes will be replayed by MPEG PC cards (see also Figure 5.2).

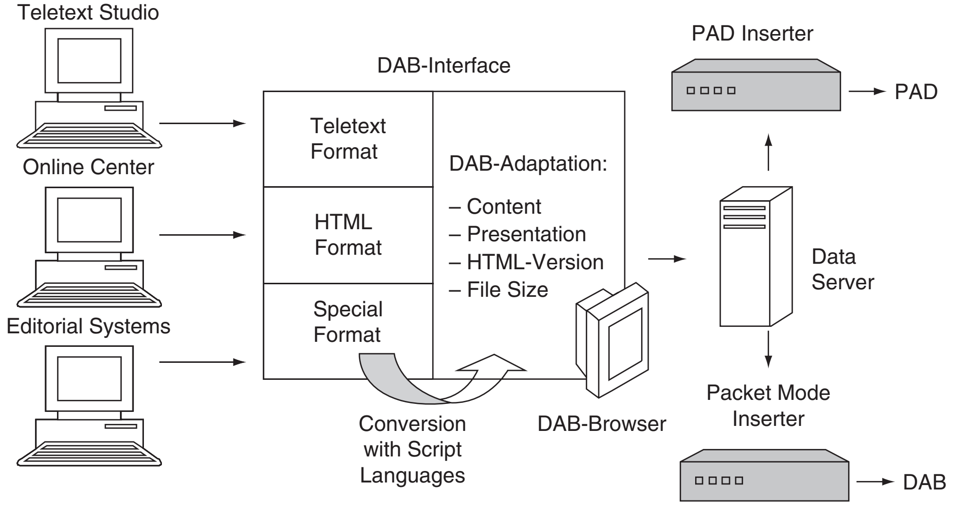

Figure 5.2 Editorial systems and DAB data interfaces

5.2.2 Editorial Systems

The editorial tools based on computer systems and servers provide large electronic information resources for the daily work of broadcasters. On completion of contributions, those ready for post-production are passed to data services.

In this context helpful script languages such as PHP3 (Hypertext Preprocessor), PERL and JAVA are employed. Figure 5.2 shows an arrangement of an editorial system with the necessary DAB interfaces.

The resulting data files are transported via local area networks to the connected data servers. Internet template tools are very popular today in on-line service centres. With these text template systems, the service design can be made with pre-defined text fields which have to be completed by the editors. This process allows one to individually generate content for specific applications such as the Internet, DAB Broadcast Web Site or DAB Dynamic Label headlines.

5.2.2.1 On-line Service Centre

The optimal information source to provide DAB data services is on-line service centres. With the success of the World-Wide Web and the increasing numbers of connected users, many broadcasters have established on-line service centres on the Internet. The on-line editors are experienced and trained in the use of modern authoring tools.

An example of on-line information, which is transported in a PAD channel, is the DAB news service of '*B5aktuell*' in Bavaria. Actual news and background information may be extracted from the on-line content in a multimedia database. Additionally, the HTML header tags of these files provide basic information for text services using the Dynamic label feature.

5.2.2.2 Teletext

The teletext service, which was introduced during the 1980s in European countries, is one of the most popular ancillary services of broadcasting systems. Owing to the increasing numbers of teletext users, broadcasters established their own teletext offices in-house. New TV stations are merging teletext and on-line centres into one common division.

Teletext is primarily the responsibility of TV editors, but if radio managers are convinced about the vision of a 'mobile teletext' over DAB, the teletext system can be used as an easy and economic data source. Such a service has been running in Bavaria, Germany, for several years. The DAB network provider uses a PC-based TV-receiver card with an integrated teletext decoder to provide a packet mode data service. The received teletext files of the *Bayerischer Rundfunk* are composed with the script language PERL into a new DAB Broadcast Web Site service. This service provides information on traffic, news, sport, business and the flight schedules of Bavarian airports.

5.3 Need for New Infrastructure

The previous section described adaptations to existing infrastructures. In most cases the requirements of programme editors can be satisfied. Sometimes such approaches lead only to spot solutions, which are limited for future expansion. In this case completely new planning is necessary. The present section shows current developments for a structured management of text, service information and multimedia content.

5.3.1 Management of Text and Service Data

With the introduction of ancillary data within the Radio Data System (RDS) for FM radio, Astra Digital Radio (ADR) and DAB in Germany, a new need for a programme-oriented administration of ancillary data arose from German broadcasters. One of the main objectives was the provision of uniform data to the data inserter. The result of this requirement was the concept of the Ancillary Data Manager and the in-house data protocol called 'Funkhaustelegramm'.

5.3.1.1 Ancillary Data Manager

The PC-based software Ancillary Data Manager was developed for the accumulation and distribution of ancillary data for a single radio service. The system concentrates, controls and provides information collected from sources like broadcasting operation systems, editorial modules and studio contacts. The concept described appears under different product names in Germany (see Figure 5.3).

Module Broadcasting Operation System. A text file interface to broadcasting operation systems allows synchronised adaptation of text information to the current audio item. The file interface stores information (i.e. title, artist, ordering information) after modification or new starts of audio takes. The contents of the file will be deleted at the end of items.

Module Editorial System. This module stores in a database pre-defined scheduled information like dynamic Programme Type (PTy), text masks, event texts or moderators. The text masks allow a more attractive presentation of titles, artists, times, moderators and magazine names.

Radiotext example:
'[MODERATOR] presenting at [TIME] the title: [TITLE] performed by [ARTIST]'.

Module News Flash. The Editorial System offers an input function for real-time news flash information. Function keys of the PC keyboard can be linked to pre-stored text fields e.g. for football matches to send quickly changing results by keyboard hit of the editor.

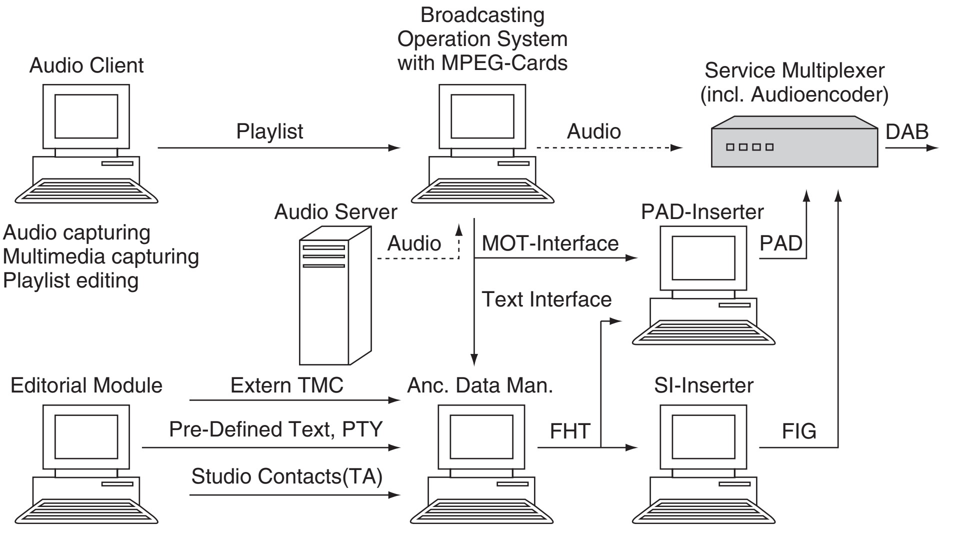

Figure 5.3 DAB studio infrastructure

Module Relay Interface. A PC-relay card detects studio switching contacts, such as a fader (signal level attenuator) contact in a traffic studio, for the generation of a traffic announcement (TA). The system supports 10 different announcement types (e.g. traffic, sport, news, transport, etc.). The relay interface also recognises the music or speech status, for generation of the M/S flag, detecting the mixing console fader.

External Data. An interface for pre-produced information allows the feeding of a formatted message called 'Funkhaustelegramm' (see section 5.3.1.2), that is TMC messages or DGPS information.

Control unit. The central system part automatically schedules and formats the outputting data. Adjustment of different priorities and broadcast cycles of the input modules described above allows an optimal presentation mix.

For different applications, such as differences in the text lengths of DAB text (128 characters) or RDS text (64 characters), the configuration of different outputs is possible.

Basic adjustments can be made to the configuration of static information, namely Programme Source (PS), Programme Identification (PI) or Traffic Programme (TP). Table 5.3 shows the information types of the system.

The ancillary data manager formats the output data according to the RDS transmission protocol, see [ARD, 1994] or UECP (Universal Encoder Communication Protocol).

Table 5.3 Ancillary Data Manager information types

Type	Information
CT	Clock/Time
DI	Decoder Identification
DGPS	Differential Global Positioning System
MS	Music/Speech
ODA	Open Data Application
PI	Programme Identification
PS	Programme Source
PTY	Programme Type
RT	Radiotext (64 and 128 characters)
RT+	Radiotext plus
TA	Traffic Announcement and 10 add. types
TDC	Transparent Data Channel
TMC	Traffic Message Channel
TP	Traffic Programme and 10 add. types
TTA	Title – Track – Artist (only ADR)

5.3.1.2 In-house Protocol

A specially defined in-house protocol, the so called '*Funkhaustelegramm*', is based at a recommendation of the ARD (German public broadcasters) for the distribution and transmission of ancillary data in broadcasting houses. One of the main objectives of the protocol definition was a uniform format for providing ancillary data to the data inserters for RDS, ADR and DAB (F-PAD and SI). The '*Funkhaustelegramm*' (see Figure 5.4) has a variable length and consists of three fragments: data head, data content and check sum. The data head contains a sync word, length, information type and source and destination addresses. The recommendation considers all control, service and text information types of RDS, ADR and DAB. The source and destination addresses allow individual addressing of a certain data inserter and recognition of the studio. The data are transmitted to the data inserter with UDP/IP via local area networks (LAN) or RS232 serial data lines.

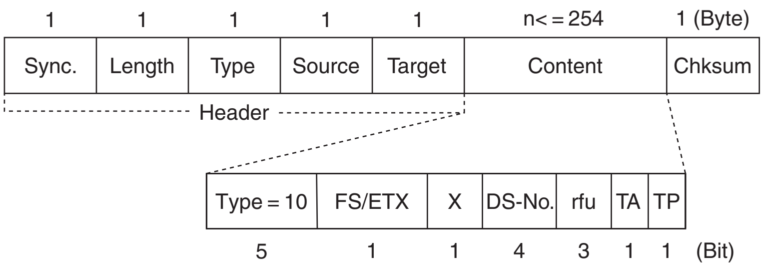

Figure 5.4 Structure of the '*Funkhaustelegramm*' (i.e. traffic announcement, TA)

5.3.2 *Multimedia Editorial Tools*

5.3.2.1 Multimedia Interface for Broadcasting Operation Systems

The broadcasting operation system (see Figure 5.3 above) allows a synchronised broadcast of audio and associated information. The capturing and administration of multi-media content requires an expansion of the existing audio database model. During hard-disk recording of the audio takes, any multimedia content (i.e. CD covers, pictures of artists or advertising pictures) can be stored in the database and combined with the associated audio parts. The recommendation [IRT, 1999] (see Table 5.4) has the same approach for multimedia content as the '*Funkhaustelegramm*' for text information. The MOT interface ensures a synchronised handling of multi-media information in the data inserters or web servers. Therefore, the broadcasting operation systems have to create a dBaseIII-compatible file, containing the MOT objects, which is read by the data inserters. The interface guarantees a fixed connection to the programme schedule. Every change in the play list is recognised by the data inserters.

The MOT interface is designed to store information ahead, over *n* titles or *n* minutes. It is the decision of the editor responsible to configure the associated information ahead of/or contemporaneously to the audio item. In the first case, the trigger time of the object, included in the MOT header, is set according to the play list time. In the second case, the trigger time is set to 'now'. This means that the MOT object will appear in the receiver immediately after transmission. Furthermore the MOT interface contains directory information for the objects and parameter files. This information indicates the archive file location. The parameter files contain pre-encoded MOT information (e.g. file size, object type, service manner). During the insertion process the PAD inserter requires all information.

Table 5.4 MOT interface for broadcasting operation systems

Name	Type	Length	Description
DATE	Date	–	Date
TIME	Char	8	Start time
TITLE	Char	36	Title
ARTIST	Char	24	Artist
H	Num H	6	Length in hours
MIN	Num H	6	Length in minutes
SEC	Num H	6	Length in seconds
PARFILE01	Char	60	Path and file name of the parameter file
MOTFILE01	Char	60	Path and file name of the MOT object file
...	...	...	...
PARFILE10	Char	60	Path and file name of the parameter file
MOTFILE10	Char	60	Path and file name of the MOT object file

5.3.2.2 Authoring Tool 'DAB Slide Show'

The 'Slide Show' application consists of sequences of JPEG or PNG pictures. A simple form of this service is the presentation of CD covers or web camera pictures. In the editorial tasks there is the desire to mix single content sources to provide an attractive service. Based on this desire, the German broadcaster '*Westdeutscher Rundfunk*' developed a special application software named 'PAD-Projector'. This allows the configuration of programme categories, such as weather, traffic, music or advertising. Every category consists of a picture and a text layer. The picture layer allows easy recognition of the topic by the use of pictograms. The text layer consists of actual information. Both layers are stored in a JPEG file and transmitted to the data inserters or web servers. The settings allow the configuration of default pictures (in the case of no available actual text content), the JPEG compression rate, the duration and the order of the slides.

5.3.2.3 Encoded Traffic Information (TMC, TPEG)

The objective of encoded traffic information is to avoid disturbances of formatted radio. Monotonous announcements and a large stock of messages have a negative influence on modern programme making with fixed speech/music portions and uniform programme flow. The objective of an exclusive coverage with encoded traffic information is, today, due to less market penetration with suitable receivers far away. Nevertheless it is important now for broadcasters to build up the necessary infrastructures to compete in future with other content traffic information bearers, like mobile telephone systems.

The RDS TMC services generated by traffic information systems currently provide a regular service in European countries. Car navigation systems are using RDS TMC for free updating of traffic status. Therefore, for identical coverage of the existing RDS market a TMC provision in DAB is essential. TMC was designed for transport in a narrow-band RDS data channel and because of that the standard is restricted for future expansion. Founded by the European Broadcasting Union (EBU), an independent expert group TPEG (Transport Protocol Experts Group) is still developing a standard for the transmission of traffic and transport information within DAB, DVB and the Internet (see also section 4.6).

5.3.2.4 Multimedia Database

A multimedia database is available for instance at the IRT for central contribution and distribution of multimedia content. The database is designed as an SQL database supported by PHP3 and JAVA scripts running on LINUX or WindowsNT®servers. The input scripts are adapted for single editorial applications and convert the arriving documents (DTP document, email) into a uniform database format. The output scripts can be optimised to the bearers (Internet, DAB, WAP, DVB) and provide applications like HTML, 'Slide Shows' or WML (Wireless Mark-up Language). The authoring tools are run easily with Internet browsers. The aim is to develop a growing multimedia toolbox for new broadcasting media with tools from the Internet market.

5.3.3 Data Inserter

Broadcasters and DAB network providers are inserting data services into the DAB multiplex with different system solutions. Described below are basic principles of DAB data service insertion following the developments by the Institut für Rundfunktechnik for the German public broadcasters.

The primary function of the data inserter is to encode data service content coming from studio sources to the transmission protocol in a given DAB data channel. The inserter processes the data according to the given transmission cycles [TR 101496] and finally feds all ancillary data streams into the DAB multiplex.

The DAB data inserter consists mainly of three parts (see Figure 5.5)

- Fast Information Channel (FIC) inserter,
- Programme Associated Data (PAD) inserter,
- Packet Mode inserter

5.3.3.1 FIC Inserter

Service Information (SI) provides supplementary information about services, both audio programme and data. Generally, SI is encoded in the Fast Information Groups (FIGS) of the Fast Information Channel (FIC) and has a static or dynamic

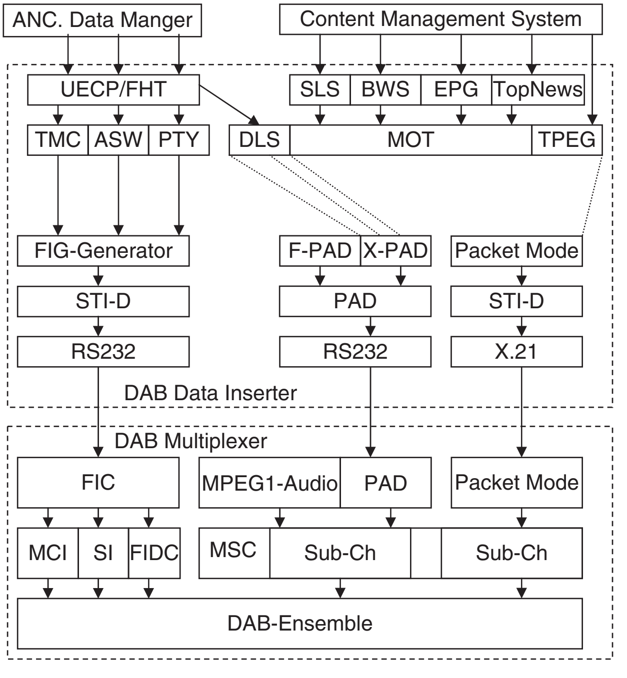

Figure 5.5 DAB data inserter

character, depending of the SI type. Dynamic FIC information includes for instance Announcement Service Switching (ASW), Programme Type (PTY) or TMC traffic data which are transported in the Fast Information Data Channel (FIDC). Static information like Programme label, Announcement support, time and country ID are signalled separately in the control software of the service or ensemble multiplexer.

To ensure a universal provision structure in broadcasting houses for dynamic ancillary data, the FIC inserter uses as input protocol the so called "Funkhaustelegramm" or the Universal Encoder Protocol (UECP). Additional parameters like Service provider ID, Service ID, Sub-channel ID and Cluster ID of each service are configured in the data inserter software.

The input buffer of the FIC inserter stacks the studio information and prioritises it to ensure a rapid sending. The SI is encoded as Fast Information Groups (FIG) and transported in STI-D format over a serial RS232 interface to the connected service or ensemble multiplexer. A cyclic repetition of the FIG data, according to the DAB Guidelines of Implementation, ensures error-free and quick recognition at the receiver.

5.3.3.2 PAD Inserter

Each DAB audio frame contains a number of bytes which may carry Programme Associated Data (PAD). PAD is information which is synchronous to the audio and its contents may be intimately related to the audio. The PAD inserter (see Figure 5.6) encodes the service content from the ancillary data manager (text information) or from content management systems (multimedia) into the PAD

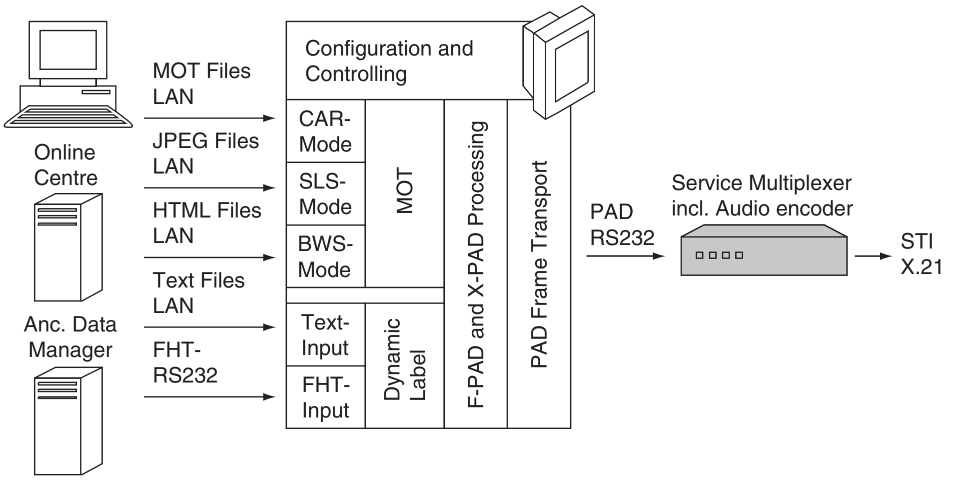

Figure 5.6 PAD inserter (IRT)

information types and transmits the PAD frames to the audio encoder via serial RS232 interface.

The DAB standard provides several PAD data services types (see section 5.3.2.1). In most DAB projects in Europe the applications Dynamic Label (DLS), MOT Broadcast Web Site (BWS) and MOT Slide Show (SLS) are used. Figure 5.5 illustrates the functionality of the PAD inserter with input interfaces for text and MOT objects. Typical PAD data rates are 8 or 16 kbps – in principal also data rates up to 64 kbps are possible.

5.3.3.3 Packet Mode Inserter

The Packet mode is defined for the purpose of broadcasting several data services using for instance the MOT or TPEG transmission protocol into a single sub-channel. The Packet mode inserter (see Figure 5.7) provides different service parameters to indicate the data service type, the network directory to receive the service content as files or streams and the packet address of the single sub-channel. In order to allow a rapid transmission of data services which consist of a high number of data files further parameters for the data carousels are adjustable. For instance to maintain fixed sequences of file types or to prioritise certain content for transmission. Data rates for the packet mode sub-channel shall be multiples of 8 kbps. Typical data rates are from 8 up to 64 kbps. Packet mode data may be carried in data groups. A single data group consists of 8191 bytes and is transmitted in one or more packets.

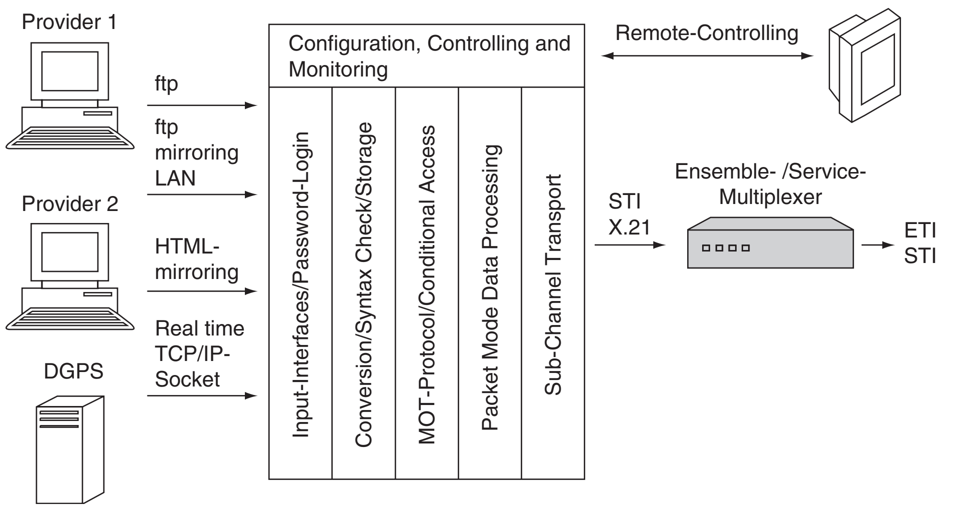

Figure 5.7 Packet mode inserter (based on the Multimedia Data Server of FhG-IIS)

5.4 Relationship between DAB Data Services and RDS

Some special features such as data services or other ancillary information are already being used in conventional radio services. One very well-known service is the Radio Data System RDS [EN 50067]. It is important to consider aspects of the implementation of the RDS and DAB systems which are directly related to each other to ensure as consistent and compatible an approach as possible. In other words, it is important to examine how the various features of both systems can best be implemented both by broadcasters and in a receiver to allow a user to benefit The listener must be able to access services and features in a similar manner regardless of whether tuned to either FM/VHF or DAB.

5.4.1 DAB SId versus RDS PI Code

Both Programme Identification (PI) codes in RDS and Service Identifier (SId) codes in DAB use a 16-bit code to identify audio services. Although the code structure may appear identical in RDS and DAB, usage of the code in terms of allocation and broadcaster and receiver implementation differs in the two systems, as illustrated in Table 5.5.

If RDS and DAB services carry the same programme at all times, common PI and SId codes may be allocated (SId = PI) and the SId code allocation will be constrained by the same rules as for RDS codes. The receiver can treat a common code as an implicit hard link. Service following between DAB and RDS can be done in this case

Table 5.5 Differences between RDS PI and DAB Sid

RDS PI	DAB SId
The second nibble of the 16-bit PI code represents the coverage area of the RDS service (Local, International, National, Supra-regional, or one of 12 Regions). This imposes constraints upon free PI code allocation. Services with PI codes that are identical in the first, third and fourth nibbles are implicitly regional equivalent services, i.e. they are "soft-linked".	The second nibble of the 16-bit SId code has no special significance, and there is no "soft linkage" implied between any two SId codes. "Soft linkage" is explicitly given by means of the linkage feature (FIG 0/6) by using common Linkage Set Numbers (LSNs).
Regional RDS services often use separate codes when originating their own programmes, but transmit a common PI code during shared programming. Broadcasters who vary PI codes in this manner are unable to provide identical codes on DAB and RDS even though services will at times be carrying identical audio.	For regional services SId codes must remain unchanged and be allocated unique codes, even if RDS services change their second nibble during common programming (in DAB SId codes must remain invariant). In this case, as a consequence, in any region the PI code and SId code will be different even though programmes are simulcast. A DAB service is explicitly linked to regional equivalent services using linkage.

without using the linkage mechanism. A receiver may use services with identical codes as alternative sources of the same programme. If at any time it is likely that different programming is carried on RDS and DAB, then different PI and SId codes must be allocated. In this case following between RDS and DAB can only be done by making use of the linkage feature.

To follow between RDS and DAB services without using linkage, the following may be noted: a receiver, on seeing that the RDS PI code has a second nibble of either '2' or '3', will know that all regions are carrying a common programme. Hence a switch is allowable to any RDS or DAB service whose PI or SId code is identical in the first, third and fourth nibble and where the generic (EG) flag is set.

5.4.2 Programme Type Codes

The Programme Type (PTy) feature in DAB has been designed to give improved functionality compared to the Programme Type code (PTY) in RDS. Consequently all related features or possibilities in DAB are not available in RDS.

Table 5.6 illustrates the differences between PTY (RDS) and PTy (DAB). In DAB it is possible to transmit PTy codes to represent both the static (service format) and dynamic (actual on-air programme item type) simultaneously. For both static and dynamic codes, up to two codes may be used. For each of the static and dynamic codes, at least one is chosen from the same international code range as used for RDS. The second, if used, may be chosen from this same set, or from the additional range of 32 coarse codes.

5.4.3 Announcements in DAB versus TA in RDS

Table 5.7 shows the differences between the Announcement feature in DAB and Traffic Announcement (TA) in RDS.

In RDS some dynamic PTY codes are used as fallbacks for DAB announcement types. Table 5.8 illustrates the coding of programme items intended to cause vectoring to a short audio message or 'announcement'.

Some broadcasters may choose to use the announcement categories 'News flash' and 'Area weather flash' instead of, or in conjunction with, programme types 'News' and 'Weather' respectively. As a consequence, in DAB, PTy codes 01 and 16 may be used for longer as well as shorter programme items, and in addition may be combined with a language filter code.

5.4.4 Dynamic Label in DAB versus Radiotext in RDS

Both RDS and DAB have the ability to carry text messages from the broadcaster to the listener's receiver. Although designed for the same purpose, Radiotext in RDS and Dynamic Label in DAB have a number of differences, see Table 5.9. In particular, control code use is different. There is a possibility to harmonise the usage to allow both broadcasters and receiver designers to implement the same text handler routines and achieve acceptable results and presentation on both systems.

Table 5.6 Differences between PTy in DAB and PTY in RDS

PTy in DAB	PTY in RDS
Besides of the basic set of 30 internationally agreed Pty codes a further set of 32 coarse codes and up to 256 PTy fine codes are available to categorise the programme content.	A list of 32 internationally agreed PTY codes is available to categorise the programme content.
Up to eight PTy codes can be allocated at the same time to a programme service.	A single PTY code only can be allocated to a programme service.
PTy codes can be signalled for the selected services and for other services via the PTy and other ensembles PTy feature.	PTY codes can be signalled for the selected service as well as for another service via EON (Enhanced Other Network).
PTy fine codes allow finer sub-divisions of some of the PTy coarse codes and can be used to watch for more specific programme types (e.g. "Tennis" instead of "Sport"). To preserve as much compatibility as possible, the DAB "fine" code should match the RDS PTYN description as much as possible.	There are no PTY fine codes in RDS but the Programme Type Name (PTYN) allows a broadcaster to transmit a short eight-character description to describe the programme content. The characters are transmitted over the air, and cannot be used for PTY Search or Watch functions.
PTy language allows PTy selection based on language criteria (e.g. "News" in "Swedish").	PTY language is not supported in RDS.
PTy downloading allows new codes to be defined "over the air". For PTys which are not agreed internationally, an accurate description can be provided. It also allows the description of PTy codes to be translated into any desired language.	PTY downloading is not supported in RDS.
PTy preview signals which PTy codes are likely to be used in the next one or two hours. This assures the listener whether a WATCH request is likely to succeed or not.	PTY preview is not supported in RDS.

Displays, which rely upon scrolling the text to show the complete message, should apply intelligent routines to present the message in the best possible way. Receivers could remove unnecessary spaces inserted by the broadcaster as 'padding', before display, to prevent scrolling through strings of blank spaces. If the resultant text length is shorter than, or equal to, the number of characters in the display, the display could show the message statically, without scrolling. Although the primary responsibility for presentation rests with the receiver manufacturer, broadcasters need to have a basic knowledge about the way Dynamic Label and Radiotext messages are being treated in receivers.

It is therefore up to the broadcasters to ensure that the data stream transmitted does not cause undesirable effects on older generations of RDS receivers. The receiver designer is responsible for ensuring the optimum presentation of the text, appropriate to the size and type of display chosen.

Table 5.7 Comparison between announcements in DAB and TA in RDS

DAB	RDS
16 announcement categories can be coded, of which 11 are currently defined. Announcement types are treated in a different way as vectored PTys in WATCH mode, whereas dynamic PTys are scheduled, announcements are unscheduled and used for short vectored interrupts.	In RDS only one announcement type (TA) is defined. The dynamic use of PTy codes 01 (News) and 16 (Weather) is often used to provide similar functionality to announcements in DAB.
Announcements are based on the cluster concept. They can be targeted to specific regions and feature a "New" flag, which is valuable in the case of a cyclic announcement channel. In the receiver announcements can be filtered according to the intended region.	TAs (or dynamic PTYs) with cluster operation is not supported in RDS. A "New" flag is not supported in RDS. Regional filtering of TAs is not supported in RDS though TAs from regional stations will inherently be regional.

Table 5.8 Coding of programme items intended to cause vectored interrupts

Description	Coded in DAB as:	Coded in RDS as:
Alarm	Anno flag b0	Dynamic PTY 31
Road Traffic	Anno flag b1	TP/TA
Public Transport	Anno flag b2	TP/TA
Warning/Service	Anno flag b3	Not available
Alarm Test	Not available	Dynamic PTY 30
News	Anno flag b4	Dynamic PTY 01
Weather	Anno flag b5	Dynamic PTY 16
Event	Anno flag b6	Not available
Special Event	Anno flag b7	Not available
Programme Information	Anno flag b8	Not available
Sport report	Anno flag b9	Dynamic PTY 04
Financial report	Anno flag b10	Dynamic PTY 1 7
Reserved for future announcement types	Anno flags b11–b15	Not available

5.4.4.1 Control Codes

A number of control characters, which the broadcaster may choose to insert in the text string, have been defined to create a particular effect on certain displays. As some of these control codes were not defined in the original RDS specification, some older RDS receivers may not be able to respond to these codes in the correct way. Most first-generation RDS receivers will substitute a space for these control codes.

Table 5.9 Differences between Radiotext in RDS and Dynamic Label in DAB

RDS Radiotext	DAB Dynamic Label
Supports a 64-character message.	Supports a 128-character message.
8-character displays on some older receivers.	Wide range of displays from $1 \times 1\ 6$ or 2×16 characters, LCD screens and PC monitors
Multiline displays should use normal word-processing techniques to provide word wrapping to show the message completely.	Because of the wide range of display types possible, it is up to the receiver manufacturer to develop software to optimise the presentation for the particular display chosen. Multiline displays will need to accommodate more than 128 characters to allow for word wrapping to show the message completely.

a) Control code '0B'

Control code 0B is used to indicate the end of the portion of text the broadcaster has designated as the 'headline'. The headline is a portion of the text meant for display in a fixed format containing information that does not require frequent updating and that is safe for viewing by driving motorists. It is physically sent as the first portion of the message and comprises all characters up to, but not including, control code 0B. If a headline is present, the 'body' is the part of the text excluding the headline and control code 0B. In compiling messages, the broadcaster ensures that the headline portion must be capable of standing alone, and that the complete message must make sense. The body may not necessarily make sense on its own. Receivers may choose to display:

- only the headline portion of the message; or
- the complete message with headline portion enhanced or differentiated; or
- the complete message without regard to the headline status;
- but *not* the body of the message alone.

If 0B occurs in the current message, it defines the headline, which comes into effect immediately. If no 0B occurs in the current message, there is no headline in effect. If 0B occurs in position '0' no headline is defined. The headline starts in position '0' and runs until the position preceding code 0B. Its effective length must not exceed 32 displayable characters. The occurrence of code 0B and hence of the headline is optional.

The content of the headline should not be changed more often than is compatible with road safety requirements. This does not impose restrictions upon the content of the body of the message which can be updated more frequently, nor on the repetition rate of the headline.

In the usual case where the broadcaster intends a space between the headline and body parts of the message, a space character must be explicitly transmitted.

- **In the case of DAB receivers and compatible RDS receivers**: When no headline is available, the receiver designer must decide upon the appropriate actions. Code 0B does not occupy a space in the displayed message. Code 0B, when ignored by the receiver (because no headline display mode is supported), does not represent a space. As the body text may not make sense on its own, it must not therefore be presented in its own right.
- **In the case of first-generation RDS Radiotext receivers**: As code 0B was not defined in the original RDS specification, some older RDS receivers may not be able to distinguish between the headline and body portions of the message. They may display headline and body as a single text string. Most receivers will substitute a space for 0B. Broadcasters who include code 0B in their Radiotext messages must be aware of this behaviour.

b) Control code '0A': preferred line break

Control code 0A is used to indicate positions in the message where the author of the text would prefer a line break to occur to structure the text in a particular way.

When control code 0A is used in the headline portion of the text, it indicates the position at which a receiver, using a two-line, 16 characters per line display, should preferably break the headline. In the case of a larger format display, the receiver may choose to ignore it.

If code 0A occurs in the headline, then the sum of displayable characters must be less than or equal to 31 (because a receiver equipped with a 1×32 character display format will insert a space character in place of the control code).

Code 0A must not occur more than once in the headline portion of the message, and may not be used if a 1F (preferred word break) is also used in the headline portion.

- **In the case of DAB receivers and compatible RDS receivers**: If a receiver uses a one-line, 32 character display to present the headline portion of the text, code 0A within the headline, if used, is replaced by a space in the display and not used to create a line break. If a receiver uses a 2×16 character display to present the headline portion of the text, the character string before code 0A is displayed in the first 16-character block; 0A codes are transmitted to aid a receiver to structure the display of the message optimally. Code 0A is replaced by a space in the display when not used to create a line break.

- **In the case of early RDS RT receivers**: Since code 0A was not defined in the original RDS standard, some older RDS receivers may not be able to use it to create a line break if desired. Most of these receivers will substitute a space for code 0A. Broadcasters including 0A in their Radiotext messages should be prepared to specifically transmit a space character in conjunction with code 0A to prevent unintentionally joining together the two words either side of the 0A code on certain displays.

c) Control code '1F': preferred word break (soft hyphen)

Control code 1F indicates the position(s) in long words where a receiver should preferably break a word between multiline (non-scrolling) display lines if there is a need to do so.

For example: If control code 1F is used in the headline portion of the text, it indicates the position at which a receiver using a two-line, 16 characters per line display should preferably break the long word. If code 1F occurs in the headline, it divides the headline in such a way that neither sub-string contains more than 16 displayable characters.

Code 1F must not occur more than once in the headline portion of the message, and may not be used if a code 0A (preferred line break) is used in the headline portion.

- **In the case of DAB receivers and compatible RDS receivers**: Code 1F, when used by a receiver as a word break, is replaced by a hyphen followed by a line break, otherwise it is ignored and does not represent a space.
- **In the case of early RDS RT receivers**: Devices produced to the original RDS specification [EN 50067] are unable to use it to create a word break. Most of these receivers will substitute a space for code 1F, so that words will be unintentionally split at the point of the 1F. For this reason, broadcasters may decide that use of code 1F is inappropriate for Radiotext.

d) Control code '0D': end-of-text

Control code 0D is used in RDS to indicate the end of the current message. It is used to help RDS receivers identify the actual length of the current message if it is less than the maximum 64 characters possible. DAB receivers do not use this control code.

Code 0D can appear only once in a message. Positions after code 0D should be disregarded and should not be transmitted or filled by spaces.

- **In the case of DAB receivers and compatible RDS receivers**: Code 0D and any characters or codes in all the following positions are ignored.
- **In the case of first-generation RDS RT receivers**: Since control code 0D was not defined in the original RDS specification, some older RDS receivers may be unable to use it. Most of these receivers will substitute a space for code OD, others may delete it from the string.

5.4.5 Cross-referencing DAB Services from RDS

It is likely to be a number of years before DAB coverage equals that of FM radio. During this period when DAB will be unavailable in all areas or for all services, it is assumed that receivers, especially vehicle receivers, will be able to tune to both DAB and FM transmissions.

Since DAB provides superior quality reception to FM, it should be the preferred choice for listening where available. Receivers may switch across from FM to DAB as soon as possible after entering a DAB served area if the DAB tuner knows in which ensemble, and on what frequency, the required service may be found.

The DAB cross-reference feature defines how to signal DAB frequencies within the RDS format to achieve a fast and effective way for a combined RDS/DAB receiver to get access to alternative programme sources in DAB transmissions.

5.4.5.1 ETSI Standard EN 301700

The European standard [EN 301700] describes the way that DAB service information can be provided from RDS by using an Open Data Application (RDS-ODA; the RDS Forum allocated the number AID = '147') allowing a receiver to find an equivalent DAB service. The ODA allows a service provider to signal not only frequency and transmission mode information about DAB ensembles but also linkage information and ensemble information about DAB services. This information can be used by a receiver to perform service following from RDS to DAB. The ODA uses an RDS type A group so that there are 37 bits available for coding. One bit is used for the *E/S flag* to differentiate between data for the ensemble table and data for the service table. The remaining 36 bits are used to express ensembles in terms of frequencies and modes, or services in terms of ensembles and service attributes (PTy, LSN, etc.).

This information will enable the tuner:

- to find the location of DAB ensembles;
- to find where the current RDS service is on DAB;
- to find a list of other DAB services;
- to know some of the attributes (PTy, LSN) of DAB services.

5.4.5.2 Relationship between DAB Services, Ensembles and Frequencies

In RDS, there is a single relationship: a service is carried on one or a number of frequencies. The RDS system allows a service provider to send all the frequencies on which the service is available to RDS receivers. The RDS receiver can build up a list of alternate frequencies (AF) which allows one (when mobile) to find the best frequency for that service. The RDS system also allows other service information to be provided, and, through the Enhanced Other Network (EON) feature, this can be extended to other RDS services.

In DAB the situation is different in that services are carried within ensembles. A given service may be carried in one or more ensembles and each ensemble may be carried on one or more frequencies. A DAB receiver is therefore required to maintain two tables, one listing ensembles and attributes for each service (*Service Table*), and a second, which lists frequencies and mode for each ensemble (*Ensemble Table*).

The requirement for cross-referencing from RDS is to deliver data to the DAB receiver to allow these tables to be constructed.

5.4.5.3 Ensemble Table

The ensemble table contains the basic information required for service following:

- the Ensemble Identifier (EId);
- the DAB transmission mode;
- the centre frequency of the DAB ensemble.

Table 5.10 Ensemble table format

PI code (16 bits in block 1)	Mode field (2 bits in block 2)	Frequency field (16 bits in block 3 + 2 bits in block 2)	Eld field (16 bits in block 4)
(PI code of tuned RDS station)	00 = unspecified 01 = mode I 10 = mode II or III 11 = mode IV	centre frequency of the DAB ensemble in the range 16 kHz–4194288 kHz	Eld of the DAB ensemble to which transmission mode and frequency apply.

Reception of the ensemble table does not mean that the tuned RDS service is carried on that ensemble. The purpose of the ensemble table is to allow the receiver to build up information about DAB ensembles. To locate an equivalent DAB service for service following requires that the receiver tune to each ensemble and inspect the service linking information. This task can be simplified by using the service table (see section 5.4.5.4).

Table 5.10 illustrates the ensemble table format. The EId being referenced occupies the last 16 bits in block 4. The preceding 16 bits in block 3 are used to signal the frequency code. In block 2, 2 bits are used to indicate the transmission mode.

NOTE: In [EN 300401] the centre frequency of an ensemble is coded as an unsigned binary number in a 19 bit field, allowing a range from 16 kHz to 8388592 kHz. As in this ODA only 18 bits are available, the frequency range will be limited from 16 kHz to 4194288 kHz.

A transmission of this RDS group is required for each EId and for each different frequency. Once all frequencies have been broadcast for a particular EId, another EId with its list of frequencies may then be broadcast. Every group is complete in its own right, and data are usable directly, without reference to any other group.

5.4.5.4 Service Table

The service table provides additional information about services available on DAB. It allows a receiver to get information via the RDS-ODA rather than having to examine each DAB ensemble for FIC information. Variant 0 allows the receiver to build up a list of ensemble identifiers that a service is available on, and variant 1 allows the linkage information for the DAB service to be stored. The service provider will signal every ensemble that a service is carried in before signalling the other attributes for that service. Once all the information is signalled for one service, the information for the next service may be signalled, and so on.

Table 5.11 shows the service table format. The SId being referenced occupies the last 16 bits in block 4. The preceding 16 bits in block 3 are used to transmit the list of Ids and all other required attributes of that service. The use of these 16 bits is indicated by the variant code that occupies the last 4 bits in block 2.

Table 5.11 Service table format

PI Code (16 bits in block 1)	Variant (4 bits in block 2)	Information Block (16 bits in block 3)	SId (16 bits in block 4)
(PI code of tuned RDS service)	(0–15)	variant 0: Ensemble information: EId variant 1: Linkage information: Linkage Actuator (LA), Soft or Hard link International Linkage Set (ILS), Linkage Set number (LSN), Rfu	SId of the DAB service to which the information block data applies.

A transmission of this RDS group is broadcast for each SId for each different EId. Once all EIds have been broadcast (using variant 0) other variants are transmitted to describe the other attributes for that particular SId. Other services may then be broadcast, each with its own list of EIds and service attributes. Every group is complete in its own right, and data are usable directly, without reference to any other RDS group. The transmission of the service table information is optional but when it is transmitted then both variant 0 and variant 1 information should be broadcast.

5.4.5.5 Operational Requirements

In order for the (RDS) receiver to know which application decoder to use and which group type is being used for carrying the ODA (Open Data Applications) data, the service provider must transmit two type 3A groups per minute.

If a service provider has many services carried on DAB in many ensembles up to two groups per second may be required to transmit all the application data. The ensemble and service table carousel may be interleaved but data will be transmitted in sequence within each carousel. All the ODA data will be transmitted within two minutes.

As already explained in section 5.4.5.3, application data from the ensemble table are transmitted such that all data for one ensemble are broadcast before data for the next ensemble are broadcast. As shown in section 5.4.5.4, application data from the service table are transmitted such that all data for one service are broadcast before data for the next service are broadcast. All ensemble information (carried in variant 0) relating to one service is broadcast before other data (carried in other variants) relating to that service are broadcast.

For service following it is recommended that the receiver use the linkage information provided via RDS for the RDS service and via DAB for the DAB service to link between equivalent services. This means that basically only the ensemble table needs to be transmitted. However, single front-end receivers have to use the linkage information provided via RDS for both the RDS and the DAB service, and therefore both the ensemble table and the service table should be transmitted. The linkage information is provided either explicitly by RDS type 14A groups and DAB FIG type 0 extension 6 (see section 5.4.2) or implicitly when the RDS PI code and DAB SId are identical (see section 5.4.1).

5.5 Electronic Programme Guide (EPG) for DAB

5.5.1 General

The Electronic Programme Guide (EPG) provides a means for a broadcaster to promote their radio station and the content on it to a wider audience. An EPG is typically an electronic version of a printed programme guide, presenting the user with a short summary of programmes that are currently broadcasting and that will be broadcast over the next few days, across a number of radio stations. As such, the EPG is an important tool for radio stations, encouraging listener loyalty and attracting new listeners by providing a mechanism for supplying visual information about the current programme, and forward promotion of future programmes.

A number of DAB receivers support the EPG standard. Figure 5.8 illustrates the typical information that is presented to a user who is tuned to a radio station carrying EPG information. The user is able to review the programmes broadcasting on that radio station and discover additional information about the individual programmes. Some

Figure 5.8 A Pure Evoke-3 DAB receiver displaying EPG information. Reproduced by permission of © James Cridlind

receivers, such as the Pure Evoke 3, enable the listener to use the information contained within the EPG to aid time-shift recording of future radio programmes; navigating through the programme list, they can choose the individual programmes and series they wish to record.

5.5.2 Development of the DAB EPG

In early 2001, WorldDMB [www.worlddab.org] set up a task force to define an open standard for a DAB EPG. From the outset the group agreed a number of fundamental key objectives, in order that the DAB EPG would meet all customer expectations:

- Display of radio schedules at varying levels of detail across a range of receivers.
- Descriptions of services and programmes.
- Selection of services and programmes by the receiver, and link to related content.
- Selection of services and programmes for recording at a given time by the receiver.
- Support both audio and data services.

It was also important that however listeners consumed radio (whether it was on a hi-fi, portable, car-radio or PC-based receiver) the EPG should be available to them.

From these objectives, WorldDMB produced a pair of specifications for the EPG that were published by ETSI in 2005 [TS 102818] and [TS 102371].

5.5.3 Data formatting

The formatting of the EPG data is important in that it has to offer the flexibility to work on a range of receivers with differing displays, resources, storage and connectivity. Ideally, it should also be backwards compatible to future XML specifications, easy to decode and efficient.

The DAB EPG uses Extensible Markup Language (XML) [W3C 2000], with the structure and content of the XML documents defined using XML Schema's. XML is ideal for this application as:

- Many applications and APIs already exist for manipulating XML.
- XML parsers are freely and readily available (which is useful for content import and export).
- XML also offers future expandability and backwards-compatibility; a well-designed XML application can be updated in the future without breaking any previous systems.

The last point in particular is important in the case of the EPG, where there are a number of potential future applications that may draw on this specification, but which were unknown at the time the first version of the specifications was published.

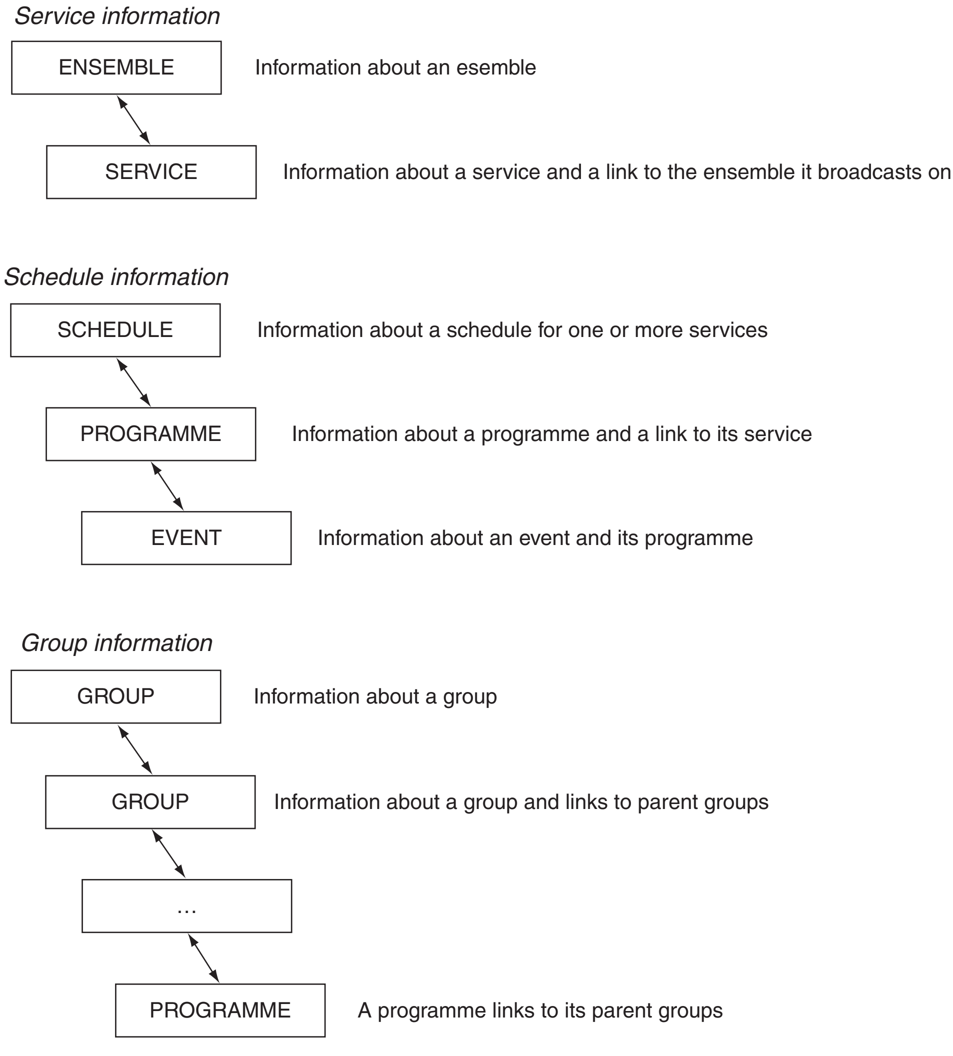

Figure 5.9 Structure of the EPG XML

The structure of the EPG is outlined in Figure 5.9. There are three main elements:

- **Service information**: This contains information relating to the ensembles and the services on those ensembles.
- **Schedule information**: This contains information about programmes and timed events that occur within those programmes and a reference to the service (or services) on which they broadcast.
- **Group information**: This enables programmes and events to be linked together into groups, such as serials or series. Additionally groups may also be linked to other

groups. For example, a programme called 'The Ultimate Radio Quiz' may be a member of the series group 'The Ultimate Radio Quiz Series'. This 'The Ultimate Radio Quiz Series' group could then be a member of the 'Quiz Programmes' group, which links together all of the programmes on the EPG that relate to quizzes.

A typical XML schedule file for a single programme is shown below:


```
<?xml version='1.0' encoding='UTF-8'?>
<epg xmlns=http://www.worlddab.org/schemas/epgSchedule/14
xmlns:epg=http://www.worlddab.org/schemas/epgDataTypes/14
xmlns:xsi=http://www.w3.org/2001/XMLSchema-instance
xsi:schemaLocation=http://www.worlddab.org/schemas/epgSchedule/14
epgSchedule_14.xsd system='DAB'
xml:lang='en'>
<schedule version='1' creationTime='2001-02-28T00:00:00' originator=
'BBC'>
      <scope startTime='2001-03-01T00:00:00' stopTime='2001-03-
      02T18:00:00'>
      <serviceScope id='e1.ce15.c221.0'/>
      </scope>
      <programme shortId='213456' id='crid://bbc.co.uk/4969758988'
      recommendation='yes'>
            <epg:mediumName>Gilles Peterson:</epg:mediumName>
            <epg:longName>Gilles Peterson: Worldwide</epg:longName>
            <epg:location>
            <epg:time time='2003-12-18T00:00:00'duration='PT2H0M0S'
            actualTime='2003-12-18T00:00:00'actualDuration=
            'PT2H0M0S'/>
            <epg:bearer id='e1.ce15.c221.0' trigger='c2213ac1' />
            </epg:location>
            <epg:mediaDescription>
            <epg:shortDescription><![CDATA[Gilles Peterson brings you
            two hours of global beats and the best of cool. Including the
            Worldwide family. KV5 are live from Maida Vale with special
            guests.]]></epg:shortDescription>
            <epg:multimediamimeValue='image/png'
            url='http://www.bbc.co.uk/radio1/r1_gilles.png' />
            </epg:mediaDescription>
            <epg:genre href='urn:tva:metadata:cs:ContentCS:
            2002:3.6.7'>
            <epg:name><![CDATA[ Rap/Hip Hop/Reggae]]></epg:name>
            </epg:genre>
            <epg:genre href='urn:tva:metadata:cs:ContentCS:
            2002:3.6.8'>
            <epg:name><![CDATA[ Electronic/Club/Urban/Dance]]>
            </epg:name>
            </epg:genre>
            <epg:memberOf shortId='1000'
            id='crid://www.bbc.co.uk/WorldwideGroup'/>
            <epg:link url='mailto:gilles.peterson@bbc.co.uk'
```

```
            description='Email:'/>
           <epg:link url='http://www.bbc.co.uk/radio1/urban/peterson/'
            description='Web:'/>
    </programme>
</schedule>
</epg>
```

The example above highlights a number of the key features of the specification:

- *'Link'* enables broadcasters to link to additional information not transported within the EPG data stream (for example links to websites, wapsites, or content transported in other DAB data channels), as well as enabling listener interaction through email and SMS links.
- *'mediaDescription'* enables the broadcaster to add additional multimedia content such as images, video or audio clips to programmes.
- *'xml:lang'* enables multi-lingual EPGs to be broadcast.
- *'recommendation'* allows the broadcaster to recommend specific programmes or groups of programmes.
- *'trigger'* enables a suitably equipped receiver to accurately record programmes, using a combination of data contained with the EPG and the PNum bytes broadcast within DAB FIG 0/16.

5.5.4 Transportation and Compression of the EPG

Whilst uncompressed XML has many advantages, it is not ideal for broadcasting over DAB, when compared with other formats:

- It is verbose and as such is not necessarily suited to the restricted bandwidth channels in which the DAB EPG will be broadcast (typically 16kbps or less).
- Parsing XML files provides additional complexity and temporary memory storage overheads for receivers.

Consequently, a companion specification [TS 102371] has been defined, which describes how the XML data is profiled, encoded and transported over DAB, such that:

- EPG data can be transported either in a standalone packet-data channel (N-PAD) or in a Programme Associated Data channel (X-PAD).
- The complexity of decoding the XML data is reduced.
- The size of the transmitted data is reduced.
- Both basic, and more advanced EPG receivers can be developed.

The encoding and transportation process contains three stages, as descibed in the specification [TS 102371]:

- data profiling;
- encoding;
- transportation.

Figure 5.10 Tag-Length-Value scheme

The XML data for each of the different data types (Service Information, Schedule Information and Group Information), is split into two profiles, the *Basic Profile* and the *Advanced Profile*. The purpose of this profiling is to enable simpler and lower-cost EPG-enabled receivers to be developed. These receivers have limited memory and decoding capabilities. The *Basic Profile* consists of a subset of the most commonly used EPG elements and attributes, the list of which is defined in the Binary Encoding specification. The remaining elements and attributes are carried within *Advanced Profile* binary data files. Some elements and attributes are contained in both files, to allow the two sets of data to be merged by advanced EPG receivers.

The *Basic Profile* and *Advanced Profile* EPG files are then binary-encoded using a tag-length-value scheme. Each element and attribute is assigned a unique tag value. This tag is then followed by a ‘length’ parameter, which indicates the length of the data contained within that element or attribute, followed by the data itself (see Figure 5.10).

The following is an example of XML containing a single programme:

```
<epg  xmlns:epg=‘http://www.worlddab.org/schemas/epg’
xmlns:xsi=‘http://www.w3.org/2001/XMLSchemainstance’
xsi:schemaLocation=‘http://www.worlddab.org/schemas/epg
epgSchedule_13.xsd’ system=‘DAB’>
      <schedule>
            <programme shortId=‘16442449’>
                  <epg:mediumName>PM</epg:mediumName>
                  <epg:location>
                  <epg:time time=‘2003-12-18T17:00:00’
                  duration=‘PT1H0M0S’/>
                 <epg:bearer id=‘e1.ce15.c224.0’/>
                 </epg:location>
            </programme>
      </schedule>
</epg>
```

The pairs of hexadecimal digits below represent the data that would be contained within a Basic Profile binary encoded data file, generated from that XML:

```
02 27 21 25 1C 23 81 03 FA E4 51 11 04 01 02 50 4D 19 16 2C 0A 80 04 33
BF C4 40 81 02 0E 10 2D 08 80 06 40 E1 CE 15 C2 24
```

Table 5.12 explains how this binary encoding is achieved

The binary encoding scheme provides a balance between efficiency of compression and complexitity of decoding. For the *Basic Profile* epg data files no further compression of the data is permitted. Due to the fact that receivers capable of decoding the *Advanced*

Table 5.12 Binary Encoding example

Bytes	Type	Description
02	TAG: <epg>	This is the tag for the top-level element.
27	LENGTH: 39	This is the length in bytes of all data contained within this file, excluding the two bytes for the <epg> *tag* and the *length* itself.
21	TAG: <schedule>	This is the tag for <schedule>.
25	LENGTH: 37	The length in bytes of all data contained within <schedule>
1C	TAG: <programme>	This is the tag for <programme>. This is the container for all information relating to a particular programme.
23	LENGTH: 35	The length in bytes of all data contained within <programme>
81	TAG: shortId attribute	This is the tag for the ShortId attribute
03	LENGTH: 3	This is the length of the value field for the ID attribute
FA E4 51	VALUE: 16442449	The decimal value for the ShortId from the XML is converted into its hex equivalent
11	TAG: <mediumName>	This is the tag for <mediumName>.
04	LENGTH: 4	This is the length of the value field for <mediumName>, including the CDATA tag
01	TAG: CDATA	This is the CDATA tag, and is used with text strings.
02	LENGTH: 2	This is the length of the content for <mediumName>
50 4D	VALUE: PM	The medium name of the programme.
19	TAG: <location>	This is the tag for <location>. The information relating to the broadcast of this programme is contained within this element.
16	LENGTH: 22	The length in bytes of all data contained within <location>
2C	TAG: <time>	This is the tag for <time>. The information relating to the time of broadcast of this programme is contained here.
0A	LENGTH: 10	The length in bytes of all data contained within <time>
80	TAG: time attribute	This is the tag value for the time attribute
04	LENGTH: 4	This is the length of the value field for the time attribute
33 BF C4 40	VALUE: 2003-12-18T17:00:00	The binary encoding scheme support two ways of encoding the time. The Short form encoding scheme uses less bytes, but does not encode the seconds.
81	TAG: duration attribute	This is the tag value for the duration attribute
02	LENGTH: 2	This is the length field
0E 10	VALUE: 3600	The length is encoded in seconds.
2D	TAG: <bearer>	This is the value for the bearer attribute. The information relating to the channel on which this programme is broadcast.
08	LENGTH: 8	This is the length of the value
80	ID attribute	This is the tag for the service ID attribute.
06	LENGTH: 6	This is the length of the value field for the ID attribute
40 E1 CE 15 C2 24	VALUE: e1.ce15.c224.0	The ID is encoded using the contentId encoding scheme as defined in the Binary Encoding specification

Profile data will have more capabilities than basic EPG receivers, additional GZIP [RFC, 1952] compression of the binary files is also permitted, to further reduce the size of these files on transmission.

The process of profiling and binary encoding the data generates a number of binary EPG files. These files are transported over DAB using the Multimedia Object Transfer (MOT) protocol (see section 4.3). Typically a DAB multiplexer will contain a single packet-mode data sub-channel containing all of the EPG data for the services carried on that multiplex.

5.5.5 EPG Data Management

Collection, warehousing, scheduling and validation of EPG data is important whether the data is transported within an individual service provider's X-PAD capacity or whether the data for all of the services is aggregated into a single data channel. In the case where the data from a number of service providers is combined into a single channel, the efficient and reliable management of this data becomes even more critical. Ensuring that the incoming data is properly formed, combining that data into a single EPG service, and encoding that data into a DAB EPG compliant format are all key functions of such an EPG management system.

To manage incoming data feeds from multiple suppliers, a process, as outlined in Figure 5.11, has been adopted by a number of commercial multiplex operators. In the situation where multiple service providers are generating and uploading their own

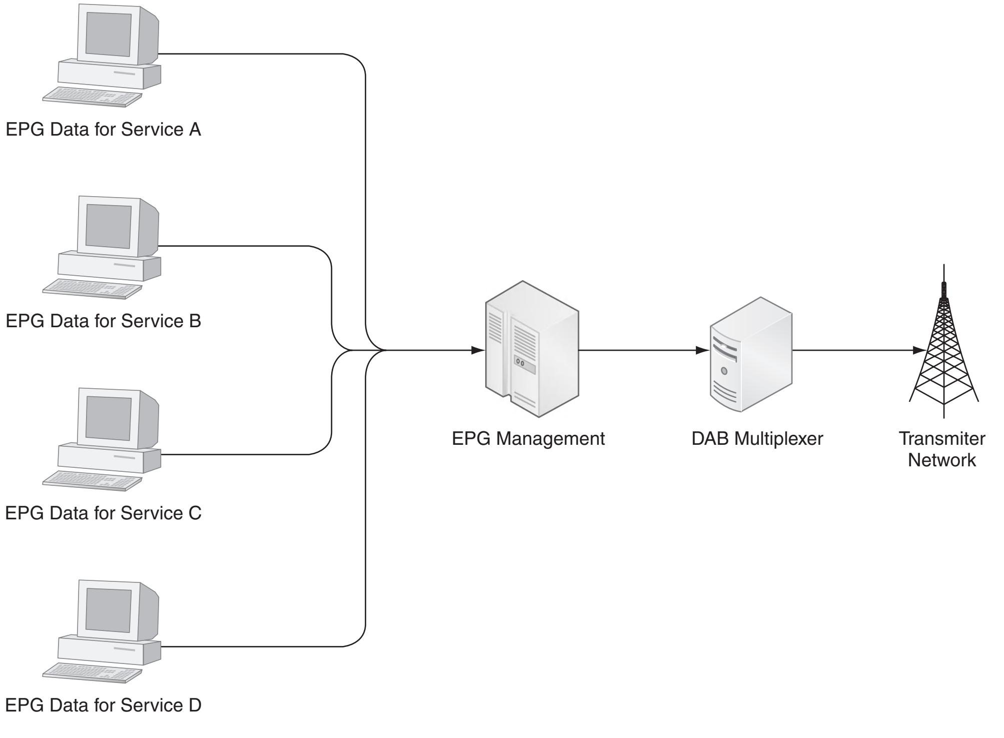

Figure 5.11 EPG data management process

content, a management server is placed in-between the content generation and the transportation of that data. The service providers still retain the capability to produce the EPG data locally, but prior to broadcast the information is validated and cached. The management of the broadcast bandwidth between multiple service providers enables the limited capacity to be used as efficiently as possible. It also provides a single point for archiving and logging the EPG data for security and licensing purposes.

5.5.6 Launch of EPG services

Through the work of WorldDMB, trials began in the UK in 2001. In London, content from several broadcasters was aggregated and broadcast on a test ensemble with PC-based DAB receivers distributed to a closed user group.

With the standardisation of the XML format a number of trial EPG services were launched in the UK in October 2002 by the *BBC, Chrysalis Radio* and *Capital Radio plc*. The launch of these services was supported by EPG software for existing PC-based DAB receivers, the *Modular Technology* PCI card and the *Psion Wavefinder*.

From 2008, a number of other countries including Germany, Ireland and Australia had trialled or launched EPG services.

5.6 Possible New Audio Services

The convergence of telecommunication networks and the integration of the next step of radio technology will lead to new dual or triple mode devices (e.g. DAB & GSM, DAB & GPS, DAB & PDA, DAB & MP3-Player).

The success of future radio with possible new audio services lies in the combination of traditional radio together with new IT technologies. Future radio therefore will be digital, with storage, internet access, display and several radio modes for DAB, DRM, DMB and FM. Following sections depict new opportunities of audio services using the DAB technology.

5.6.1 Dynamic Reconfiguration for Temporary Service Splitting

A main objective of the STI Standard [EN 300797] was to submit a dynamic reconfiguration of the DAB multiplex. This is necessary if services are added or removed within a DAB ensemble, or data rates of the services are changed (see section 6.2). The feature allows to extent the service structure at a particular time. An audio service can be split into two or three sub-services. In the case of low audio bit rates the appliance of half-sampling-rate coding (LSF) will improve the audio quality. Temporary reconfiguration may be reasonable for multilingual channels in multi-cultural programmes, for different regional programmes or during parallel sport events (e.g. Football leagues, Olympic Games etc.).

5.6.2 Secondary Service Components for Main and Sub-Services

The DAB standard provides primary and secondary service components for the organisation of services (see section 2.3.6). Primary components can contain a number of until 12 secondary service components. One secondary service component can be linked to several primary components. It's the listeners decision to switch to any sub-service. The service is realised by the service provider in collaboration with the ensemble provider. The availability of a secondary service at the receiver is signalled acoustically or visually to the listener.

5.6.3 Announcement Channels for a "Near Radio on Demand"

Radio listeners want to get information just-in-time he's interested in. Announcement channels are based on the idea to transport audio information (e.g. news, sport, weather, traffic update etc.) with narrow bit rates (e.g. 48 or 64 kbps) in parallel to audio channels. Beside the new source of information for the listener, the benefit for broadcasters is to discharge regular radio services from endless monotone announcements.

Possible audio sources are voice response systems, text-to-speech-systems and auto-record-/replay-systems. A voice response system assembles speech phrases, pre-stored in a database, to a whole message and replays this information over the broadcasting channel. An example is VERA (Verkehr in Real Audio) – a system for traffic information of the Westdeutscher Rundfunk WDR (German public broadcaster). Another alternative is to auto-record spoken messages from existing radio programmes (e.g. news channel) in a digital audio server and to auto-replay this in durable audio loops. Most flexibility offer text-to-speech-systems. New speech synthesis technologies improve the former poor speech quality of text-to-speech-systems and allow natural language processing.

5.6.4 Announcement Switching for Personal Preferences

The announcement switching is known from the FM/RDS Traffic Announcement (TA) feature. The TA service supports traffic information updates while listening to other radio programmes or local audio sources (e.g. CD, MP3). DAB at present offers 10 announcement types more than FM/RDS (see Table 5.8). A benefit for broadcasters is to achieve listeners who are using local audio sources. Additional mechanisms like new and region flags enhance the feature in case of update for same topics and regional interests.

5.6.5 Mailbox Radio

Mailbox Radio represents a project of IRT and Bayerischer Rundfunk to study the combination of traditional Radio with new type of media consumption arising from Internet and MP3 technology. Basic requirement is a storage functionality embedded in the radio receiver. Furthermore, the broadcaster sends inaudible tags using the standard Dynamic label plus (see 5.1.3.1) as description for every single programme item. Mailbox Radio receives all radio signals in the traditional way as linear radio programmes. The innovation lays in the personal recorder function which allows to record and to replay every programme item by using the content description from the Dynamic label plus feature. Now the listener is able to choose previously broadcasted audio pieces, which are held in the local storage of the receiver ("mailbox"). The feeding of the "mailbox" is possible either with DAB audio sub-channels or with MOT file casting in Packet Mode data channels.

6

Collection and Distribution Networks

6.1 General

6.1.1 Basic Requirements

Operation of the broadband DAB system requires the continuous collection of service component data, the formation of the respective ensemble multiplex data and their distribution to the transmitter sites of the single frequency network. This leads to a distributed, synchronised real-time system encompassing both a collection and a distribution network. These networks are connected by the ensemble multiplexer device which is situated in between.

Major differences between FM/RDS and DAB network architectures are outlined in Figure 6.1. In contrast to FM/RDS, DAB broadcasters, the service providers, send service component and additional data not directly to the transmitter sites, but on to the ensemble provider who operates the ensemble multiplexer. Tasks to be fulfilled by a DAB ensemble multiplexer are numerous owing to expanded functionality in DAB as provision for the dynamic change of the ensemble configuration, support of multicomponent services and enlarged on-air signalling compared to FM/RDS. With respect to signalling it should be emphasised that all service providers and the ensemble provider have to share the FIC, the common DAB signalling channel. Therefore provisions for dynamic signalling by service providers, but also avoidance of interference or inconsistency of signalling data are important issues.

In general, cooperation among the different contributors to the DAB transmission signal has to be managed. This will normally be done based on contracts between service providers and the ensemble provider. Resources are allocated which comprise not only transmission capacities but also coding resources (unique identifiers) necessary for the signalling of multiplex configuration, service and other information to DAB receivers. When sub-divided into management categories usually applied to telecommunication

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

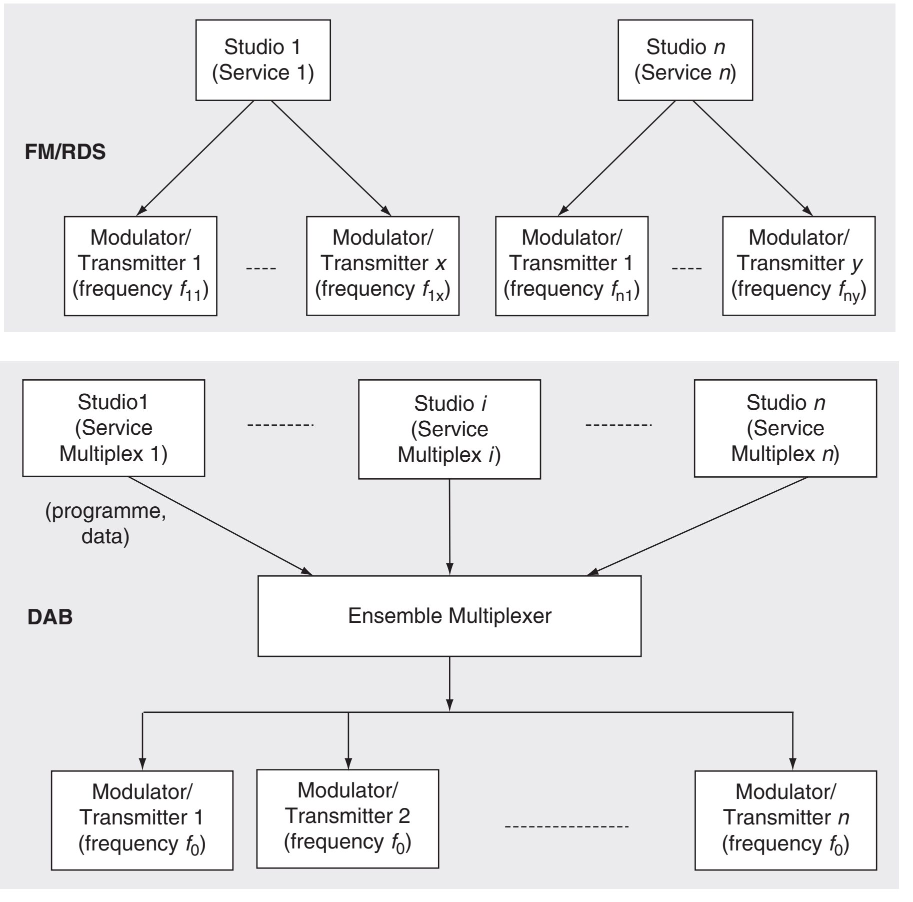

Figure 6.1 Network architecture of DAB versus FM/RDS

networks, general management tasks to be realised in DAB networks are as shown in Table 6.1.

Obviously the extent of technical support of these tasks depends on the specific DAB network scenario and its operational requirements. In a more static and regulated environment there is of course lower pressure to provide for technical support of management tasks than in the case when frequent changes are very common. Normally DAB service providers should be able to control a virtual ensemble of their own and to signal accompanying static and dynamic service information comparable to FM/RDS.

Concerning DAB networks, timing aspects have to be taken into account carefully. Not only do the single data streams of the service providers have to be synchronised, but also their actions have to be coordinated in a timely fashion to prevent any recognisable impairments at the DAB receivers. Different network propagation delays between the ensemble multiplexer and the transmitter sites in

Table 6.1 General management tasks in DAB networks

Management Category	Task
Configuration	(Re-)Configuration of ensemble multiplex
	(Re-)Configuration of equipment (e.g. DAB mode, TiI, physical interfaces, network and device delay)
Service	Co-ordination of service provider actions
	Monitoring and signalling of service information
Fault	Alarm and error signalling
Account	Admission of service providers
	Allocation of resources
	Billing
Security	Support of closed user groups
	Blocking of unauthorised control access

the distribution network have to be compensated in order to exclude destructive RF signal superposition in the single frequency network. Therefore a common time and frequency reference is needed. The satellite-based Global Positioning System (GPS) is largely used for that purpose.

DAB standards dedicated to the collection network, that is the Service Transport Interface (STI) [EN 300797] as well as the distribution network, the Ensemble Transport Interface (ETI) [EN 300799], have been defined. These standards take into account the implementation of DAB networks based on public telecommunication networks. Specific error protection methods are defined to ensure safe and cost-efficient data transport from studios to transmitters. Although data transport issues are mainly dealt with, essential management tasks related to DAB operation are also specified by these standards. Consequently, the following sections are closely related to these standards.

6.1.2 Ensemble Multiplexer

The ensemble multiplexer forms the key element in the DAB signal generation chain. Its implemented functionality determines to a large extent how the collection and the distribution network can be operated, for instance which type of connections can be used and which DAB-related controlling and signalling possibilities are available for service providers. To prevent complete ensemble drop-outs, the requirements for error-free operation of the ensemble multiplexer are especially high.

Figure 6.2 presents a conceptual block diagram of the ensemble multiplexer. Incoming data streams have to be synchronised first and then usually demultiplexed because the preferably used STI data frames are designed to include several service components. According to the actual configuration, the remultiplex of service components has to be accomplished thereby creating the MSC part of the ETI output frame. With regard to the 24 ms frame timing, these processes take place

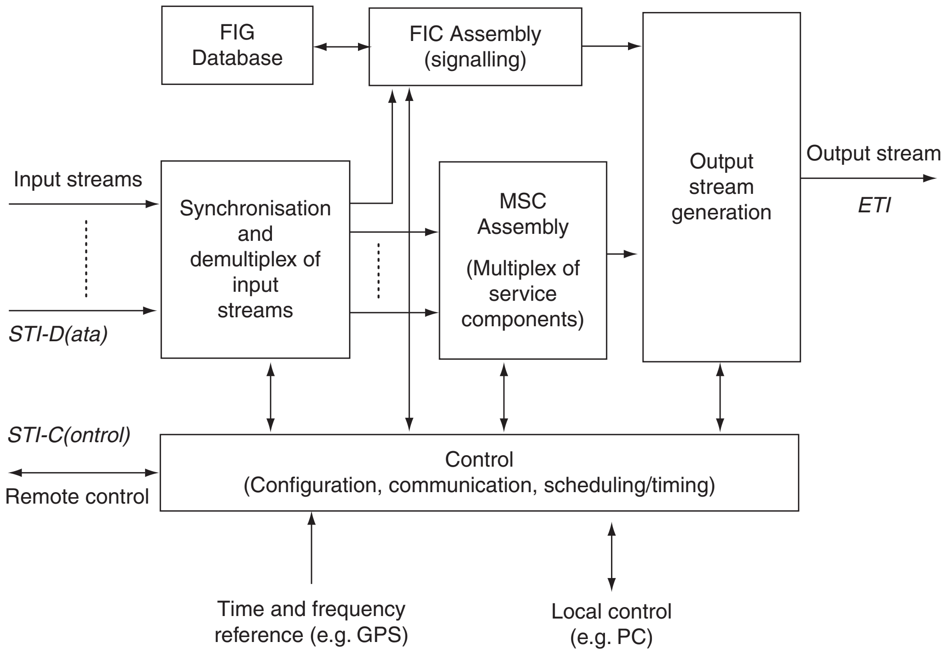

Figure 6.2 Conceptual block diagram of ensemble multiplexer

synchronously, which means imposing a fixed frame relation between the output and each of the contributing input data streams.

Concerning the signalling channel (FIC) which is part of the ETI output frame as well, two different sources of data have to be used in general.

One is the local FIG database. It comprises not only the MCI FIGs corresponding to the MSC configuration in use, but also FIGs defined by the ensemble provider (e.g. ensemble-related information like frequency or region definitions). Additionally service information FIGs delivered by service providers only once via control connection can be present there (FIG files). It is necessary to feed all these FIGs periodically into the FIC. The respective repetition rates are individually different and defined in [TR 101496].

The other source of FIGs to be transferred into the FIC are the STI input data streams which can carry so-called FIG streams too, that is FIGs provided and, if necessary, repeated by the service provider itself. These FIGs have to be transparently passed through the ensemble multiplexer.

The whole FIC contribution to the 24 ms ETI frame has to be assembled into Fast Information Blocks (FIBs) of 30 bytes length first. Concerning MCI, specific assembling rules have to be met [TR 101496]. The FIC assembling process is asynchronous by nature; this means that in contrast to MSC contributions no fixed frame relation or delay exists between FIGs delivered by service providers and their appearance in the ETI stream. Additionally consistency of FIG contents regarding resources allocated to service providers should be checked and ensured.

Because they are both reliable and cost efficient, the 2 Mbps [G.704] connections of public telecommunication networks are used in preference to the distribution network. It is the task of the ETI frame generation block to form transport data frames. Control information regarding final coding is included. ETI frames are normally error protected and also time-stamped in order to cope with delay compensation among connections of different length and type from ensemble multiplexer to the transmitter sites.

Beyond the necessary device configuration and subtle scheduling, the control block functions comprise a local control interface, for example a proprietary Application Programming Interface (API), as well as remote communication to service providers via STI. Management functions, as mentioned in the preceding section, are thereby supported including the data transfer from and to both remote sites and locally used tools supporting primary specifications of resources, configurations or FIGs and the monitoring of the ensemble operation as well.

To ensure synchronised data transfer and control actions a common time and frequency reference is indispensable. GPS provides for both a reference clock and absolute time at all locations and is therefore used in preference. Taking into account the signal delay up to the transmission antenna, the ensemble time can be derived from the GPS time string. This time is signalled via the FIC to receivers and is also needed for coordination of activities in the collection network. Time-stamps are related to the 1 pps GPS signal.

It is obvious that this potentially large functional complexity of the ensemble multiplexer will lead to a wide variety of implementations. To ensure interoperability of devices from different manufacturers, compliance with the corresponding standards and implementation guidelines is most important.

6.1.3 Conceptual Broadcast Network Outline

Figure 6.3 presents the DAB conceptual broadcast network outline as described earlier. At the service provider side, in general a primary multiplex of service components (encoded audio, packet or stream data, fast information data channel) and service information will be provided and delivered via STI. Service providers in the collection network can also be replaced by sub-networks comprising further STI service providers.

Both the collection and the distribution network are dealt with in detail in sections 6.2 and 6.3. Concerning the delay in the distribution network, refer also to section 7.4. A complete implementation example is described in section 6.4.

6.1.4 Implementation Issues

Many aspects have to be considered in the course of planning and implementing a DAB broadcast system. With regard to collection and distribution networks some important aspects will be listed here briefly. The coverage area of a Single Frequency Network (SFN) is dealt with in Chapter 7. Figure 6.3 shows a DAB conceptual broadcast network outline.

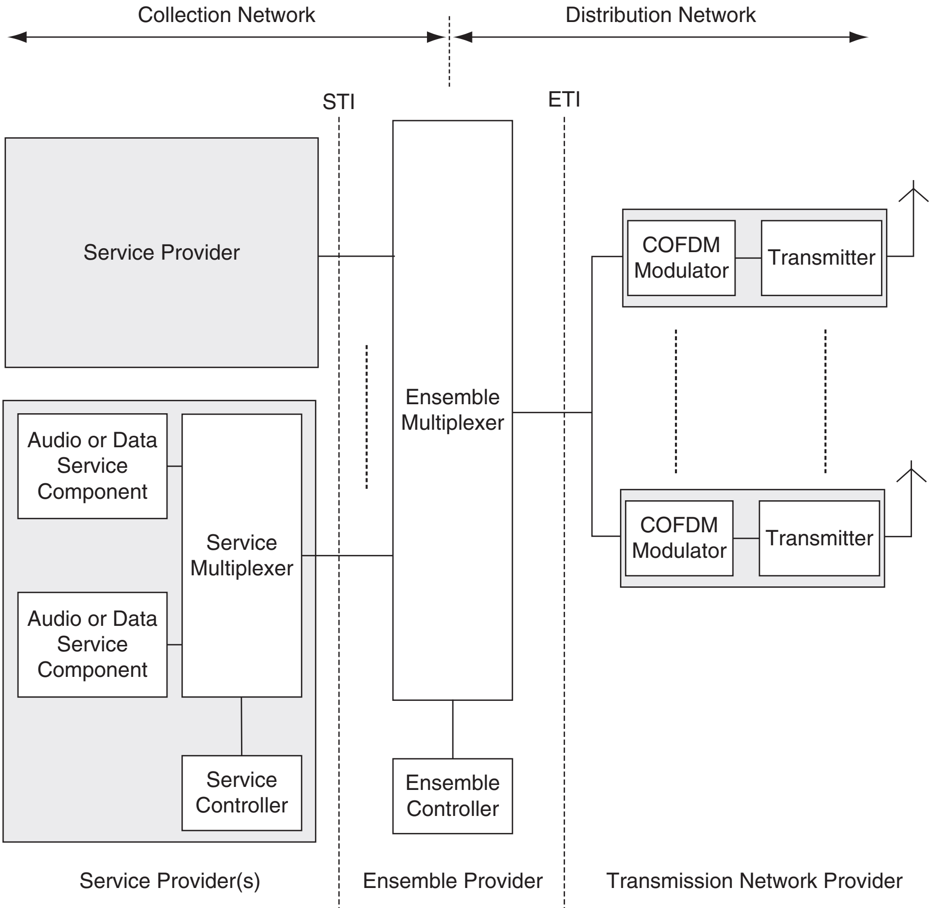

Figure 6.3 DAB conceptual broadcast network outline

From an operational point of view the following requirements are essential:

- Reliability of operation, based on technical stable networks and correct manual specifications created as much as possible 'right by construction'.
- Ease of use, for example tool support related to different user classes, for instance (pre-) specification in contrast to simple scheduling of configurations.
- Automated operation, including scheduling of control actions, error and alarm reactions, remote operation.
- Provision of flexibility, for example ability of service providers to control their part of ensemble resources based on constraints individually allocated by the ensemble provider who has to avoid erroneous interference due to the commonly used FIC.
- Integration into existing infrastructure, for example reuse of available RDS data sources and connection to automated play-to-air systems (see Chapter 5).

- Cost efficiency, for example alternative use of telecom connections (for instance, use of bundled ISDN channels instead of 2 Mbps. G.704 PDH lines by service providers).

Consequently, the following requirements for the respective equipment can be derived:

- Reliability, that is of course a high MTBF (maybe redundancy) but also supervision of regular operation including error and alarm handling.
- Variety of (configurable) physical interfaces to support different types of connections.
- Automated control, for example by means of scheduling functions integrated in service and ensemble controllers.
- Technical support of management tasks, for example resource allocation.
- Open system architecture supporting interworking in existing infrastructure (e.g. API).
- Standard compliance of equipment to ensure interoperability between devices of different manufacturers.
- Ability to integrate non-standard devices (e.g. available non-STI audio codecs).

Of course, not all of these requirements will be of the same importance in every case. According to evolving requirements, evolution strategies are preferred. This comprises feedback and evaluation of practical experiences.

6.2 The Collection Network

6.2.1 The Service Transport Interface (STI)

The STI was originally defined within the Eureka 147 project and published by ETSI as [EN 300797]. The standard is rather complex because it covers both a large variety of data transmission issues and also DAB specific management tasks related to the collection network.

The basic building blocks of an STI-based collection network are point-to-point connections established between devices supporting STI at their respective connectors. From a logical point of view the STI connection is sub-divided into a unidirectional data connection, mainly conveying the data streams to be broadcast, and a bidirectional control connection. Of course, this does not imply strict separation with respect to physical connections.

Regarding the single logical STI connection as shown in Figure 6.4, STI devices can be depicted in general as abstract entities which are distinguished according to the direction of the main data flow. The upstream entity delivers the data and therefore represents a service provider in every case. The entity at the receiving side of the data connection is called the downstream entity and can represent the ensemble provider or alternatively an intermediate service provider in the collection network.

The conceptual model of the STI is outlined in Figure 6.5. Beyond the separation of data and control, layers are introduced to reduce complexity and to provide for further internal interfaces.

At the top layer the basic data frame structure and control messages are defined. Consequently it consists of two logical interface parts: the data part STI-D(LI) and the

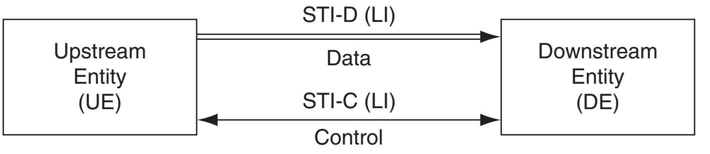

Figure 6.4 Logical STI connection

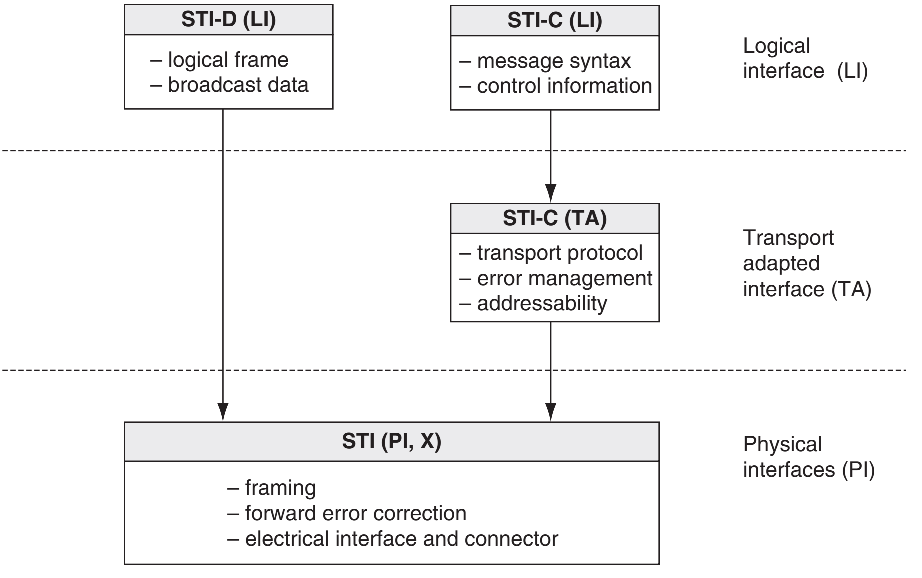

Figure 6.5 Layer structure of STI

control part STI-C(LI). Unlike broadcast data transport, error-free transport of control data is indispensable but not time critical.

Therefore an additional transport adaptation layer STI-C(TA) ensuring safe transport of control data was defined. This includes the ability to address the respective entities, too. Based on the definition of a common generic transport frame, the physical interface layer STI(PI,X) standardises how STI should apply widely used physical interfaces. Thereby aspects of framing, forward error correction, delay compensation as well as electrical parameters and connectors had to be considered.

The different parts of the STI definition will be dealt with in the following sections.

6.2.1.1 The Logical Data Interface

The frame structure of the logical data interface STI-D(LI) is represented in Figure 6.6. To provide for flexible use, the STI-D(LI) frame is related to the common interleaved frame (CIF) as defined in [EN 300401].

FC	STCEOH	EOH	MST	EOF	TIST
Frame characterisation	**Stream characterisation**	**End of header**	**Main stream data**	**End of frame**	**Time-stamp**
– SP identifier – Frame counter – Number of streams	per stream: – Stream length – Stream type – Stream Id – Stream CRC flag	– CRC	per stream: – Stream data – Stream CRC (if signalled in STC)		– Time stamp (optional)

Figure 6.6 Frame structure of the STI-D(LI)

Table 6.2 STI-D(LI) stream types

Stream	Type/Extension	Comment
MSC audio stream*	0/0	Independent of sample frequency
MSC data stream*	0/1	
MSC packet mode stream*	0/2	
MSC packet mode contribution	3/0	Needs to be adapted to sub-channel
FIC service information*	4/0	No MCI FIGs allowed
FIC data channel*	4/1	TMC, paging, EWS
FIC conditional access information	4/2	EMM and EMC FIGs (only an alternative to usage of MSC)
FIB (asynchronous insertion)	5/0	32 bytes per frame (FIGs and CRC)
FIB (synchronous insertion)	5/1	FIBGRID message defines insertion
In-house data	7/X	User definable (no broadcast)

* Basic stream types as defined in [TR 101860], see section 6.2.4.

Apart from the source identification, the unique service provider identifier, each single frame can be identified additionally within time slots of 2 minutes by its modulo 5000 frame counter value. In general, the STI-D(LI) frame will comprise several streams each individually described in the stream characterisation field. Possible stream types are listed in Table 6.2.

Independent of the contents, two basic classes of streams have to be distinguished: continuous and discontinuous. Continuous streams have a fixed amount of data per STI-D(LI) frame. According to initial synchronisation they have to experience a constant delay being passed through the DE. Thereby synchronous transmission is ensured. All of the MSC-related stream types with the exception of the MSC packet mode contribution are of that type. The latter requires pre-processing, that is an additional pre-multiplex step to merge several streams of this type and to form a complete MSC packet mode stream with a sub-channel bit rate of n $\times$ 38 or n $\times$ 332 kbps as required by [EN 300401].

Unlike continuous streams, discontinuous streams will not convey the same amount of data, or data at all, in each of the consecutively provided STI-D(LI) frames and, in general, data will not be delayed, but is constantly passing through the DE.

The actual number of bytes per stream and frame is called stream length and is dynamically signalled in the stream characterisation field. Stream length equals zero if a stream is logically opened but has no data in the respective frame.

A stream identifier (stream Id) is used to refer to streams by STI-C(LI) configuration data. It should be noted that the stream data extraction process from STI frames received is performed under control of the current configuration at the ensemble multiplexer (or DE in general). Further stream type/extension coding values are reserved for future definition.

6.2.1.2 The Logical Control Interface

The logical control interface is based on ASCII-coded messages which can be transferred asynchronously. A bidirectional connection is assumed to support typical request response communication between the two entities.

The unified message format of STI-C(LI) consists of a seven-character command word followed by a three-character extension and an individual set of parameters. All fields are separated by blanks and the message is terminated by a semicolon. The whole set of messages and its usage are described in detail in [EN 300797]. The standard also defines which kind of entity (upstream or downstream) is entitled to use a certain message and how to respond on its reception. For the sub-set definition of the message set with regard to STI implementation levels refer to section 6.2.4.

So-called data exchange sessions are used at logical interface level to provide for complete block transfer of logically related messages, for example an MCI configuration or a FIGFILE. In Table 6.3 a survey is given of the spectrum of STI-C(LI) messages. Each message class comprises a number of messages, specified for the different purposes.

6.2.1.3 The Transport-adapted Control Interface

The transport adaptation is aimed at ensuring an error-free transmission of STI-C(LI) messages including the addressing capability. Unlike for audio channels repeated transmission is not a principal problem. Therefore STI-C(TA) consists of a TCP/IP-like transport protocol which can be replaced by real TCP/IP under certain conditions (see section 6.2.5). Differences from TCP/IP concern the reduced length of packet numbers and addresses.

Table 6.3 Message classes of STI-C(LI)

Message Class	Messages	Function	Message Example
ACTION	6	Control of reconfiguration	RCONFIG REQ name time framecount;
CONFIG	10	Ensemble configuration	CONFDEF DEF name <spec>;
FIGFILE	8	Static FIG data	FIGFILE REC <FIG data>;
FIBGRID	6	FIB insertion (into FIC)	FIBGRID REC <data>;
RESOURCE	27	Resource allocation	CHANCAP DEF MSC capacity;
INFORMATION	16	Information about status	COUNTER INF;
SUPERVISION	9	Error and alarm information	ALARMST DEF status time;

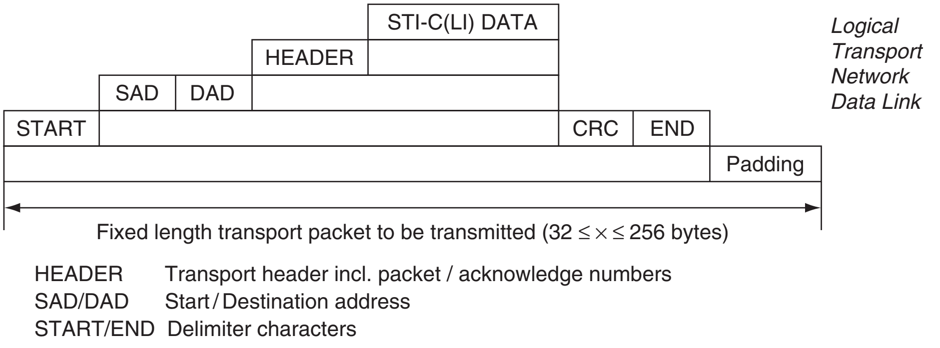

Figure 6.7 Layer structure and format of STI-C(TA) packets

Figure 6.7 illustrates how STI-C(TA) data are formed. This process is again based on a layered approach. The whole functionality could be assigned to the transport layer of the OSI reference model because the STI-C is an end-to-end connection between two STI entities. Nevertheless functionality analogous to lower OSI layers is used to describe the different functional parts.

At the receiving side the CRC of the data packet will be checked first. In case there is no error then the useful data will be submitted to the layer above. Otherwise the data will be ignored and re-sent by the source entity later on owing to missing acknowledgement. At the next layer the destination address of the packet is compared with the address of the receiver. If there is no match, the data packet will again be ignored. In the opposite case, the data were intended for the receiving entity and were transmitted error-free.

Consequently the next layer will acknowledge its reception using the respective packet number under the condition that no packets have been lost in between. The acknowledgement requires sending a message back to source and can be combined with the transmission of useful information. Finally the STI-C(LI) DATA, a substring of predefined length of the overall message sequence at source side, is passed to the logical layer.

It should be emphasised that the received data packet in general will not be aligned to one or more complete STI-C(LI) messages. Message detection and interpretation takes place in the resulting character string formed by appending the data packets consecutively as received. Overhead per transport packet is 32 bytes. The transmitting side behaves in the opposite way.

The protocol requires an explicit open/close of connections. Owing to the multiplex of logical connections corresponding to SAD/DAD addresses, a common STI-C channel can be used to communicate with several destinations.

This will be particularly useful in the case of a common reverse channel from the ensemble multiplexer or in the case of multicasting the same forward channel from the service provider (e.g. together with STI-D via satellite).

6.2.1.4 The Physical STI Interface

Physically transported frames are all derived from a generic transport frame as shown in Figure 6.8.

SYNC	TFH	DF	DFPD	CF	FRPD
Synchronisation	**Transport frame header**	**Data frame**	**Data frame padding**	**Control frame**	**Frame padding**
– Error status – Frame synchronisation	– Data frame size – Control frame size	– STI-D(LI) data (see Fig. 6.6)		– STI-C(TA) data (see Fig. 6.7, a constant number of transport packets)	

Figure 6.8 Frame structure of the generic transport frame STI(PI,X)

Compared to STI-D(LI) it contains the additional synchronisation field including an error status field, marking the severity of errors detected and set by the error-detecting unit. Further on, a transport frame header is inserted which signals the fixed size of the following data and control part. Either STI-D(LI) or STI-C(TA) or both can be transported depending on the respective field size described in the transport header.

The generic transport frame has to be adapted to the specific physical interface in use. Frame assembly details as well as connectors are described in [EN 300797]. In total the standard defines eight physical interfaces. For synchronous transmission the widely used 2 Mbps interfaces G.704 (with and without FEC as in ETI) and G.703 are available.

Further interfaces concern bundled ISDN channels including delay compensation according to H.221 (J.52), V.11 (RS422), IEC 958 and WG1/2 from Eureka 147. V.24 (RS232) can be used for asynchronous, low-bit-rate transmission. Although it is appropriate in most cases to transport STI-D and STI-C within the same connection, the standard provides possibilities of transmission using separate connections. Figure 6.9 demonstrates both cases.

6.2.2 Network Architecture

According to [EN 300797] the collection network is logically based on point-to-point connections forming a tree-like hierarchy of entities. Figure 6.10 outlines this by means of an example. It should be recalled that diverse physical connections between entities can be implemented.

Each entity can be classified by a type according to its role in the network [TS 101860]. As already introduced in section 6.1, the essential roles are service or ensemble provider.

Based on implications to specific STI functionality to be implemented, service providers are further distinguished into those who are in the UE position and those who are situated intermediately in the tree, that is, those who are both UE and DE (see also section 6.2.4).

As outlined in Figure 6.10 each entity needs to be identified uniquely by a service or ensemble provider identifier. These identifiers are used for the addressing of control messages and marking of logical data frame sources as well. No specific numbering scheme is pre-defined. The total of transmission and coding resources allocated to an entity have to be shared between this entity and all of the entities connected to its inputs.

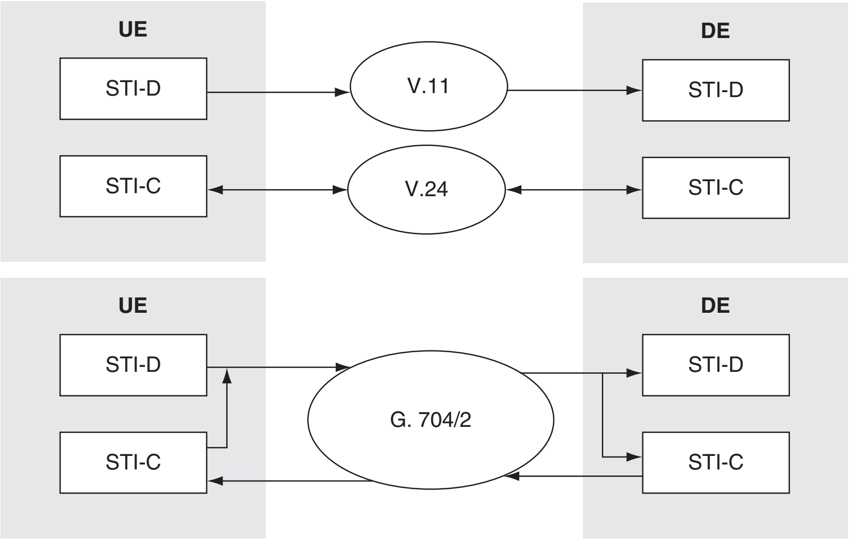

Figure 6.9 Physical connections of STI-D and STI-C (examples) upstream

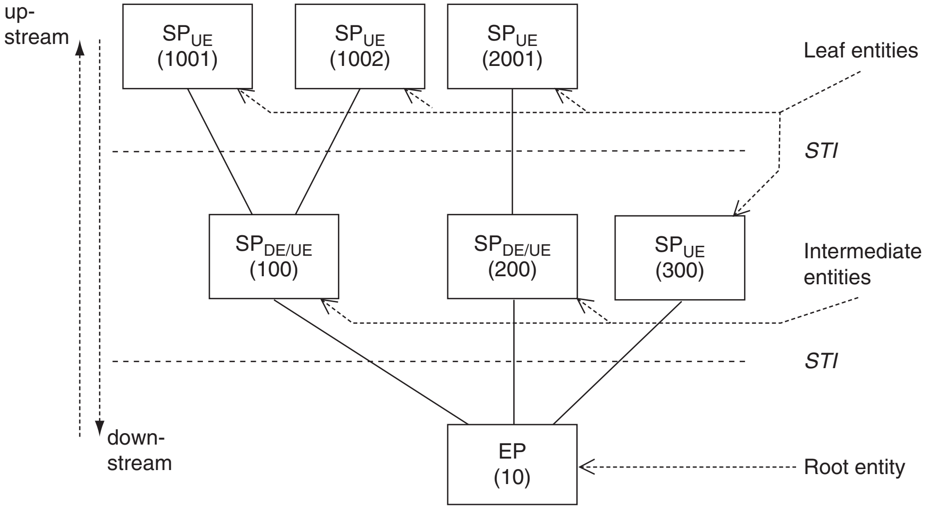

Figure 6.10 Tree-structured DAB collection network (example)

For instance, in Figure 6.10, resources allocated to SP 100 comprise resources reserved for local use by SP 100 itself (e.g. due to service components or information inserted by non-STI sources at this level) and those propagated to its subordinated SPs, that is to SP 1001 and 1002.

At least during the migration phase real networks may additionally comprise non-STI SPs (see also section 6.2.4). Section 6.4 presents a corresponding example.

6.2.3 Network Operation

The requirement of both flexible and safe regular operation of the complete DAB broadcast network constitutes a challenge due to the complexity of the distributed real-time system. In general, service providers should be able to operate independently of each other, but be coordinated by the ensemble provider, based on respective contracts.

Assuming collection network has been established according to section 6.2.2, STI-C(LI)-based management tasks and subsequent actions have to be considered concerning dynamical operation. STI-D stream processing is consequently controlled. For simplicity, the focus will be on the most important first hierarchical level of the collection network, that is EP and directly connected SPs. The STI defines an interface of machine-readable data and no user interface for operators. In practice there will be a need for additional tools performing the translation tasks from manual specifications to STI control messages or FIGs and vice versa.

In order to enable service providers to start broadcasting, the ensemble provider has to provide some prerequisites. The control connections to the service providers have to be opened using the provider identifiers as negotiated. Via these connections information about individually allocated resources could be transmitted to the service providers if implemented. Otherwise this information has to be passed to them outside of STI. Essential data, which can be delivered via STI-C by RESOURCE messages, comprise:

- identifiers of sub-channels, services, FIC data channels and linkage sets;
- transmission capacities of MSC and FIC;
- maximum number and bit rates of specific stream types;
- restrictions concerning usable FIGs (regarding FIG type/extension);
- entitlement and parameters to send announcements.

Based on these, service providers are able to specify and check activities of their own.

Further on, an empty ensemble can be generated, which means no service components are included before the first reconfiguration initiated by a service provider. Nevertheless basic ensemble-related FIGs can already be sent as defined by the ensemble provider. Essentially this FIG set will consist of

- ensemble identification and label (FIG 0/0, FIG 1/0);
- date and time (FIG 0/10);
- ensemble-related tables to be used, country with local time offset (time zone) (FIG 0/9);
- frequency lists (FIG 0/21);
- TII lists (FIG 0/22);
- regional definitions (FIG 0/11).

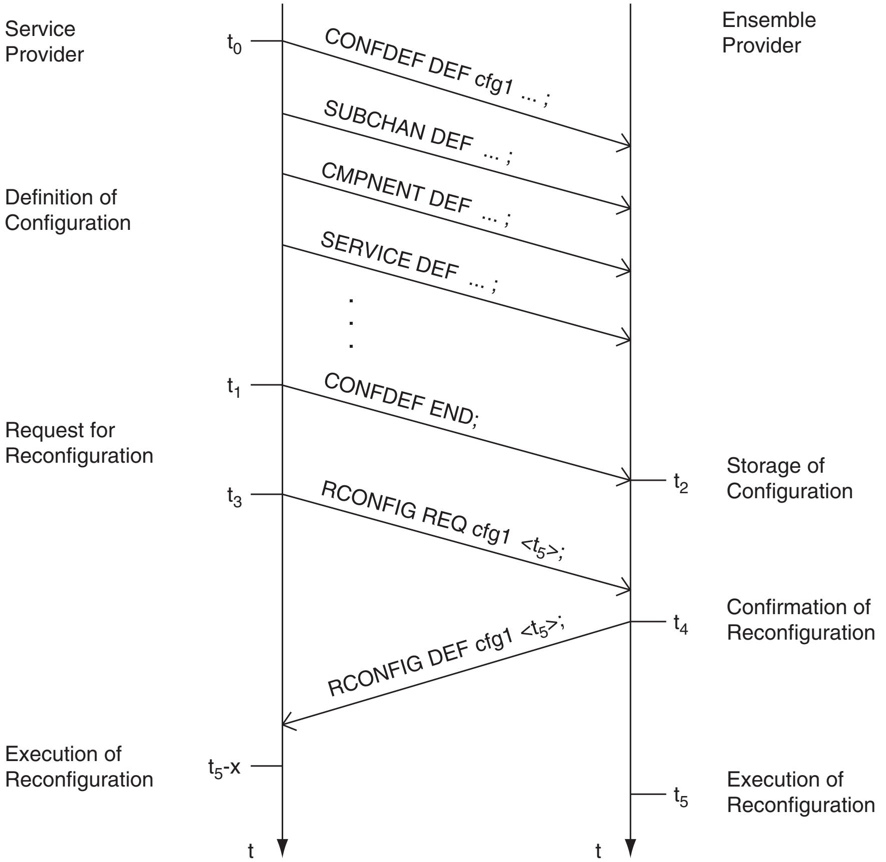

Figure 6.11 Control message flow for definition and activation of virtual ensemble configuration (example)

Service providers will normally start transmission by performing a reconfiguration of the virtual ensemble allocated to them, for instance by means of RESOURCE messages as mentioned above. Figure 6.11 shows this process in chronological sequence.

First of all the multiplex configuration has to be defined in the frame of a data exchange session. Therefore the first message block will be enclosed by CONFDEF DEF and CONFDEF END messages. The CONFDEF DEF message is the header of the block providing the name of the configuration and numbers of messages of specific type included.

The body of the message block contains the following definitions (one message per item):

- Sub-channels to be used for transmission of service components including identifiers and protection levels (SUBCHAN DEF).
- Service components (CMPNENT DEF) including service component types and corresponding stream and sub-channel identifiers (for both MSC and FIDC), also including packet addresses in the case of packet mode service components.

- Service definitions (SERVICE DEF) comprising one or more service components and based on the specifications given before.
- Further optional definitions, for instance to specify by name that predefined FIG files (signalling data) have to be enabled at reconfiguration instants (USEFIGF DEF) or that FIG streams out of STI-D have to be inserted into FIC (USESTRM DEF).

Firstly, the ensemble provider needs these data for generation of MCI FIGs to signal the multiplex on-air. Secondly, these data are necessary for controlling stream extraction from STI-D to form the MSC and to enrich the FIC in case FIG streams are in use.

The messages received will be checked for plausibility and compliance to the allocated resources. If the checks are passed, the configuration is stored and can be referred to from then on. Otherwise an error message is sent back to the service provider. How many configurations an ensemble provider is able to store depends on implementation and usage. Service providers can be informed about that via STI-C. Next the service provider will request a reconfiguration from the ensemble provider specifying the name of the configuration and the execution time wanted (RCONFIG REQ). To ensure seamless reconfiguration the time specification (<*t*5> in Figure 6.11) must be frame related. It is therefore sub-divided into a UTC value uniquely addressing 2 minutes out of 24 hours and a data frame count value specifying the respective 24 ms frame within the 2 minutes.

According to [EN 300401] two subsequent reconfigurations must have a time distance of at least 6 seconds. Because there is no coordination required between service providers to agree on reconfiguration times, only the ensemble provider is entitled to define the instant of execution. Thereby the latter is able to merge the requirements of several providers in such a way that only execution times equal to or later than the requested ones are allocated.

Correspondingly the ensemble provider will check that a correct configuration of the received name has been defined previously by the service provider and will define the execution time for reconfiguration as close as possible to that requested. This information is sent back to the service provider (RCONFIG DEF). From now on a reconfiguration is pending at the ensemble multiplexer which will lead to the corresponding on-air signalling of the next ensemble configuration starting automatically about 6 s before reconfiguration. Per provider, only one reconfiguration can be pending at a time.

Assuming that the service provider does not cancel the reconfiguration request right in time (RCONFIG CAN), it is expected that contributions are delivered via STI-D precisely timed and in accordance with the activated configuration. Otherwise error messages would be generated by the ensemble provider. Some further remarks are needed regarding the right instance to change streams or bit rates. As depicted in Figure 6.11 the service provider needs to know the propagation delay x of the STI-D frames, that is the time offset between departure of a frame at the service multiplexer and the appearance of its contents in the respective ETI frame leaving the ensemble multiplexer. By means of the message COUNTER INF a service provider can ask the ensemble provider about the frame-counter-related time difference and has to take this into account to support seamless reconfiguration. Moreover, decreasing the bit rate of a sub-channel needs to be accomplished 15 frames in advance to comply with the 16 frame interleaving process of the on-air signal.

Provision of service and ensemble information is essential. As mentioned earlier, there are two basic mechanisms to be used in conjunction or alternatively: FIG streams (transported in STI-D) and FIG files (transported in STI-C). The main difference concerns the responsibility for FIG repetition. Required repetition rates are defined in [TR 101496]. In the case of FIG streams, which are transparently passed into the on-air FIC, this must be performed by the source, for example by a service provider. Otherwise the ensemble provider is responsible for repetition of FIGs.

It should be emphasised that special treatment of so-called list FIGs is necessary. In the main, information is concerned with

- service linking sets;
- region definitions;
- frequency lists;
- transmitter identification lists;
- announcement support;
- other ensemble services;
- satellite database.

These FIGs comprise small databases. Well-defined parts of them should be updated selectively in order to avoid time-consuming recovery of complete data by receivers in case of changes. That is, with regard to key data fields the change event concerning a data subset has to be signalled by means of specially defined FIGs sent for 6 seconds between the old and the new data versions (SIV/CEI, see also [TR 101496]).

Additionally or alternatively FIG files can be used to signal supporting information. As with ensemble configurations, the corresponding file, which consists of one or more FIGs, has to be transferred to the ensemble multiplexer first. It would be checked and stored analogously under its name.

Further on, two alternative activation mechanisms are standardised for FIG files. As mentioned above, the FIG file to be activated at the instant of reconfiguration should be named by means of a USEFIGF DEF control message which is part of the respective ensemble configuration. Alternatively, FIG files can be activated or deactivated explicitly using FIGFILE SEL(elect) or DES(elect) messages respectively at arbitrary time points. Thereby activation/deactivation can be triggered in principle either based on pre-defined schedules or derived from events like keystroking by a moderator. Repetition of FIGs as part of FIG files including the SIV/CEI treatment is up to the ensemble provider.

Intermediate service providers exist in hierarchical collection networks (Figure 6.10) consisting of more than one level of service provider. As a downstream entity they should communicate with its upstream service providers like an ensemble provider. In the opposite direction, as an upstream entity, they should behave like other service providers (leaf entities). Consequently, intermediate service providers have to share resources with and act on behalf of their subordinated service providers.

Because STI only covers management issues very close to DAB, there is a need for support beyond STI to operate real networks. For example, section 6.4 will briefly cover such requirements and their practical implementation.

Important requirements are:

- automation of operation (scheduling);
- coding support for ensemble configurations and FIGs guided by allocated resources;
- event logging;
- alarm handling;
- connection to existing infrastructure (e.g. studio operation system, databases/ sources).

The problem of interoperability due to devices from different manufacturers who may each implement different subsets of the STI is considered in section 6.2.4.

6.2.4 STI Implementation Levels

As demonstrated above, the STI standard [EN 300797] is rather complex and therefore the question arises how the interoperability of equipment produced by different manufacturers can be ensured. It is quite reasonable that not all STI-compliant devices have to provide for the complete functionality as defined in the standard. Owing to the different extents of dynamic operation of the specific collection network and the number and kind of providers involved, there will of course be varying requirements leading to different implementation costs too.

Faced with this situation, a WorldDAB task force derived subsets of functionality from the standard. It was assumed that the standard itself should not be changed. A technical specification defining 'STI Levels' has been standardised by ETSI [TS 101860].

Analysis has shown that definitions of basic stream types to be processed as well as management functionality related to the STI control functions are most important considering useful sub-sets of functionality. Therefore sub-division of functionality has been done with regard to these criteria. Three hierarchical levels of STI functionality have been defined. This means that a higher STI level fully encloses the lower ones. But even the highest level, the third level, does not comprise the whole functionality of [EN 300797]. Some special functions are declared to be optional and level independent.

Also physical interfaces are beyond the scope of the STI level definition because it is much easier to reach interoperability between devices on this level than on a functional level. A wide variety of terminal adapters and other interface converters can be used for that purpose. Table 6.4 outlines the STI levels as defined in [TS 101860].

Implementation of the control channel is not required at STI Level 1. Only STI-D has to be supported regarding basic stream types. In contrast to the ensemble provider, or a downstream entity in general, service providers are only enforced to generate at least one out of the basic stream types. Implementations at the downstream side will be similar to those aimed at supporting nonSTI providers (refer to section 6.2.5).

To be compliant with STI Level 2 the availability of the control channel is assumed and processing of a basic set of STI-C(LI) messages has to be implemented. This leads to the ability for seamless dynamic reconfiguration initiated by the remotely situated service provider. FIG files can be used for FIC signalling. Both FIG file activation mechanisms as described in section 6.2.3 have to be implemented.

Table 6.4 STI implementation levels

Level	Interface	STI-D(LI) Stream Types	STI-C(LI) Messages	Comment
1	Restricted		(no control channel)	Local control proxy for UE at DE needed
2	Regular	Processing of basic stream types: MSC audio MSC stream data MSC packet data FIG (SI) FIG (FIDC)	ACTION, CONFIG, FIGFILE, INFORMATION, SUPERVISION	Seamless dynamic re-configuration initiated by remote SP (UE), FIG file-based signalling, status and error messages
3	Advanced		RESOURCE	Resource allocation and consistency checks, more status and error messages
	Options	Processing of other stream types: FIB, PMC, FIG(CA)	FIBGRID Reconfiguration enforcement of UE by DE	Selection of physical interfaces according to STI standard by both users and implementers

The advanced STI Level 3 additionally provides for ensemble coordination by means of RESOURCE messages. Nearly the full functionality of STI-C(LI) has to be realised to be compliant. Upstream entities working at the same level get useful information via STI-C to guide their specification process and to make it right first time. At downstream entities consistency checks will be carried out on all directly connected upstream entities based on the respective resources allocated. This concerns supervision of used stream rates and transmission capacities as well as the checking of FIG contents. Severe interference between service providers can thereby be excluded.

For example, this relates to service and service component identifiers used in a couple of FIGs to signal service labels, programme type, announcement support and so on. Also linkage and announcement provision data (e.g. cluster Id, announcement type and, if applicable, sub-channel Id) should be checked. In case of errors or inconsistencies, the generation of STI-C(LI) error messages takes place, for instance STERROR DEF messages pointing out respective stream Id and error type.

Optional functions which are beyond the scope of the STI level definition mainly concern:

- Ensemble output of time, respectively frame, window related FIGs as needed for low-power-consuming receivers dealing with paging or emergency warning systems or for entitlement messages to be signalled in FIC to support conditional access.
- MSC sub-channel contributions, as PMC, which need to be pre-processed before mapping into ETI frames.

- Use of FIB instead of FIG streams leading to specific FIC assembling requirements.
- Enforcement of upstream entity reconfigurations by the downstream entity.

These functions, if needed, have to be negotiated between users and implementers individually because they cannot be assumed as supported by STI devices in general.

STI levels should be assigned to single interfaces (input and/or output) of STI devices. Connecting interfaces of different levels by an STI connection is possible. In order not to lose functionality, at the paths from the leaf entities to the root entity the following rule should apply to each point-to-point STI connection: the STI level of the upstream entity should be lower than or equal to the STI level of the downstream entity. In case this does not hold and the upstream entity uses an STI-C(LI) message not implemented at the downstream side, the downstream entity should answer with a message informing about the reception of an unknown message.

6.2.5 Integration of Non-STI Service Providers

DAB collection networks according to Figure 6.10 are exclusively based on STI and therefore do not take into account service providers not compliant with STI. Integration of these so-called non-STI service providers supports migration to DAB and is motivated by the following requirements.

At least in the introductory phase of DAB there are broadcasters who intend to start DAB transmission without substitution of available but not STI-compliant equipment. They will accept some operational restrictions instead. Secondly, in special cases of cooperation among ensemble providers the ability to extract service components from one ensemble output data stream (ETI, see section 6.3) and insert them into another ensemble could be useful.

A principal solution can be based on splitting up both the control and data parts included in the STI. The non-STI service provider owns no STI control channel and delivers only the service component data, in the simplest case only one DAB-formatted audio stream according to [EN 300401], using arbitrary physical interfaces supported at the input of the receiving downstream entity. No FIG streams can be delivered. Necessary configuration and service signalling data have to be provided by the downstream entity using its local control interface. In other words, the downstream entity completely takes over ensemble control including the signalling of service information as proxy for a non-STI service provider. This is close to support of service providers at STI Level 1.

Surely some restrictions will be imposed regarding regular operation where service and ensemble provider typically belong to different organisations and where an ensemble provider will not normally be able to perform frequent and timely coordinated changes in service signalling or dynamic reconfigurations on behalf of a remotely situated nonSTI service provider. Therefore, application of this approach will be restricted to special cases of static kinds of operation without dynamically signalling service information.

The network example given in section 6.4 shows that the approach described above has been implemented successfully.

6.2.6 Advanced Features and Further Development

In the early stage of DAB introduction, not all features defined in the respective standards are of the same importance and will be implemented immediately. The most important features related to collection networks have been described earlier, also taking into account different STI implementation levels.

Based on the changes in the broadcast landscape in general, and experiences gathered on collection networks, implementations are developed with respect to three categories:

- use of advanced features already standardized;
- evolution of standards, especially the STI standard;
- evolution outside DAB standardisation.

The first category mainly comprises features such as:

- extended use of satellites in the collection networks (multicast of STI data);
- dynamic change of local services and respective signalling of local service areas;
- dynamic post-processing of received signalling data by the ensemble provider, for example to form ensemble preview on programme types;
- use of the FIC overflow channel (Auxiliary Information Channel, AIC);
- provision for time window related FIC output (by paging, conditional access, EWS).

Evolution of the STI standard could mean for instance:

- acknowledgement of every transaction executed to support closer supervision;
- additional information messages, for example to ask for FIG files which are currently active;
- provision of ensemble-related global signalling data as, for instance, table, country and regional definitions to upstream entities thereby supporting specification of related FIGs;
- support for cross-signalling between cooperating ensemble providers or even cascading of ensembles;
- standardisation of conformance tests for STI devices.

All in all, open and reliable system solutions providing easy and seamless adaptation to the embodying infrastructure, as well as offering trade-offs between functionality and expenditure will be developed stepwise.

6.3 The Distribution Network

6.3.1 The Ensemble Transport Interface (ETI)

Originally defined within the Eureka 147 project, the ETI was published by ETSI [EN 300799]. It is intended to be used for the distribution of ensemble data from the ensemble multiplexer to the transmitting sites of the single frequency network (SFN).

The ETI is conceptually based on layers differentiating between the logical interface (LI) and, at the physical layer, network-independent interfaces (NI) and network-adapted interfaces (NA). For physical transport in public telecommunication networks, it has been decided not to exceed the bit rate of 2 Mbps with regard to cost efficiency. Consequently, the process of convolutional encoding, which leads to higher ensemble data rates of 2448 kbps (DAB transmission mode III) or 2432 kbps (DAB mode I, II, IV), must be shifted from the ensemble multiplexer to transmission sites. The data to be broadcast is transmitted un-coded via ETI and normally another means of error protection is needed. Forward Error Correction (FEC) based on Reed Solomon (RS) block coding has therefore been defined in ETI(NA). In the following, the different ETI layers will be described in more detail.

The frame structure of the logical data interface ETI(LI) is presented in Figure 6.12 and corresponds to those of STI-D(LI) as presented in Figure 6.6. As STI-D(LI), the ETI(LI) frame carries data that are related to the 24 ms Common Interleaved Frame (CIF) formed in the channel encoder at the transmission site.

The frame characterisation field comprises the lower part of the complete frame count (modulo 250, i.e. at 6 s periodicity) and a flag to point out that FIC data are present at the beginning of the main stream data part. The amount of FIC data, if present, is determined by the DAB mode signalled as well (three or four FIBs with 32 bytes each depending on the transmission mode). Further on, the number of streams contained, and the overall frame length, is described in the frame characterisation field. The frame phase (FP) consists of a modulo 8 counter, incremented at each frame, and controls the TII insertion in the channel encoder. On starting the ensemble multiplexer it must be ensured that FP zero is aligned to CIF count zero.

Each stream to be broadcast in the MSC is specified by its single stream characterisation (SSTC) which thereby commands the channel encoder, see section 6.3.3. As depicted in Figure 6.12, the start address in MSC (in CU, 0 . . . 863), the sub-channel identifier as well as service component type and protection level to be applied by the channel encoder are given in addition to the length of the respective stream data in the MST field. Of course, data included in the SSTC have to be compliant with those signalled via FIC (i.e. the MCI).

In contrast to STI-D(LI), only streams constituting complete MSC sub-channels can be transported using ETI(LI).

The subsequent end of header (EOH) field comprises not only a CRC for error detection but also 2 bytes of the multiplex network signalling channel (MNSC) to be used for management purposes (see section 6.3.3).

Frame data are completed by a further check sum of the MST contents and the ETI(LI) time-stamp. This time-stamp is for managing delay compensation in the transport network and defines the notional delivery time of the frame at the input of the channel encoder. Time is represented as (always positive) offset to a common time reference, mostly the one pulse per second (1 pps) of GPS. Time resolution is 61 ns. But, in most cases, accuracy will be determined by the accuracy of the time reference. An additional time-stamp is defined at the NA layer (see below).

At the physical layer, ETI(NI) is defined without error protection and therefore is only applicable for restricted use, such as for local connections or test purposes. To form an ETI(NI) frame, frame sync pattern and error status (analogous to STI generic transport

FC	STC	EOH	MST	EOF	TIST
Frame characterisation	**Stream characterisation**	**End of header**	**Main stream data**	**End of frame**	**Time-stamp**
– Frame count – FIC flag – Number of streams – Frame phase – DAB mode – Frame length	per stream (SSTC): – Sub-channel Id – Start address in MSC – Type and protection level – Stream length	– MNSC – CRC	– FIC data (if FIC flag set) per stream (MSC sub-channel): – Stream data	– CRC	– Time-stamp

Figure 6.12 Frame structure of Ensemble Transport Interface ETI(LI)

frame, see Figure 6.8) as well as frame padding bytes (according to the bit rate used) are added to the ETI(LI). The interfaces ETI(NI,V.11) and ETI(NI,G.703) are defined in [EN 300799] to support RS432 and PDH-based connections respectively.

For operational distribution networks, ETI(NA) is normally used owing to the provision of FEC. The interfaces defined in [EN 300799] primarily concern two versions of interfaces compliant to G.704, but adaptation to PDH based on the first hierarchical level of 1544 kbps instead of 2048 kbps is given as well.

With respect to G.704, the mapping of 24 ms ETI(LI) frames into the corresponding multiframes of PDH is defined. There is a trade-off between redundancy used for FEC and the maximum useful bit rate available.

The first version, ETI(NA,G.704)5592, provides for 5592 bytes of useful data (ETI(LI) data) corresponding to 1864 kbps. Further capacity up to the maximum bit rate of 1920 kbps is occupied by 40 kbps allocated to FEC and 16 kbps used for management and supervision purposes. These data include block counts used for frame synchronisation, a further time-stamp related to an NA link and defining the frame delivery time at the output of the respective converter from the NA layer to the NI or LI layer as well as a Network-adapted Signalling Channel (NASC) (see also section 6.3.3).

The respective figures of the alternative, the ETI(NA,G.704)5376, are 1792 kbps for LI data, 112 kbps for FEC and again 16 kbps for management.

6.3.2 Network Architecture

According to [EN 300799], the DAB distribution network is unidirectional and logically based on point-to-multipoint connections. Figure 6.13 outlines this principle.

Because the radiated signals in the SFN have to be synchronised not only in frequency but also in time with tolerances of a few microseconds, automatic delay compensation is one of the most important issues. Therefore GPS receivers are usually located at each site involved to cater for the common time reference. Using the ETI(LI) time-stamps, it can be ensured that the overall delay throughout the transport network is always the same for every path between the ensemble multiplexer and a transmission site. Together with the information about the constant transmitter delay the ensemble multiplexer is able to evaluate the right time to schedule service providers and to signal time in FIC with regard

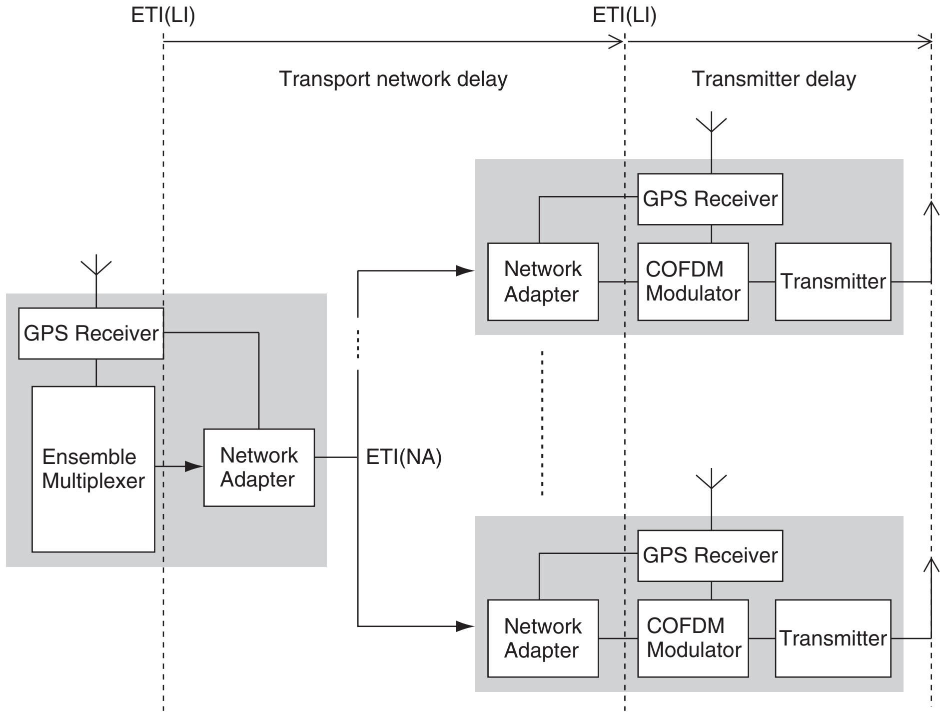

Figure 6.13 Typical DAB distribution network

to absolute time outside of the DAB network. See Chapter 7 (section 7.6.3) for details about delay definitions and treatment.

Network adapters as depicted in Figure 6.13 are needed to convert ETI(LI) into ETI(NA) and vice versa. This functionality is often integrated in the respective devices. The additional NA time-stamp in principle allows for the retiming of NA links, for example in the case of cascading ETI devices. Although support for distributed multiplexing would be useful in some special scenarios, this leads to considerably extra effort regarding synchronised actions and signalling among the sites involved and is therefore not considered further here.

The transport network in between the ETI network adapters can of course make use of further bearer services than PDH as long as G.704 payload can be transported transparently. For instance, additional terminal adapters to comply with SDH or ATM allow for the use of the corresponding networks. Satellite transmission or radio links are also used alternatively to terrestrial connections.

6.3.3 Network Operation

In the basic configuration, operation of the distribution network has to be performed fully automatically under control of the ensemble multiplexer. In contrast to the collection network, control is straightforward without response from transmission sites via ETI due to unidirectional communication. The basic configuration of the distribution network can be taken from the ensemble multiplexer using the signalling channels (MNSC, NASC, see also section 6.3.1) to channel encoders.

The MNSC provides for 2 bytes per 24 ms frame, that is 667 bit/s at maximum. Messages for frame synchronous and asynchronous signalling are defined formally in Annex A of [EN 300799]. Frame synchronous signalling can be used to transfer time information. Important applications of asynchronous signalling concern predefinition of the individual TII and transmitter offset delay (see also section 7.6.3).

In the case when ETI(NA) is applied, the NASC can be used additionally to communicate between ETI(NA) network adapters. It has a capacity of 24 bytes per 24 ms, that is 8 kbps. As for the MNSC, frame synchronous as well as asynchronous signalling have been defined in general, but no application is predefined so far.

During regular operation dynamic delay compensation and especially reconfiguration of the MSC controlled via ETI(LI) have to be managed automatically. Delay compensation requires careful definition of delay figures and time reference in the network (see above) and will then work reliably.

Dynamic reconfiguration needs control of both the channel encoder and the receiver operated in the coverage area of the respective SFN. Normally, changes have to be accomplished inaudible to the listener and therefore require careful coordination.

Figure 6.14 outlines how this can be performed based on frame-related actions carried out by the ensemble multiplexer, see also [EN 300401] and [EN 300799].

The signalling of multiplex reconfiguration starts in advance in the so-called preparation phase lasting up to 6 seconds. During this time the ensemble identification (FIG 0/0) is extended by a byte signalling the instant of planned reconfiguration in terms of the frame count (occurrence change). Additionally the type of reconfiguration (sub-channel or service organisation or both will be changed) is signalled by this FIG at each fourth frame.

In parallel, the current and the next MCI will be signalled at least three times to the receiver using the respective FIGs marking its scope by means of the current/next flag. In particular, only the relevant part of the next MCI, which depends on the type of reconfiguration in turn, is required to be signalled. For instance, only the next subchannel organisation FIGs (0/1) has to be sent in case the reconfiguration is restricted to that part of the MCI.

Service information related to the next configuration can also be optionally provided in advance during the preparation phase.

The channel encoder is controlled by means of the SSTC included in the header of the ETI(LI) frame. As depicted in Figure 6.14, it is important to note how a certain sub-channel changes in the course of reconfiguration. Owing to the time interleaving process, stream data delivered via ETI(LI) will be spread over 16 consecutive CIFs formed by the channel encoder. Therefore, sub-channels which will be removed or reduced in capacity at reconfiguration instant must be changed in ETI(LI) 15 frames before then.

The ensemble multiplexer has to ensure this by communicating with the respective service provider(s) and taking into account resource allocation. Bit error measurements aimed at single sub-channels, full transmission channel or telecommunication lines are usually based on pseudo-random binary sequences [EN 300799].

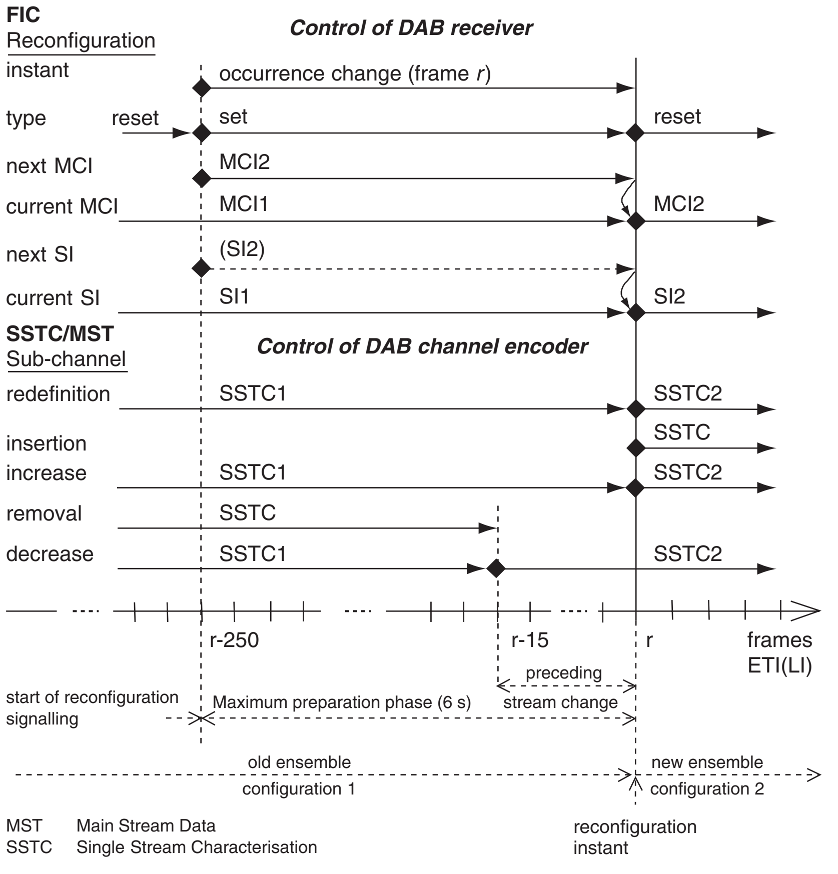

Figure 6.14 Commanding the ensemble reconfiguration via ETI(LI)

6.4 Example of Implementation

As shown in previous sections, a good design of the entire network for the conveyance of all necessary contributions is a prerequisite for successful DAB operation.

In 1999 a field test in Berlin proved that the designed system and its components were working properly. On the basis of that field test a simple example of implementation focusing on the more challenging collection network will be presented in the following sections, see also [Nowottne, 1998] and [Peters, 1999].

6.4.1 Operational Scenario

In the trial, STI service providers had to be allowed to operate their own service contributions and supporting signalling. The extent of the latter was comparable to FM/RDS, that is, both static information such as service labels, programme types and announcement support, but also dynamic changing information such as supporting

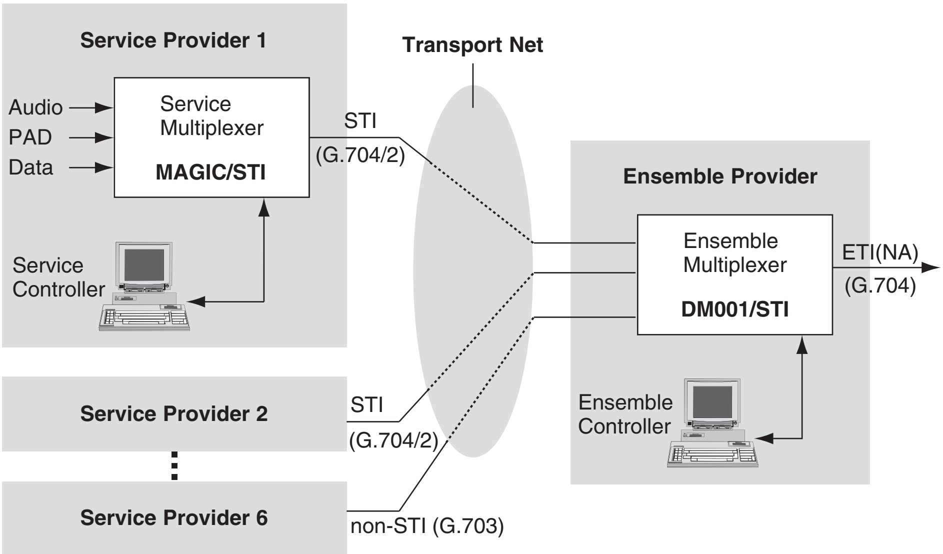

Figure 6.15 Structure of the collection network

traffic announcements via STI. Decentralised initiation of reconfiguration was needed to start broadcasting. To prevent failures, all specifications from the STI service providers had to be based on STI resource allocations provided via STI by the ensemble multiplexer and had to be supported by appropriate tools. Additionally, non-STI service providers had to be integrated into the collection network.

Figure 6.15 shows the STI-based collection network as used in the field test. For instance, two STI service providers operate audio encoders MAGIC/STI for the generation of audio streams, optionally including insertion of PAD. Other service providers feed their service components to the ensemble multiplexer without using STI. Moreover, externally provided packet mode data stream, delivered in STI-D format, can be fed to the ensemble multiplexer directly or inserted by the MAGIC/ STI service multiplexer. The MAGIC/STI interface to the ensemble multiplexer corresponds to STI(PI,G.704/2) and comprises both STI-D and STI-C with transport adaptation. As the ensemble multiplexer the DM001/STI by Rohde & Schwarz was used. MAGIC/STI as well as DM001/STI are connected via RS232 with PCs, working as service controller and ensemble controller, respectively.

As required, the system is also able to cater for non-STI service providers. In this case the ensemble provider takes the role of a proxy relating configuration and signalling.

6.4.2 *The Service Provider Profile*

The efficient treatment of resource allocation and consistency checking in an operational collection network is based on so-called Service Provider Profiles (SPPs). This has to be done by the ensemble provider in arrangement with service providers.

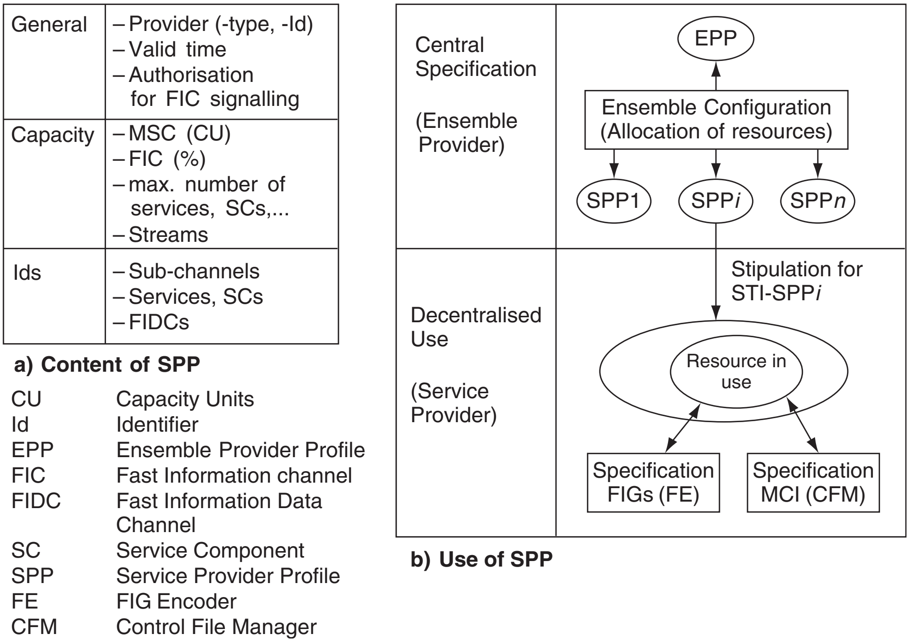

Figure 6.16 Management by means of service provider profiles

Each service provider is allowed to manage services independently within the constraints of the SPP. It is the task of the ensemble multiplexer to supervise the service provider's actions to ensure impact-free running. The management processes are performed and monitored by means of the STI control part, on which the service controller and ensemble multiplexer communicate.

Figure 6.16a outlines the contents of the applied SPP, as stored within the database of the ensemble multiplexer. These data comprise information on the service provider's address and basic entitlements, about capacity and coding resources, as well as other parameters.

Figure 6.16b gives an idea of the way it works. It has to be distinguished between central specification and decentralised application. SPPs specified and stored at the ensemble provider are downloaded to the service providers and define their frame of action. The ensemble multiplexer supervises their operation, responds to change requests and carries out the changes if possible. Otherwise it rejects requests or reacts with error messages.

6.4.3 Equipment in Use

Two basic devices form the hardware basis in the 1990s of the introduced collection and distribution network: the MAGIC/STI Audio Encoder and Service Multiplexer by Fa.

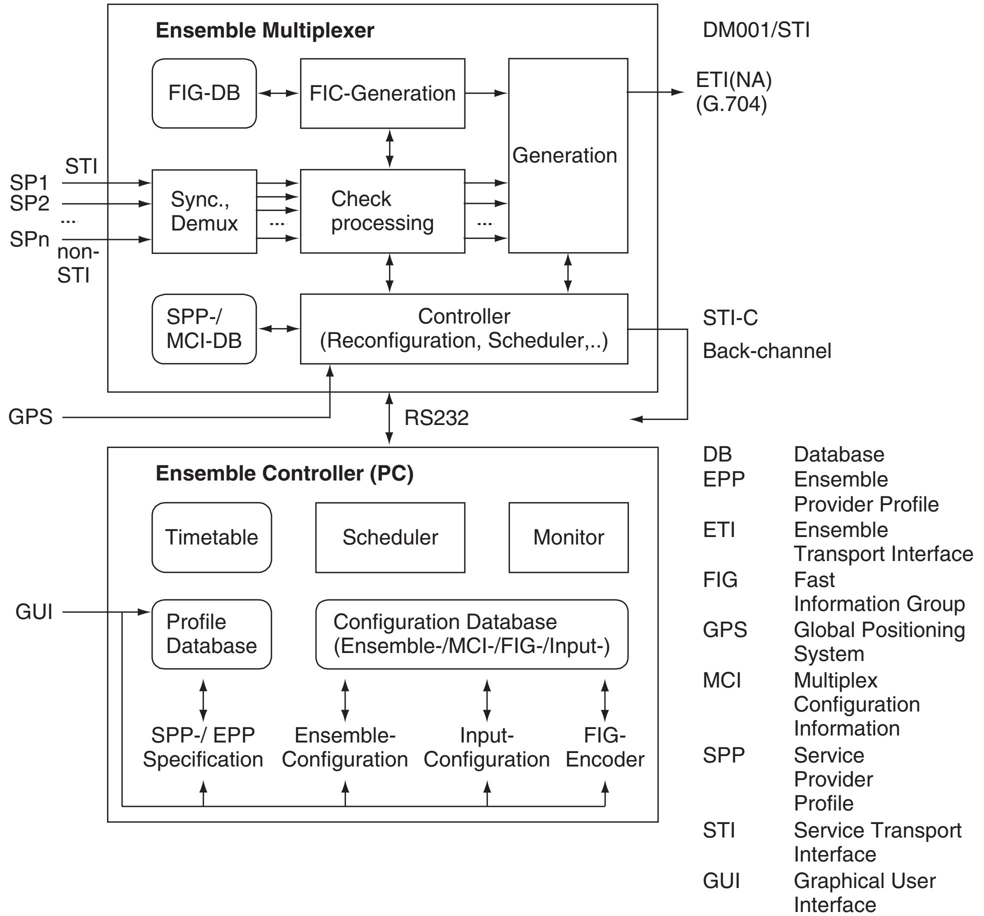

Figure 6.17 STI solution at the ensemble provider

Audio Video Technologies and the DAB Ensemble Multiplexer DM001/STI by Fa. Rohde & Schwarz complemented by their Frame Decoder FD1000 for monitoring and analysis of both STI and ETI.

The structure and components of the Ensemble Multiplexer DM001/STI are depicted in Figure 6.17. The common clock and time reference is provided by an external GPS receiver. The DM001/STI is connected via an RS232 interface to its ensemble controller. Using this computer-based device the ensemble controller software, running under WindowsNT®, supports STI-C-like communication with the multiplexer.

It is possible for the operator to define and download complete ensemble configurations each consisting of an EPP (Ensemble Provider Profile), several SPPs and all other ensemble-relevant parameters. To illustrate operation, the steps necessary to start broadcasting from the very beginning at the ensemble multiplexer are listed in the following:

1. Basic configuration of the DAB ensemble multiplexer, including:
   - definition of DAB mode and external clock source (2.048 MHz from GPS);
   - definition of transport network and transmitter delay;
   - definition of ETI type, for example ETI(NA,G.704)5592.
2. Specification of an ensemble configuration consisting of one EPP and several SPPs:
   a) Specification of the EPP:
      - allocation of ensemble identifier (EId) and ensemble label;
      - stipulations for the FIC (FIC capacity and FIGs allowed to send);
      - specification of a FIG file including FIGs for signalling of ensemble information.

   b) Specification of SPPs for all service providers:
      - allocation of service provider identifier (SPId);
      - allocation of resources (MCI and FIC capacities, coding ranges, streams, etc.);
      - stipulations for the FIC (FIGs allowed to send, announcement parameters);
      - specification of physical input;

   c) Additional specification of MCI and SI (for non-STI service providers only):
3. Manual or scheduled download of the ensemble configuration specified (including valid time).

The DM001/STI will then generate the ETI stream according to the specifications downloaded. A reconfiguration will be carried out and the STI-C(TA) control connections to the STI service providers will be opened. After that, the STI service providers are able to define their virtual ensemble and to initiate reconfigurations of their own.

Monitoring of the ensemble multiplexer can be done by means of the status windows at the ensemble controller. Additionally a log file will be generated to store messages about all relevant events. A message browser for filtered evaluation of the log file is also provided.

Figure 6.18 displays the conditions at the service provider side. The MAGIC/STI system provides not only sufficient audio encoding functionality but also realisation of service multiplex, that is several services or service components can be handled.

MAGIC/STI basic units can be expanded by up to seven further encoders, if several audio signals are to be encoded at the same location. Encoders are also able to insert PAD into the audio data stream. Audio signals are fed either as analogue signals or as digital AES/EBU signals.

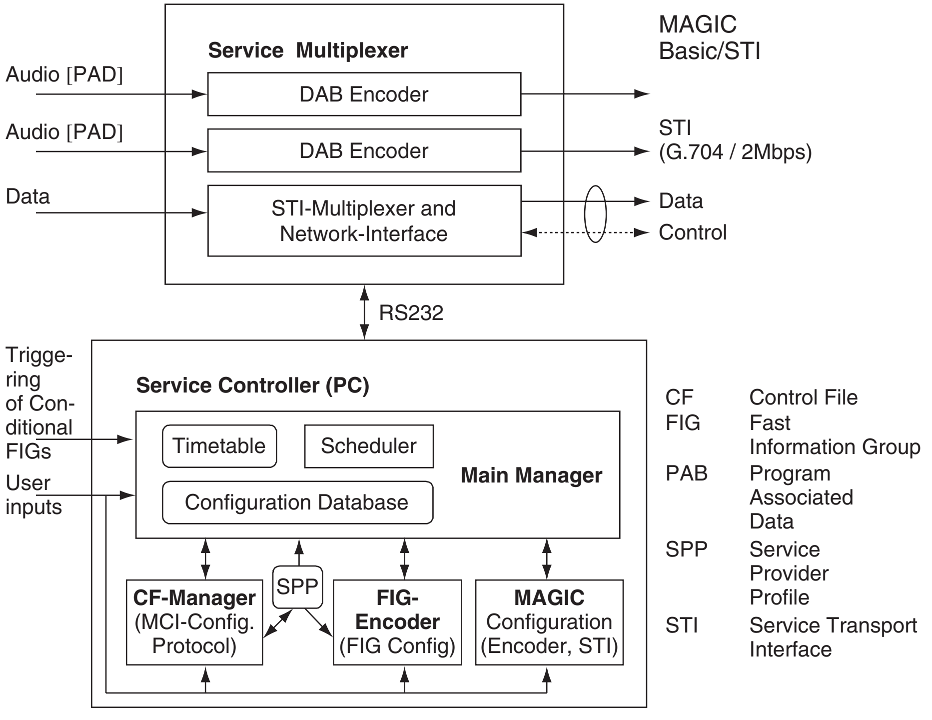

Figure 6.18 STI solution at service provider

The output interface complies with G.703/G.704 (2 Mbps). Additionally data services in packet mode format can be inserted via the STI-D input. Running under WindowsNT®, the service controller software consists of three major modules:

- main manager;
- control file manager;
- FIG encoder.

The main manager controls, in a central function, collaboration with the MAGIC/STI audio encoder(s) and service multiplexer. It accepts inputs from the service provider operator and serves to define and administrate the streams to be handled. It is also used to specify scheduled actions, to establish event-triggered signalling and coordinate collaboration with the control file manager and FIG encoder.

In its database the main manager stores all information relating to configurations and schedules. The control file manager complements the main manager in specifying and monitoring the respective MCI configuration, the specification of the virtual ensemble. Based on the SPP the service provider operator is able to specify sub-channel and service organisation, that is, assignment of bit rates, start addresses, protection levels and so on. Consistency of the defined configuration is checked immediately.

The task of the FIG encoder is to provide FIG data for insertion into the FIC of the DAB signal. Since FIC data originate at the service provider and at the ensemble provider as well, the FIG encoder can be used by both sides equally. At the service provider side the FIG encoder supports the encoding of static and quasi-static FIG data. Resulting FIGs are embedded within FIG files and transported via STI-C to the ensemble provider in order to be stored there and passed into the ETI data stream if activated. Conversely, FIG data from the ensemble multiplexer database can be recalled and depicted for monitoring purposes. In order to initiate signalling via the FIG file mechanism, the service provider software can be coupled to the studio control process. For instance, announcement switching can be event triggered accordingly.

Furthermore, precautions have been implemented to adapt the whole network solution to a separate quality supervision system of the DAB network operator.

7

The Broadcast Side

7.1 General

This section explains the broadcast side of DAB without touching upon the issue of signal delivery to the different transmission sites of the network since this was covered in the previous chapter. DAB transmission networks can have national, regional or local coverage. Depending upon specific national, geographical and financial requirements, Band III or L-Band frequencies may be used for these networks.

The rest of this chapter is structured as follows: Section 7.2 gives an introduction to DAB networks, highlighting the major differences of DAB networks compared to conventional FM networks and explaining the concept of a single frequency network (SFN). The particularities of SFNs are then described in section 7.3. The equipment needed on the transmitter site to set up SFNs and the associated specifics are presented in section 7.4. Section 7.5 deals with the transmission channel limitations due to RF signal propagation, illustrates the reasons behind the choice of 1.5 MHz bandwidth for the DAB system and describes the propagation model used with the DAB system. The following section 7.6 treats the aspects to be considered when planning networks with SFN coverage and section 7.7 discusses the issues of coverage evaluation and monitoring for SFNs. Finally, aspects of frequency management and frequency allocation for DAB networks are covered in section 7.8.

7.2 Introduction to DAB Networks

7.2.1 Difference between FM and DAB Networks

Planning of transmission networks for FM broadcasting is traditionally based on the concept of multiple frequency networks (MFNs). In an MFN, adjacent transmitters radiate the same programme but operate on different frequencies to avoid interference of the signals where the coverage areas of different transmitters overlap. Basic FM receivers cannot cope with interfering signals from other transmitters of the same network using the same or nearby frequencies. Coverage planning for an FM network requires frequency planning for the different transmitter sites, to optimise use of the scarce resource: RF frequencies.

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

DAB in contrast allows single frequency networks (SFNs), where all transmitters of the network transmit exactly the same information on the same frequency. The main condition for a working SFN is that all transmitters are synchronised to each other in frequency and fulfil certain time delay requirements which will be explained later in this chapter. Coverage planning for a DAB network requires time delay planning between the different transmitters instead of frequency planning as in the case of FM. The SFN capability of DAB allows complete coverage of very large regions without the receiver having to tune to a different frequency while moving around in the area.

In contrast to FM broadcasting, DAB transmits typically five to seven different programmes in one single ensemble on one frequency and all programmes contained in that multiplex share the same coverage area. Distinction by coverage area is therefore not possible for radio stations whose programmes share the same multiplex.

Although possible in principle, it is not advisable in an SFN to introduce local windows, that is, areas where some transmitters of the SFN radiate a slightly different multiplex to achieve local programme variation. Local windows cause problems for the receiver in the overlap area of the differing programmes of the multiplex since it cannot determine which programme to select. The gain in planning flexibility for local programmes would not compensate for the overall loss of coverage in the network.

7.2.2 Single Frequency Networks

7.2.2.1 Why SFNs are Possible with DAB

DAB is a digital broadcasting system which was developed especially for the challenging transmission characteristics of the mobile radio channel. Typical phenomena of this channel like Doppler shift and multipath propagation with the resulting time and frequency selective fading had to be taken into account while developing DAB. To cope with these problems, the guard interval was introduced between consecutive data symbols, time and frequency interleaving techniques were applied to the data stream, a choice of sub-carrier spacings in the multicarrier modulation scheme was introduced and channel coding techniques were applied to correct for transmission errors. Table 7.1

Table 7.1 Problems of mobile radio transmission systems and their solution in DAB

Problems	DAB Solution
Time-dependent fading (multipath while driving)	Time interleaving
Frequency-dependent fading (stationary multipath)	Broadband system with frequency interleaving
Doppler spread (speed dependent, while driving)	Sub-carrier spacing increased as a function of the transmission frequency
Delay spread (due to multipath)	Guard interval (allows SFNs)
Transmission errors	RCPC (Rate Compatible Punctured Convolutional) codes and Viterbi decoding to reconstruct the original bit stream

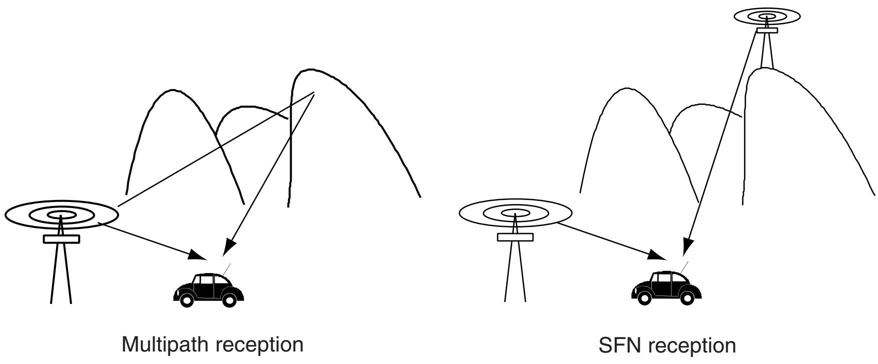

Figure 7.1 SFN capability of DAB

shows the most important problems of mobile radio transmission systems and how they are alleviated in DAB.

The DAB system is a very robust and frequency economical transmission system which enables correct decoding of information despite Doppler spread and multipath reception. The effect of multipath reception is depicted graphically in Figure 7.1. Both the direct signal from the transmitter and reflected signals arrive at the antenna of a receiver.

All signals contain identical information but arrive at different instances of time. Owing to the measures previously described, the DAB receiver is able to cope with these multipath signals.

Going a step further, it is now irrelevant for the receiver whether the delayed signals were originated by the same transmitter or come from another transmitter that transmits exactly the same information synchronised in time as shown in the second part of Figure 7.1. This means that DAB allows coverage of any area with a number of transmitters that transmit the identical programme on the same frequency. Such broadcast networks are called single frequency networks.

The SFN capability of DAB transmission networks can be seen as an extra benefit which is an implicit result of the initial requirement during the development of DAB to cope with multipath phenomena typical of mobile radio reception.

7.2.2.2 Power Economy of SFNs

Owing to the properties of the system, DAB receivers can use all signals received in an SFN in a constructive manner. This works as long as all signals arrive within the guard interval. Signals with longer delays create self-interference problems in the SFN and must be avoided by careful network planning.

Moreover, SFNs also bring the benefit of space diversity due to the fact that transmitters are located at different locations in the network. The diversity effect, resulting from the fact that the probability of simultaneous shadowing in the presence of several signals is much lower than the probability for shadowing for a single signal, contributes an additional statistical network gain.

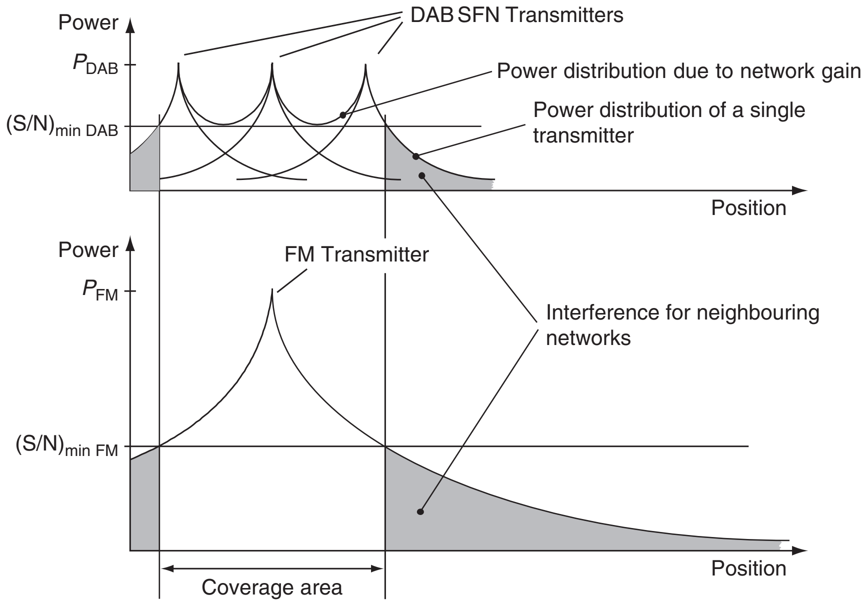

Figure 7.2 Power economy of DAB SFNs in comparison to FM

DAB networks are therefore very power economical. The advantages offered by SFN operation and the properties of the digital transmission system itself, allow lower transmitter powers compared to FM for the same coverage quality. Power savings can be as high as 10 dB. Figure 7.2 shows this effect in a qualitative manner. To cover the same area with one programme on the same frequency, FM needs one high-power transmitter, whereas DAB with an SFN network with several transmitters needs much lower transmitter power in total. Another effect of the DAB SFN compared to FM networks is the much lower spill-over resulting from the steeper field strength slope at the edge of the coverage area as illustrated in Figure 7.2 which reduces unwanted interference in neighbouring networks.

7.2.2.3 Frequency Economy of SFNs

The fact that SFNs allow coverage of large areas occupying only a single frequency in the spectrum results in highly frequency economical planning possibilities for complex broadcast landscapes. The SFN technology also allows successive improvement of coverage quality without having to re-plan frequency allocations. Coverage problems within the network can be solved by simply putting up additional transmitters. The additional planning that must be done is to check the timing constraints of the SFN to avoid violation of the timing budget given by the guard interval of the chosen DAB transmission mode. Note that extra care must be taken to avoid an impact on reception in distant parts of the network because of spill-over from the additional transmitter during abnormal propagation conditions. Frequency planning for DAB is dealt with in detail in section 7.8.

7.3 Particularities of Single Frequency Networks (SFNs)

7.3.1 Review of COFDM Principles

As stated earlier in the chapter, COFDM was originally intended to provide successful reception in multipath propagation conditions arising from reflected signals, but it works equally well for reception of multiple transmitters carrying the same signals. This gives rise to the possibility of an SFN, in which all transmitters carry the same information at the same time (or nearly the same time). Important factors for successful implementation of SFNs are the accuracy of the frequency and the timing of each transmitter. In addition, the length of the guard interval is important for an SFN implementation, because it influences the allowable range of transmitter spacing (see also section 7.6.6).

7.3.2 Time and Frequency Synchronisation

For an SFN to operate effectively, the transmitters must deliver the DAB signal to the receiver at the same time, or nearly the same time, and at the same frequency.

Frequency errors between the transmitters cause a loss of orthogonality between the received carriers, and also reduce the receiver's tolerance to the Doppler spread effects experienced in mobile reception.

Timing errors between transmitters erode the guard interval of the composite received signal, and can therefore disrupt the performance of the SFN.

For these reasons, transmitter networks are normally specified to keep transmitter frequencies within 1% or so of the carrier spacing, and timing within a few per cent of the guard interval. In order to achieve this, an independent and ubiquitous time and frequency reference is required. Global Positioning System (GPS) receivers are commonly used for this purpose. These receivers offer time and frequency references, typically 1 pulse per second (pps) and 10 MHz signals, with an accuracy well in excess of that required for DAB SFNs. The 1 pps signal is used to define the transmission time of the data, and the 10 MHz signal is used as a reference for LO synthesisers that determine the final radio frequency.

7.3.3 Network Gain

If two or more transmitters serve the same area, their signal strengths are, in general, not strongly correlated. The signal strength of the transmitters varies with location, but because the signals are not strongly correlated, an area of low signal strength from one transmitter may be 'filled' by higher signal strength from another transmitter. In RDS (Radio Data System, used in FM networks to carry additional information), this is exploited by allowing the receiver to retune to alternative frequencies carrying the same programme if reception of a particular frequency is poor, which is called frequency diversity. However, unless two receiver front-ends are used, the receiver has to retune without prior knowledge of whether the alternative frequencies will offer greater signal strength.

In a DAB SFN, retuning is not necessary. For multiple transmitters on the same frequency, it remains true that an area of low signal strength from one transmitter may be 'filled' by higher signal strength from another transmitter. This is a form of 'on-frequency' diversity, in which the receiver does not have to retune, although it may adjust its synchronisation to make best use of the available signals. Moreover, beyond this statistical effect, with DAB the individual field strengths add up constructively to form a wanted sum field strength which exceeds the field strength of each individual signal.

Another way of looking at this phenomenon is to consider the aggregate strength of the composite signal from a number of transmitters. This varies less with location than the signal strength from any of the individual transmitters, or in statistical terms, the variance of the strength of the composite signal is lower.

This effect offers a major advantage. In a multiple frequency network, a number of transmitters may provide signal strength to an area without providing adequate coverage. This is less common in SFNs because of the tendency of two or more transmitters to fill each other's coverage deficiencies. This results in the coverage of an SFN being greater than the sum of the coverages of its individual transmitters, and is often known as 'network gain'. A detailed discussion of the additive and statistical components of network gain can be found in [BPN 066].

7.3.4 Self-Interference in DAB Networks

Nearby transmitters on the same frequency have a constructive effect, but in a large SFN, the more distant transmitters, whose signals may arrive outside the guard interval, can act as interferers. This effect is called self-interference. Although it is possible for signals arriving just beyond the guard interval to contribute some useful energy, they will at the same time contribute to the interfering power and this latter effect will in most cases be predominant. This is a complicated issue influenced by the design of the receiver, and therefore the effects are somewhat variable. More information about signal summation, signal synchronisation and the treatment of constructive and destructive signal components can be found in [Brugger, 2003], [BPN 066].

Transmitter spacing is therefore a factor in network design, but in practice the availability of suitable transmitter sites, topography and population density are also major influences. For Band III SFNs, using Mode I, transmitter spacings are usually somewhat smaller than the distance corresponding to the guard interval (around 75 km). This may result in many transmitters contributing useful energy under favourable circumstances, but in SFNs above a certain size there will always be potential for interference from distant transmitters. Because of this, SFNs have to be planned taking very careful account of the interference caused by transmitters to distant locations, as well as the coverage provided in their immediate surroundings. More information about maximum transmitter spacing as function of the DAB Mode is given in Table 7.10.

7.3.5 Optimised Coverage of SFNs

Although the transmitters in an SFN need to deliver the signals with very precise timing, it is not always necessary for them to be exactly co-timed, and in some circumstances it can be advantageous to offset the timing of particular transmitters by significant fractions of the guard interval. This is particularly true at the extremities of coverage, or where low-power transmitters are used to fill gaps within coverage provided primarily by high-power transmitters. Transmitter timing is a variable in network design and can be used in combination with transmitter powers and directional radiation patterns to optimise the coverage of the network. An example is given in section 7.6.4.

7.3.6 Closure of Coverage Gaps Using Gap Fillers

In conventional MFNs, as used for FM broadcasting, coverage gaps are closed with additional transmitters, which need their individual frequency assigned. This requires careful planning to ensure that incoming and outgoing interference is correctly managed. This is often not economical from the frequency spectrum point of view.

SFNs, however, allow relatively simple filling of areas not well served by the main transmitters, that is, gaps in the coverage area, by installing low power on-channel repeaters located inside the coverage area which operate on the same frequency as the rest of the SFN (see Figure 7.3). Typical under-served areas include zones shadowed by natural or man-made obstructions such as valleys, tunnels and in cities blockages behind tall buildings.

These additional re-transmitters with a typical output power of the order of a few watts are called gap fillers or repeaters. A gap filler is simple to construct and install since it requires relatively small power and can be mounted on a small tower or on the roof of a building. The receiving antenna of the gap-filler should be highly directional with reduced back lobes, while the re-transmitting antenna will generally be tailored to the specific characteristics of the shadowed area. Gap fillers must be located at points in the network where there is sufficient incoming field strength and where the re-transmitting antenna can be directed towards the as yet uncovered area of the SFN.

In the Canadian DAB networks more powerful re-transmitters with an output power of up to 100 W are also used. This type of device is called a coverage extender since it is typically located at the fringe of the network and fires beyond the coverage area of a main

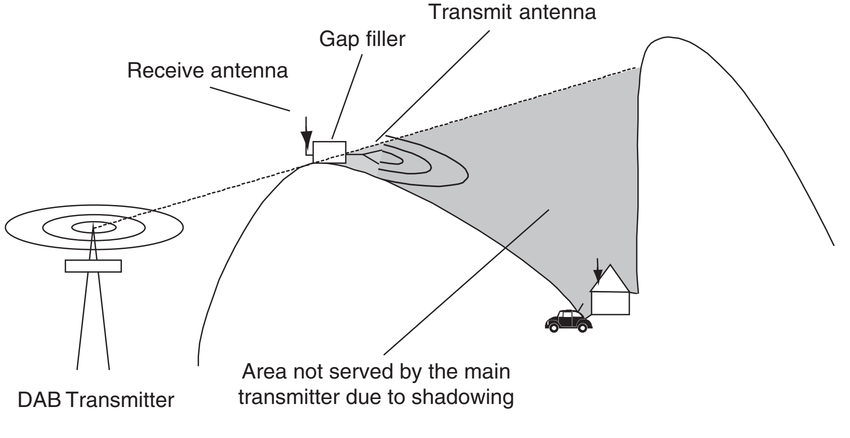

Figure 7.3 Typical application of a DAB gap filler

transmitter. Coverage extenders enlarge the total area of the SFN instead of serving regions not well covered owing to local shadowing as gap fillers do. Regarding planning and synchronisation aspects, the same rules apply to both sets of devices. However, gap fillers must meet more stringent delay and transmission power requirements since they are typically not located at the fringe of the coverage area and their signal can more easily interfere with the signal of the main transmitter(s).

Figure 7.4 illustrates the rule that governs the use of gap fillers within the coverage area of a single main transmitter, rated in this example at 25 kW effective radiated power (e.r.p.). In the case of DAB, this rule is closely related to the size of the guard interval (in Figure 7.4, Mode II is used with a 62 μs guard interval). An omni-directional re-transmitting antenna is assumed at the gap filler to cover the worst case situation. The propagation model of Recommendation ITU-R P.1546 was used for this exercise, assuming flat terrain with a roughness factor of $\Delta h = 50$ m. The domain of operation is under the curves. If the e.r.p. of the gap-filler exceeds the values shown by the curves at a given distance from the main transmitter and for a given gap-filler, an unserved area starts to appear between the main transmitter and the gap-filler due to the presence of the destructive echo in this area.

In some cases, an alternative to gap filling would be to increase the power and/or the antenna height of the main transmitter. Apart from the cost implications, this would also increase interference to co-channel services in other areas, and thus, limit implementation

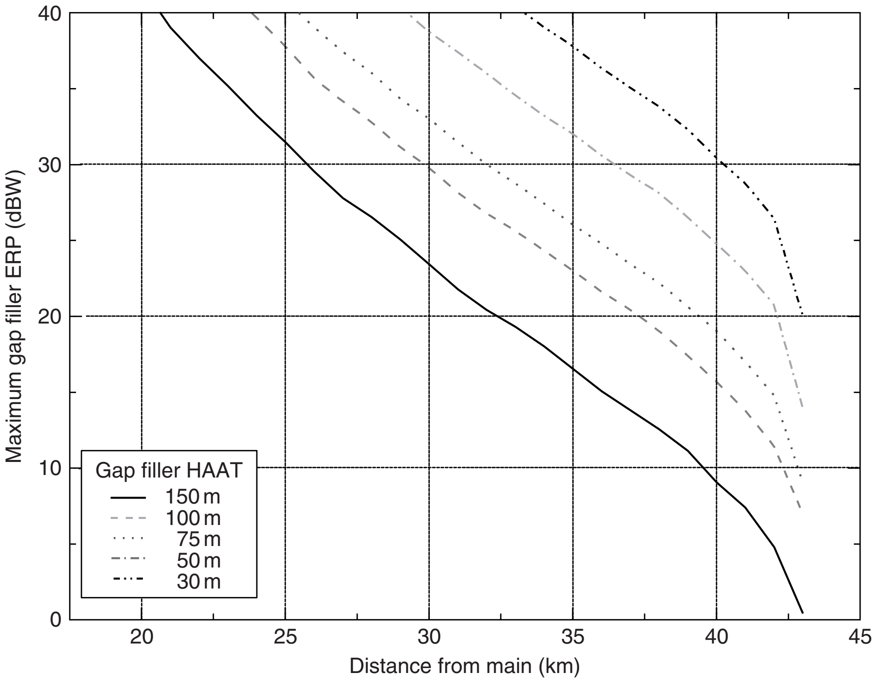

Figure 7.4 Domain of operation of a gap-filler as a function of distance, HAAT and e.r.p.

of such a high-power DAB service, or reduce spectrum reuse efficiency. The use of gap filling transmitters therefore contributes to spectrum conservation.

Gap fillers do not have to be exactly time synchronised to the other transmitters in the SFN. The gap filler simply amplifies the received DAB signal. To achieve good performance the received signal may be downconverted to an IF or even baseband for signal conditioning before it is upconverted again for final amplification and filtering. De- and re-modulation for signal improvement are not done in gap fillers because the processing delay inherent in the DAB system means that the retransmitted signal would lie outside the guard interval irrespective of the DAB mode chosen. When a gap filler is installed, it must be ensured that the transmit and receive antennas are sufficiently decoupled to avoid unwanted feedback and blocking effects. In general, the necessary isolation between receiving and transmitting antennas determines the upper limit for the amplification gain. At higher frequencies such as L-Band, large shadowing buildings in urban areas can provide some of the necessary isolation.

In areas where several DAB blocks are in use, the input stage of the gap filler must be block selective to guarantee that only the wanted signal is amplified. It is also possible that the gap filler could be used to amplify a number of blocks to cover the same gap. Proper filtering will be needed to avoid intermodulation. It should also be noted that in areas with gap fillers in a DAB network, geographical position estimation of the receiver using the TII feature will not be possible for two reasons: first, because the transmitter time delay signalled in FIG 0/22 (TII field) is only valid for directly received signals, and not for signals which suffer from the additional delay due to the signal processing in a gap filler; and secondly, if the signal from the main transmitter and one or more gap fillers is simultaneously received, distinction of the different signal sources will no longer be possible since they will all have the same TII identification. (Gap fillers cannot change the TII information of the signal.) For more information on TII in DAB see section 7.3.7.

7.3.7 Application of the TII Feature in SFNs

DAB allows identification of individual transmitters in an SFN with the TII feature. The TII signal is transmitted every other null symbol to allow the receiver to perform channel state analysis in null symbols without the TII signal. The TII signal consists of a certain number of pairs of adjacent carriers of an OFDM symbol and the actual pattern of the carrier pairs identifies the individual transmitter.

The identification of each transmitter is given by two parameters: the pattern and comb number, also called the main and sub-identifier of a transmitter. FIG 0/22 in the FIC of the DAB signal describes a set of parameters, the TII field, which contains all information necessary for the unique description of a transmitter. These parameters are transmitter identifiers, geographical location of the transmitter and the time offset of the transmitter (see section 7.6.4).

The main identifier is used to describe a cluster of transmitters in a certain region and each transmitter within a cluster has its own sub-identifier. Table 7.2 gives the number of possible main- and sub-identifiers as a function of the DAB mode.

Each comb number identifies a number of carrier pairs of which only half is used in a TII symbol. Which of the carrier pairs are used is determined by the associated pattern

Table 7.2 TII *parameters for different DAB modes*

Mode	Number of Main Identifiers (= diff. patterns)	Number of Sub-identifiers (= diff. combs)	Number of Carrier Pairs Used per Comb
1, IV, II	70	24	4 of 8
III	6	24	2 of 4

number. Since each comb number identifies a unique set of carrier pairs (i.e. each carrier pair is only used by a specific comb), the DAB receiver can simultaneously identify the signals of all transmitters with the same main identifier, that is, pattern number. To distinguish between transmitters with different main identifiers, the sub-identifier must be chosen carefully to avoid ambiguities. The exact relationship between comb and pattern is given in references [EN 300401] and [TR 101497].

The TII feature of DAB allows the receiver to calculate its position if signals from at least three transmitters are received and FIG 0/22 (TII field) is signalled in the FIC [Layer, 1998]. The knowledge of the receiver position can be used for intelligent change of frequency when leaving the coverage area of the network. It can also aid the automatic selection of information, for example only information relevant for the current region is displayed.

7.4 DAB Transmitters

7.4.1 General Aspects

Figure 7.5 shows the block diagram of a DAB transmitter. Each transmitter consists of a number of functional blocks which will now be explained. The ETI output signal from the ensemble multiplexer is delivered to the transmitter site via the DAB distribution network. At the input of the transmitter the signal is buffered and a precise delay is inserted to synchronise the SFN in time. After COFDM encoding the baseband output signal of the COFDM encoder can be subjected to further signal processing for non-linear pre-distortion or crest factor manipulation before it is converted from digital to analogue. After conversion to the analogue domain the signal is upconverted to the desired final radio frequency. Finally the RF signal is amplified and filtered to fulfil the relevant spectrum masks before it is radiated.

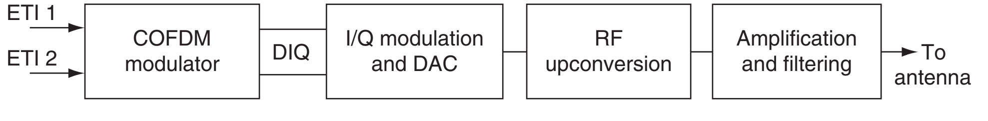

Figure 7.5 Block diagram of a DAB transmitter

7.4.2 *Signal Processing Blocks of a COFDM Modulator*

COFDM modulators usually contain not just the pure DAB signal processing part but also an input stage to process the different variants of the ETI signal and to insert the required signal delay. The output signal of the modulator is either the DIQ (Digital In-phase and Quadrature) baseband signal according to [EN 300798] or an RF signal at a convenient IF or RF if an I/Q modulator is included. The signal processing blocks of a COFDM modulator are shown in Figure 7.6 and described in the following paragraphs.

The input stage strips the ETI signal (both ETI(NI) and ETI(NA), see Chapter 6) down to the ETI(LI) level, the logical interface level, as described in [EN 300799]. Since the ETI signal is transmitted in HDB3 format, it is also converted to TTL level in the input stage. If the modulator has two inputs to allow networks with redundant distribution paths, both inputs are monitored for signal quality in this stage of the modulator and selection of one of the two input signals for further processing is carried out here.

In the delay compensation section the input signal is delayed within a range of zero to sometimes more than one second, typically in steps of 488 ns (488 ns are conveniently available in the system since they are the period of the ETI signal which is delivered at a rate of 2,048 Mbps). For dynamic delay compensation, the time-stamp of the ETI(NI) or the ETI(NA) is evaluated using the 1 pps (pulse per second) signal from a GPS (Global Positioning System) receiver as time reference. The information for an automatic static delay compensation is transmitted in the MNSC (Multiplex Network Signalling Channel) of the ETI. This delay is called transmitter offset delay and can be set for each transmitter individually making use of the unique encoder Id. The user can also set a separately adjustable delay for each input which is referred to as manual delay compensation. Combinations of the different methods of delay compensation are possible. A detailed treatment of the different delays in a DAB network is given in section 7.6.4.

The channel coding block performs all the encoding necessary to achieve a high level of signal robustness and to allow error correction in the case of bad transmission. Energy dispersal, convolutional encoding, MSC (Main Service Channel) time interleaving, MSC multiplexing, transmission frame multiplexing and frequency interleaving according to [EN 300401] are performed in this block.

In the OFDM modulation block, the output bit stream from the channel coding block is mapped on to DQPSK (Differential Quadrature Phase Shift Keying) symbols, before $x/\sin x$ pre-correction for the digital-to-analogue conversion is performed. Finally the

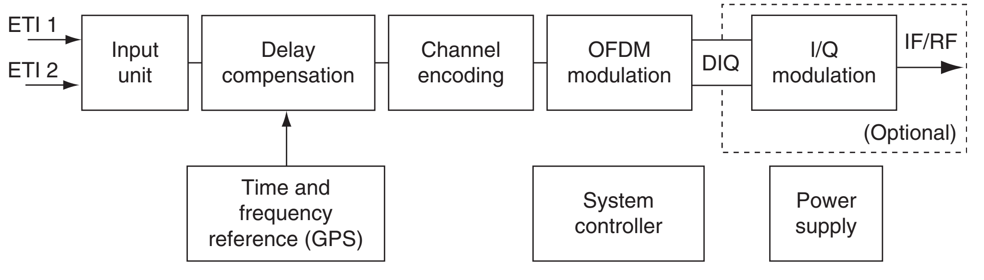

Figure 7.6 Block diagram of a COFDM modulator

IFFT (Inverse Fast Fourier Transformation) with generation of phase reference symbol, TII (Transmitter Identification Information) and guard interval are performed to generate the DIQ baseband signal.

The DIQ baseband signal can be used to perform further signal processing like non-linear pre-correction or crest factor manipulation. To complete the COFDM modulator, a system controller and power supply are needed and an I/Q modulator may be incorporated.

7.4.3 Digital-to-analogue Conversion

The output from the digital processing, and thus the input to analogue processing in the transmitter, is nominally the DIQ signal defined in [EN 300798]. This standard specifies a resolution for the I and Q signals of 8 bits, and a 'clip-to-RMS' ratio for these signals of 12 dB. This specification is intended to allow digital and analogue sections from different manufacturers to work together. However, many manufacturers supply both sections, and internally may use slightly different settings (e.g. higher resolution to reduce the quantising noise floor, or a different clip-to-RMS ratio).

The example transmitter in Figure 7.7 shows individual baseband I and Q chains, each consisting of DAC and baseband filtering, followed by upconversion to a final frequency, amplification, and band-pass filtering.

In this arrangement it is important that DACs (digital-to-analogue converters), filtering and upconversion paths for I and Q signals are very closely matched, in amplitude and phase, across the bandwidth of the baseband I and Q signals. Any imbalance will result in distortion of the final RF signal, resulting in additional in-band components that will raise the in-band noise floor.

Further considerations include the phase accuracy of the in-phase and quadrature local oscillator signals, and the linearity of the upconversion mixers. Low-level

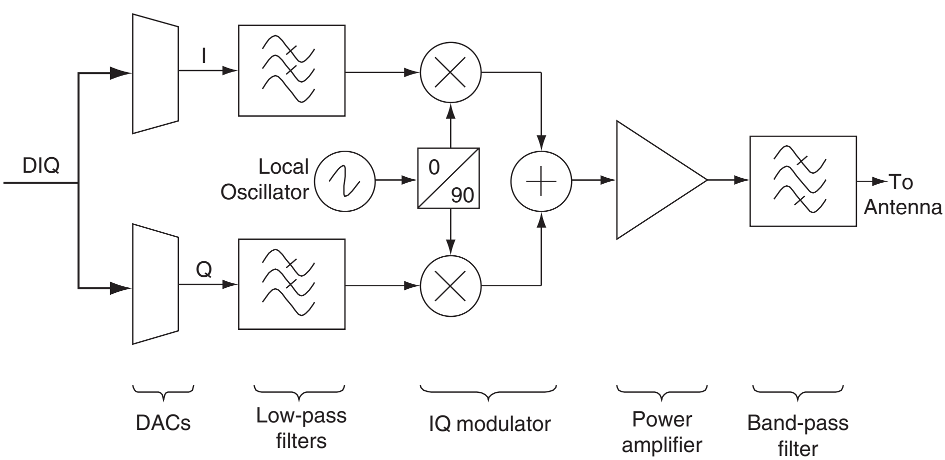

Figure 7.7 Example block diagram of a DAB transmitter with DIQ baseband input

breakthrough of the local oscillator, at the centre frequency of a DAB signal, can be tolerated because this carrier position is not modulated.

Most early implementations of DAB transmitters used this approach, and with careful alignment very good results can be achieved. One of the advantages of the approach is that transmitters at Band III can be implemented by a single up-conversion to the final frequency, avoiding the need for IFs requiring filtering and further upconversion.

An alternative technique, in which the upconversion to a low IF (normally a few MHz) is performed in the digital domain, is illustrated in Figure 7.8. This is also common and avoids the need for closely matched DACs and filters. In addition, highly accurate local oscillator (LO) quadrature can be achieved, and LO breakthrough is eliminated (although this is not a major problem for DAB). However, further upconversion is required in this case.

With the advent of ever faster DACs it is at the time of writing becoming possible to develop direct conversion transmitters as shown in Figure 7.9. Band III transmitters do not need an IF any more; for L-Band, up-conversion from a convenient Band III frequency will still be necessary for a while. The advantage of the direct conversion concept is its simplicity and the minimum amount of circuitry needed. However, LO-pulling and carrier leakage must be addressed in the design. LO-pulling occurs since the oscillator in the frequency synthesiser operates at the same frequency as the power amplifier. Due to the limited reverse isolation, the transmitted output signal couples back into the oscillator thus degrading its performance.

One further factor to be taken into account is the spectral purity of the oscillators used in the transmitter, most commonly expressed as single-sideband phase noise (or 'phase noise' for short). OFDM systems are tolerant of phase noise to some extent, but beyond certain limits suffer from adverse effects.

The first effect is mainly due to oscillator phase noise at frequency offsets equal to, or greater than, the carrier spacing of the OFDM system. This contribution to the phase noise causes leakage of the carriers into one another. The carriers are no longer genuinely

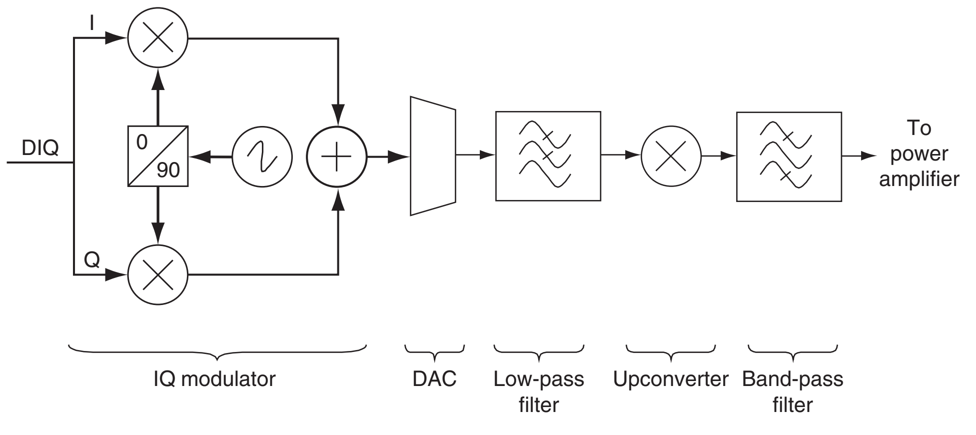

Figure 7.8 Block diagram of DAB transmitter using low IF

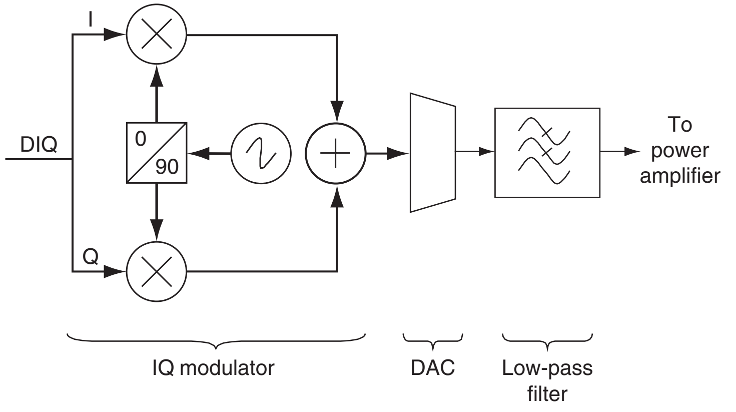

Figure 7.9 Block diagram of a direct conversion DAB transmitter with fast DAC

orthogonal, and cause a small amount of interference to one another. This phenomenon is therefore often termed 'intercarrier interference'.

The second effect is mainly due to oscillator phase noise at frequency offsets less than the carrier spacing. Because the data are carried by changes in phase between successive symbols, the phase noise causes a reduction in the inherent signal-to-noise ratio. This effect is often termed 'common phase error', because it affects all carriers equally.

The effects of oscillator phase noise in OFDM systems can be analysed in detail for individual oscillator characteristics using weighting functions (e.g. [Stott 1996]), but for practical purposes, common phase error is normally the dominant effect in DAB systems. Good engineering practice requires that the degradation of signal-to-noise due to the transmitter from these effects is minimised, and a common rule of thumb is that the phase noise should be no more than –60 dBc/Hz at an offset of 25% of the carrier spacing [TR 101496].

7.4.4 RF Upconversion

As noted above, upconversion is required if the in-phase and quadrature signals are modulated onto an IF rather than the final RF. This applies for some Band III transmitter implementations, and most if not all L-band implementations. Most of the considerations discussed above apply equally to the upconversion process, including mixer linearity, spectral purity of oscillators, and frequency response of band-pass filters used to remove LO breakthrough and image products.

7.4.5 Amplification and Filtering

Because COFDM is a multicarrier system, an important cause of signal distortion is amplifier non-linearity. This causes intermodulation between the carriers, giving rise to RF products both within the bandwidth of the COFDM signal itself, and also in adjacent channels (Figure 7.10).

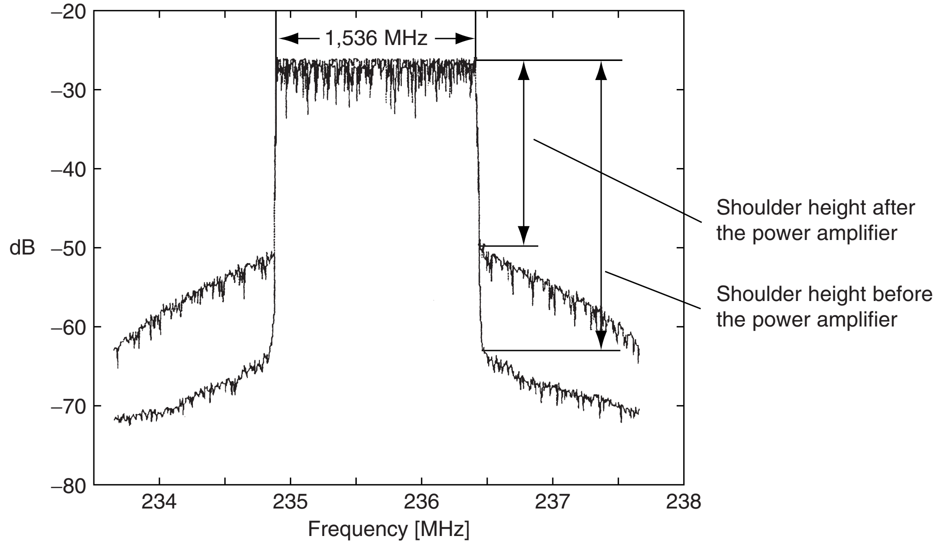

Figure 7.10 Spectrum of DAB signal with amplifier non-linearity

The difference between the spectral density of the COFDM signal and the out-of-band intermodulation products is often referred to as 'shoulder height' (in Figure 7.10, for example, the signal distortion due to the non-linearities of the power amplifier has resulted in a shoulder height of 26 dB). Control of out-of-band products is important, because their level affects the performance of other signals in the adjacent channel.

The in-band intermodulation products are hidden by the COFDM signal itself, but their level is typically around 3 dB higher than the shoulder height of the out-of-band products immediately adjacent to the ensemble. These products limit the performance of the DAB transmission, because they represent a 'noise floor' that cannot be removed in the receiver. Accordingly, transmitters are designed to ensure that the in-band 'floor' is sufficiently low that its impact on system performance is minimal.

Having chosen the acceptable level of in-band 'floor', a similar level of out-of-band products will be generated. However, to ensure efficient use of spectrum, [EN 300401], [EN 302077] and other standards require a much lower level of out-of-band emission by the transmitter, and this can be achieved only by external band-pass filtering. The attenuation requirements of the filter depend on the level of out-of-band emissions at the output of the amplifier, and therefore on the linearity of the amplifier and preceding stages. As a result, one of the major design decisions in a DAB transmitter is the trade-off between filter performance and amplifier linearity.

The most efficient amplifiers tend to be highly non-linear, and are not suitable for DAB. Conversely, highly linear amplifiers tend to be inefficient. Most DAB transmitters are designed so that the final power amplifier is the dominant non-linearity in the system, which allows it to be as efficient as possible. Even so, this amplifier requires a 'back-off' of several dB (i.e. the output power of the DAB signal is several dB below the saturated output power of the amplifier when passing an unmodulated RF signal). For solid state amplifiers, the back-off is typically between 6 and 8 dB.

Operating amplifiers in this fashion results in a shoulder height of 25 to 30 dB. In order to meet the out-of-band spectrum requirements of [EN 300401] and [EN 302077], high-order passband filters are required (e.g. in Band III eighth-order filters are commonly used). These filters are normally the dominant cause of frequency selective effects in the DAB transmitter, in terms of both amplitude and phase (although baseband filters can also have an influence). Phase effects in particular need to be kept within acceptable limits. If this is not done, the performance of the DAB system can be degraded (e.g. a large spread of group delay across the ensemble can reduce the useful guard interval at the receiver).

Owing to the expense of high-power filters and amplifiers, recent years have seen much interest in improving the efficiency of DAB amplifiers without adversely affecting the spectrum. Analogue pre-correction techniques, in use in broadcast transmitters for many years, are now being replaced by more sophisticated techniques such as closed-loop systems and various pre-conditioning or adaptive pre-correction methods implemented largely in the digital baseband domain. One of the techniques is to apply non-linear pre-correction to the signal to linearise the nonlinear power amplifiers. As a result the in-band ‘noise floor’ and the out-of-band emissions are reduced, the latter giving much better shoulder heights of the output spectrum. Another technique is crest factor manipulation which can be used to achieve significantly higher output power of the transmitter. However, crest factor manipulation degrades the overall system performance due to additional signal distortion. It should only be applied in combination with nonlinear pre-correction techniques since they improve the overall system performance.

7.5 Radio Frequency Propagation Aspects

The design, the complexity and, as a consequence, the cost of a DAB system is strongly dependent on the factors affecting the propagation characteristics of the transmission channel to the vehicular receiver, the indoor receiver, the portable receiver and the stationary receiver; in decreasing order of difficulty. The propagation path is subject to attenuation by shadowing due to buildings, trees and other foliage and to multipath fading due to specular reflections and diffuse scattering from the ground and nearby obstacles such as trees and buildings (see section 2.1). The degree of impairment to the received signal depends on the operating frequency, the receiving antenna height and the type of environment in which the receiver is operating: whether it is an open, rural, wooded, mountainous, suburban or dense urban environment.

This section deals first with the characteristics of an impaired RF channel before it describes the propagation and channel models used to obtain a coverage prediction.

7.5.1 The Impaired RF Channel

The radio frequency channel encountered in DAB reception can be characterised as a time-variant multipath channel. Apart from the direct signal, additional reflections arrive with a certain time delay at the location of the receiver, superpose and build a more or less complex wave field with a varying power density which can be described by a Rice or Rayleigh distribution (for more details see section 7.5.2). This multipath field is time-variant, even if the receiver is not in motion since the environment in which the signals propagate and which give rise to reflections cannot be regarded as fixed. Normally, this time-variance is slow and the Doppler spread going along with it is not remarkable. However, it accounts for the fact that a receiver will experience all possible configurations of the Rayleigh field even if not in motion.

The usual channel response is illustrated by a two-dimensional representation of the time delay spread of the channel, where multipath results in peaks of various amplitude, present at specific excess delays relative to the original channel excitation impulse, as illustrated in Figure 7.11. Each delayed peak corresponds to a reflected signal.

When the receiver is in motion, additionally, various Doppler shifts appear for each of the signals received. The Doppler shift on an individual multipath component depends on the angle of its arrival with respect to the direction of the vehicle displacement. The cumulative effect of these shifts is called the channel Doppler spread. The procedure involved in analysing such spread is to line up a series of successive impulse response snapshots, take a slice through them at a given time delay, and then do a Fourier transform on the resulting samples. As a result, the broadcast channel can be illustrated by a three-dimensional representation of its impulse response.

The effect of the spreading of the received signal in both time and frequency is illustrated as scattering diagrams in Figures 7.12 and 7.13 where the channel impulse response is given on a three-dimensional plot of amplitude versus time and frequency. This is a very useful tool for visualising the effects of the surrounding environment on the signal received by a mobile receiver. Figure 7.12 shows an example of a scattering diagram obtained from a measurement run that took place in the downtown area of a large city. The maximum Doppler shift that can occur is a function of the radio frequency and the vehicle speed. In terms of the normalised frequency scale used here, the maximum shift at 1.5 GHz is ±5 Hz/m. This corresponds to a maximum shift of 139 Hz at a typical highway speed of 100 km/h.

In an urban area such as the one depicted here, the scattering diagram tends to be limited to small excess delays but is rather complex due to the multitude of echoes. In this case, the scattering is from objects close to the receiver, and the scattered signal arrives from many different angles. Particularly in dense urban environments, the arrivals tend

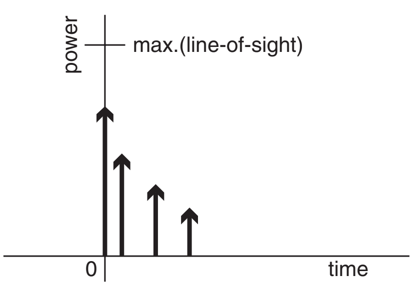

Figure 7.11 Simplified 2-dimentional channel impulse response

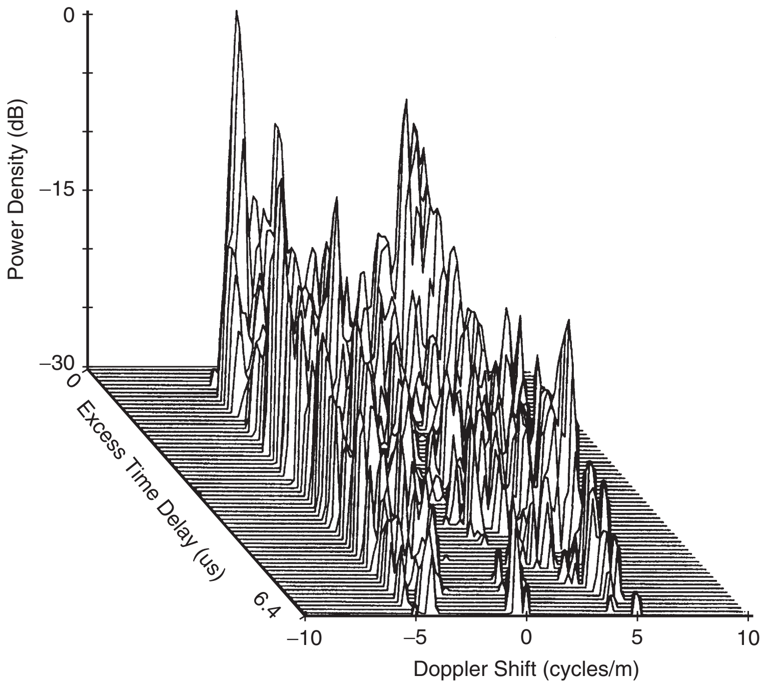

Figure 7.12 Scattering diagram – downtown/large city

towards a uniform angular distribution, which results in a 'U' shaped Doppler spectrum. At larger excess delays in such an environment, the components with small Doppler shift (nearly perpendicular to the direction of travel) tend to disappear. In urban environments, these components are frequently blocked by the buildings along the street. The Doppler spectrum then divides into two groups, one corresponding to the signal components arriving from the direction in which the vehicle is travelling (positive Doppler shift), and the other from components arriving from behind the vehicle (negative Doppler shift).

Figure 7.13 shows a scattering diagram derived from a measurement in the downtown area of a small city. The division of the Doppler spectra into two groups near the maxima is more evident here. The diagram clearly shows the funnelling of the signal by the street along which the vehicle is moving, with many components lined up near the maximum Doppler shift. There is a sizeable component having about 5 μs excess delay that seems to be caused by a reflection from a building located some 750 m (2.5 $\mu s \times c$) behind the transmitter and reaching the receiver from behind (negative Doppler shift). These two figures are each the result of a group of 128 successive channel impulse response snapshots representing some 5 m of travel.

Four different transmission modes were developed to cater for a wide range of speed and frequency requirements in DAB systems as shown schematically in Figure 7.14. Dependent on the frequency used and the maximum vehicle speed envisaged, the appropriate DAB transmission mode can be chosen.

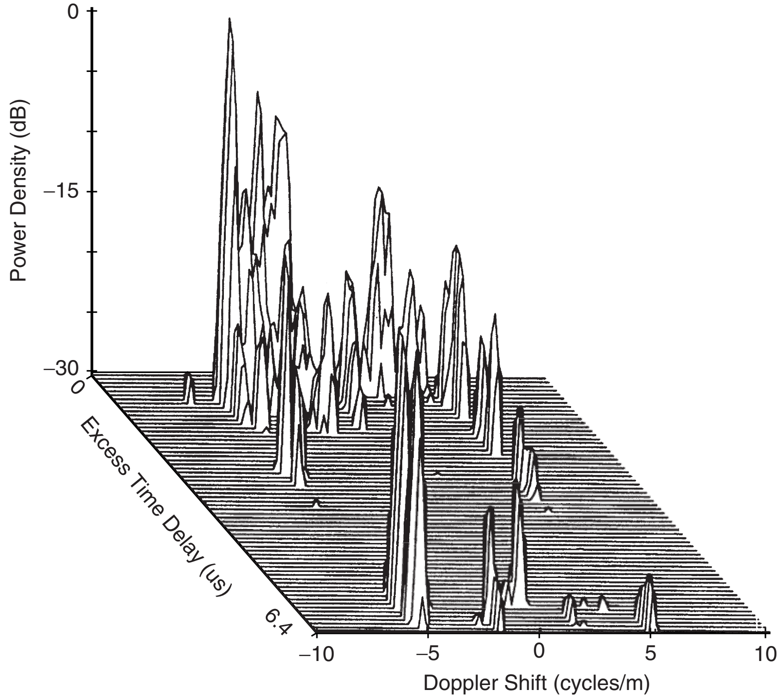

Figure 7.13 Scattering diagram – downtown/small city

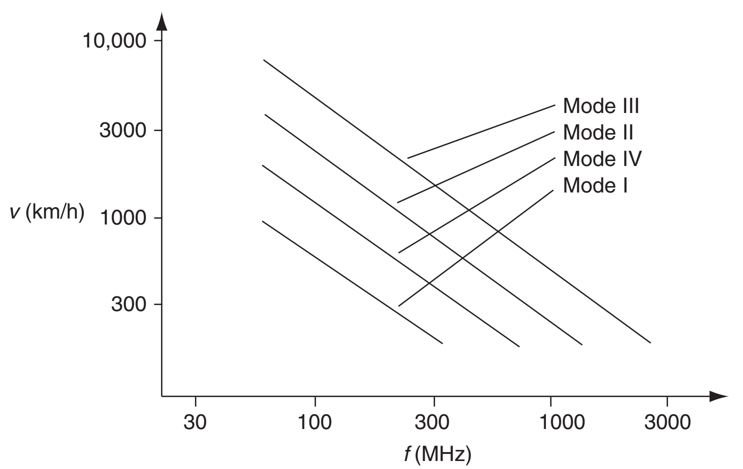

Figure 7.14 Interdependencies of maximum vehicle speed, frequency and DAB transmission mode

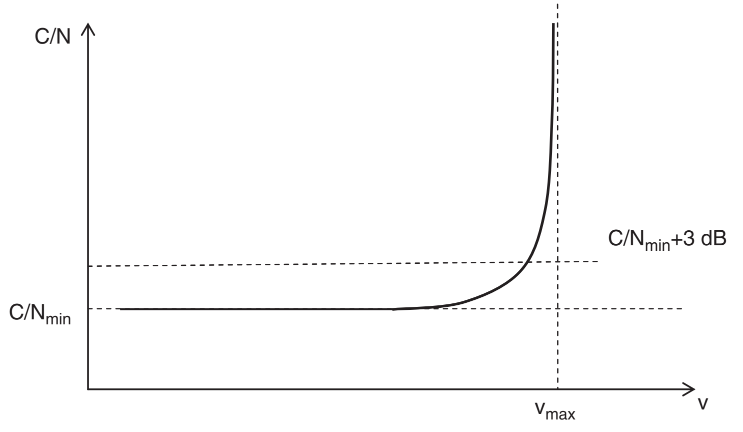

Figure 7.15 Impact of Doppler spread due to receiver velocity v on the required C/N of the receiver

A critical situation arises if the vehicle velocity is higher than recommended for the given DAB transmission mode and carrier frequency. Figure 7.15 describes in principle the C/N behaviour of the mobile receiver with velocity. Clearly, a limiting speed can be identified beyond which reception is no longer possible. Often the value $C/N_{min} + 3$ dB is taken for planning of DAB for mobile reception.

7.5.2 Propagation and Channel Models

Propagation models describe the propagation of electromagnetic waves with the aim to predict the field strength level in a certain area. These models take into account propagation losses and shadow fading effects. Propagation models are in general independent of the specific transmission system. Channel models take care of the specific properties of the transmission system such as certain types of fading, bandwidth, and effects that occur when the receiver is in motion.

The probability distribution functions of the received field strength relevant to the reception of DAB signals were found to correspond to a number of statistical distribution models related to the specific environment. These distribution models are generally different for so-called 'small areas' and 'large areas'. The large areas are usually defined as locations extending over many wavelengths (λ), usually 200 m by 200 m, whereas the size of small areas extend only over few wavelengths. Both terms 'small area' and 'large area' will subsequently be used in this section.

7.5.2.1 Propagation Models

Shadow Fading

The variation of the signal power due to reflections, scattering and diffraction of the signals by obstacles in the near or more distant vicinity of the receiving location is called shadow fading. It takes place on 'large areas' once the signal levels have been averaged to remove the signal variations within the 'small areas'. Being multiplicative processes these damping mechanisms give rise to a log-normal distribution of the received signal field strength. In measurements typical values between 3 dB and 6 dB are found for the standard deviation of this distribution.

This corresponds to the model adopted in Recommendation ITU-R P.1546 [P.1546] which describes a statistical field strength prediction model. In this recommendation a value for the standard deviation of 5.5 dB is adapted.

Statistical Models

Recommendation ITU-R P.1546 describes the log-normal model used by the ITU-R in predicting 'large areas' propagation losses. The model is appropriate for rural areas as well as for built-up areas, giving close agreement with the well-known Okumura/Hata propagation model [Okumura, 1968], [Hata, 1980] for the latter environment. The validity of these models is illustrated by Figures 7.16 and 7.17 which compare the results of actual field measurements conducted at 1.5 GHz in the Montreal area in Canada with these propagation models for a given effective isotropic radiated power (e.i.r.p) and height above average terrain (HAAT). (In Figure 7.16 a comparison with the predecessor of ITU-R P.1546, Recommendation P.370, is made).

Deterministic Models

Although the ITU-R prediction model is widely accepted, it is preferable to augment it with more precise prediction methods based on topographic databases and land occupation data [Voyer, 2000], [Voyer, 1996], [Whitteker, 1996], [Grosskopf, 1991], [Grosskopf, 1997]. Since 2007 also a deterministic ITU-R propagation model to be used with topographic data is available [P.1812]. In this way, a DAB system more precisely tailored to the given service area can be planned and specific requirements, such as repeaters to cover specific hard to reach areas, can be assessed.

Intersymbol Interference

In DAB, a particular multipath situation is encountered if some signal echoes fall outside the range that can be corrected by the DAB receiver. Multipath reflections

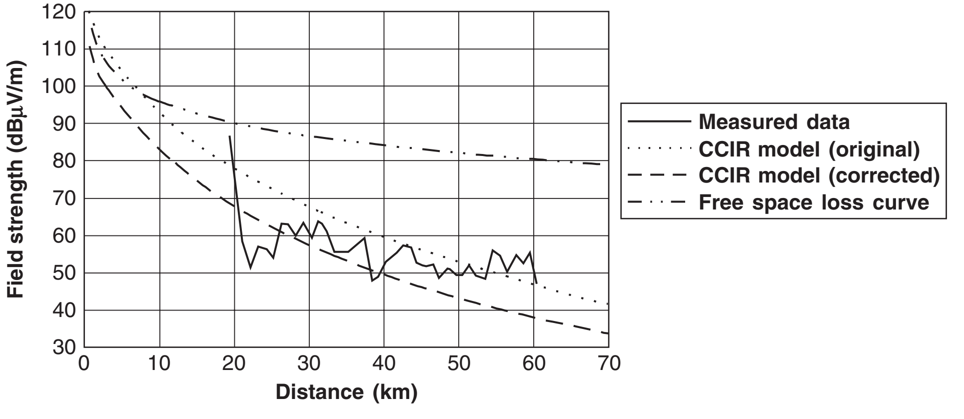

Figure 7.16 Comparisons of measured data with Rec. ITU-R P.370 model and free-space loss curve in Montreal for 50% of locations (e.i.r.p. = 41.1 dB(W), HAAT = 235.5 m)

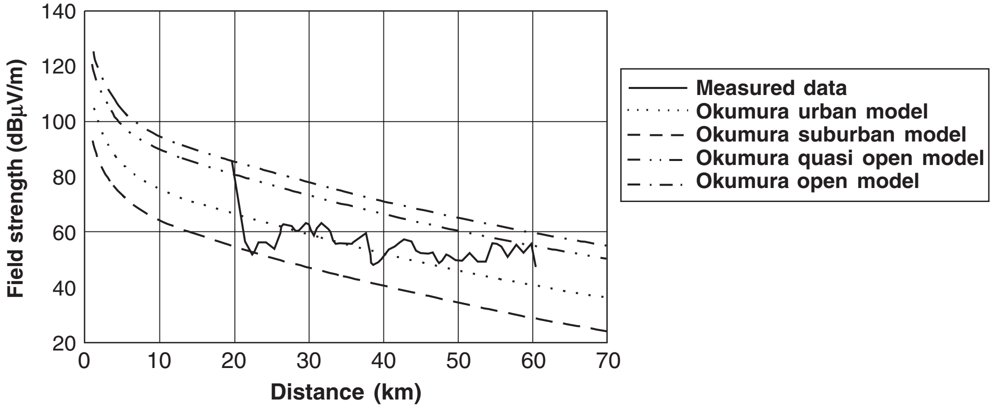

Figure 7.17 Comparisons of measured data with the different Okumura models in Montreal for 50% of locations (e.i.r.p. = 41.1 dB(W), HAAT = 235.5 m)

which fall beyond the symbol guard interval create inter-symbol interference. The reception performance will rapidly be impacted by this intra-system interference and reduce the service availability.

In coverage prediction models, normally, destructive components leading to inter-symbol interference are taken into account explicitly, at least for active echoes, i.e., for direct path signals coming from different transmitters in an SFN, see for example [BPN 066]. Alternatively, inter-symbol interference, in particular that arising from reflections, could be accounted for in channel models. However, this is normally not done since a more complex model would need to be used in this case for the performance simulation.

7.5.2.2 Channel Models

Typical properties of a transmissions system which can be taken into account by channel models are described in this sub-section. Typically, the various effects taken into account by the channel model result in an increase of the C/N in order to obtain a certain system performance.

Effect of Multipath Fading

The variation of the field strength over a 'small area' due to the constructive and destructive superposition of electromagnetic waves is called fast fading (see also section 2.1). In most cases one wave is dominant. Such a wave field can be modelled by a Rician distribution (constant vector plus Rayleigh distributed vectors). If there is no dominant wave present the field can be modelled by a Rayleigh distribution (only Rayleigh distributed vectors).

Therefore, the total probability density function of the received power should combine log-normal and Rice or Rayleigh distribution in order to take account of both large-area variations and small-area variations. When modeling this transmission channel in detail, the distribution of instantaneous values in a small area is obtained by considering a Rice or Rayleigh variable whose mean value is itself a random variable having a log-normal distribution.

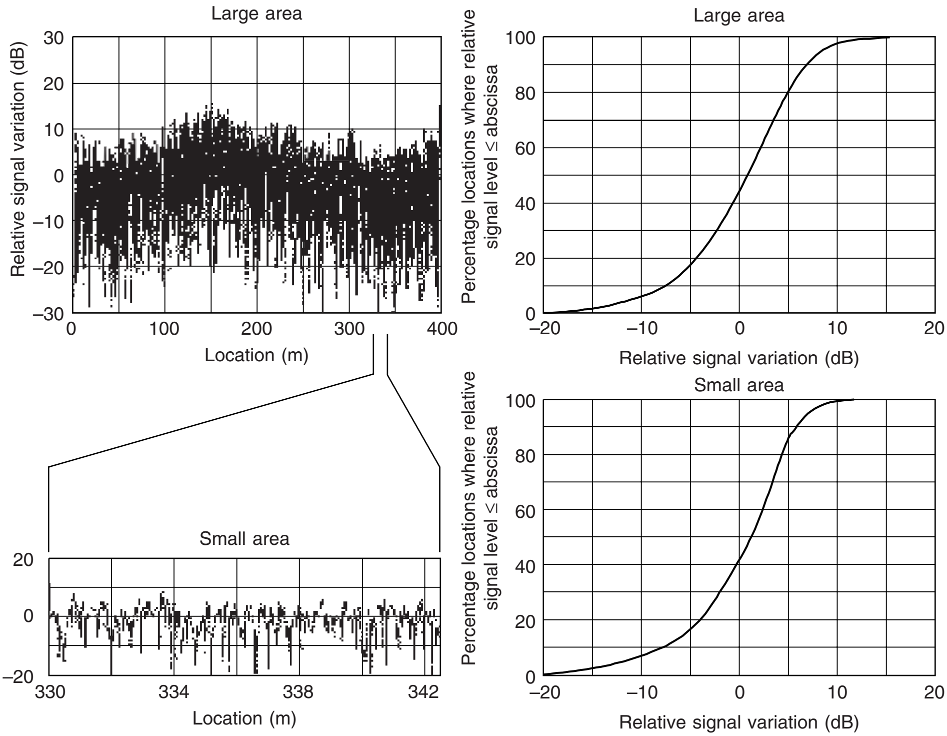

Figure 7.18 Relative signal variation (dB) with the corresponding cumulative distribution function of a CW signal as seen by the mobile unit in a large area and a small area

Figure 7.18 confirms these conclusions and depicts the variation of a CW signal received by a mobile receiver in an urban area. The relative field strength (normalised to the mean) is shown as a function of the receiver location along the measurement route. It can be seen that multipath causes fast and very deep fades, while a rather slow variation of the signal envelope in the large area plot in Figure 7.18 reveals the presence of shadowing due to tall buildings.

A magnified view of the received signal as a function of location, presented as the small area, shows that multipath causes large signal variations resulting from signal cancellation between the various scattered signal components. This variation of the resulting signal field strength usually corresponds to a Rayleigh distribution. The CDF functions of the measured signal levels are presented on the right side of Figure 7.18. Due to the time interleaving used by DAB (see section 2.2.5), these small scale fluctuations will not affect reception in vehicles moving at sufficiently high speed, however, they will be more relevant at pedestrian speed or when the receiver is not in motion.

As will be seen later in this section, except for a cost of about 4–8 dB of carrier-to-noise ratio for operation of the DAB system in a multipath environment, it can be assumed that the performance of the DAB reception will correspond to the averaged field strength found over the ‘small area’. This means that, instead of using the combined log-normal and Rice/Rayleigh propagation model, the reception performance for the DAB system can be modelled following the simpler ‘large area’ log-normal model.

However, the validity of this approach is limited in particular reception situations where micro-reflections with excess delays below about the reciprocal of the DAB channel bandwidth occur. These micro-reflections result in so called flat fading over the channel which can wipe out most of the signal. If the receiver is not moving to take advantage of the time interleaving this could result in loss of service.

For coverage predictions, normally, such particular receiving situations are accounted for by requiring an increased carrier-to-noise ratio in excess of that required for fast fading channels while continuing to apply the log-normal model. This increase of C/N is typically in the order of 3 dB.

Effect of Channel Bandwidth

The effect of the transmitted signal bandwidth on the variability of the received signal is illustrated by comparing Figures 7.18 and 7.19. As can be seen from these figures, the level of the 1.47 MHz bandwidth DAB signal shown in Figure 7.19 exhibits much less multipath fading than the CW signal from Figure 7.18, and the cumulative distribution function (CDF) of the wideband signal is closer to the reference log-normal CDF. This is due to the fact that the power is integrated over the 1.47 MHz bandwidth, thus averaging out most of the sharp frequency selective fading occurring in small areas. At the limit,

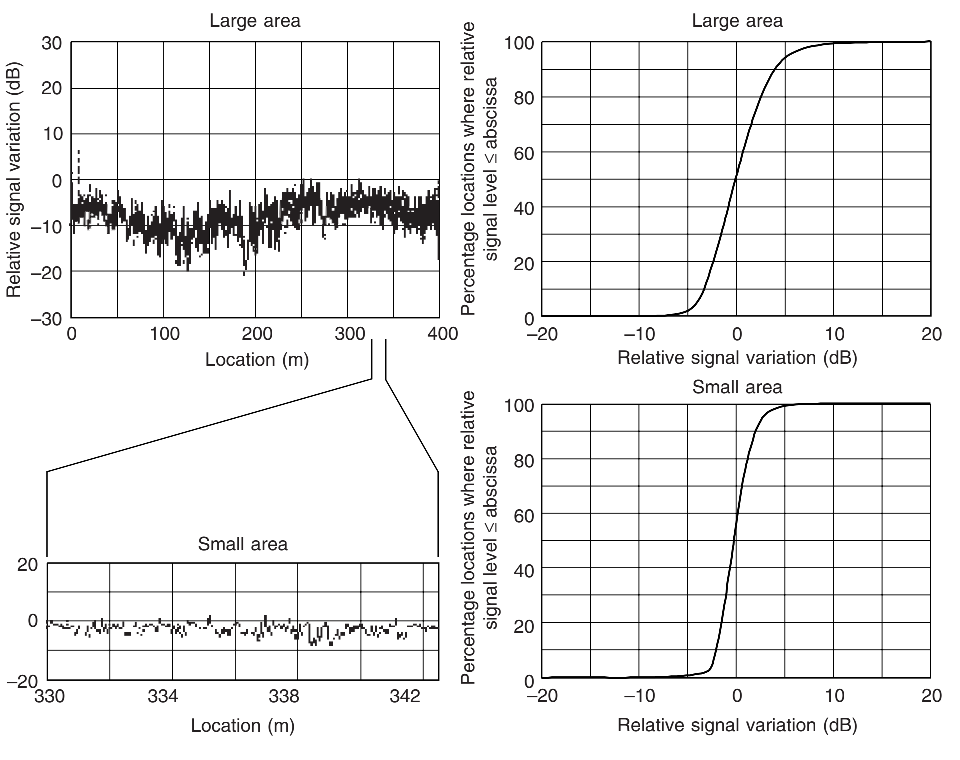

Figure 7.19 Relative signal variation (dB) with the corresponding cumulative distribution function of a 1.47 MHz signal as seen by the mobile unit in a large area and a small area

such averaging has the same effect as the averaging over a small area, therefore eliminating the Rayleigh portion of the combined channel propagation model.

As long as the signal echoes fall within the symbol guard interval, the assumption of integrating the power within the channel bandwidth, thus averaging out the sharp frequency selective fades, is a useful approximation for the channel model, although the underlying mechanisms (channel coding and interleaving) would have to be considered in detail for a precise analysis. In fact, DAB reception in a Rayleigh fading environment suffers an apparent raise of the threshold C/N of about 4 -8 dB compared to a Gaussian channel but once it is in a Rayleigh environment, the specific number and relative level of echoes becomes irrelevant.

This behaviour was verified in field measurements conducted to characterise the sensitivity of the received signal variation as a function of channel bandwidth. Figure 7.20 shows the increasing multipath fade margin as the channel bandwidth is increased from

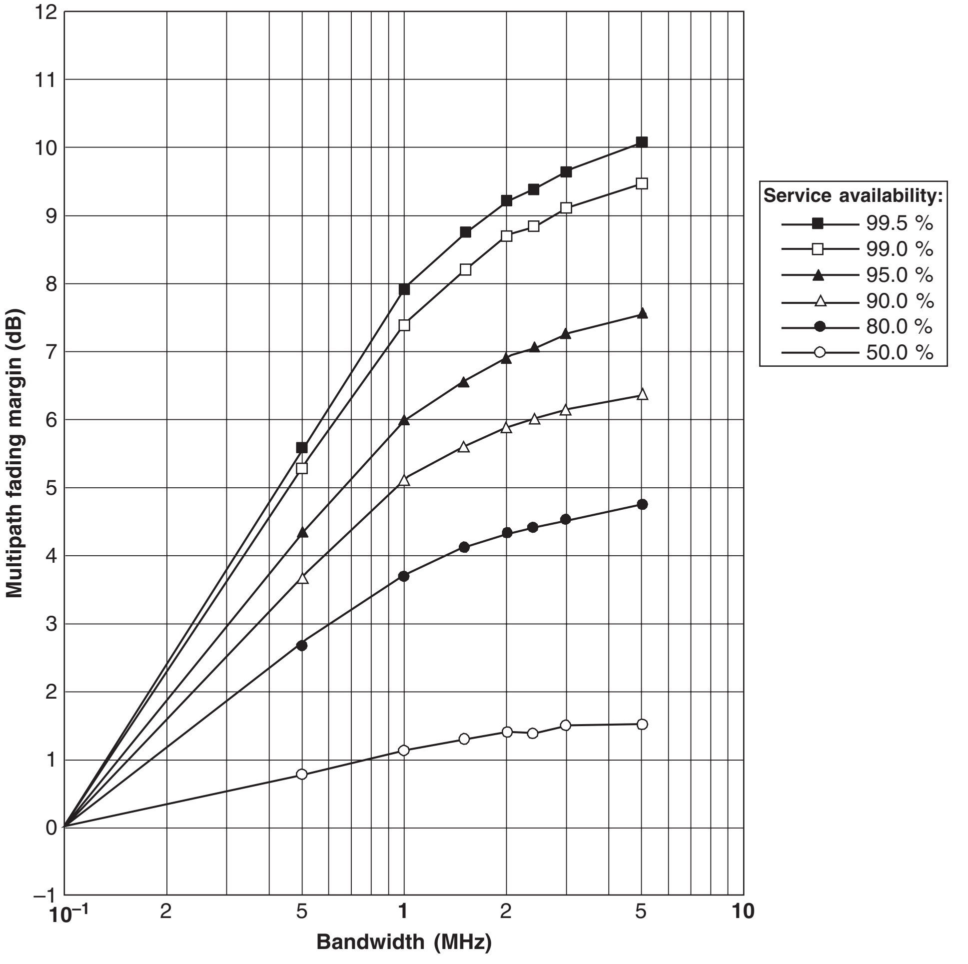

Figure 7.20 Improvement in multipath fade margin, dense urban environment

100 kHz to 5 MHz in a dense urban environment. The fade margin can be interpreted as the possible saving in transmit power relative to that needed for a 100 kHz channel bandwidth system, for an equivalent service availability objective.

Figure 7.20 shows that, for service availability objectives lower than 50%, the improvement in fade margin remains in the order of 1.5 dB in a dense urban area. Significant improvement is observed for service availability objectives of 90% or greater. Each curve can be divided into two sections, the first part being from 100 kHz to a bandwidth value that corresponds to a knee in the curve, the second part being from the knee position to the 5 MHz bandwidth value.

It seems that the position of the knee on the curves falls between 1 and 2 MHz, confirming the validity of the choice of 1.5 MHz bandwidth for the DAB system. Below 1 MHz, the multipath fading increases abruptly while above 2 MHz the improvement in fade margin is generally not very significant. This also means that most of the 'small area' Rayleigh fading is removed when a channel bandwidth of more than about 1 MHz is used in combination with the DAB modulation.

7.6 Coverage Planning

The coverage planning of a DAB service comprises a number of aspects. Firstly, it has to be decided which kind of service reception is intended. The choice could be, e.g., mobile reception or portable reception for handheld or standard receivers. Then the service availability has to be specified. Planning criteria have to be identified relating to the receiver characteristics, like carrier-to-noise ratio, receiver noise figure, antenna gain and, in particular, minimum required field strengths. Then, propagation issues have to be specified and interference from inside and outside the planned DAB network has to be taken into account. Finally, the particular aspects of the operation of a single frequency network have to be considered, like time delays between transmitters or wanted signal summation in order to account for network gain.

7.6.1 Reception Modes

Traditionally, coverage of analogue broadcast systems has been planned on the assumption that reception would make use of fixed antennas at roof height, nominally 10 m above ground level (AGL). For DAB it is clear that this assumption would not be valid.

DAB is mainly used in a mobile and/or portable reception environment where the receiving antenna is either integrated in the receiver or is located near the receiver. In both cases the reception is assumed to be at a height of 1.5 m AGL.

In [Tech 3317], [BMCO, 2007] four reception classes have been defined to describe typical receiving scenarios for DAB and DMB. They are given in Table 7.3.

For the frequency plans GE06 and MA02, which are introduced in more detail in section 7.8.2, class B has been used to describe portable reception and class C has been used to describe mobile reception.

Table 7.3 Reception classes for DAB and DMB

Reception mode	Class	Description
Portable reception	Class A	– hand-held portable outdoor reception – with external (for example telescopic or wired headsets) or integrated antenna – at no less than 1.5 m above ground level, at very low speed or at rest
	Class B	– hand-held portable indoor reception – with external (for example telescopic or wired headsets) or integrated antenna – at no less than 1.5 m above ground level, at very low speed or at rest – on the ground floor in a room with a window in an external wall (*)
Mobile reception	Class C	– hand-held reception inside a moving vehicle (car, bus etc.) – with the receiver connected to the external antenna of the vehicle – at no less than 1.5 m above ground level, at higher speed
	Class D	– hand-held reception inside a moving vehicle (e.g. car, bus, etc.) – without connection of the receiver to the external antenna of the vehicle – with external (for example telescopic or wired headsets) or integrated antenna – at no less than 1.5 m above ground level, at higher speed

(*) in [BMCO, 2007] it is additionally distinguished between a 'light indoor' (B1) and a 'deep indoor' (B2) mode

7.6.2 Planning Criteria

Coverage planning for broadcasting systems has to be based on certain planning criteria which comprise receiver characteristics, propagation aspects and coverage quality considerations. A central role in coverage planning plays the minimum required field strength for reception. This chapter gives an overview of these planning criteria and describes which figures are used in coverage planning and which have been used when the frequency plans GE06 and MA02 have been established. A comprehensive treatment of planning criteria for DAB/DMB in the VHF and L-Band can be found in [BPN 003], [Tech 3317], [RRC06, 2006], [CEPT, 2007b].

7.6.2.1 Receiver Characteristics (Noise Figure, Carrier-to-Noise Ratio, Antenna Gain)

The technical aspects of receiver characteristics are dealt with in Chapter 8. Here, only the values for the receiver noise figure, the required carrier-to-noise ratio and the antenna gain are given and it is explained how they are used for coverage planning. As a guideline the EBU paper [Tech 3317] is referred to.

For the receiver noise figure a value of 7 dB is assumed for both VHF and L-Band. For the required carrier-to-noise ratio C/N a value of 15 dB is used for DAB in the Ge06 planning and the same value has been used for the MA02 planning in the L-Band. The EBU paper [Tech 3317] proposes a value of 13.5 dB for DMB for both mobile and portable reception. However, it is pointed out in this chapter that portable reception is more demanding since time-interleaving is not thus effective which could lead under adverse conditions to C/N values of up to 16.5 dB, see also section 7.5.2.

A further criterion to be taken account of in coverage planning is the antenna gain of the receiving device. A distinction has to be made with regard to the kind of antenna under consideration. The antenna of the receiver can be integrated – which will often be the case for handhelds, it can be an external antenna or the antenna can be adapted as it is, for example, assumed for mobile reception in a car. Table 7.4 summarises the antenna gain values from [Tech 3317]. In the planning for GE06 and MA02 the figures for adapted antennas have been used.

7.6.2.2 Height Loss

Portable and mobile DAB reception takes place at a lower height than roof-level which has been the planning standard for analogue broadcasting. For coverage planning purposes a nominal height of 1.5 m AGL is assumed. Therefore, for those field strength prediction methods which calculate the field strength at roof-level, a correction called height loss has to be applied. This height loss varies with the environment and the frequency. The values proposed in the EBU paper [Tech 3317] are given in Table 7.5. For the GE06 planning a value of 12 dB (for the VHF band) has been used [RRC06, 2006] assuming a rural/suburban environment as representative. In MA02 10 dB (for the L-Band) has been used for the height loss. In the EBU planning paper more conservative values are proposed for the L-Band as can be seen from Table 7.5.

Table 7.4 Antenna gain for the different antenna configurations and reception classes

	Band III	L-Band	Classes
Integrated antenna	−17 dBd	−4 dBd	A, B, D
External antenna)(*)	−13 dBd	−1 dBd	A, B, D
Adapted antenna	−2.2 dBd	0 dBd	C

(*) Telescopic or wired headsets

Table 7.5 Receiving antenna height loss

Environment	Band III	L-Band
Urban	19 dB	27 dB
Suburban	12 dB	21 dB
Rural	12 dB	19 dB

7.6.2.3 Building Penetration Loss

In the case of indoor reception (reception class B) the DAB signal suffers from a penetration loss when entering into the building. Field strength measurements were performed to derive typical figures for the building penetration losses in the frequency bands relevant to DAB.

Around 230 MHz, building penetration loss was measured in the UK [Green, 1992] and Germany [Schramm, 1996]. The UK results show that the building penetration loss varies between 2 dB and 18 dB on the ground floor of domestic buildings. Measurements on the first floor gave about 6 dB more field strength. The average loss was found to be 8 dB ±1.2 dB. The German results basically support these figures. The penetration loss measured ranged from 3 dB to 20 dB and the median value for typical buildings was found to be 9 dB at 220 MHz and 8.5 dB at 223 MHz with a standard deviation of 3.5 dB.

As a figure for network planning a mean value of 9 dB and a standard deviation of 3 dB is given in the EBU paper [Tech 3317] and has also been used for the GE06 plan [RRC06, 2006]. However, a general observation is that the building penetration loss depends strongly on the kind of buildings and their environment. For example, recent measurements [Mason, 2004] in London gave values between 9 dB and 16.4 dB for the mean value and 3 dB to 5.5 dB for the standard deviation. The large variety of values is the reason why different figures for planning can be found in the literature.

At 1.5 GHz, measurements in Australia have shown that the average building penetration loss for DAB in domestic dwellings averages 6.7 dB (ranging from 6.1 dB to 9.4 dB, depending on construction materials used) and is approximately 18.6 dB in reinforced concrete commercial buildings [DSB, 2002]. Measurements in a 1.5 GHz SFN were performed in the DAB pilot project at Dresden, Germany [Michler, 1998]. The field strengths in rooms at different floors of seven different buildings were measured. The buildings were all located in a zone where two or three transmitters contributed to reception. In most buildings the level difference between outdoor and indoor measurements was found to be 0 dB to 5 dB at upper floors and 8 dB to 15 dB at the ground floor. In a modern office building (a concrete-steel construction with metal coated windows), however, the corresponding values were 20 and 30 dB, respectively.

As a figure for planning in [Tech 3317] a mean value of 11 dB and a standard deviation of 6 dB is taken for the L-Band, whereas in [BMCO, 2007] 13 dB and 4.5 dB can be found.

In order to cope in network planning with the large variation of the penetration loss figures, in [BMCO, 2007] a second indoor reception mode, called ‘deep indoor’, is proposed. For this reception mode the mean value of the building penetration loss is increased by 6 dB and the standard deviation by 1 dB with respect to the figures of the standard ‘light indoor’ reception mode as described above.

For reception class D (handheld reception inside a vehicle) penetration loss figures are needed, too. Up to now, no measured values are available. In the EBU paper [Tech 3317] values of 8 dB and 2 dB are assumed for the mean value and the standard deviation, respectively. In [BMCO, 2007] respective values of 7 dB and 0 dB are proposed. In both cases VHF and L-Band values are identical.

7.6.2.4 Reception Quality

In section 7.5.2 the statistical aspects of signal propagation and field strength distribution have been discussed. For coverage planning, a choice has to be taken for the minimum location probability needed in order to regard a location as covered. In [Tech 3317], [BMCO, 2007] the classification described in Table 7.6 is given. For GE06 and MA02 a minimum location probability of 95% for portable reception and 99% for mobile reception has been assumed.

As a consequence, minimum required field strengths for coverage planning have to be increased by a certain amount, the so-called location correction factor, in order to ensure the higher coverage probability. This correction factor is necessary since in coverage planning field strengths are always expressed in terms of their mean values. The location correction factor is a function of the intended coverage probability and the standard deviation of the field strength distribution. In Table 7.6 the last row gives the corresponding location correction factors for a field strength standard deviation of 5.5 dB. A more detailed treatment of this aspect can be found in [BPN 066].

7.6.2.5 Man-made Noise

Man-made noise has to be taken into account when the minimum required field strength for a service is to be determined. Man-made noise is measured relatively to thermal noise. Recent measurements in Band III have shown relatively high values of up to 8 dB whereas in L-Band values 0 dB can be assumed. The full man-made noise the value, however, are only valid for receiving antennas with a gain greater than 0 dBi

Table 7.6 Coverage quality (probability in %) and location correction factor

Reception Class	Coverage quality	Location correction factor	
		Band III	L-Band
A	Acceptable (> 70%)	3.0 dB	3.0 dB
	Good (> 95%)	9.0 dB	9.0 dB
B	Acceptable (> 70%)	3.0 dB	4.0 dB
	Good (> 95%)	10.0 dB	13.0 dB
C	Acceptable (> 90%)	7.0 dB	7.0 dB
	Good (> 99%)	13.0 dB	13.0 dB
D	Acceptable (> 90%)	7.6 dB	7.6 dB
	Good (> 99%)	13.7 dB	13.7 dB

Table 7.7 Allowance for man-made noise depending on the antenna type in urban areas

	Band III	L-Band	Classes
Integrated antenna	0 dB	0 dB	A, B, D
External antenna)(*)	1 dB	0 dB	A, B, D
Adapted antenna	8 dB	0 dB	C

(*) Telescopic or wired headsets

(-2.2 dBd). For antennas with a gain less than 0 dBi it is important to distinguish between the pure antenna gain and the efficiency of the antenna. As a consequence, in the calculation for the minimum required field strength only an effective allowance for man-made noise is be taken into account, which only in the case of adapted antennas amounts to the full value of the man-made noise itself. Table 7.7 summarises the allowance for man-made noise in urban areas as given in [Tech 3317] where also more details regarding the treatment of man-made noise can be found.

7.6.2.6 Minimum Field Strength

Minimum required field strengths for reception can be deduced from the above discussed planning criteria for the different frequency bands and reception scenarios. Traditionally, in frequency planning minimum field strengths are expressed in terms of minimum median equivalent field strengths required at 10 m AGL for 50% of the time and 50% of locations. Table 7.8 gives the required minimum field strength values which have been used for the frequency plans MA02 and GE06.

In the EBU paper [Tech 3317] more conservative minimum field strength values are recommended for DMB network planning. They are given in Table 7.9. The figures for F_{min} in Table 7.9 refer to a receiving antenna height of 1.5 m AGL.

7.6.3 Interference Considerations

The previous section described the approach taken to predict coverage on the basis of signal strength from nearby transmitters in a network. Strictly speaking, this applies

Table 7.8 Minimum field strengths F_{min} used in GE06 and MA02 for T-DAB planning at 10 m AGL for 50% of time and 50% of locations

	Band III (GE06)		L-Band (MA02)
Reception mode	mobile, (RPC 4) (=class C)	portable indoor, (RPC 5) (=class B)	mobile, (= class C)
Location probability	99 %	95 %	99 %
C/N	15 dB	15 dB	15 dB
Effective standard deviation	5.5 dB	6.3 dB	5.5 dB
Minimum field strength F_{min}	60 dB(μV/m)	66 dB(μV/m)	69 dB(μV/m)

Table 7.9 Minimum field strengths F_{min} for T-DMB network planning for different reception classes at 1.5 m AGL for 50% of time and 50% of locations (from [Tech 3317])

Reception class	A		B		C		D	
Reception mode	portable outdoor		portable indoor		mobile, adapted antenna		mobile, integrated antenna	
Location probability	99 %		95 %		99 %		99 %	
C/N	13.5 dB		13.5 dB		13.5 dB		13.5 dB	
	Band III	L-Band	Band III	L-Band	Band III	L-Band	Band III	L-Band
Effective standard deviation	5.5 dB	5.5 dB	6.3 dB	8.1 dB	5.5 dB	5.5 dB	5.9 dB	5.9 dB
Minimum field strength F_{min}	55.6 dB(μV/m)	60.1 dB(μV/m)	65.9 dB(μV/m)	75.3 dB(μV/m)	49.5 dB(μV/m)	59.8 dB(μV/m)	68.2 dB(μV/m)	72.7 dB(μV/m)

only in the situation where noise is the factor limiting reception. In practice, interference from other networks, or distant transmitters within the same network, may be equally or more important, and it is necessary to plan services for adequate protection against such interference. The interfering field strength from distant transmitters can vary significantly according to the type of terrain that lies between the interfering transmitter and the receiver, the land occupation as well as the diurnal variation and climatic conditions. Because of these variations, interfering field strengths are often predicted using terrain-based models. Interference from a number of distant transmitters may need to be taken into account, and this is normally built into coverage planning software.

Although terrain-based modelling is favoured for network planning, it is not always suitable, for example if detailed terrain data are not available, as may occur if significant interferers are located in neighbouring countries or overseas. Under these circumstances, other methods, such as the one described in Recommendation ITU-Rec P. 1546 [P.1546], are used. Also for the purpose of establishing a frequency plan, where a very large number of field strength predictions are needed, a statistical approach as given in Rec. P.1546 is advantageous. Again taking the rapid failure characteristic of DAB into account, the criterion for coverage is that the wanted signal strength should be protected against interference for 99% of the time and locations in Europe (for, e.g., for the case of mobile reception). Put another way, given an interfering field strength predicted to be present for 1% of the time and locations, the wanted signal strength must exceed the level of the interferer by a sufficient margin to allow successful reception. The required signal-to-interference ratio to avoid noticeable channel errors as measured in the laboratory is called the protection ratio and is used to establish the maximum level of allowable interference.

In the GE06 plan for DAB in Band III an intra-service protection ratio, i.e. for interference of a DAB service into another DAB service, of 15 dB has been assumed, whereas in MA02 for DAB in the L-Band an intra-service protection ratio of 10 dB was chosen. The EBU paper [Tech 3317] proposes a protection ratio of 13.5 dB for DMB interfered with by DMB.

Adjacent channel protection ratios as well as inter-service protection ratios of DAB/DMB with regard to other broadcasting and relevant non-broadcasting services can be found in [RRC06, 2006], [CEPT, 2007b], [Tech 3317].

As was already discussed in section 7.6.2, in order to provide for the higher location availability needed for DAB, a location correction factor has to be taken into account when the system protection ratios mentioned above are applied. This correction factor differs by a factor of '$\sqrt{2}$' from that mentioned in 7.6.2 since now the statistical variation of both the wanted as well as the unwanted signal has to be taken into account. More details can be found in [BPN 066].

7.6.4 Delay Considerations

As already mentioned in the introduction to this chapter, coverage planning in a DAB single frequency network means, to a large extent, delay planning. All transmitters in a DAB network must be time and frequency synchronised for proper operation of the SFN. It is common practice to use GPS receivers to provide a highly stable frequency and time reference at different points in the network, typically the ensemble multiplexer and transmitter sites. The transmitter output frequency is locked to the 10 MHz signal of the GPS receiver and the GPS 1 pps signal serves as a reference for the delay compensation at each transmitter.

Time synchronisation is performed by introducing an artificial delay in the distribution network of the SFN or at the different transmitter sites as shown in Figure 7.22. The total duration for which the signal must be delayed is the sum of four different types of delay [EN 300799]:

1. **Network compensation delay**: The time by which the ETI (Ensemble Transport Interface) output signal of the multiplexer is delayed on its way through the distribution network to the transmitter site is called the network path delay. To ensure that the overall delay to each of the transmitter sites is constant and of known value, a network compensation delay is added for each path through the distribution network. The largest network path delay to a certain transmitter site determines the network compensation delays necessary for all the other network paths. In a correctly adjusted SFN, the sum of network compensation and network path delay must be the same for each network path.
2. **Transmitter compensation delay**: The time the signal is delayed in the equipment at the transmitter site through channel encoding (e.g. COFDM modulation) and RF modulation is called the transmitter processing delay. This also includes delay due to signal conditioning and RF processing in the amplifier and associated filtering. The processing delay of the equipment differs from manufacturer to manufacturer. To ensure that the overall delay between transmitter input and antenna output at each transmitter site is constant and of known value, a transmitter compensation delay is added individually for each transmitter in the network. The transmitter with the longest processing delay determines the transmitter compensation delay for all the other

transmitters in the network. In a correctly adjusted SFN, the sum of transmitter compensation and transmitter processing delay must be the same for each transmitter in the network.

3. **Transmitter offset delay**: This delay can be set for each transmitter site individually and is necessary for network planning aspects to achieve optimised coverage in an SFN. Owing to the surrounding topography and differing effective output power, the coverage areas differ from transmitter to transmitter. The transmitter offset delay ensures that the signals from individual transmitters arrive approximately at the same time in the overlap zone of their coverage areas. The transmitter offset delay can be set individually for each transmitter from the ensemble multiplexer using the MNSC (Multiplex Network Signalling Channel) in the ETI stream.

   An example of such a network is given in Figure 7.21. It shows the 38-transmitter DAB network in Bavaria, where most of the transmitters have individually set

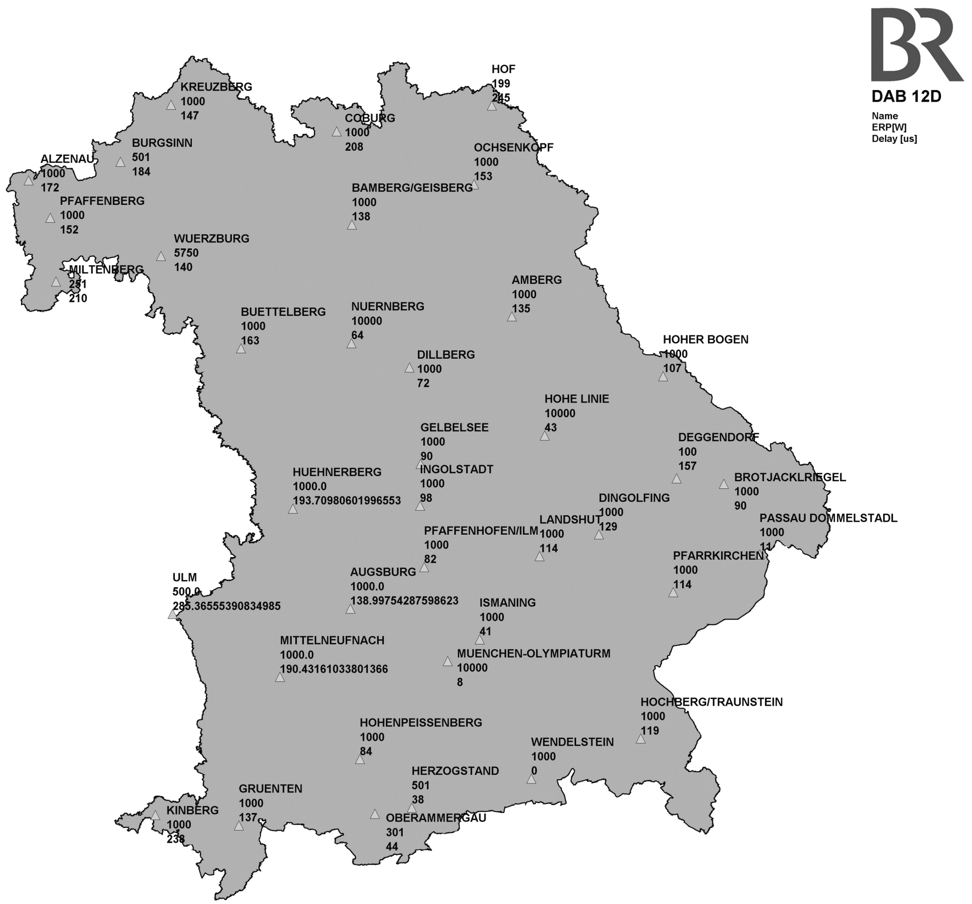

Figure 7.21 DAB SFN in Bavaria operated by BDR with transmitter powers (W) and offset delays (μs) (Reproduced by permission of © BDR)

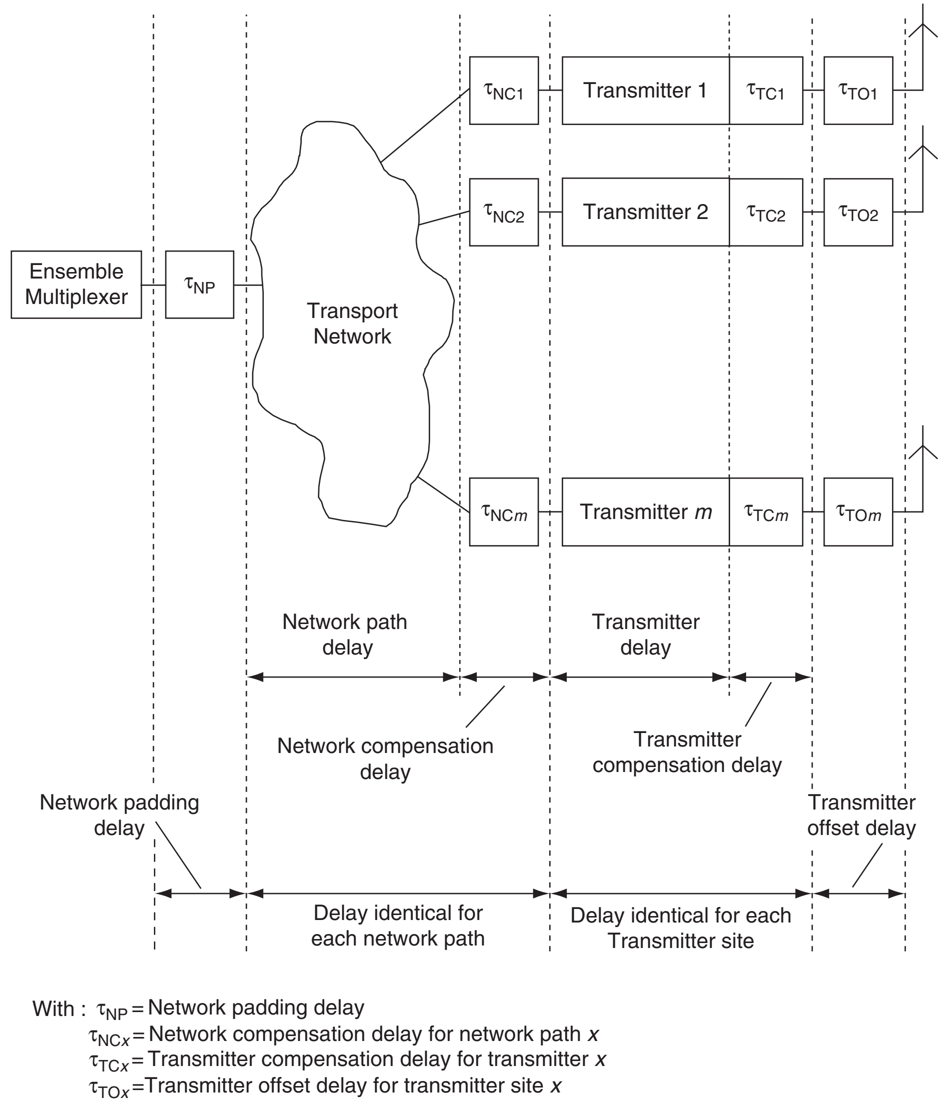

Figure 7.22 Types of delay in an SFN

transmitter offset delays in order to optimise the coverage. This measure is necessary since the transmitter characteristics with regard to distance from the next transmitter, antenna height or transmitter power are very heterogeneous.

4. **Network padding delay**: This delay is used by the network operator or the broadcaster to adjust the overall delay of the network. It allows equalisation of the overall delay of the DAB network with that of other networks such as FM or DTV. It can also be used to synchronise different DAB networks to allow radiation of co-timed

synchronising symbols for speedier locking of DAB receivers in the case of tuning to a different frequency.

In summary, network and transmitter compensation delays ensure that the DAB signal is theoretically transmitted at the same point in time from each transmitter site in the network, whereas the transmitter offset delay takes care of the topology of the SFN. The padding delay is only relevant with respect to other broadcasting networks. The different types of delay and how they interrelate are given in Figure 7.22.

In modern distribution networks, the network path delay may change as a result of the necessity to reroute the signal from time to time. To automatically manage this varying network path delay, dynamic delay compensation on the basis of time-stamps is used in DAB which allows synchronisation of the delivery of ETI frames at all transmitter sites. This requires a common time reference (e.g. GPS) at the ensemble multiplexer and all transmitter sites in the network. The time-stamp defines the instant of time at which the frame should be delivered to the channel encoder. The time-stamp can be carried in all variants of the ETI signal.

7.6.5 Coverage Prediction

When planning a broadcast network, the prediction of the expected coverage is an elementary part. Propagation prediction methods and coverage criteria applied in this context have been discussed earlier in sections 7.5 and 7.6.2. This section deals with the evaluation of the coverage prediction.

In networking planning, a location is regarded as covered if the coverage prediction for this location exceeds the minimum coverage probability that has been chosen as a coverage target – see section 7.6.2.4. For an analogue broadcast system this assessment is uniquely determined by the comparison of the wanted field strength with the noise and interference environment at the location under consideration. For an OFDM system like DAB this is no longer valid since at the receiving location the wanted signals from the various transmitters of the DAB SFN add up to an effective wanted signal power whose statistical properties depend on the particular composition of the signal situation and which vary from location to location. For the evaluation of the coverage probability, therefore, more elaborate statistical methods for signal summation have to be applied.

Mathematically, the task consists in the summation and combination of log-normally distributed statistical variables. Since no analytical solutions exist for this task approximations have been developed most of which fall into the class of 'log-normal method' (LNM) approximations and which are widely used. An alternative approach to log-normal summation is the application of Monte-Carlo methods. However, because of their high numerical demand these latter methods are still restricted to the treatment of selected problems.

As a consequence, the minimum field strength criterion is no longer sufficient for the assessment of broadcast coverage; however, it still gives a rough estimate what coverage can be expected. A comprehensive treatment of signal summation and network gain is given in [BPN 003], [BPN 066].

As a standard, in coverage planning for DAB networks deterministic field strength prediction methods are employed. These methods use topographic and morphographic terrain data and allow for a more detailed field strength prediction than is possible with a statistical method like ITU-R P.1546. Various approaches for deterministic field strength prediction methods exist in the broadcasting field, an overview is given, e.g., in [Grosskopf, 1987], [Grosskopf, 1988]. An internationally agreed method has been established by ITU-R with Rec. P.1812 [P.1812] which is in force since end of 2007.

7.6.6 Further Planning Considerations

When planning networks in detail, the planner is subject to a number of constraints. These include the frequency range to be used, the receiving environment to be served, the availability of transmitter sites, and the allowable radiated power from these sites.

The frequency range will determine which of the possible DAB transmission modes is most appropriate (see also Figure 7.14).

The type of receiving environment has to be chosen, such as mobile, portable or indoor for example. As described in the preceding section the planning criteria regarding service quality, coverage quality and minimum field strength have to be adapted accordingly.

The availability of transmission sites, and the allowable transmitted power, will have a strong influence on the design of the network and the distance between transmitters. As a rule of thumb it is usual to ensure that the spacing between adjacent transmitters is no more than the distance determined by the guard interval of the chosen mode, as set out in Table 7.10. However, in many circumstances a smaller spacing between transmitters is found to be necessary.

A frequency plan entry which is the basis of the implementation of the network may turn out not to perfectly fit with the intended coverage of the service. Bilateral coordination with neighbours may therefore become necessary together with a modification of the plan entry. This aspect is described in more detail in section 7.8.

For an efficient broadcast network planning, several powerful software packages are available from commercial companies which integrate the various aspects of coverage prediction, see for example [www.lstelcom.de], [www.irt.de], [www.atdi.com].

More information on implementation aspects can be found in [BPN 066].

Table 7.10 DAB modes and approximate maximum transmitter spacing

DAB Mode	Guard interval (μs)	Approximate Maximum Transmitter Spacing (km)
I	246	74
II	62	18
III	31	9
IV	123	37

7.6.7 Example of an SFN in Band III

In the United Kingdom, the BBC's DAB network is an example of an SFN providing large-area coverage, and is illustrated in Figure 7.23. The network operates in Band III (Block 12B), and by 2006 it comprised 66 transmitters, with radiated powers between 1 kW and 10 kW. This provided coverage to more than 85% of the UK population,

Figure 7.23 DAB SFN in UK (Block 12B) (Reproduced by permission of © Arquiva)

and the network has since been enhanced with additional transmitters. In addition to achieving a high population coverage, the network was also designed to cover the major road routes between centres of population. Many major cities are served by more than one transmitter, so that the coverage would benefit from network gain. The majority of the transmitters are co-timed, but for a number of them the signals are time-delayed in order to avoid self-interference thus optimising the coverage.

7.6.8 Example of an SFN in L-Band

In 1996 in Ottawa, an L-Band DAB network operating in Mode II with one main transmitter, one coverage extender and two gap fillers has been designed to serve the National Capital Region [Paiement, 1996], [Paiement, 1997], [Voyer, 1996]. This is an area of approximately 30 km in radius, comprising Ottawa and several smaller adjacent municipalities with a total population of close to one million people.

It was required to provide 90% service availability to the city core, the suburbs, portions of the main roads and highways surrounding the region. This objective is represented by a service polygon drawn as shaded contour over a simplified road map of the area (see Figure 7.24, covering an area of 92 km by 64 km). The predicted coverage achieved by the main transmitter is given in Figure 7.24 (availability at 90% of locations). It can be seen that there is a need for a second transmitter in the network to cover the south-west area of the specified service polygon.

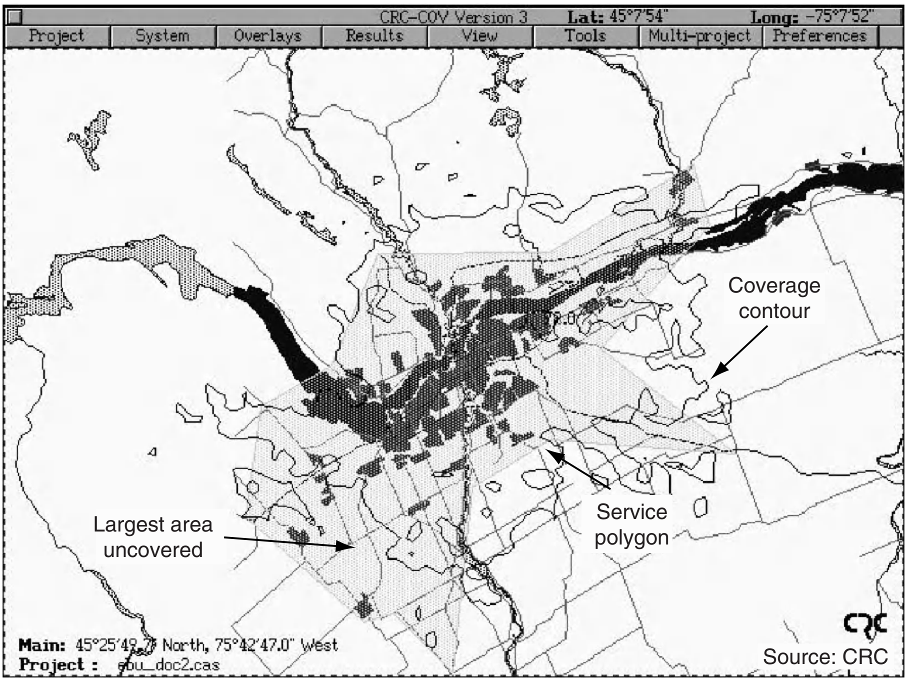

Figure 7.24 Service polygon for the Ottawa region and coverage of the main transmitter

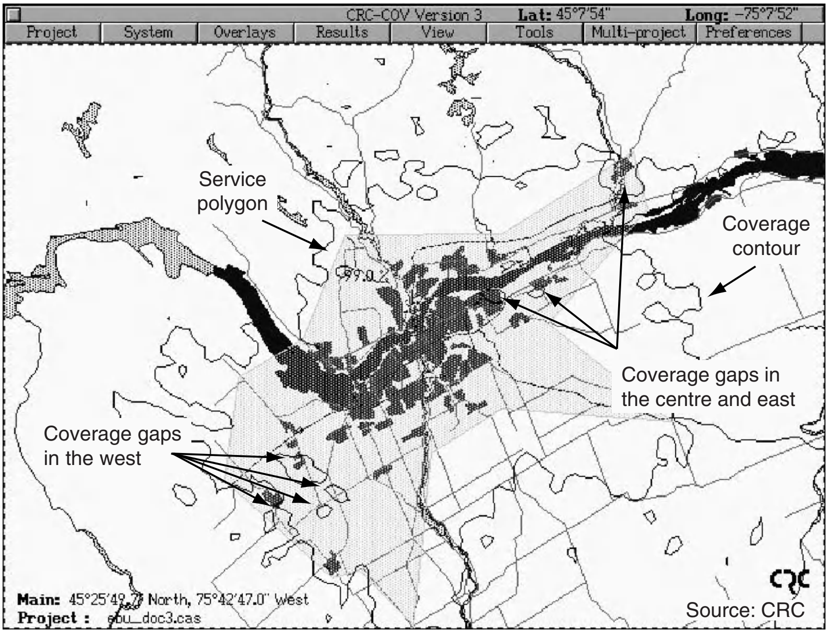

Figure 7.25 Coverage of main transmitter plus coverage extender

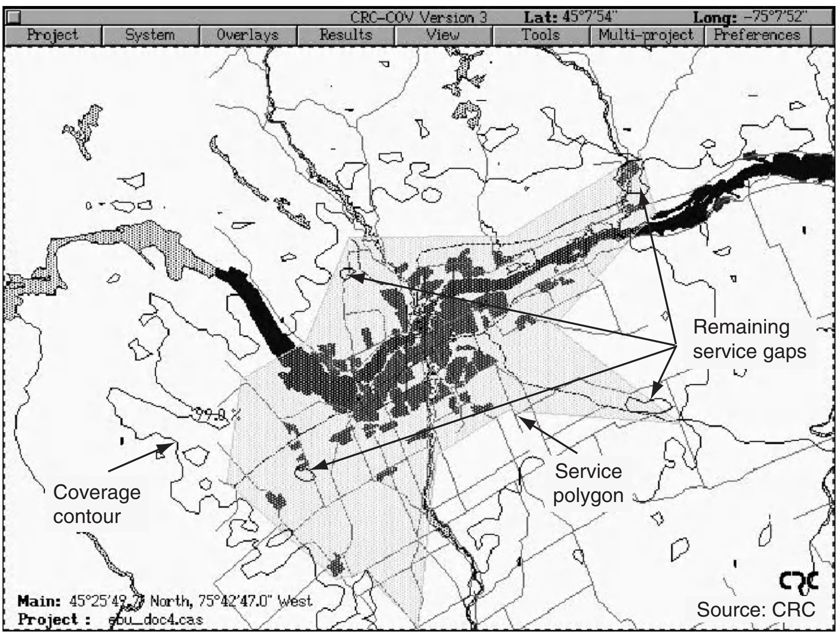

Figure 7.26 Coverage of main transmitter, coverage extender plus two gap fillers

Table 7.11 Transmitter parameters of the Ottawa DAB system

	Main Transmitter	Coverage Extender	Gap Filler – West	Gap Filler – East-centre
ERP	1000 W	500 W	1000 W	1200 W
Amplifier power	100 W	50 W	30 W	12 W
Antenna gain	13 dBi	13 dBi	19 dBi	22.5 dBi
Antenna pattern	Omnidirectional	Omnidirectional	Directional, 90° beamwidth	Directional, 40° beamwidth
Antenna height	105 m	105 m	50 m	50 m
Distance from main transmitter	N/A	5.4 km	7.1 km	10.7 km

The network planners then decided to use a coverage extender to enlarge the coverage area. The coverage predicted for the main transmitter plus coverage extender is given in Figure 7.25.

To close the coverage gaps in the west and east-centre of the required service area, two additional gap fillers were put into place. The complete predicted coverage of the Ottawa system is shown in Figure 7.26. The contour lines show four remaining small holes in the requested coverage area where three of them are not in densely populated areas.

The transmitter parameters are summarised in Table 7.11. It can be seen that the total service area of about 30 km in radius can be covered by four transmitter locations with a total amplifier output power of less than 200 W.

7.7 Coverage Evaluation and Monitoring of SFNs

7.7.1 Parameters for DAB Coverage Evaluation

In order to evaluate the actual coverage in a DAB network, field measurements have to be taken. In an MFN for FM, field strength is the main parameter for evaluating the quality of the coverage. For DAB, however, more parameters are necessary to evaluate the quality of the SFN coverage. The following sections describe the main parameters and what they indicate:

1. **Field strength:** This parameter indicates whether there is signal coverage at all in this area of the SFN. However, sufficient field strength does not necessarily mean that proper reception of the DAB signal is possible. This signal level could be made out of contributions from several transmitters which are not properly synchronised in time and/or frequency.
2. **TII:** Analysis of the TII with a measurement system delivers the identity of the received transmitters, i.e. their pattern and comb numbers and the relative signal strength as shown in Figure 7.27. The corresponding FIG 0/22 gives additional information about transmitter location and time offset.

TII-Information				
ENSEMBLE		**LABEL**	**FREQUENCY**	
00004258		DAB Bayern	223936 kHz	
TRANSMITTER			**COMB**	**PATTERN**
1	49%		4	1
2	30%		7	2
3	21%		6	2

Figure 7.27 Typical analysis of TII information

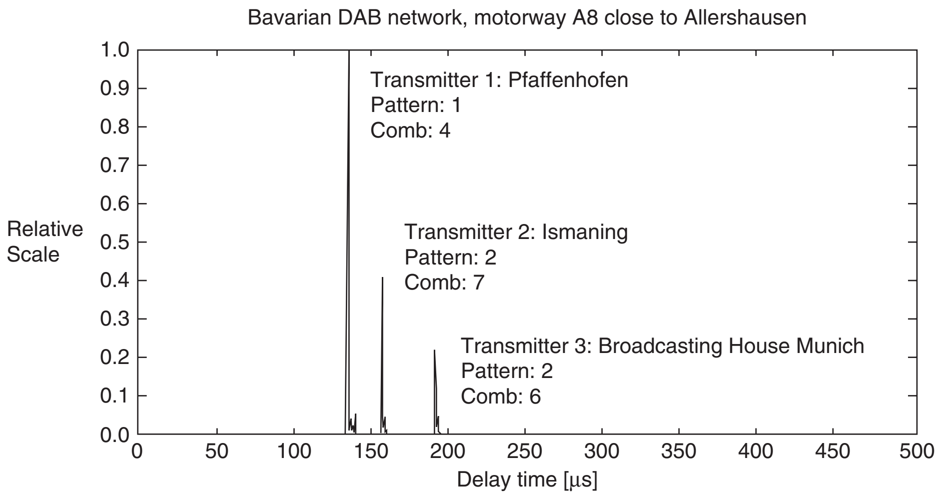

Figure 7.28 Typical plot of a channel impulse response

3. **Channel impulse response:** The channel impulse response shows in the time domain the different signal contributions which arrive at the receiver (see Figure 7.28). Evaluation of the channel impulse response shows whether the guard interval is violated or not. This parameter is therefore an indicator of the quality of the time synchronisation of the network. The channel impulse response together with the TII allows the identification of contributions from each transmitter. It must be noted that possible contributions from signal reflections due to the geographical topography must be taken into account in the evaluation process.
4. **Transmit frequency:** All transmitters in an SFN must operate exactly on the same frequency. To achieve a performance degradation of less than 1 dB, the minimum required accuracy of the output frequency is 10% of the sub-carrier spacing, that is 100 Hz in Mode I, 200 Hz in Mode IV, 400 Hz in Mode II and 800 Hz in Mode III

[TR 101496]. This parameter can be measured at the output of each transmitter in the SFN.

5. **Bit error ratio (BER):** The BER is an indicator of the quality of the network. High BERs can be caused by not enough field strength or synchronisation errors in time and/or frequency. For good audio quality the BER should be 10^{-4} or better. The different types of BER are discussed in section 7.7.2. To indicate the quality of the received signal, CRC error measurements were discussed as alternatives to BER measurement at the early stages of DAB development. The DAB audio signal contains CRCs for the audio header and the scale factor which could be used. However, it turned out that these CRC errors are not sensitive enough since they only occur once the audio signal degrades audibly.

Coverage evaluation of a DAB network can be done by measuring and evaluating the above parameters. If the coverage is below the required level (e.g. a certain BER for typically 99% of the time at 99% of the area) careful analysis of this set of parameters and their interdependencies allows the planner to derive the measures necessary to improve the coverage quality.

All the parameters described so far represent well-measurable effects that do not vary much with time. However, to complete the coverage evaluation, investigation of long-distance interference must also be made. This type of interference depends heavily on propagation conditions and is only measurable for a small percentage of time. These effects are therefore usually modelled statistically in advance and taken care of in the planning phase of the network.

7.7.2 A Closer Look at Bit Error Ratio Measurements

There are two different types of BER in a DAB system. Depending on the application, the channel BER or the net BER can be used as a measure for the quality of the coverage. In comparison, the net BER is the exact method for BER measurement whereas the channel BER stands for the pragmatic approach. The BER is defined as the ratio of number of errored bits received to total number of bits received, in contrast to the 'bit error rate', which is defined as the number of incorrect bits in a certain time interval.

7.7.2.1 Channel BER

The channel BER can be calculated by comparing the re-encoded bit sequence in the receiver to the actual bit sequence received. To measure the exact channel BER, the complete bit stream would have to be re-encoded in the receiver. An alternative to this concept is to re-encode only part of the received signal, that is only the first bit of the 4-bit mother code is re-encoded. Most receiver decoder chips contain this feature and generate a 'pseudo-channel BER'. Figure 7.29 gives the block diagram for the channel BER calculation using an MSC sub-channel as the reference bit stream. An alternative to the MSC BER is the FIC BER which uses the non-time-interleaved FIC data stream as the reference bit stream.

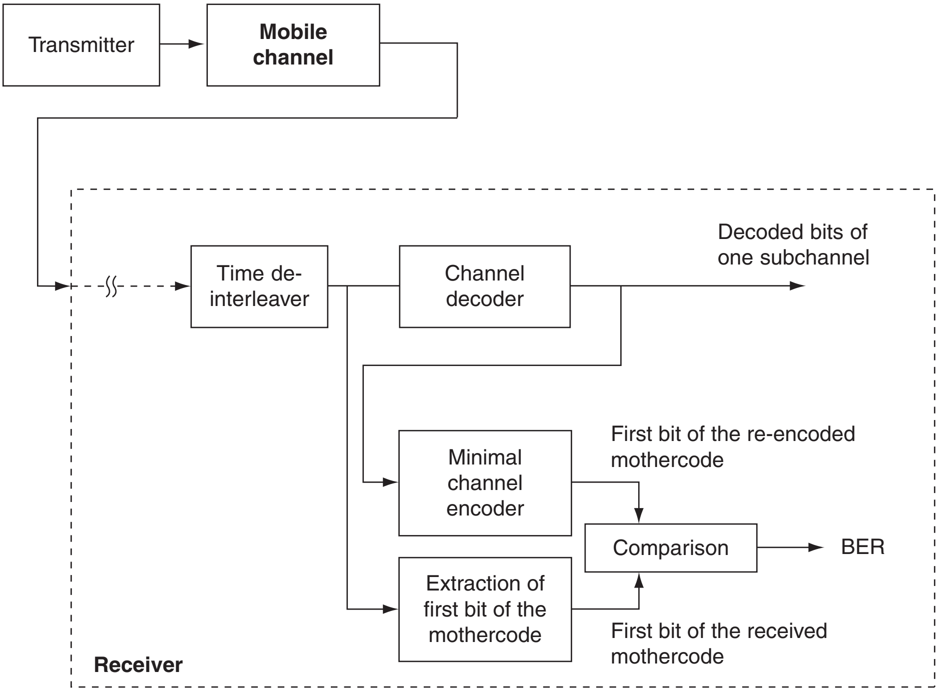

Figure 7.29 Generation of the pseudo-channel BER using a MSC sub-channel

The shortcomings of this method can be clearly seen in Figure 7.29. The calculation of the (pseudo-)channel BER is only correct if the channel decoder is able to correct all transmission errors, i.e. it produces an ideal reference bit stream at the output. This is true for low bit errors and channel bit error measurement give good indications for BER in the order of 10^{-4} and lower. The advantage of this method lies in its simple implementation [Schramm, 1997].

7.7.2.2 Net BER

The net BER is calculated by comparing the received and decoded bit stream directly to the reference bit stream. In this case, the DAB transmission system is treated as a black box and a comparison of the output and the input signals of this black box gives the net BER (Figure 7.30).

To calculate the net BER, the reference signal must be known. For on-line measurements, a known test sequence must be transmitted, be it in a small subchannel or in the PAD which takes up some of the capacity of the multiplex.

An advantage of the net BER measurement is that the result is always correct irrespective of the quality of the transmission [Frieling, 1996]. The net BER is interesting for service providers who are only interested in the quality of their service and treat the DAB system simply as another distribution channel for that service.

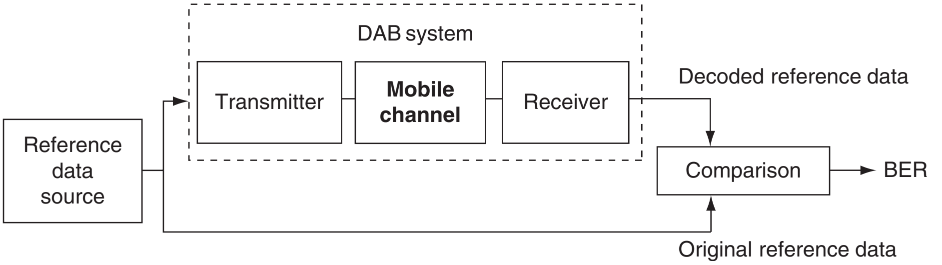

Figure 7.30 Generation of the net BER

7.7.3 Timing Adjustment of SFNs

The only way to check for correct time synchronisation in an SFN is to evaluate the channel impulse response together with the TII. If the timing within the SFN is not correct, the delay times for certain transmitters must be adjusted. This can be done centrally in a DAB network from the ensemble multiplexer site. The ETI output signal of the multiplexer allows each transmitter site to be addressed and the delays of each transmitter to be set individually. This convenient feature avoids the time- and labour-consuming alternative of having to send people directly to the different transmitter sites to perform the necessary adjustments.

7.7.4 Monitoring of SFNs

Four parameters determine the quality of an SFN and must be constantly monitored: the correct transmit signal according to [EN 300401], level of field strength, accuracy of transmit frequency and accuracy of timing between the different transmitters.

The correct transmit signal and the accuracy of transmit frequency can be monitored at each transmitter site using a reference receiver and a frequency counter. Monitoring the level of field strength must be done at two locations: firstly, at the transmitter site by monitoring the output power of the transmitter and, secondly, in the field by constantly measuring the field strength and comparing it to the required values. If all is correct at the different transmitter sites, an alarm by a field strength measurement probe in the field would be triggered only by a change in propagation conditions. The timing of the SFN can only be monitored through units that measure the channel impulse response in the field. Changes of the channel impulse response can be caused by wrongly timed transmitters or a change in propagation conditions.

7.8 Frequency Management

7.8.1 General Aspects

In order to introduce a broadcasting system, international agreement on frequency bands and a channel raster for this system is necessary. All DAB transmissions have a centre frequency on a nominal 16 kHz lattice, and two frequency bands are, in effect, standardised for DAB, though their availability varies from country to country. These are Band III (174–240 MHz) and L-Band (1452–1492 MHz). Within these bands, individual centre frequencies for DAB transmissions are specified, which simplifies management of the spectrum as well as receiver design. The spectrum occupied by an ensemble centered on a particular frequency is referred to as a ‘block’. A full list of blocks and corresponding centre frequencies is given in Appendix 3 and [EN 50248].

In Band III, DAB frequencies are specified to accommodate four DAB ensembles within a 7 MHz television channel. This is illustrated schematically for Channels 11 and 12 in Figure 7.31. The four blocks in each channel are numbered A to D. Korea adopted a slightly different channel raster in Band III.

In the L-Band the 23 frequency blocks are labeled LA through LW in Europe; however, not all of these blocks are available for terrestrial DAB. In Canada the frequency blocks are labeled by the numbers 1 through 23. In a number of countries such as Canada where Band III is not available, only the L-Band spectrum can be used for terrestrial DAB.

In addition to frequency bands and channel raster, international agreement on a frequency plan is required before broadcasters can start to transmit services on a particular frequency or block. In order to reach such agreement, it is necessary to define criteria for sharing the use of frequency blocks in different geographical areas, i.e. to ensure spectral and/or spatial decoupling of transmissions. This can be achieved by defining a maximum allowed interfering field strength at the border of a neighbouring service area which uses the same frequency or by restricting the power of a transmitter or network to an allowed limit (or a combination of both). Such international agreements are fixed in frequency plans with their associated procedures to rule the implementation of the networks and the modification of the plans. The latter aspect of modification of a frequency plan is of utmost importance since it allows for an adequate development of the broadcasting services.

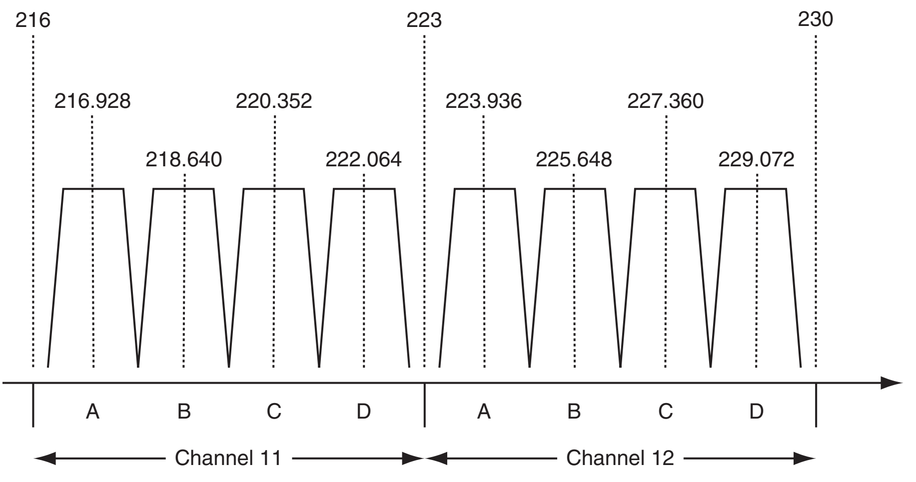

Figure 7.31 Block allocations in Channels 11 and 12 (all frequencies in MHz)

The remaining sections of section 7.8 describe the existing frequency plans for T-DAB, how frequencies are allocated and the practice of frequency coordination.

7.8.2 The Frequency Planning Conferences for DAB in Europe

This section describes the history and development of frequency plans for terrestrial DAB in Europe. The principles of the currently valid plans and rules for coordination are described in section 7.8.4.

7.8.2.1 Wiesbaden 1995 (WI95)

In 1995 CEPT established in Wiesbaden the first frequency plan for terrestrial DAB in Europe [CEPT, 1995]. The plan is based on the allotment concept: Geographical areas are identified where a frequency block may be used by an administration with certain interference restrictions, however, without specifying the detailed characteristics of the network. For more details see section 7.8.4. The allotment concept is in contrast to the traditional assignment planning concept where the detailed characteristics of any transmitter entering in the plan are fixed.

Two coverage layers were planned for each country, i.e. a homogeneous coverage concept was realised where in each location of the planning area two DAB multiplexes can be received. One layer was planned in Band III, mainly in channel 12. The other layer was planned in the L-Band, where nine frequency blocks (LA – LI) were available. This is valid for most of the CEPT countries with only a few exceptions, e.g. France, where at that time no Band III usage by DAB was possible and therefore two L-Band layers were realised. As reception mode a mobile environment had been assumed and for each frequency range one reference network was defined. For the concepts of layers and of reference networks see also section 7.8.4. The Wiesbaden 1995 frequency plan and agreement is called WI95.

7.8.2.2 Maastricht 2002 (MA02 and WI95revMA02)

In 2002 CEPT extended the L-Band frequency plan for terrestrial DAB by another seven frequency blocks (LJ – LP). For each CEPT country a further DAB layer was made available in this band. In order to accommodate also the needs for smaller service areas two additional reference networks were defined which cover a smaller area and whose interference potentials are reduced with regard to the original first L-Band reference network. Details can be found in [BPN 030] and [CEPT, 2002b]. This extended DAB frequency plan for the L-Band is called MA02.

Since with MA02 the L-Band part of the WI95 plan was shifted into a separate frequency plan the remaining part relating to Band III was formally revised and called WI95revMA02. However, no modifications were made with regard to the plan entries and procedures [CEPT, 2002a].

7.8.2.3 Geneva 2006 (GE06)

In 2006 the ITU Regional Radiocommunication Conference (RRC-06) in Geneva established a frequency plan in Region 1 (more exactly: ‘in parts of Region 1 and Iran’) for the introduction of digital broadcasting, called GE06 [RRC06, 2006], [O'Leary, 2006]. This frequency plan covers both a large planning area (Europe, Africa, RCC countries, Middle East and Iran) and the broadcasting Bands III (174 MHz - 230 MHz) and IV/V (470 MHz – 862 MHz) which form the major part of the spectrum used by broadcasting. In Geneva, T-DAB was assumed as the standard system for audio broadcasting and DVB-T for television broadcasting. Whereas Band IV/V was reserved for DVB-T only, Band III could be used in a mixed way for both T-DAB and DVB-T according to the wishes of the administrations. The GE06 plan allows allotment plan entries as well as assignment plan entries; however, in Europe for T-DAB only allotments have been entered.

A 7 MHz channel raster for DVB-T was assumed in Band III in Europe and the identical frequency block raster of WI95 was adopted for T-DAB. Throughout Europe a mixed T-DAB/DVB-T usage was assumed for Band III with one DVB-T layer and three T-DAB layers. In most cases the former WI95 layer in Band III now forms one of the GE06 T-DAB layers. In total, more than 8000 DAB plan entries are found in the GE06 plan.

The technical criteria used in GE06 for DAB are to a large extent taken from WI95 (the technical annex of which has been transferred to the ITU context in [BS.1660]). An important improvement has been achieved in GE06 by introducing a portable indoor reception mode for DAB as an additional receiving environment. In fact, most of the European DAB plan entries now are based on this reception mode which allows for higher transmitter powers. Accordingly, an additional reference network with increased transmitter powers by 9 dB was introduced. More technical details are given in section 7.8.4.

An interesting new frequency planning mechanism was introduced in GE06 with the so-called envelope concept. It allows for the operation of a different transmission system – technically as well as regulatory – as indicated in the plan entry as long as certain regulatory and technical constraints are fulfilled.

7.8.2.4 Constanţa 2007 (WI95revCO07 and MA02revCO07))

In 2007 CEPT adapted in Constanţa the WI95revMA02 frequency plan relating to Band III to the fact that the major part of this band is now ruled by GE06. Apart from minor parts – channel 13 plan entries (230 MHz – 237 MHz) and one plan entry in Band I (47 MHz – 68 MHz) – WI95 relating to VHF was abrogated. A transition period was established until 2012 within which WI95 plan entries are protected [CEPT, 2007a].

With regard to the L-Band, Constanţa 2007 confirmed the procedures and the frequency plan of Ma02 (MA02revCO07). In addition, a concept similar to that of GE06 was agreed which allows also for the implementation of other mobile multimedia systems than DAB, including the aggregation of frequency blocks. However, no indoor reception mode has been defined for the L-Band as was introduced in Band III in GE06 [CEPT, 2007b].

7.8.3 The Frequency Plan for DAB in Canada

As an example of a non-European frequency plan for DAB the case of Canada is described.

In Canada, the 23 blocks of the 1.5 GHz Band were allotted throughout the country except in western and eastern Canada where the upper five blocks had to be left unused to accommodate Mobile Aeronautical Telemetry systems operating in various locations within the USA. The centre frequency of these blocks is slightly different from those in Europe since no duplication of the Band III spacing was needed. The raster is centred with respect to the band (i.e., centre frequency of the middle block 12 is exactly at 1.472 GHz, with the L-Band frequency allocation ranging from 1.452 GHz to 1.492 GHz) and the blocks have their centre frequency separated by 1.744 MHz, leaving a guard band of 48 kHz at both ends of the band. The resulting DAB plan allowed all current AM and FM broadcasters to be accommodated.

7.8.4 Allocation of Frequencies and Practice of Coordination

This section describes the concepts that are applied in the present frequency plans for DAB, gives an overview of the technical aspects that underlie these concepts and gives information about the regulatory aspects for the modification of a plan entry and the implementation of a network.

7.8.4.1 The Allotment Concept

The allotment concept was introduced for the first time in the broadcast field in the Wiesbaden 1995 frequency plan for DAB [CEPT, 1995]. In the allotment concept, geographical areas are identified where a frequency is used with certain restrictions regarding the outgoing interference produced by the network implemented in this area. Different from the previous broadcast approach – the assignment approach – no detailed specification of the characteristics of the individual transmitters of the network is necessary. It is left to the particular network implementation to keep the overall restrictions regarding the outgoing interference.

The allotment concept is the adequate frequency planning approach for transmission systems which allow for single frequency networks. It gives more freedom to the network planner and allows for a better adaptation of the technical aspect of frequency planning to the design of broadcast service areas which are determined by social, cultural and political aspects.

In frequency planning the impact of the outgoing interference originating from a network is relevant. Since with the allotment concept the real network implementation is not yet known at the stage of the establishment of the frequency plan – and it is not even desirable to fix it once and forever – a representative network is defined in order to describe the expected and, under a regulatory aspect, allowed outgoing interference. The outgoing interference of a network is called its interference potential and the representative, artificial network is called reference network. On the basis of these assumptions and definitions a frequency plan can be synthesised where certain frequency re-use conditions have to be obeyed.

As an example, Figure 7.32 shows the GE06 plan for frequency block 12D in Central Europe. The figure clearly illustrates how the different national approaches have resulted in different uses of the same frequency, including the mixed use of this channel or frequency block by DAB and DVB-T, respectively.

Reference networks are described in more detail in the next section. A comprehensive treatment of reference networks and their interference potentials can be found in [ECC 049], [Brugger, 2005].

Figure 7.32 Allotments in frequency block 12D in the GE06 plan in Central Europe

7.8.4.2 The Concept of Reference Networks

The criteria for sharing a frequency are based around the predicted levels of interfering signals from one or more service areas into other areas using the same block. These signal levels must be kept below an acceptable threshold. Calculation of the levels is performed on the basis of specified service areas and 'reference networks' of transmitters that are felt to be sufficiently representative of a practical network. For the assessment of compatibility the interfering signal levels arriving in an area, from the reference network of another co-channel area, are calculated, and if the levels are sufficiently low, then sharing of that block is possible. Planning models are given in [BPN 003], [Brugger, 2005] for VHF networks and in [BPN 030] for L-band networks and in addition in the conference texts [RRC06, 2006], [CEPT, 2007b].

Figure 7.33 shows an example of such a reference network in Band III for mobile reception as it was used for GE06. The main transmitter in the reference hexagon has 100 W output power and the six peripheral transmitters have 1 kW each. Since directional antennas are assumed for the peripheral transmitters, this set-up is called a closed

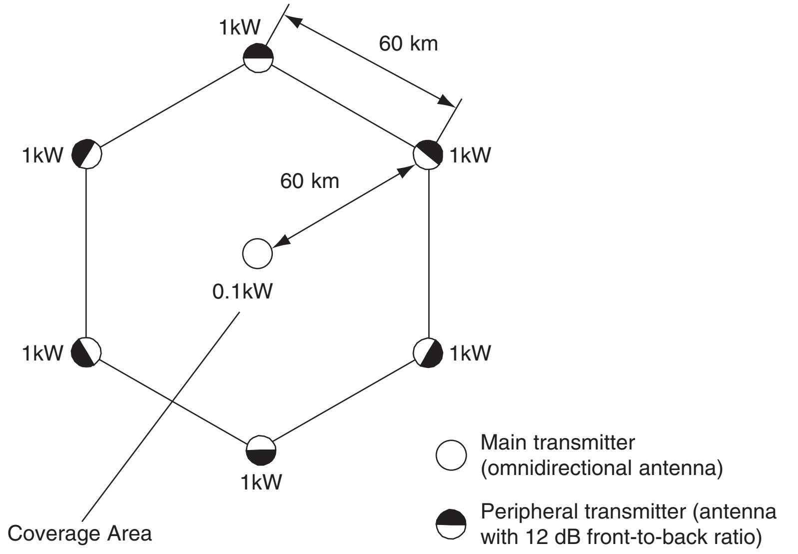

Figure 7.33 GE06 reference hexagon for Band III

hexagon structure. An effective antenna height of 150 m is assumed and the distance between each of the transmitters is 60 km. GE06 used this reference network also as the second reference network with an identical geometry but 9 dB higher transmitter powers for a portable indoor reception environment. The properties of the reference networks are summarised in Table 7.12.

Table 7.12 Reference networks for mobile and portable indoor reception in Band III (GE06)

Reference Network	RN – mobile reception	RN – portable indoor reception
Reception mode	mobile	portable indoor
Type of network	closed	closed
Geometry of service area	Hexagon	Hexagon
Number of transmitters	7	7
Geometry of transmitter lattice	Hexagon	Hexagon
Distance between transmitters d	60 km	60 km
Service area diameter D	120 km	120 km
Transmitter effective antenna height	150 m	150 m
Peripheral Transmitter antenna pattern	Directional 12 dB reduction over 240°	Directional 12 dB reduction over 240°
Central Transmitter antenna pattern	Non-directional	Non-directional
Peripheral Transmitter e.r.p.	30.0 dBW	39.0 dBW
Central Transmitter e.r.p.	20.0 dBW	29.0 dBW

Because of the different propagation conditions in the L-Band, the reference networks for this frequency range have different properties. The most prominent difference is the reduced distance between the transmitters. Table 7.13 summarises the properties of the three reference networks applied in MA02.

7.8.4.3 The Layer Concept

In both the CEPT and the ITU frequency planning conferences a layer concept was applied for Europe. In this approach a homogeneous coverage concept is adopted where in each location of the planning area the same number of programmes or multiplexes are made available. In this way full area coverage of all countries is achieved. This concept is preferably applied in frequency planning conferences since it allows for a simple realisation of the principle of equitable access to the radio spectrum – the holy grail of international frequency planning. However, in practice – when a frequency plan is going to be implemented – often a mixture of full area coverage and population centre coverage with aggregation of frequencies in the latter is realised.

The MA02 plan provides two DAB layers for each country in the L-Band and GE06 provides three DAB layers for each country in Band III.

7.8.4.4 The Envelope Concept

As well in the GE06 as in the MA02revCO07 agreement the envelope concept was introduced as a new frequency planning mechanism. It allows for the operation of a different

Table 7.13 Reference networks for mobile reception in the L-Band (MA02)

Reference Network	RN 1	RN 2	RN 3
Reception mode	mobile	mobile	mobile
Type of network	open	closed	closed
Geometry of service area	Hexagon	Hexagon	Hexagon
Number of transmitters	7	7	4
Geometry of transmitter lattice	Hexagon	Hexagon	Square
Distance between transmitters d	15 km	26 km	16 km
Service area diameter D	60 km	45 km	23 km
Transmitter effective antenna height	150 m	150 m (central) 50 m (peripheral)	50 m
Peripheral Transmitter antenna pattern	Non-directional	Directional 12 dB reduction over 240°	Directional 12 dB reduction over 225°
Central Transmitter antenna pattern	Non-directional	Non-directional	n.a.
Peripheral Transmitter e.r.p.	30.0 dBW	37.0 dBW	30.0 dBW
Central Transmitter e.r.p.	27.0 dBW	31.0 dBW	n.a.

transmission system as indicated in the plan entry as long as not more interference is produced and not more protection is claimed as allowed and given by the original plan entry.

As an example, this concept is, in particular, of interest for DAB since it allows for the operation of T-DAB multiplexes under a DVB-T plan entry. The possible applications are investigated in [ECC 116]. The conclusion is that DVB-T plan entries for portable indoor reception are well suited for a conversion to DAB, but also with DVB-T plan entries for portable outdoor reception a conversion is possible.

In particular, it allows for the operation of four T-DAB multiplexes under a DVB-T plan entry in Band III. In fact, currently, several European administrations think about using their DVB-T plan entries in Band III for DAB/DMB implementations. However, the reception mode associated with the DVB-T plan entry should comply with the intended DAB reception mode. This is not a matter of course, since several administrations have chosen fixed roof-top reception as the reception mode for DVB-T.

7.8.4.5 Multilateral Coordination and Modification of a Frequency Plan

A frequency plan for a broadcasting system starts its existence with the establishment by a planning conference. However, time going on, it does not remain in its original state. Provision has to be made for its modification and development in order to adapt it to future needs. Actually, this is a major and important feature of a broadcasting frequency plan. Both the GE06 and MA02revCO07 plans allow for this possibility, although they are already relatively dense. Nonetheless, they give room for addition of further allotments or extension of existing allotments.

From a regulatory point of view, a modification of a frequency plan – either addition of a plan entry or modification of a plan entry – has to be agreed on by all potentially affected countries. Traditionally, the procedures for such a modification are treated in Article 4 of an international frequency plan agreement. This is why such rules are called 'Article 4 procedures'. The first step in such a procedure is the identification of the potentially affected administrations. Secondly, a certain time frame for agreement/disagreement to the proposed plan modification has to be kept with certain duties and responsibilities of the involved administrations and the respective plan management body, which is in the case of GE06 the ITU Radiocommunication Bureau (BR) and in the case of MA02revC007 the European Radiocommunication Office (ERO) of the CEPT.

In MA02revCO07 a country is regarded as potentially affected by a proposed modification of the plan if the interference potential of the respective – new or modified – plan entry exceeds the value of 38 dB(μV/m) at the border of the country. In GE06 the criterion is similar: a trigger field strength contour at 12 dB(μV/m) is constructed from the reference network of the plan entry to be modified or newly introduced. All countries lying inside this contour are regarded as potentially affected.

The trigger field strength value in GE06 is lower, i.e. more sensitive, for several reasons. Firstly, for the GE06 plan, a more conservative approach was taken with regard to the identification of potentially affected administrations, as a principle. Secondly, in GE06 not only DAB services are to be protected in Band III, but also analogue television services (for a transition period until 2015) and DVB-T services. For these services, cases exist which are more sensitive to inference from a DAB service than is intra-service interference of DAB.

In particular for GE06, the coordination rules and procedures are much more complicated as described in this short section. This is due to the fact that GE06 is a mixed assignment-allotment plan where the principles of both approaches have been combined. Moreover, additional rules and procedures apply to other, non-broadcasting services in Band III and L-Band.

A detailed description of the procedures can be found in the agreement texts [CEPT, 2007b] and [RRC06, 2006] and for GE06 in addition also in [BPN 083].

7.8.4.6 Bilateral Coordination of a Frequency Plan

In addition to multilateral international agreements, bilateral agreements between countries can allow additional use of frequency blocks, particularly where the geography and the topography are favourable. Administrations can agree on more sophisticated methods for the determination of interference, e.g., topography-based field strength prediction methods, or agree on higher interfering field strengths in particular cases. In practice, bilateral talks and agreements are a powerful means of improving the usage of the spectrum in border regions. As an example, already during the planning conference RRC-06 many special agreements between administrations have been established which rendered the overall planning easier.

7.8.4.7 Regulatory Aspects of the Implementation of Networks

In the past, for analogue broadcasting systems, the regulatory aspects of network implementation were relatively simple since in frequency plans for analogue systems, which were assignment plans, the transmitter characteristics were already determined. This is different from digital broadcasting systems where the allotment concept has been applied. Here only an upper bound for the outgoing interference of the network to be implemented is defined, and a separate regulatory mechanism is needed in order to ensure that the interference potential of the network implementation does not exceed the allowed upper bound.

Such a mechanism can be realised in various ways. In the CEPT agreement MA02revCO07 each allotment having been entered into the frequency plan is supplied with a set of so-called ‘calculation test points’ which lie at a certain distance from the border of the allotment area. When a network is implemented the interference potential of the network is calculated at the calculation test points and checked against its maximum allowed field strength fixed in the frequency plan agreement. If the interfering field strength of the network implementation lies below this upper bound the implementation conforms to the plan.

A similar mechanism is implemented in the GE06 agreement, called ‘conformity check’. There the calculation points are generated by a specific geometrical algorithm and the interference potential of the network implementation is compared with the interference potential of the reference network that is associated with the allotment plan entry. If the latter is not exceeded by the network implementation conformity with the plan is achieved.

These checks are performed by the respective plan management bodies, ERO for MA02revCo07 and ITU-R BR for GE06. Details of the technical and regulatory procedures can be found in the agreement texts of these frequency plans [RRC06, 2006], [CEPT, 2007b] and also in [BPN 083].

8

The Receiving Side

8.1 General

DAB is different from the traditional analogue audio broadcasting systems like AM and FM. For example, DAB is a broadband transmission system, transmitting several audio programmes and data channels over the same frequency. The frequency bands assigned for DAB broadcasting are different from the traditional broadcasting bands and are separated almost by a decade. The transport of the information, audio and data also employs new concepts like audio compression (see Chapter 3). Therefore, new receiver concepts had to be developed.

Since the DAB system is fairly complex in terms of computational requirements, it was evident that there was a need to design specific, highly integrated chip-sets covering both the analogue and digital parts of the system. These chip-sets are the building blocks for all kinds of DAB receivers and are the vital basis for cost-effective solutions. However, owing to the rapid developments in PC technology, modern PCs are able to handle the digital part of a DAB receiver.

Today, the various types of DAB receivers can be categorised as follows:

- Home receivers including HiFi tuners, kitchen radios, clock radios and portable stereo systems ('boomboxes').
- Car radios, either a head unit with a digital radio black box or all-in-one.
- Portable receivers and mobile phones including DAB/DMB.
- PC-based receivers whereas the receiver is a DAB-enabled notebook or a USB device.
- Monitor receivers for network monitoring.

The design of a good and inexpensive DAB receiver is not a trivial task. Expertise from different engineering fields is required for a successful development. System expertise of the DAB standards and also the increasing number of other broadcast standards the receiver supports is a prerequisite. Since nowadays more and more features of digital

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

signal processing are implemented in software, the increasing complexity requires advanced software engineering capabilities. A good receiving performance as one of the major headstones of a broadcast receiver mainly depends on the quality of the RF circuitry design and consequently on the knowledge of the RF experts. Last but not least, HMI (human machine interface) experts shall ensure an easy and comfortable operation of the receiver.

This chapter delivers an insight into these different aspects of receiver technology.

8.1.1 Normative Receiver Requirements

Several normative requirements related to DAB receivers have been standardised during the last few years. An overview of these standards is provided in Table 8.1. The scope of the various normative standards is referred to in subsequent sections as indicated in Table 8.1.

The most important among these standards are the 'minimum receiver requirements', standardised by CENELEC [EN 50248], which define the minimum and typical

Table 8.1 Normative receiver requirements

Title	Scope	Reference	Section
CENELEC: EN 50 248 (2001). Characteristics of DAB receivers.	Minimum receiver requirements	[EN 50248]	8.1
EACEM: TR-004 (2000). Application of the EMC Directive 89/336/EEC for Digital Audio Broadcast receivers.	EMC directive	[TR-004]	8.1
ETSI: EN 300 401 V1.4.1 (2006). Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers.	DAB standard	[EN 300401]	8.1
ETSI: TS 102 563 V1.1.1 (2007). Digital Audio Broadcasting (DAB);Transport of Advanced Audio Coding (AAC) audio	Audio coding	[TS 102563]	8.4
CENELEC: EN 50 255 (1997). Digital Audio Broadcasting system – Specification of the Receiver Data Interface (RDI).	Data interface	[EN 50255]	8.5
CENELEC: EN 50 320 (2000). Digital Audio Broadcasting system – Specification of the DAB command set for receivers (DCSR).	Data interface	[EN 50320]	8.5
ETSI: EN 301 700 V1.1.1(2000). Digital Audio Broadcasting (DAB); VHF/FM Broadcasting: cross-referencing to simulcast DAB services by RDS-ODA 147.	FM cross-referencing	[EN 301700]	8.8
ETSI: TS 102 368 V1.1.1 (2005). Digital Audio Broadcasting (DAB); DAB-TMC (Traffic Message Channel)	TMC	[TS 102368]	8.8

Table 8.1 (*continued*)

Title	Scope	Reference	Section
ETSI: TR 101 758 (2000). Digital Audio Broadcasting (DAB); Signal strength and receiver parameters; Targets for typical operation.	Receiver requirements	[TR 101758]	8.1
ETSI: TR 101 496-2 V1.1.2 (2001). Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 2: System features.	Implementation requirements	[TR 101496-2]	8.1
IEC: EN 62 516 (2008). Terrestrial digital multimedia broadcasting (T-DMB) receivers (Draft)	DMB receiver requirements	[EN 62516]	8.1
ETSI: TS 101 757 (2000). Digital Audio Broadcasting (DAB); Conformance Testing for DAB Audio.	DAB audio decoder	[TS 101757]	8.4

performance of DAB receivers. EN 50248 is the result of a joint effort of the receiver industry and experts from Eureka 147. Some important details are discussed in paragraph 8.1.2. In addition, EACEM [TR-004] identifies specific references, requirements, methods and test conditions for EMC tests for DAB receivers. Two more standards with helpful information are available from ETSI. First, [TR 101758] describes the general principles for deriving values for field strength and the compatible receiver sensitivity for satisfactory operation of a DAB system. Second, [TR 101496-2] gives guidelines and rules to comply with the system features which should also be taken into account by receiver manufacturers. Both support the interpretation of the on-air signal. Additionally, for DMB receivers a requirement specification [EN 62516] is available as a draft version considering the special needs for portable DMB services. It includes implementation guidelines, receiver requirements for the audio, video and data signal as well as RF performance specification.

8.1.2 Discussion of the Minimum Receiver Requirements

The standard [EN 50248] describes the receiver characteristics for consumer equipment with focus on the functional performance requirements, interfaces and the minimum performance levels including the measurement methods. For a long time, the performance levels were up for discussion because of the different points of view of the involved parties. On the one hand very strict requirements for the receiver reduce the costs of the transmission network regarding the connection between network coverage and the sensitivity of the receiver. But on the other hand they also cause expensive receivers. In contrary, too weak requirements could cause bad receivers. But the time showed that some manufacturers brought much better receivers onto the market in comparison to the requirements of the standard.

In the following, the most important functional performance requirements are listed. For a detailed description see [EN 50248].

- The audio decoder shall conform to the relevant standards (see Chapter 3) and shall be able to decode audio streams with 24 kHz and 48 kHz sampling frequency as well as bit-rates up to 256 kbps. Higher bit-rates are optional.
- The receiver shall detect the transmission mode (see section 2.2.1) of the DAB signal and switch automatically to the appropriate mode.
- To gain access to the desired service, the receiver must decode the MCI (see sections 2.3, 2.5), make the information available to the human machine interface for selection, and then output the selected service.
- The receiver shall follow the in advance signalled multiplex re-configurations (see section 2.3.7). For services which are unchanged, the reception should continue without adverse effect.
- In order to allow service following of a particular DAB service, the ability of a mobile DAB receiver to switch automatically to another ensemble is mandatory.

The performance levels and measuring methods are listed in the following. The measuring shall be performed for all frequency bands covered. Using standardised measuring methods ensures some comparability between receivers. But the standard does not provide a description of how the measurement results should be presented. This leads to misunderstandings since some manufacturers average several measurement runs, others take the worst result. Most of the tests consider the RF performance of the receiver. As criterion the BER is used which should be below 10^{-4} after the convolutional decoder with a code rate of ½ which ensures a nearly undisturbed audio signal.

- The sensitivity gives an indication of the lowest RF-input level and field strength of the DAB receiver for the given BER. In other words, it measures the ability of the receiver to decode weak signals from transmitters far away.
- In contrast to this, the maximum input power gives the maximum input level the receiver can properly decode, that means its ability to decode strong signals from transmitters very close.
- The selectivity of a receiver is a measure of its ability to discriminate between a wanted signal to which the receiver is tuned and unwanted signals, e.g. interfering signal entering the RF-Input. Two kinds of measurements are considered, adjacent channels selectivity and far-off selectivity. For the first case, the unwanted signal is located in the neighbouring channel of the wanted signal. For the second case, the unwanted signal has a frequency offset higher than 5 MHz. For the interfering signal DAB, DVB-T, analog TV, FM and other systems have to be taken into consideration.
- The measurement in Rayleigh channels provides a reference to the performance in a dynamically changing environment. The channel profiles were used from [COST 207] and simulate the behaviour in typical urban, rural and SFN environments at different speeds. The two considered parameters are the sensitivity and the acquisition time after synchronisation loss.

An evaluation of 15 consumer receivers based on the EN 50248 standard was performed [Schiphorst, 2008]. The publication gives a good overview of current receiver performance in Band III and L-band. The experiments reveal that there is a large difference in performance between consumer receivers: The 25% worst performing consumer receivers fail most experiments and the 75% best performing consumer receivers pass all experiments. Most difficulties appeared with non-adjacent channel selectivity in Band III and with decoding a DAB signal applying the [COST 207] rural area channel model in L-band.

8.1.3 Receiver Architecture Overview

Figure 8.1 presents a block schematic of a typical DAB receiver. The signal received from the antenna is processed in the radio frequency (RF) front-end, filtered and mixed to an intermediate frequency or directly to the complex baseband. The resulting signal is converted to the digital domain by corresponding analogue-to-digital converters (ADCs) and further processed in the digital front-end to generate a digital complex baseband signal. This baseband signal is further OFDM demodulated by applying an FFT (Fast Fourier Transform), see sections 2.2.1 and 8.3.2.

Each carrier is then differentially demodulated (DQPSK, see section 8.3.3) and the deinterleaving in time and frequency is performed. Finally, the signal is Viterbi decoded, exploiting the redundancy added at the transmitter side for minimising the residual error caused by transmission errors. After the Viterbi decoder, the source coded data, like audio and data services and FIC information, are available for further processing. The selected audio sub-channel is decoded by the audio decoder, whereas a data stream might be transferred to an external decoder through the receiver data interface (RDI, see section 8.5) or other interfaces.

Details of each processing step are provided in the subsequent sections 8.2 (RF Front-end), 8.3 (Digital Baseband Processing), 8.4 (Audio Decoder) and 8.5 (Interfaces).

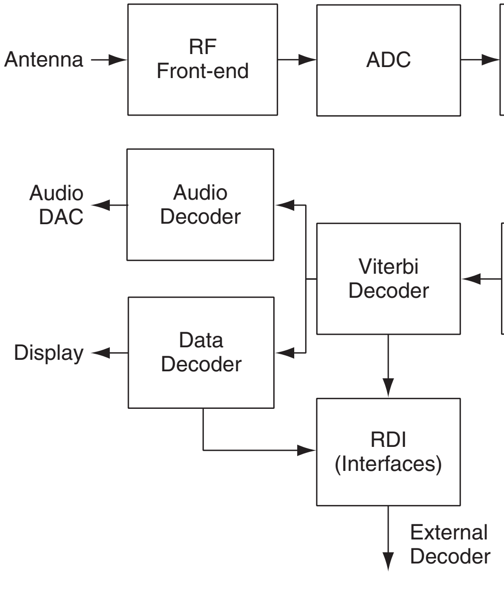

Figure 8.1 Receiver block schematic

8.2 RF Front-end

Since a COFDM-based DAB system has some special properties, traditional broadcasting receiver designs that have been designed for analogue transmission standards cannot be used. The new digital system imposes some special requirements which led to new receiver architectures.

8.2.1 Requirements

A COFDM signal can more or less be viewed as band-limited white noise. This is because the signal contains many individual carriers that are independently modulated according to the slow symbol rate, but as the individual carriers are completely uncorrelated, the signal behaves like Gaussian noise. In the frequency domain, the signal spectrum is limited according to the allocation of the sub-carriers and looks like a 'Simpson's head', very similar to a WCDMA (Wide-Band CDMA) signal. In the real world, insufficient filtering and transmitter non-linearities cause 'sidelobes' in the signal spectrum, limiting the adjacent channel rejection of the system (see section 7.4.5). Figure 8.2 presents the spectrum of a received DAB signal.

The special properties of COFDM impose some requirements on the receiver's RF circuits as described in the following subsections.

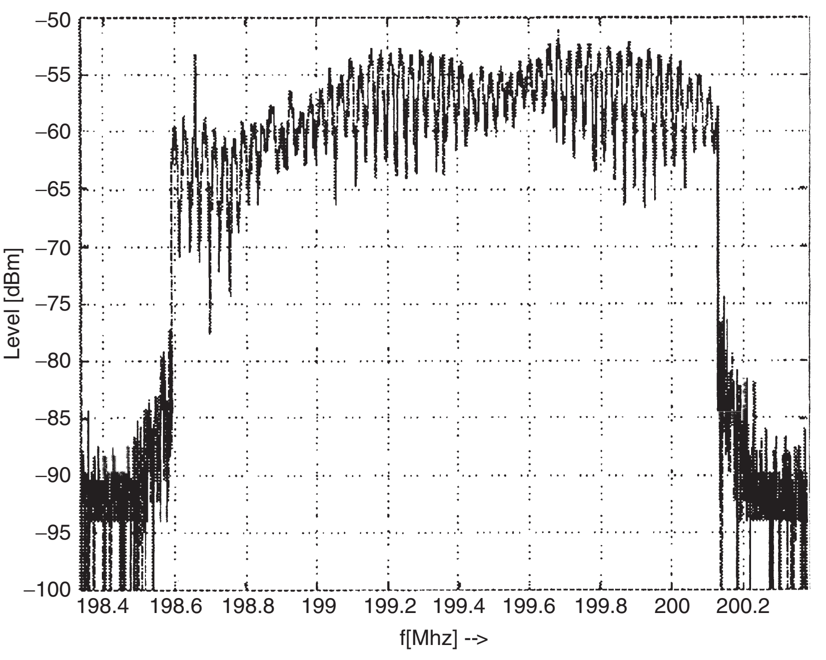

Figure 8.2 DAB signal in frequency domain. The ripple is caused by multipath propagation

8.2.1.1 Noise Influence

Different noise sources influence the reception performance of a receiver. Man-made noise arises from electrical appliances in industry and homes and will not be considered here. Thermal noise exists in every receiver due to the random thermal motion of the electrons. It can be modeled as additive white Gaussian Noise (AWGN). Crucial for the decoding quality in the receiver is the signal-to-noise ratio (SNR). The noise figure is defined as the ratio between the SNR at the antenna input and the SNR at the demodulator. In the design phase of a receiver, the total noise figure is calculated from the noise figures of its components as amplifier, mixer etc. using Friis formula for a cascade of stages. A low noise figure corresponds to good receiver sensitivity and the ability to decode weak signals. Figure 8.3 shows the bit error ratio (BER) in the AWGN channel over the input power for a typical DAB receiver. Both curves, with and without channel decoding, are diagramed. The BER after Viterbi decoding has to be below 10^{-4} to achieve an undisturbed audio signal.

8.2.1.2 Non-Constant Envelope

Unlike many phase-modulated systems like FM, GSM and AMPS, limiting amplifiers cannot be used, since the signal would be clipped and the amplitude part of the information would be lost. Limiting amplifiers would eliminate the requirement for gain control,

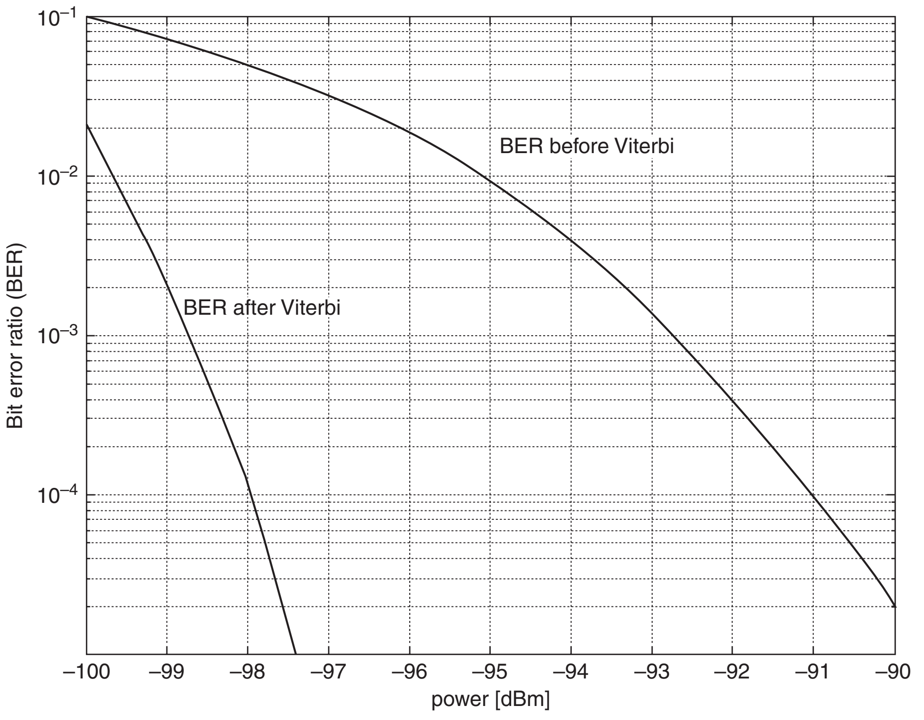

Figure 8.3 Bit error ratio in the AWGN channel for a typical DAB receiver

greatly simplifying systems design. A COFDM receiver requires a highly linear signal path from the antenna to the demodulator.

Because of mobile operation, the average signal amplitude constantly varies by about 20 dB. Hence, for COFDM receivers, careful AGC (Automatic Gain Control) design is a vital feature.

A special scheme called a null symbol (see section 2.2.2), a break in the transmitted signal of 1 ms in DAB transmission mode I, for example, is used for synchronisation. Special care has to be taken to pause the AGC circuit during reception of the Null-symbol.

8.2.1.3 Phase Accuracy

Another noise source is phase noise which is a random perturbation in the phase of a steady sinusoidal waveform. It is caused by the oscillator in the receiver. According to [Stott, 1998] phase noise produces two effects in OFDM-systems. The common phase error arises simultaneously on all carriers and generates a rotation of the whole signal constellation. It can be compensated in the receiver by clever implementations in the digital signal processing part. The other effect can be described as thermal-noise-like and amounts to inter-carrier interference which degrades the sensitivity. The overall phase noise is usually expressed in units of dBc/Hz in various offsets from the carrier frequency. Figure 8.4 shows an example of such a phase noise mask for DAB. The region near DC is

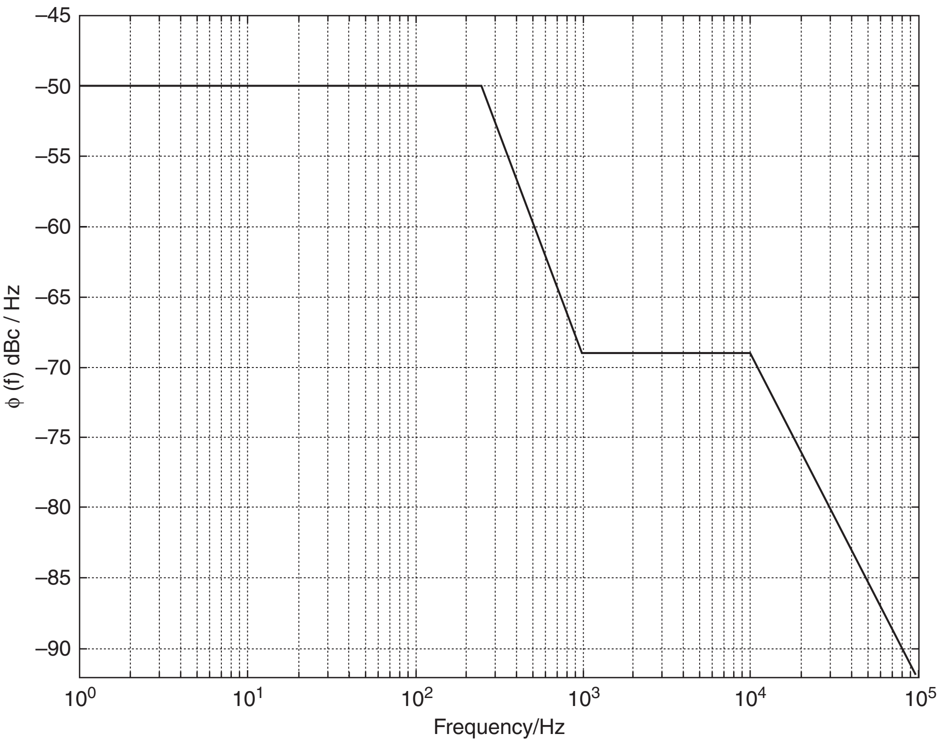

Figure 8.4 Typical phase noise mask for DAB

induced by the crystal itself and the plateau at medium frequencies is determined by the PLL bandwidth. The compliance with the values in the Figure 8.4 guarantees negligible performance degradation by phase noise.

Since the Eureka 147 DAB system partly uses the same frequencies as classical analogue TV signals, conventional TV tuners might be an option for a DAB receiver for Band III.

For TV tuners, VCO (Voltage-Controlled Oscillator) phase noise is usually of no concern. The signal information is contained in the amplitude; the signal phase contains no information. Even for the signal-to-noise ratio of FM sound, phase noise is not important since in most analogue receivers the stable signal carrier of the transmitted signal is used to generate the 5–5.5 MHz sound carrier according to the CCITT TV standard used in many countries.

The COFDM signal consists of many individual carriers which are modulated in phase. A VCO with low phase noise (jitter) is essential for downconverting the signal from the input frequency to a proper IF. Although absolute phases are not important because of the differential phase modulation, only a total RMS phase error of less than 10 degrees can be tolerated without affecting the BER of the system. Therefore, high-performance VCO and PLL circuits are required, exceeding the specifications of those for analogue systems.

8.2.1.4 High Linearity

All active components as amplifiers and mixers in the RF-front-end show a weakly nonlinear behaviour. It is typically modelled using a low order polynomial and described by the third-order intercept point IP3 related to the 3rd order term of this polynomial. The nonlinearities influence the reception performance in several ways. Firstly, the spectrum of the signal is broadening which leads to a limited performance when strong adjacent channels exist. Figure 8.5 shows a wanted DAB signal and an adjacent signal in both diagrams. In the left picture, no interference occurs. In the right picture, the unwanted signal interferes with the wanted signal which can be seen in the shoulder of the adjacent channel signal. The effect relates directly to the selectivity value of the receiver characteristic standard described in section 8.1.2. Secondly, two strong signals at the frequencies f1 and f2 also generate an interference signal at the frequencies 2f1-f2 and 2f2-f1. This case could happen in crowded DAB bands. Thirdly, the received signal itself will be disturbed by nonlinearities. This effect is negligible since the requirements for receiver linearity for the first effect are much stronger.

8.2.1.5 Wide Dynamic Range

DAB is a broadcasting system. An SFN may be viewed as a cellular network with each cell reusing the same frequency. Unlike cellular phone networks, where typical cell sizes are as small as a single building, a typical ‘cell’ for Band III is 50 km in diameter, and for L-band 15 km. Transmission powers are up to several kW, for example 10 kW for the Canadian DAB system, while base stations for cellular networks only typically use up to 10 W. This means that the receiver has to accommodate larger signals at the input, according to this difference in transmission power. While CDMA receivers are designed for a maximum input power of –25 dBm, a DAB receiver should be operational for up to

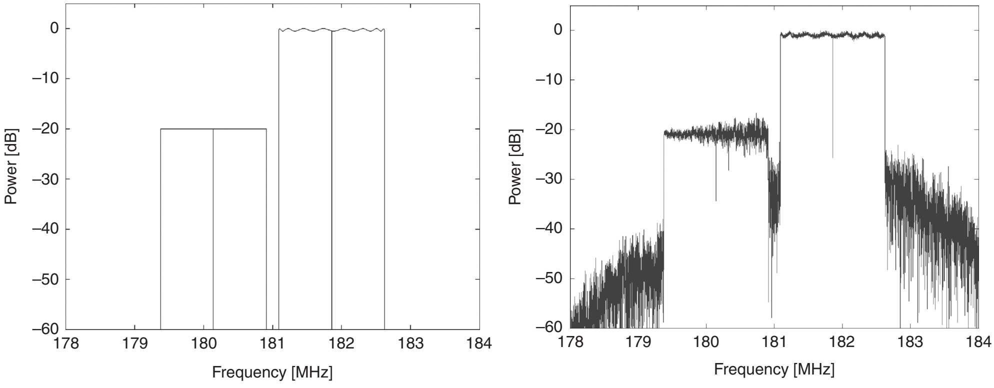

Figure 8.5 Spectrum of a DAB signal with adjacent signal, left picture without interference, right picture with interference

–15 dBm at the input in L-band. In Band III, maximum input levels are even more critical. In this case, not just DAB transmitters with moderate power levels about 1–10 kW are present. Strong TV stations occupy the same frequency band, with effective transmission powers of typically 100 kW. So it should be assumed that a nearby interferer may be up to 10 dB stronger than the strongest signal for L-band. [EN 50248] assumes a maximum input level of –5 dBm with linear circuit operation (see Table 8.2).

Although requirements are much harder than for mobile phone handsets, maximum signal levels are not as high as for FM reception where 2 V_{pp} at the antenna input are not uncommon in critical reception situations. Since the antenna aperture and hence the received signal level become smaller for higher frequencies, the situation is better for DAB, where the input frequency always exceeds 174 MHz.

High input levels require high linearity, and high linearity is a constraint not in accord with low-power consumption or good sensitivity. A DAB receiver must be optimised more for wide dynamic range and high selectivity, while a satellite receiver (e.g. GPS system) can be entirely optimised for a low noise figure only.

Table 8.2 Overview of the required values for the different receiver categories [EN50248]

Minimum requirement	Mobile receiver	Stationary receiver	Portable receiver
Minimum input power (Sensitivity)	−81 dBm	−81 dBm	−81 dBm
Maximum input power (VHF / L-Band)	−10 / −25dBm	−15 / −25 dBm	−20 / −25 dBm
Adjacent channel selectivity	≥30 dB	≥30 dB	≥30 dB
Far-off selectivity	≥40 dB	≥40 dB	≥40 dB
Rayleigh channel performance	−75 dBm	−75 dBm	−75 dBm
sensitivity /acquisition time	<3000 ms	<3000 ms	<3000 ms

8.2.2 *Analogue Front-end Concepts and Architectures*

Besides the special signal properties of OFDM signals, the two widely separated frequency bands (see Table 8.3) bring about certain constraints for receiver design.

The centre frequencies of both bands are separated by a factor of 7–8. Unlike TV or DVB-T tuners which can basically cover the required frequency range with the same single-conversion architecture, it is not possible to extend a UHF tuner up to 1.5 GHz, for example.

On the other hand, the two bands are quite narrow and it is not necessary to support reception at any frequency in between. This allows the design of special 'dual-band' receiver architectures.

8.2.2.1 Direct Conversion/Zero-IF

A common approach for multiband receivers is to use no IF at all and to down-convert the signal to an IF of zero (Figure 8.6). By doing this, any intermediate frequency translation steps that always bring about spurious reception frequencies, and hence special filter requirements, can be avoided. This approach is very promising and is presently adopted in triple-band GSM handsets, completely eliminating the need for IF filters [Strange, 2000]. Two or three switchable band select filters are used in front of the LNA (Low-Noise Amplifier). A similar architecture may be used for DAB and is presently being investigated.

Table 8.3 DAB Frequency Band

Band	Minimum	Maximum
L–band	1452 MHz	1492 MHz
Band III	174 MHz	239 MHz

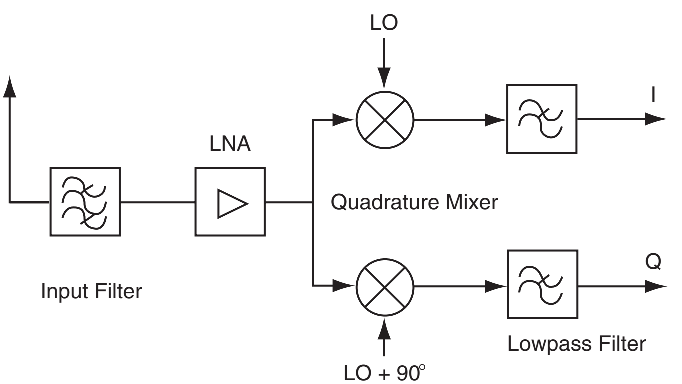

Figure 8.6 Zero-IF DAB receiver

This architecture, however, requires perfectly matched (with respect to amplitude and phase) low-pass filters in the I and Q signal path. For transmission systems with relatively small bandwidth (GSM: 20 kHz, IS95: 30 kHz), the required sample rates for the low-pass filters and ADCs are fairly low. For wide-band signals, more sophisticated low-pass filters are required. In addition, it has to be mentioned that this architecture requires two completely independent ADCs with corresponding good match. Usually the power consumption and costs are low but the performance is limited.

The most severe technical problem of zero-IF receivers is LO feedthrough to the antenna input, causing a DC offset in the IQ signal. This DC component has to be removed for proper signal processing, for example by capacitive coupling. This is not feasible with every modulation scheme, but COFDM is perfectly suitable for this approach.

8.2.2.2 Receivers Based on TV Tuner Concepts

Since TV tuners for Band III are readily available, most DAB receivers basically use a modified 'TV tuner' for Band III. This implies a similar IF of 38.912 MHz. The choice of the IF is determined by the required image rejection. With an IF of 38 MHz, the spurious image reception frequency for the lower end of Band III (174 MHz) would be above the upper band of the TV band, at

$$f_{image} = 174 + 2 \times f_{IF} = 250\,\mathrm{MHz} : \tag{8.1}$$

This frequency assignment prevents strong TV/DAB stations at the upper edge of Band III from acting as interferers for stations at lower frequencies. The spurious image reception frequency moves away from Band III if a frequency towards the upper end of Band III is tuned in.

Of course, if a traditional TV tuner is used for DAB reception, the IF filter has to be replaced by a proper DAB channel selection filter and the VCO has to be 'cranked up' for better phase noise performance.

Support for L-band can be added by a block downconversion. The width of L-band is smaller than the width of Band III, so it is possible to downconvert the complete L-band with an LO set to a fixed frequency down to Band III.

This concept, depicted in Figure 8.7, was originally developed inside the JESSI project [Jongepier, 1996]. It is often used in commercial receivers because several chip-sets based on this concept are available.

This tuner concept brings about some problems, however, especially at L-band:

1. The IF of 38.912 MHz is fairly high for direct processing by an ADC; usually a second (third in the case of L-band) IF is required.
2. The first IF frequency for L-band reception (which actually is Band III) is occupied by possibly very strong interferers. If for example a –95 dBm L-band station should be tuned with a –5 dBm blasting TV station present on the same frequency, more than 90 dB isolation is required. This is technically hard to achieve.
3. Tuneable image rejection filters ('tracking filters') have to be used. These filters are bulky and are subject to parameter variations. Therefore, filters have to be manually tuned or electronically adjusted, based on stored alignment data.

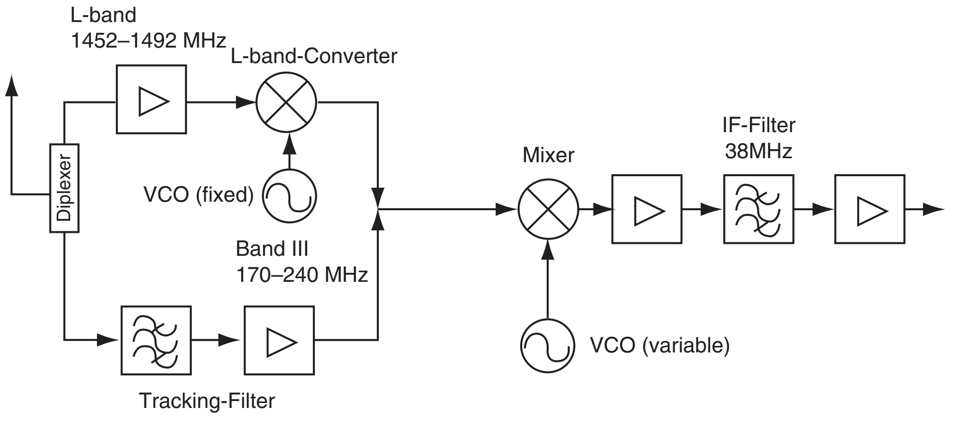

Figure 8.7 JESSI tuner

Since these tuners are highly optimised for TV reception and have a long design history, performance for Band III reception is usually good.

8.2.2.3 High-IF Tuner

Some limitations of the 'JESSI' tuner can be overcome by using upconversion for Band III. This concept has also been employed for TV tuners and set-top boxes [Taddiken, 2000]. A block diagram of a tuner derived from this tuner concept is given in Figure 8.8. The high-IF technique is used for Band III. L-band is down-converted, but also to high IF.

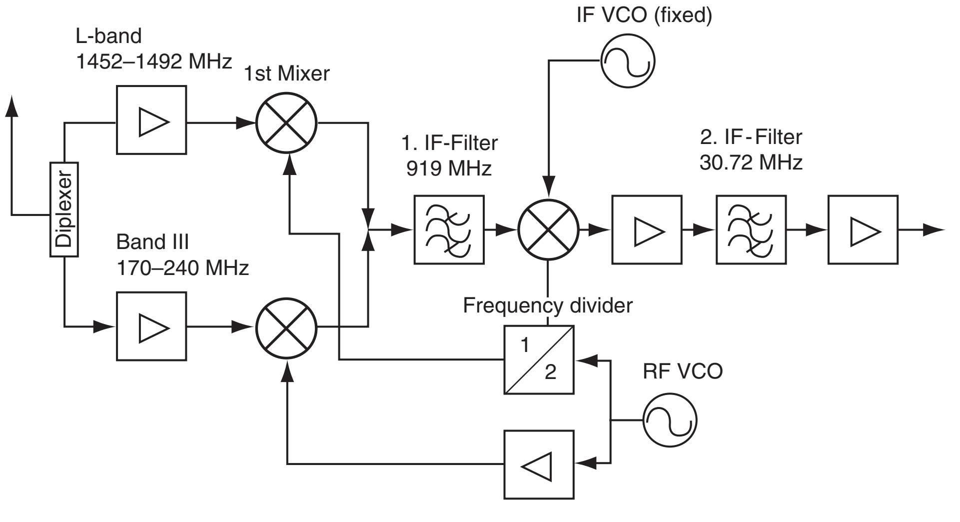

Figure 8.8 Receiver with high-IF for Band III

The most important advantage of this concept is the fact that the spurious image reception frequency for Band III is moved to very high frequencies. Therefore, no tuneable pre-selection filters are required, eliminating the requirement to manually/electronically adjust any components. With careful circuit design of the Band III LNA, tracking filters can be replaced by a band-pass filter. This enables a dramatic reduction of the form factor of the tuner, allowing compact designs for portable applications.

The first IF may be chosen according to two aspects:

1. IF should not be used by high-powered wireless services.
2. A common VCO with a limited tuning range should be used.

A possible approach might be to use an IF at 650 MHz, exactly in the middle of both bands. This frequency is, however, also used by strong UHF TV stations, but most importantly, this arrangement would place Band III at the image frequencies for L-band reception and L-band at the image frequencies for Band III reception, again causing isolation problems. A solution is to use frequency dividers for deriving the LO frequencies from a common VCO.

In the AD6002/6003 chip-set [Titus, 1998], [Goldfarb, 1998], [Clawin, 2000], an IF of 919 MHz is used. This frequency allows optimum Band III image suppression, is not used by a major service (actually, it is in the gap between GSM uplink and downlink), and only requires a VCO with a small tuning range of about 10%.

Based on the concept of the AD6002/6003 chip-set, a single chip DAB tuner was presented at IFA 99 by Bosch, sized about 2 cm × 3 cm (see Figure 8.9). This high-IF architecture even supports FM reception with the same receiver architecture.

It is obvious that the number of components of such a design is already lower than for a contemporary high-performance FM tuner. It can be expected that, once high volume is reached, a DAB tuner may be manufactured for a lower cost than a state-of-the-art FM tuner.

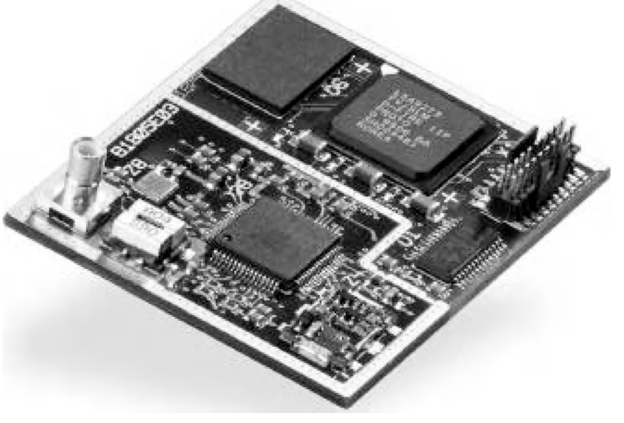

Figure 8.9 DAB module based on single chip tuner

8.2.3 Trends and Future Developments

The wireless technology boom of recent years has created some new technologies which have – according to 'Viterbi's law' – created some very low cost solutions for wireless systems. Technical progress is targeted at the reduction of system cost by reducing the number of components and the form factor.

With the introduction of FBAR-filters, a new low-cost competition is entering the filter market for wireless systems. FBAR-filters are a new option for preselection filters or IF filters with low cost and small form factor [Aigner, 2002].

Two chip implementations, based on an RF frontend chip and a digital baseband processor are available for almost every wireless system. Integrated VCOs have been successfully introduced in products and have eliminated external VCO modules. SAW filters may be either eliminated by zero-IF concepts or replaced by FBAR filters that are suitable for hybrid integration ('System in a Package'). Since passive SAW/FBAR filters provide the inherent advantage of not requiring an active power supply, they should persist in advanced receiver designs.

For Bluetooth, single chip implementations with both RF transceiver and digital baseband processor on the same chip already dominate the marketplace (CSR, Broadcom). This is possible since the minimum sensitivity requirement is only –70 dBm for Bluetooth. Nevertheless, single chip solutions for DAB with acceptable performance are announced.

For good DAB signal coverage, a DAB home receiver might also work reasonably well with a sensitivity of just –70 dBm, but since most people are listening to radio while driving to work, such simplified receiver designs will not provide sufficient performance for mobile reception.

The feasibility of pure CMOS front-ends has been demonstrated for most wireless applications, including Bluetooth, GSM, CDMA and WLAN (e.g. IEEE 802.11). Several 5 GHz transceivers for IEEE 802.11a, working at a frequency as high as 5.2 GHz have been reported [Su, 2002], [Cho, 2003], [Etz, 2003]. CMOS has become the mainstream technology for receiver front-ends up to 5 GHz, in spite of some severe deficiencies compared to SiGe implementations. The economy of scale of large scale CMOS processes is simply unbeatable. Nowadays, DAB front-ends used in mobile phones are also often based on CMOS. The advantage is the lower power consumption; the disadvantage is a worse linearity in comparison to the BICMOS technology.

The performance requirements for a good FM receiver are a lot higher than for a digital broadcasting receiver. An FM receiver requires a dynamic range of 120 dB while DAB only requires about 90 dB of dynamic range. According to the general trend of 'Viterbi's Law', a DAB receiver in volume production would already outperform a state-of-the-art FM receiver both in perceivable reception quality and in price and performance. 'Moore's law' predicts that the implementation of the complex digital algorithms will be possible at about no cost.

Similar developments can be expected for digital TV receivers, since the deployments of DVB-T, ISDB-T and ATSC are gaining momentum.

The ultimate challenge of receiver design is the implementation of a system that can cope with the 'balkanisation of standards' arising for mobile wireless services, both for the analogue front-end and for the digital baseband processor. For digital sound broadcasting, there are already six different systems in service worldwide (see section 1.6), creating a market opportunity for multi-standard receivers. A future receiver for the European market will have to support FM, DAB, DRM and eventually DVB-T in a single system, integrated with a Bluetooth or WLAN interface to other systems in the home or in the car. The system concept of 'software radio' has been proposed for solving the Gordian knot of integration of multiple systems, but progress in this area is limited, mostly by the limited progress in the area of development of high speed, high resolution ADCs.

8.3 Digital Baseband Processing

Digital baseband processing is the generic term for all signal processing steps starting directly after digitisation of the IF signal using an ADC until the source coded data become available after Viterbi decoding. In the case of DAB, baseband processing includes the following processing steps:

- generation of the complex baseband signal;
- OFDM demodulation, possibly combined with compensation for the frequency drift of the baseband signal;
- demodulation of the DQPSK modulated carriers;
- time and frequency deinterleaving;
- channel decoding using the Viterbi algorithm;
- synchronisation of time, frequency and phase.

8.3.1 Digital Front-end

RF front-ends provide two different types of interface for the baseband processing depending on the overall receiver architecture chosen (see Figure 8.10a, b).

I/Q interface: In this architecture the generation of in-phase (I) and quadrature (Q) components of the complex baseband signal is done in the analogue domain. This type of interface naturally occurs in zero-IF receiver concepts, which are believed to be a path for the high integration of RF front-ends for DAB (see section 8.2.2). A major disadvantage of this approach is the quality requirements for amplitude and phase balance over the required signal bandwidth of 1.536 MHz. In the context of the JESSI project, these problems led to the decision to focus on the IF concept.

IF interface: Here, the last IF of the RF front-end is fed into the ADC and frequency correction is provided via a complex multiplier fed by a numerical controlled oscillator (NCO). After shifting the signal towards zero frequency low-pass filters are employed to provide image and adjacent channel suppression. Finally, the signal is decimated towards the sampling rate of Fc = 2.048 MHz. Owing to the digital implementation of this architecture, phase and amplitude imbalance requirements can easily be solved. In the following section we focus on the IF interface.

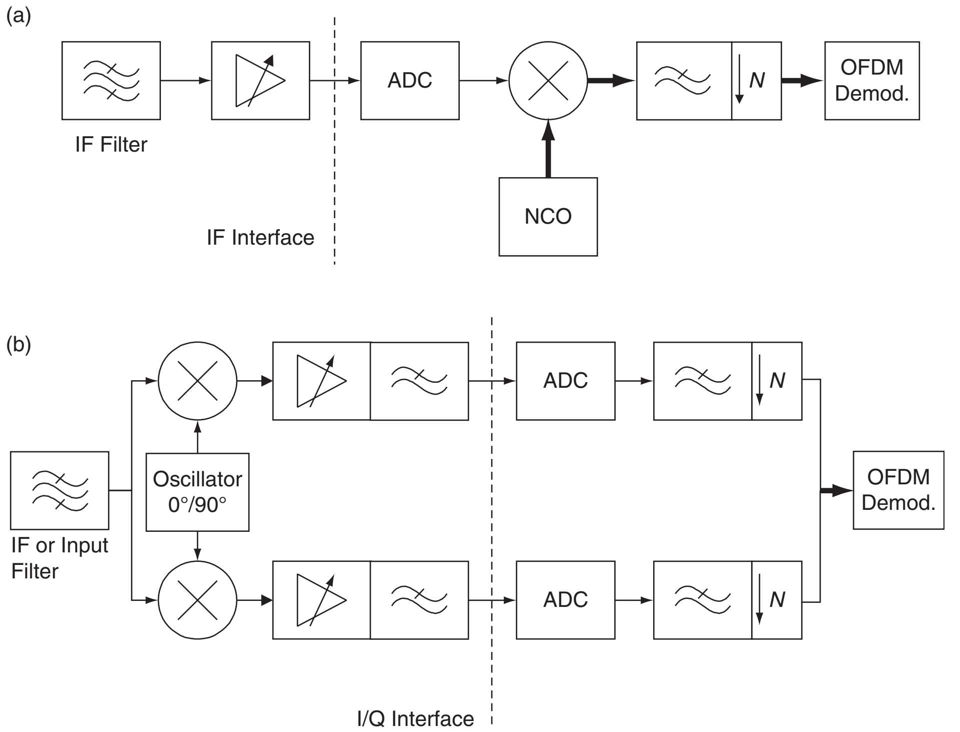

Figure 8.10 Receiver architectures and interfaces: a) intermediate frequency interface; b) in-phase (I)/quadrature (Q) interface

In order to minimise the hardware implementation cost, the sampling rate of the ADC is usually chosen as an integer multiple, N, of the sampling frequency of the complex baseband signal $F_c = 2.048$ MHz, that is

$$F_{\mathrm{ADC}} = NF_c. \tag{8.2}$$

The actual choice of the ADC sampling frequency is a trade-off between filtering in the analogue and the digital domains. The minimum possible ADC sampling frequency for an IF concept occurs for N = 2 with $F_{\mathrm{ADC}} = 4.096$ MHz, a choice which can be found in the early JESSI concepts. An advantage of the IF concept in Figure 8.10a is the fact that the IF can freely be chosen according to the needs of the special front-end architecture. Any IF can be supported as long as it is compatible with the ADC chosen regarding input bandwidth and sampling rate. However, more attractive solutions – regarding hardware implementation costs – can be found if certain relations between ADC sampling rate and IF are fulfilled. This will be shown in section 8.6.3.2.

8.3.2 OFDM Demodulation

The demodulation of the OFDM symbols is performed by applying FFTs to calculate the complex amplitudes of the carriers of the DAB spectrum. These amplitudes contain the

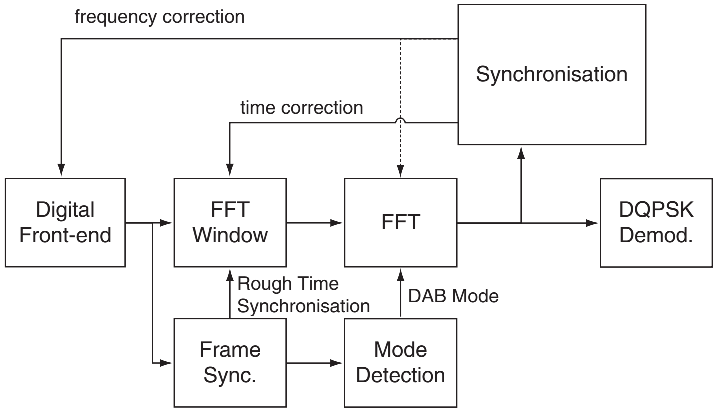

Figure 8.11 OFDM demodulation

Table 8.4 DAB transmission modes and FFT length

Mode	Carriers	FFT Length
I	1536	2048
II	384	512
III	192	256
IV	768	1024

information of the modulated data by means of a DQPSK modulation. A complete overview of OFDM demodulation including the synchronisation function is given in Figure 8.11. According to the various DAB transmission modes I–IV (see section 2.2.1), FFT lengths varying from 256, 512, 1024 and 2048 have to be implemented as indicated in Table 8.4. This can be realised very efficiently by the well-known Radix-2 FFT algorithm using a simple control of the FFT memory addressing.

To cope with the possible frequency drift of the baseband signal, an AFC (Automatic Frequency Control) is necessary. One possible realisation is indicated in Figure 8.10a, where the frequency shift is compensated by means of a complex mixer stage and an NCO. An even more attractive solution, which avoids the complex mixer stage, takes advantage of a modified FFT algorithm and is described in section 8.6.3.2.

8.3.3 DQPSK Demodulation

Differential demodulation of the carriers used is usually performed by applying a complex multiplication by the stored complex conjugated amplitude of the last OFDM

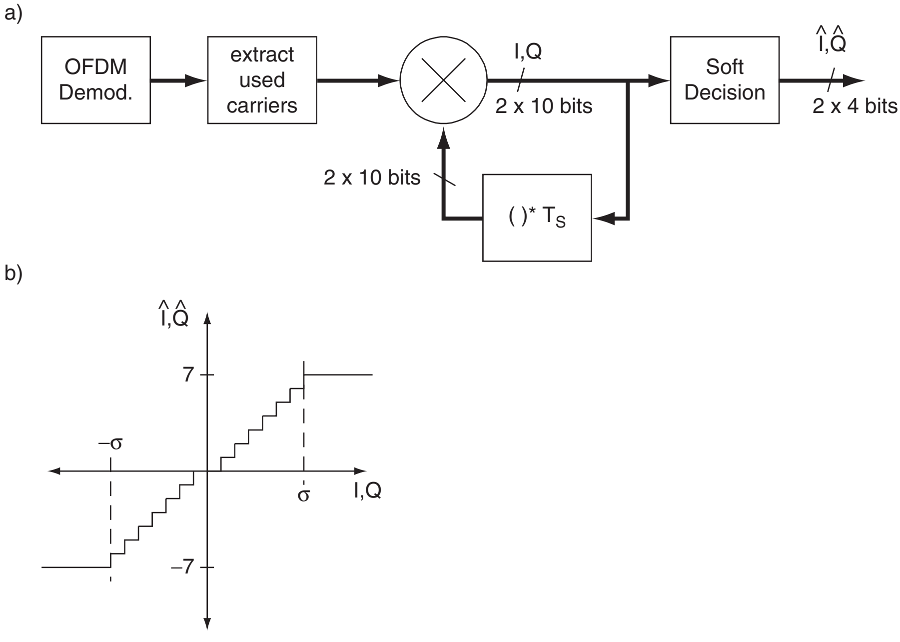

Figure 8.12 Differential demodulation: a) block diagram; b) possible soft decision mapping

symbol. Initialisation of this process is done using the phase reference symbol (TFPR, see section 2.2.2). Figure 8.12a gives an overview of the algorithmic processing steps and indicates typical word widths encountered in hardware implementations.

Figure 8.12b gives an example of a possible mapping of the demodulated amplitudes to soft decision values suitable as inputs to the Viterbi decoding algorithm, see section 8.3.5. The parameter s is used to adapt the characteristic curve to the actual signal level in the receiver chain.

8.3.4 Deinterleaving

In order to cope with transmission disturbances, two interleaving mechanisms are used (see sections 2.2.4, 2.2.5):

- Frequency interleaving is a rearrangement of the digital bit-stream over the carriers, eliminating the effects of selective fades. Frequency interleaving operates on one OFDM symbol only.
- Time interleaving is used to distribute long error bursts in order to increase the channel decoder's error-correcting capability.

The frequency deinterleaving can be implemented by addressing the output of the FFT according to the interleaver tables.

The time deinterleaving is a task that requires a substantial amount of memory. As presented in Chapter 2, the data of each sub-channel are spread over 16 CIFs, whereas each CIF represents the information of 24 ms. Thus the interleaving process requires a memory that has 16 times the capacity of the data to be decoded.

As an example, we examine an audio sub-channel with a typical bit rate of 192 kbps. One audio frame of 24 ms duration equals 576 bytes. Since the time deinter-leaving is located prior to the Viterbi decoder, each information bit is represented by its soft decision value, typically a 4-bit number. Thus, the memory required for deinterleaving this sub-channel works out at 36864 bytes.

The maximum amount of memory needed for time interleaving and assuming the storage of 4-bit soft decision output values of DQPSK demodulation works out at 442 kbytes or 3.54 Mbits. This amount can be halved by using appropriate in place usage of this memory leading to a necessary amount of 221 kbytes or 1.77 Mbits for a full-stream DAB decoder.

8.3.5 Viterbi Decoding

To combat errors due to channel distortions, DAB employs a powerful punctured convolutional code (RCPC) with constraint length 7 and mother code of rate ¼ for channel coding. This mother code is punctured (see section 2.2.3.1) to obtain a wide range of possible code rates, so as to adapt the importance of the information bits to the channel characteristics. For decoding these codes, the Viterbi algorithm is used [Proakis, 1995], which offers the best performance according to the maximum likelihood criteria.

The input to the Viterbi decoder can be hard-decided bits, that is '0' or '1', which is referred to as a hard decision. A better performance (2.6 dB improvement) is achieved if the uncertainty of the input is known to the Viterbi decoder, by using intermediate values. The optimum performance for this soft decision is reached when each input value is represented by a 16-bit number. However, the degradation is still negligible if the number of bits is reduced to 4 bits [Proakis, 2008].

The energy dispersal de-scrambling is another task that can easily be assigned to the Viterbi decoder module. The BER (Bit Error Ratio) on the channel can be estimated by re-encoding the decoded sequence or a sub-set of the sequence and comparing this sequence with the received bit-stream (see section 7.7.2). This information can be used as additional reliability information.

8.3.6 Synchronisation

Synchronisation of a DAB receiver is performed in several steps:

1. Coarse time or frame synchronization.
2. Coarse frequency synchronisation on carrier accuracy.
3. Fine frequency synchronisation on sub-carrier accuracy.
4. Fine time synchronisation.

Frame synchronisation. The Null-symbol of the DAB transmission frame provides a simple and robust way for the coarse time synchronisation, which is also called frame synchronisation. The underlying idea is to use a symbol with reduced signal level which can be detected by very simple means. In practice a short time power estimation is calculated which is then used as input to a matched filter. This filter is simply a rectangular window with a duration according to the Null-symbol length. Finally a threshold detector indicates the beginning of a DAB frame. It is also possible to calculate an AGC value for optimal scaling inside the following FFT signal path (FFT stages).

Coarse and fine frequency synchronisation. Coarse and fine frequency synchronisation can be performed using the TFPR symbol (see section 2.2.2) in the frequency domain. This step clearly requires a sufficiently exact coarse time synchronisation. Frequency offsets are calculated using the various CAZAC (Constant Amplitude Zero Autocorrelation) sequences inside the TFPR symbol. These sequences provide a tracking range of the AFC of about ± 32 carriers. This is a sufficiently large value to cope with cheap reference oscillators used in RF front-ends.

Fine time synchronisation. Fine time-synchronisation is performed by calculating the channel impulse response based on the actually received TFPR symbol and the specified TFPR symbol stored in the receiver.

All the described steps are subject to algorithmic improvements and contain various parameters that reflect the receiver manufacturer's experience in the field. Thus in all concepts which are on the market today synchronisation is mostly performed in software on a digital signal processor (DSP).

8.4 Audio Decoder

The introduction of DAB+ resulted in the enhancement of the audio decoder implementation. As presented in Chapter 3, two versions of MPEG audio coding schemes are defined for DAB now. These are the traditional MPEG-1 and MPEG-2 Layer II [IS 11172], [IS 13818] and the new MPEG 4 HE AAC v2 [IS 14496]. With the help of the FIC, the receiver gets knowledge about the currently used audio coding scheme. For DAB use, the first ones have been extended to provide further information for the detection of transmission errors in those parts of the bit-stream with the highest error sensitivity. This is useful for error concealment. The second one, the AAC, already includes all necessary error detection mechanisms. But an additional error protection based on Reed-Solomon coding and virtual interleaving is used to improve the reliability of audio decoding in the receiver. To support this mechanism a super frame which consists of several audio frames, the so-called access unit, is used.

Furthermore, the dynamic range tool which provides a mechanism to reduce the decoded audio signal at the receiver and is especially helpful in noisy environments like vehicles was added within DAB for Layer II decoding and is already part of the AAC standard.

For Layer II audio coding section 8.4.1 is relevant, audio coding according to DAB+ is described in section 8.4.2 and 8.4.3. For DMB additionally to AAC a modified version called BSAC is standardised which is described in Chapter 9.

8.4.1 Layer II Audio Decoder Architecture

The DAB audio decoder is based on an MPEG-1 and MPEG-2 Layer II decoder, but can additionally calculate and utilise error status information of the audio bit-stream like ISO-CRC and SCF-CRC (see Chapter 3) which is necessary for a powerful error concealment. Furthermore, the decoder should be able to decode dynamic range control (DRC) information. However, the DAB specific extensions to the MPEG audio standard are not normative and the receiver manufacturer can even decide not to exploit this information. The block schematic of a DAB audio decoder is depicted in Figure 8.13.

As presented in Chapter 3, an audio frame consists of header and side information data, sub-band samples, error detection information and additional data that are intimately related to the audio programme, like PAD and X-PAD.

Firstly, the header is decoded and information like bit rate and audio mode are extracted. Based on this information, the table is selected that is necessary to interpret the side information correctly. The side information contains the quantisation that was employed in the encoder and the scaling factor that was used for normalisation for each sub-band. In MPEG Layer II audio coding, the audio signal is transformed into the frequency domain and is decomposed into 32 equally spaced sub-bands. The inverse process is the synthesis filterbank. The reconstructed 32 sub-band samples in the frequency domain are transformed into 32 consecutive samples of one channel in the time domain, applying a 32-point IMDCT followed by a windowing process. Finally, the audio signal is transferred to a digital-to-analogue converter (DAC) using the I2S protocol (see section 8.5).

The synthesis filter is the most demanding task with respect to the computational effort. Consequently, there exist a number of algorithms that offer more efficient solutions in terms of multiply and add operations, but the drawback is that the addressing becomes more complex.

8.4.2 Reed-Solomon Decoding and Virtual Deinterleaving

Before starting the audio decoding process, the receiver has to decode the FIC to decide about the current used audio coding scheme. According to DAB+ Figure 8.14 shows the

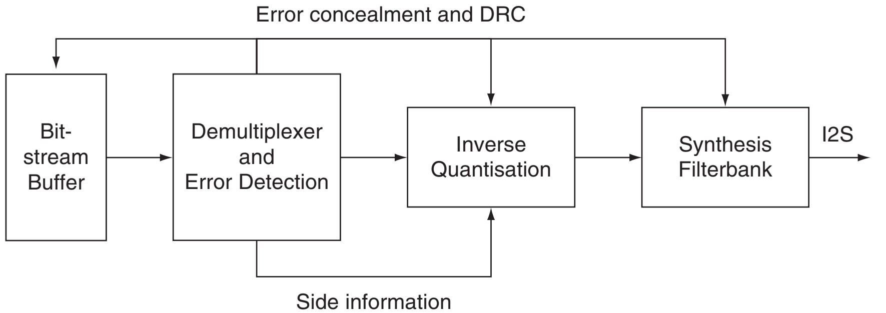

Figure 8.13 Audio decoder block schematic

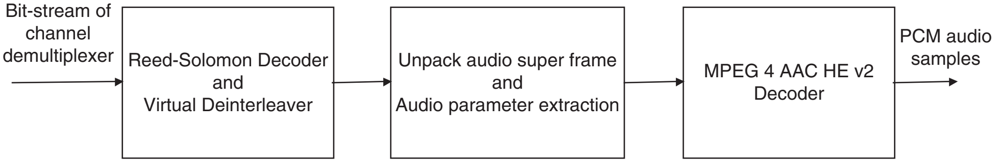

Figure 8.14 Audio decoding for DAB+

procedure for extracting the audio frames from the sub-channel bit-streams in the receiver. Details can be found in [TS 102563].

The bit-stream is provided from the main service channel demultiplexer. The following Reed-Solomon decoder and virtual deinterleaver allow a further reduction of the number of bit errors (see section 2.2.3.2). The receiver has to collect five successive DAB logical frames to start the deinterleaving process. It is called a virtual deinterleaver since the bytes are transmitted in the same order than they are generated in the transmitter. Only the RS parity bytes of the code are calculated from interleaved data. The mechanism in the receiver is to write the received bytes in an array of 110 columns before adding the parity bytes in additional 10 columns. The code words are presented as rows. This results in a Reed-Solomon RS(120, 110, $t = 5$) shortened code which is derived from the systematic RS(255, 245, $t = 5$) code. The additional redundancy allows the correction of up to five erroneous bytes of the 120 transmitted bytes. For a better understanding of the decoding of RS codes a short description is given. For further details the interested reader is referred to [Moon, 2005].

The algebraic decoding of RS codes can be described in the following process sequence:

1. Lengthening of the received code words: By adding zeros the original length of the systematic RS code should be achieved.
2. Calculation of the syndrome: Multiplication of the received vector with the parity check matrix.
3. Determination of the symbol error locations: With the Berlekamp-Massey algorithm or Euclid's algorithm an error locator polynomial can be calculated. With the Chien search algorithm the roots of this polynomial can be found which gives an indication of where the errors are.
4. Finding the error values: This involves solving equations simultaneously which is typically done with the Forney algorithm.

The complexity of the Reed-Solomon error correction should not be underestimated. But suitable hardware and software solutions for consumer products are available nowadays.

In the next step, the audio parameters from the audio super frame can be extracted before individual audio frames can be forwarded to the MPEG 4 audio decoder. The audio parameters are protected with an additional Fire code. Only when this code shows an error free bit sequence can the audio super frame be decoded. The audio parameters provide the configuration of the audio decoder as sampling rate or the use of MPEG surround.

8.4.3 AAC Audio Decoder Architecture

The DAB+ audio decoder is based on the MPEG-4 High Efficiency AAC v2 profile, level 2 [IS 14496] often abbreviated as HE AAC v2. In this section only a rough overview about this audio coding mechanism is given to understand the behaviour of the audio decoder in the receiver, details about audio coding can be found in Chapter 3. HE AAC v2 combines three MPEG technologies: Advanced Audio Coding (AAC), Spectral Band Replication (SBR) and Parametric Stereo (PS). The AAC is the core codec which provides transparent audio quality at a typical bit rate of 128 kbps [Meltzer, 2006]. For lower bit rates the enhancement tools SBR and PS support the achievement of a good audio quality. When using SBR, the original AAC audio codec considers only a small audio bandwidth. The signal components at higher frequencies with less psychoacoustic impact are harmonically repeated from the lower frequency range. So for higher frequencies only the spectral envelope of the original input signal has to be transmitted to allow the reconstruction of the signal at the receiver side. When using PS in addition only a mono downmix of the stereo signal is transmitted together with side information containing stereo image parameters. The HE AAC v2 decoder is depicted in Figure 8.15.

The bit-stream at the input of the decoder is split into the AAC decoder part and, if existent, into the side information for SBR and PS. For the AAC decoder two configurations exist. Firstly, without the use of SBR and PS the AAC decoder directly produces the time domain signal of the sample-rate f_s marked with a) in Fig. 8.15. Or second, the AAC decoder only outputs the low-band signal at a sample-rate of $fs/2$ indicated by b). The following 32-band Quadrature Mirror Filter (QMF) bank transforms the signal into the QMF domain to allow the reconstruction of the high-band in the SBR block. The low and the high band are added to get the representation. When using PS, the stereo signal is generated in the QMF domain with the help of the PS side information. In Fig. 8.15, this is marked with c) in contrary to d) where PS is omitted. The time domain output signal at sample rate f_s is generated in a last step by a 64-band QMF synthesis bank.

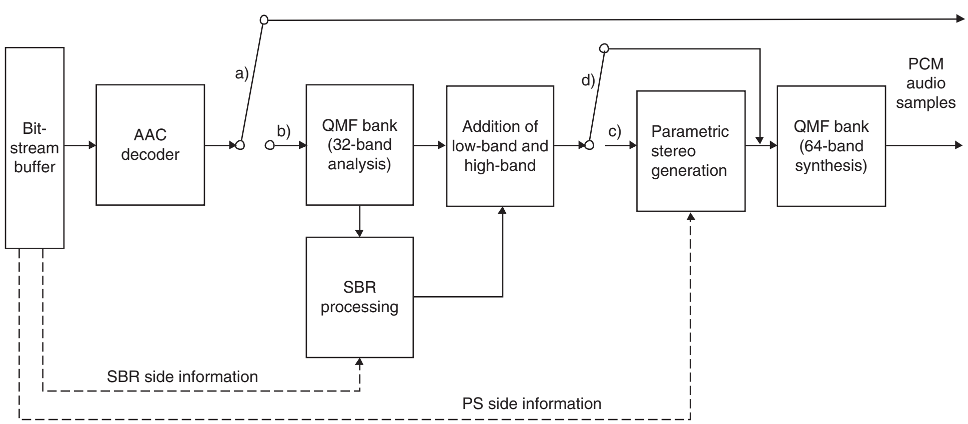

Figure 8.15 Block diagram of HE AAC v2 decoder

The complexity in the receiver for audio decoding is significantly increased in comparison to Layer II decoding. A rough estimation of the complexity can be found in [IS 14496]. For the processing power a number of 9 MOPS (million operations per second) is given, the memory usage corresponds to about 10 kwords. These numbers do not pose any greater challenge to state-of-the-art consumer technology. Additional complexity has to be considered for multichannel decoding according to MPEG Surround (MPS), standardised in MPEG-D, Part-1 [IS 23003]. The decoder shall support either level 2 or level 3 of the Baseline MPEG Surround profile. That means a 5.1 channel configuration for loudspeaker presentation can be achieved. The MPEG Surround decoder has to distinguish between two different modes, the normal mode works with a downmixed signal of the original multichannel signal and associated spatial parameters and the enhanced matrix mode enables a multi-channel upmix from a stereo signal where the spatial synthesis is derived from an analysis of the received signal only. Furthermore it has to be mentioned that for the receiver the two features MPS and PS are mutually exclusive. That means that to the parametric stereo block in Figure 8.15 an optional MPEG surround block can be added, but only one of them can be active for a received signal.

8.4.4 Normative Requirements

Conformance testing for DAB Audio [TS 101 757] defines the normative requirements for a DAB audio decoder which is based on the procedures defined by MPEG as Part 4 of [IS 11172] and [IS 13818] respectively. [TS 101 757] defines a test procedure and associated bitstreams that can be used to verify whether an audio decoder meets the requirements defined in [EN 300401]. The basic principle of the test is depicted in Figure 8.16.

For calculation of the conformance criteria, the definition of the difference of the two related output samples ($d(\mathrm{n})$) and their error energy (RMS) are given below:

$$d(\mathrm{n}) = x_{\mathrm{ref}}(\mathrm{n}) - x_{\mathrm{test}}(\mathrm{n}) \tag{8.3}$$

$$RMS = \sqrt{\frac{1}{\mathrm{N}} * \sum_{\mathrm{n}=0}^{\mathrm{N}-1} d(\mathrm{n})^2} \tag{8.4}$$

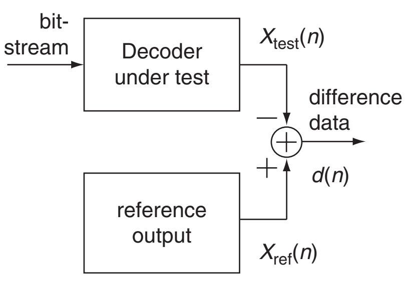

Figure 8.16 Audio conformance test procedure

The decoder under test decodes the test bit-stream. The output (x_{test}) is digitally recorded and compared on a sample-by-sample basis with the reference output (x_{ref}). Two certificates for a DAB audio decoder are defined:

1. 'full accuracy DAB audio decoder', which is more strict;
2. 'DAB audio decoder', which is more relaxed.

To be called a 'full accuracy DAB audio decoder', the average error energy as defined in equation (8.4) must be below $2^{-15}/\sqrt{12}$ and in addition the first 14 bits of each output sample must be bit exact. If the average error energy does not exceed $2^{-11}/\sqrt{12}$, the criteria for a 'limited accuracy DAB audio decoder' are fulfilled.

8.5 Interfaces

In principle, DAB receivers can be equipped with two types of interface: those that carry data to or from DAB receivers and those that carry control information.

Table 8.5 Overview of interfaces to DAB receivers

Interface	Area	Description
RDI (Receiver Data Interface)	Data and audio	This interface is defined in [EN 50255]. It is the standard interface of DAB receivers that can transmit the content of the complete DAB ensemble of up to 1.8 Mbit/s.
I2S (Inter IC Sound)	Audio	Audio interface for digital audio data in the time domain (PCM format). It is used typically as the interface to the audio DAC.
AES/EBU S-P/DIF	Audio	Audio interface for PCM coded data. AES/EBU is the professional format, whereas S-P/DIF is the consumer format. Physically, both protocols are based on [IEC 60958]. The difference is in the interpretation of some of the "side information bits". For further details we refer the reader to [IEC 60958] and [Watkinson, 2000].
IEC 61937	Audio	This interface allows compressed audio to be carried over an interface, that is physically identical to [ISO/IEC 60958]. This allows for example a multichannel audio programme received over DAB to be feed to an external multichannel decoder. For further details we refer the reader to [IEC 61937].
I2C (Inter IC)	Front-end control	This is a typical three wire interface to control the analogue front-end of the DAB receiver. The upper limit for the data rate is about 400 kbit/s.
SPI	Front-end/ data	An alternative to I2C allowing higher data rates. It can in addition be used to transfer any data (like data services) to any external device.
MOST	General	In-car bus system. One application is to link the DAB receiver to the front unit that controls the receiver.
IEEE 1394 (FireWire)	General	Alternative bus system to MOST that allows higher data rates. IEEE 1394 is even suitable for uncompressed video distribution.

Data interfaces. The RDI (Receiver Data Interface) [EN 50255] is a data interface which was specifically developed for DAB. It is suitable for connecting an external data decoder to a DAB receiver. It carries the full DAB stream of up to 1.8 Mbps and is thus useful for high data-rate applications, like video broadcast through the DAB system. Nowadays, this interface is less important, since a PC-card receiver transfers data through the PC bus system, like PCI, and for car radios there are other specific solutions, like Media Oriented Systems Transport (MOST).

Control interfaces. The I2C (Inter IC) interface belongs to the second category. It is used to exchange control information, such as the frequency, that the front-end should tune in. The bit rate of this I2C is up to 400 kbps. In this context, this interface is internal to the DAB receiver and connects the digital part or a micro-controller with the analogue front-end.

Control interfaces are also used to control a black-box-type DAB receiver (see section 8.7) from the car radio. A common DAB command set for receivers (DCSR) was standardised for this purpose [EN 50320]. It is intended for the use on different physical bus systems. An API to control a DAB receiver box through the car bus systems MOST (Media Oriented Systems Transport) is also defined in the MOST function catalogue.

Table 8.5 provides an overview of the most important interfaces of DAB components and receivers.

8.6 Integrated Circuits for DAB

Chip-sets are the building blocks for the various kinds of DAB receivers. From the very beginning of DAB standardisation in 1986 it was clear that the delivery of key components for receivers, that is highly integrated circuits – ICs, is substantial for the success of the new standard. Today the integration is at a level where a complete DAB broadcast receiver can be realised even in combination with other systems using two highly integrated circuits (RF chip and baseband chip) or even using only one chip. Nowadays, for baseband processing also a DSP (Digital Signal Processor) instead of an ASIC (Application Specific Integrated Circuit) is a convenient solution to reach more flexibility.

Today, many highly integrated solutions from different chip-set manufacturers are available. In the following, we will give an overview before we present two different concepts for DAB chip-sets. The first one is the result of a joint effort of the European semiconductor and receiver industry. The second one is an example for a more modern implementation in this sector and differs in many respects from the first concept.

8.6.1 Overview of DAB Chip-sets

With the increasing number of DAB chip-set manufacturers the variety of different receiving solutions has also risen. Three trends can be recognised. Firstly, multi-standard RF and baseband ICs covering several broadcast systems were developed.

Relevant systems are the two analogue ones AM and FM as well as the digital ones DRM and DVB-H/T. Additionally, Wi-Fi to enable internet radio can be found in some implementations. These solutions help to serve the world market with its different standards. The second trend is a dramatic reduction in power consumption. It can be explained by the need to integrate DAB/DMB into mobile phones to enable mobile TV. The reduction of power consumption can be achieved by the technical progress in the semi-conductor world. Also solutions where the RF-Tuner and the baseband processing are implemented in one package are available. The third trend is an improved reception performance. It is driven by the automotive domain with its more difficult reception environment. Integrated multistandard RF-Tuners and low power chip-sets behave worse in comparison to DAB dedicated solutions with higher power consumption. Since most car manufacturers concentrate on high performance systems, the traditional partitioning of RF and baseband has its right to exist. To allow the support of several broadcast systems dedicated ICs are used.

Many silicon vendors from different countries entered the market. Some provide only RF or baseband ICs, but most offer both. Market relevant solutions are provided by the companies Frontier Silicon, Analog Device, Texas Instruments, Atmel, NXP, Infineon, Future Waves, Siano, Media Phy and Maxscend Technologies. In addition, there are many companies using these ICs to build small modules which can be easily used by the receiver manufacturer.

8.6.2 History of DAB Chip-sets – the JESSI Programme

The European JESSI (Joint European Submicron Silicon Initiative) programme was a suitable umbrella for the necessary pre-competitive activities of the European semiconductor and equipment industry. Inside JESSI the AE-14 programme was one of the application-specific programmes and was started in 1991 at a time when the final DAB specification, now officially known as [EN 300401], was not available. The system specification and verification, however, had already been done during the Eureka 147 project. Most of the partners active in the Eureka 147 project also played an important role during the JESSI AE-14 project. The overlap of the two groups ensured that AE-14 had early access to the DAB system specification and an extremely thorough understanding of the DAB system.

The JESSI AE-89 DDB (Digital Data Broadcasting) project was launched in 1995 in order to support the introduction of data services into DAB by delivering an IC at a very early stage. In addition to this, the combination of DAB and FM and the further integration of DAB have been a focus of this project.

The following paragraphs give an overview of the final silicon delivered by AE-89 which was the basis of numerous receiver designs in the past few years.

8.6.2.1 Analogue IC – JESSI AE-89

As mentioned above, most DAB tuners are based on the ‘JESSI concept’ as shown in Figure 8.7. In the first implementation, four ICs were required. The signal strips for

L-band and Band III were each integrated in a special IC, and both the L-band downconverter and the Band III signal strip required a separate PLL IC that had to be compliant with the Eureka 147 system requirements. In general, an external low noise front-end transistor was required for both bands. The formerly popular TEMIC chip-set presented in Table 8.6 relied on this concept.

8.6.2.2 Digital IC – JESSI AE-89

The final generation of AE-89 DAB ICs achieved the following technical features:

- decoding capabilities for the full DAB bit-stream (up to 1.7 Mbps);
- digital I/Q generation and filter for adjacent channel selectivity;
- digital AFC;
- combined analogue and digital AGC;
- RDI capable of transporting up to 1.7 Mbps.

For realisation of the DAB channel decoder two architectures have been produced by TEMIC and Philips. These architectures are the result of trading flexibility against level of integration. Architecture I, developed by Philips, is shown in Figure 8.17 and offers a high level of integration.

Table 8.6 JESSI analogue IC chip-set, by TEMIC

U2730 B	L-band downconverter
U2750 B	Band-III- tuner

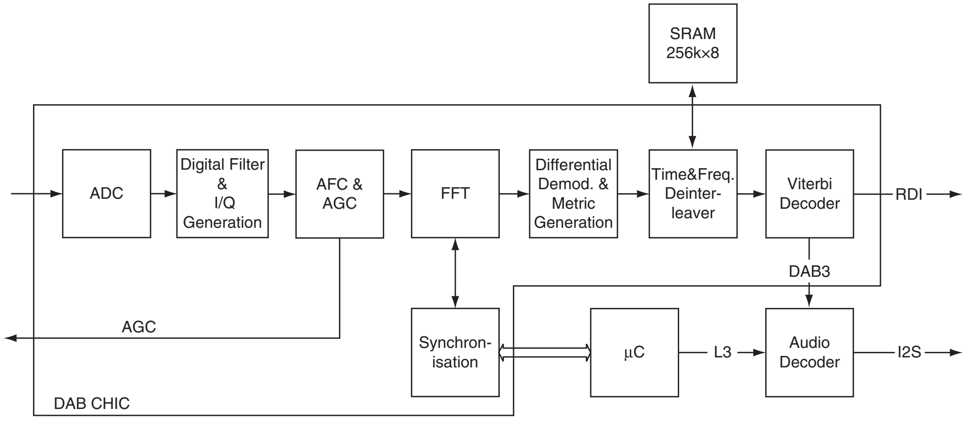

Figure 8.17 JESSI AE-89 DAB Channel Decoder Architecture I – Philips

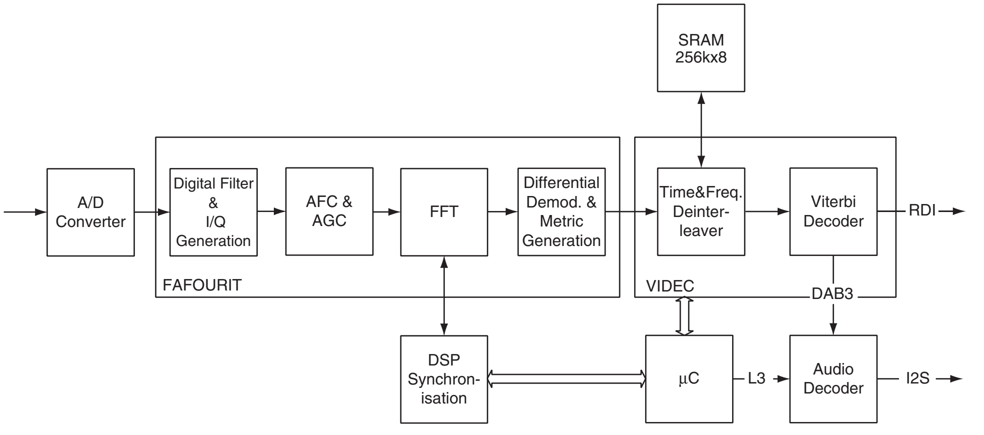

Figure 8.18 JESSI AE-89 DAB Channel Decoder Architecture II – TEMIC

The synchronisation functions are completely integrated and realised mainly in hardware. Closing of the synchronisation loops, however, is under the control of an external microcontroller (µC). As external components only a low-cost µC and 2-Mbit SRAM are needed.

Architecture II, developed by TEMIC and depicted in Figure 8.18, offers a lower level of integration, but has the advantage of offering more flexibility for realising different synchronisation strategies, due to the fact that the synchronisation is done in software on an external DSP.

These chip-sets have been the basis for numerous receiver designs and have been available as prototypes since the end of 1996.

8.6.3 D-FIRE Chip-set (Bosch)

The following paragraphs give a detailed description of the Bosch D-FIRE chip-set [Bolle, 1998], [Clawin, 1998], [Clawin, 2000], which differs in many aspects from the chip-sets which are based on the JESSI concept (see sections 8.6.2.1 and 8.6.2.2). This concept had some properties which hindered a low-cost, low-power and small-size DAB receiver. These properties are:

- The analogue VCXO required for the synchronisation is a cost-intensive component, which furthermore requires a DAC. In addition, it is well known that the VCXO control signal itself is vulnerable to interference.
- The use of tracking filters for interferer and image suppression in Band III is a major cost-producing factor: bulky, high-Q, tuneable inductors have to be used in connection with varactor diodes, which have to be driven by a control voltage generated by DACs.
- Manual or automated tuning has to be employed during manufacture of the DAB receiver.

- Supply voltages in excess of 5 V have to be used.
- A large number of VCOs are required, for example four VCOs for a receiver covering Band III and L-band.
- The L-band is sensitive to interferers at frequencies up to 300 MHz, which are ubiquitous from digital equipment (on-board DSP, PCs, TV stations).

Derived from this the following major goals for the development of this DAB receiver engine (D-FIRE: DAB–Fully Integrated Receiver Engine) chip-set were identified as:

- Use of supply voltages <5 V for the analogue part and 3.3 V for the digital part.
- Design of an adjustment-free receiver.
- Full digital synchronisation (time, frequency and phase).
- Minimisation of the total number of VCOs.
- Low power consumption in comparison to state-of-the-art receiver concepts.

8.6.3.1 Front-End Chip-Set

The commonly used receiver for DAB signals employs a Band III (174–239 MHz) heterodyne architecture, preceded by a 1.5 GHz to 210 MHz block converter for L-band (1452–1492 MHz) reception. This architecture requires tuneable pre-selection filters to meet the stringent selectivity and dynamic range requirements [EN 50248].

The alternative architecture used in the D-FIRE concept exploits a high level of integration by using two separate receiver stages for the two bands, with one PLL operating both mixers to the common first IF, on a single RF chip (Figure 8.19). The

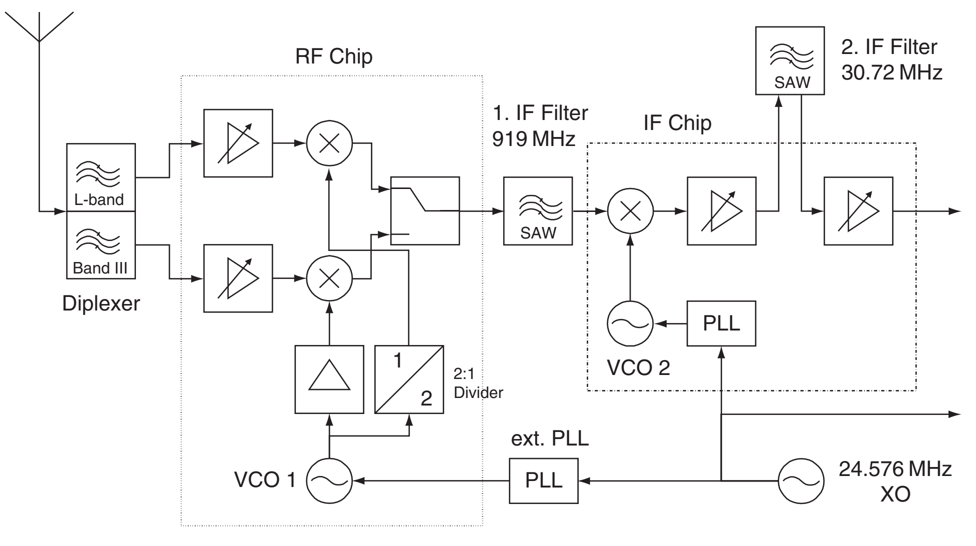

Figure 8.19 Front-end architecture of the D-FIRE chip-set

common IF is chosen between the two receiving bands (Band III and L-band). This choice allows the elimination of all tracking filters in Band III due to the underlying upconversion principle.

This separation allows independent optimisation of receiver designs for noise and intermodulation (IM) performance. A second PLL then performs the downconversion to the second IF of 30.72 MHz which is finally sub-sampled by the ADC using a sampling frequency of 24.576 MHz.

A next development with comparable architecture can be found in the newest receiver generation. It uses an integrated filter for the first IF to further reduce the number of components. In addition, the ADC and the digital conversion block were included on the same piece of silicon to enable the easy combination with digital ICs without ADC. So this new front-end IC became a milestone, it is the first one with integrated ADC for DAB.

8.6.3.2 Digital IC (D-FIRE)

The digital IC (D-FIRE) has still, despite its long availability, a modern architecture. The main difference to its near successor IC is the missing support of DAB+.
The architecture consists of two main parts (see block diagram in Figure 8.20):

- The signal path, which provides all the DAB functions for signal processing and decoding.
- A control block consisting of an OAK DSP core, a MIPS RISC core, 42 kbytes fast internal RAM for the processors and a cross-connector which realises flexible links between all components.

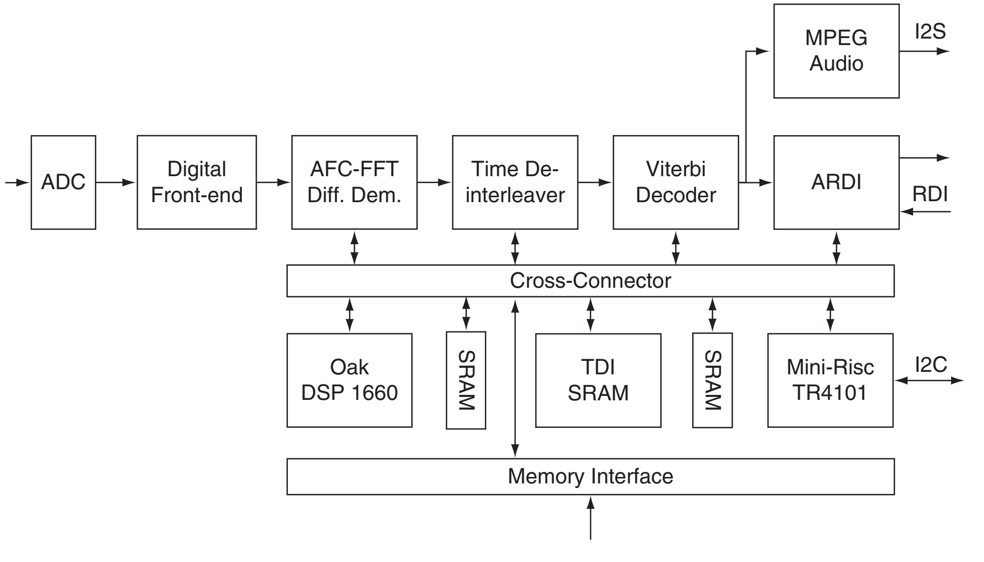

Figure 8.20 Architecture of the digital part (D-FIRE) of the DAB receiver system

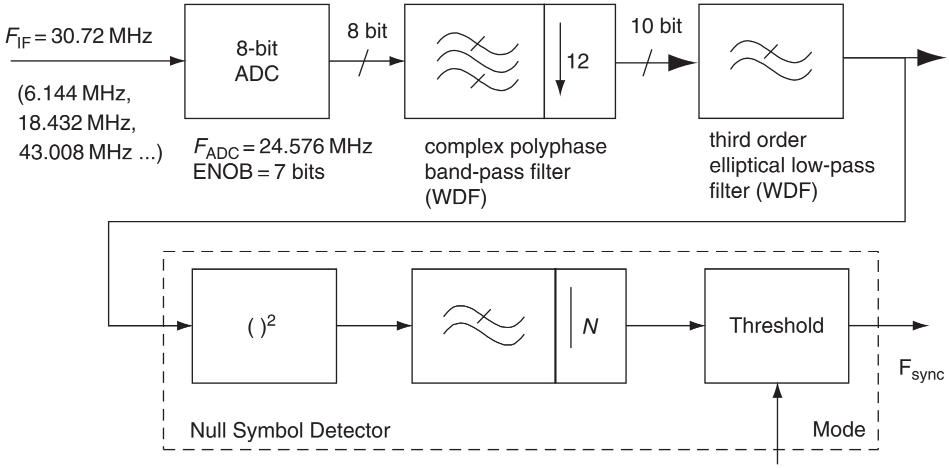

Figure 8.21 D-FIRE digital front-end

The digital front-end. In contrast to the generic front-end architecture of Figure 8.10a, the D-FIRE digital front-end employs a close relation between ADC sampling rate and IF in order to drastically reduce the hardware implementation effort. The basic idea is to use a complex recursive band-pass filter and subsequent decimation to generate the complex baseband signal and to provide simultaneously adjacent channel suppression. Very little hardware implementation effort is necessary if the following relation holds:

$$\mathrm{F_{IF}} = \frac{F_{\mathrm{ADC}}}{4}.(1 + 2m) \text{ for } m = 1, 2, \ldots \tag{8.5}$$

For D-FIRE the choice $m = 2$ leads to an F_{IF} of 30.72 MHz which is sub-sampled by the ADC working at a sampling rate of $F_{\mathrm{ADC}} = 24.576$ MHz, cf. Figure 8.21. Besides the chosen IF the front-end is of course able to cope with IFs of 6.144 MHz, 18.432 MHz and 43.008 MHz respectively. Care, however, has to be taken in the RF front-end to provide sufficient filtering at these unintentional IFs.

Because of the differential modulation scheme chosen in DAB it is possible to employ recursive filters (Infinite Impulse Response – IIR) in order to meet the demanding adjacent channel suppression requirements imposed by [EN 50248]. In order to cope with the stability requirements encountered in recursive filtering the well-known wave digital filter (WDF) concept [Fettweis, 1986] has been employed. In particular, for the first filter a complex polyphase filter has been chosen, which can be very efficiently combined with the subsequent decimation by a factor of 12.

After decimation, an additional low-pass filter is necessary to suppress the remaining interferences. For this filter a third-order elliptical low-pass WDF has been designed. Figure 8.22 shows the effective signal transfer function of the digital front-end. The design goal of this filter architecture has been to provide a minimum attenuation to possible adjacent DAB channels of at least 48 dB. It can easily be observed in Figure 8.22 that this goal has been achieved.

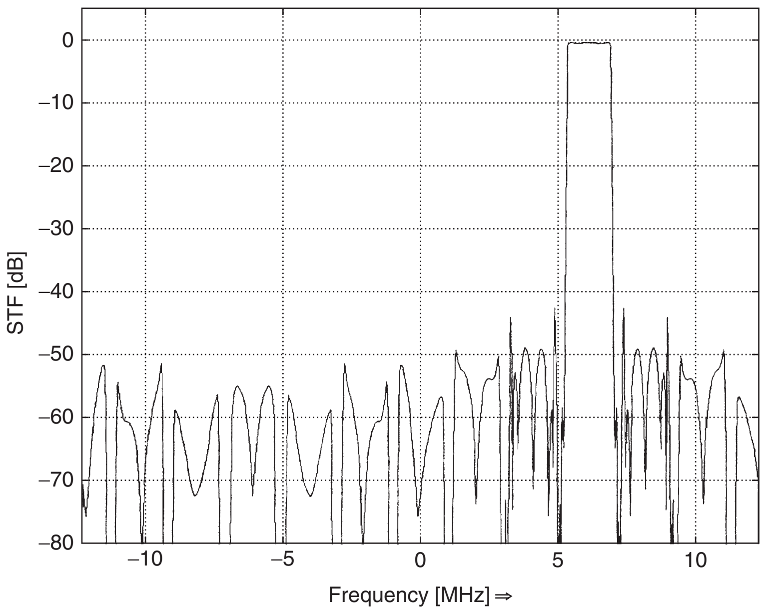

Figure 8.22 Effective transfer function of the D-FIRE digital front-end

OFDM demodulation. The demodulation of the OFDM symbols is performed by applying FFTs to calculate the complex carriers of the DAB spectrum. These carriers contain the information of the modulated data. A Radix-2 FFT with decimation in time is implemented so that the specified DAB transmission modes can be realised by a simple control of the FFT addressing. To cope with the frequency drift of the baseband signal, an AFC (Automatic Frequency Control) is necessary for which a new approach has been chosen. Let $x(\mathrm{k})$ be the complex output signal of the digital front-end, N the FFT length and r be the normalised frequency deviation, that is the measured frequency deviation normalised to the sub-carrier distance. In this case the following frequency correction has to provide the corrected signal:

$$y(\mathrm{k}) = x(\mathrm{k})e^{-j2\pi k\rho/N} \tag{8.6}$$

The Fourier transform Y of y is given by

$$Y(1) = \sum_{k=0}^{N-1} y(\mathrm{k})e^{-j2\pi kl/N} = \sum_{k=0}^{N-1} x(\mathrm{k})e^{-j2\pi k(l+\rho)/N} \tag{8.7}$$

It is important to note that $Y(l)$ is now calculated by a modified discrete Fourier transform, where the running index l is replaced by $\rho + l$. The interesting point is that now fast algorithms can be derived for this modification which leads to a hardware architecture which is a slight modification of the well-known Radix 2 decimation in time FFT architecture [Bolle, 1997].

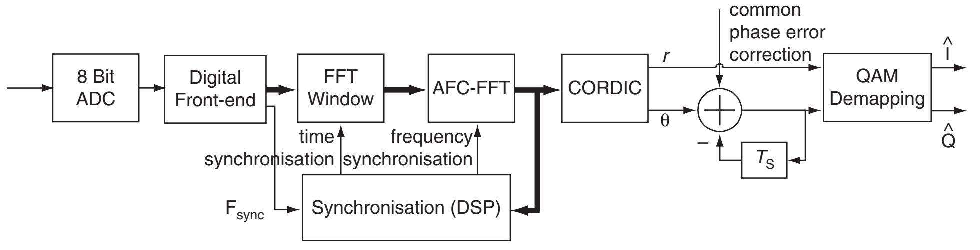

Figure 8.23 D-FIRE OFDM demodulation

The baseband energy, calculated by the Null-symbol detector (Figure 8.22), is used for optimal scaling of the fixed-point FFT implementation and the following signal path.

Demodulation (Figure 8.23). In contrast to the approach taken by JESSI (see section 8.6.2.2) the DQPSK demodulation is performed by calculating the polar representation (r, θ) of the carriers using the CORDIC algorithm, which can be implemented very effectively in hardware compared to the costly implementation of the complex multiplier which is needed in the traditional approach. In doing so, the differential phases of two DAB symbols can easily be calculated by a subtraction of the carrier phases. In addition the phase representation allows additional functionalities such as the removal of a constant phase error which is introduced by the phase noise contribution of the RF front-end oscillators. To determine the metrics for the following soft-decision channel Viterbi decoder, the actual amplitude and the difference between two following phases is used.

Deinterleaving. The time deinterleaver is implemented in combination with a subchannel demultiplexer. Only DAB capacity units with a size of 64 bits, belonging to selected subchannels, are deinterleaved and transferred to the Viterbi decoder. This procedure permits a multiplex reconfiguration to be followed for all selected sub-channels in the ensemble simultaneously without any interruption or disturbance.

Viterbi decoding. To combat errors due to channel distortions, DAB employs a convolutional code with constraint length 7 for channel coding. It can be punctured to obtain a wide range of possible code rates so as to adapt the importance of the information bits to the channel characteristics. Decoding is done using the Viterbi algorithm (traceback version). The Viterbi decoder is able to decode the full data rate of 1.8 Mbps which is the maximum data rate specified by ETSI. All specified code rates between 8/9 and 1/4 are supported, that is all equal error protection (EEP) and unequal error protection (UEP) profiles can be decoded. The decoder operates with 4-bit soft-decision metrics and closely approaches the theoretical coding gain. Furthermore, the Viterbi module includes the calculation of the CRC for the FIC data, an energy dispersal descrambler and a re-encoding unit for estimating the BER on the channel.

Audio decoder. The architecture of the implemented audio decoder is depicted in Figure 8.24. Three major blocks can be identified: demultiplexing, reconstruction of (frequency domain) sub-band samples and synthesis filtering. The latter blocks share an MAC (Multiply Accumulate) unit that allows one MAC operation per cycle.

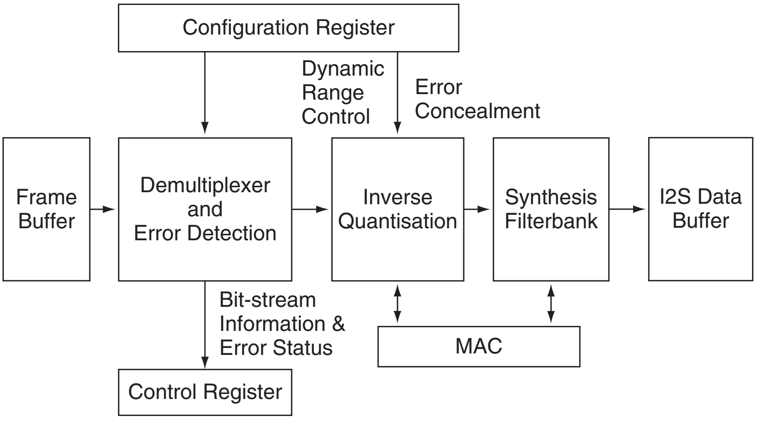

Figure 8.24 Audio decoder block schematic

A RAM is located before the decoder that stores the MPEG coded audio data. This buffer handles one part of the error concealment strategy (see section 3.7.2): in the case when the audio frame is not decodable owing to transmission errors the audio decoder requests another audio frame so the previously decoded frame is repeated. The communication between buffer and audio decoder is realised by an intelligent and flexible interface. With this buffer concept it is possible to decode ancillary data information carried in the DAB specific data field in the audio frame. Flexible concealment is guaranteed since the sub-band reconstruction process can be influenced on a sub-band basis, that is the audio signal can be shaped in the frequency domain prior to transformation into the time domain. Using this mechanism, the audio decoder can be configured in such a way that very annoying 'birdies' caused by an error-prone environment are avoided while the reproduction of the audio signal is maintained.

The audio decoder is able to use directly the information for the reduction of dynamic range which is part of the coded audio data. These data are applied during the decoding process in the frequency domain so the effect of the DRC process is smoothed by the filterbank. The synthesis filter is the most demanding task with respect to computational effort. The reconstructed sub-band samples in the frequency domain are transformed into the time domain by applying a 32-point IMDCT (Inverse Modified Discrete Cosine Transformation) followed by a windowing process.

8.7 Receiver Overview

DAB receivers have been available for more than one decade. In the last years, the variety of different receivers increased in order to serve the needs of the customer. Also the prices dropped so that DAB reached the status of a household product used at home, in the car and on the way to work. The miniaturisation and lower power consumption was an important step to integrate DAB into a variety of technical products.

8.7.1 History of Receiver Development

The first commercially available receivers were presented at the Consumer Electronics trade fair Internationale Funkausstellung (IFA) in Berlin in 1995. These so-called fourth-generation receivers, consisted of a car radio that provides a tuner for the traditional analogue broadcast systems (FM and AM) and the control interface to a black box, which facilitates the complete DAB specific processing. An antenna capable of coping with both DAB frequency bands completed these receivers. They were the first DAB receivers to decode an arbitrary multiplex configuration and some were even able to follow a multiplex reconfiguration. These first car receivers, manufactured by Bosch/Blaupunkt, Grundig and later by Pioneer, were used for DAB pilot projects in Germany and other European countries.

Two years later, at the IFA in 1997 about 10 manufacturers presented DAB car radios and even the first tuners fitting a 1-DIN slot have been displayed.

At the IFA in 1999 and on other occasions, the first HiFi tuners and PC-card-based receivers were presented.

8.7.2 Receiver Categories

Today, DAB receivers for a wide range of markets are available including home receivers, car radios, portable receivers, PC receivers and professional devices. While some manufacturers have developed DAB-only products, others have developed combined DAB/FM/AM/CD/MP3 units. Also combinations with other digital broadcast systems as DVB-H/T and DRM are available. An up-to-date overview of DAB products can be found on several Internet web-sites, for example [www.worlddmb.org], [www.digitalradio.de], [www.digitalradionw.com] and [www.dab-digitalradio.ch].

Home receivers. The 'at home products' provide the biggest plurality of different receiver types and they are the most successful category nowadays. Home radios on the market include hi-fi tuners, kitchen radios, clock radios, portable stereos ('boomboxes') and midi-systems. Many of these products look like traditional radios, but also provide new features such as pause/rewind to stand out from the crowd of competitor products.

Car radios. Two main categories for car radios prevail on the market. Firstly, complete all-in-one digital radios combined with AM/FM. These are also called 1-DIN tuners because the single housing fits into the standard mounting frame. The second category consists of car radios with an extra black box. The black box can be stored in the boot, under the seat or behind the dashboard. Both concepts are of interest for car manufacturers that are offering standard fittings for cars as well as products for retailers. In addition, HMI including DAB receivers which are mounted on the dashboard and can be connected to the car radio entered the market. These devices are also available with bigger displays to combine the DAB functionality with navigation or TV-like services. An example is shown in Fig. 8.25.

Portable receivers. Real hand-held receivers and accessories are now a reality. These are based on smaller, low-powered chips to enable the creation of truly portable DAB radios. The plurality of products is increasing fast. Some are very small and optimised for music listening on the way. Others are combined with bigger displays to allow presentation of

Figure 8.25 Newest generation DAB Car receiver VW RCD 510 DAB (Bosch)

data services and videos. Other features such as the different audio compression algorithms MP3, AAC, WMA and broadcast systems FM, AM, DRM or picture and video viewing with JPEG and MPEG are often included. A very successful combination is the mobile phone with DMB/DAB functionality. A detailed description for this can be found in Chapter 9.

PC products. DAB receivers plugged to a computer allow listening to radio and having access to data services while working at the computer. PC products rely on USB as interface between receiver and computer. The flexibility of this category is huge since software updates for new data services are simple to obtain. In addition, first manufacturers produced DAB-enabled notebooks with the advantage of a very easy use.

Reference receivers. Reference receivers are an indispensable necessity for network monitoring and coverage evaluation (see section 7.7). Information such as signal strength, BER or channel impulse response is displayed by such receivers.

8.7.3 Receiver Market

Several market research institutions cope with the prognosis of the receiver market. The trend is unambiguous, the sales numbers are increasing distinctly. This tendency can also be seen in the figures from the last five years shown in Figure 8.26. Apart from the four million DMB receivers that have been sold in Korea between the commercial launch of T-DMB in November 2005 and mid 2007, more than five million DAB radios have been sold in Europe in the last years. Most of these DAB radios can be found in UK households, but other countries started to catch up.

Beside the improved programme choice, the variety of products and lower prices are the reasons for the increasing market. DAB/DMB receivers are available in a wide range of prices, from a kitchen radio which costs less than 50 Euros to a hifi set for hundreds of

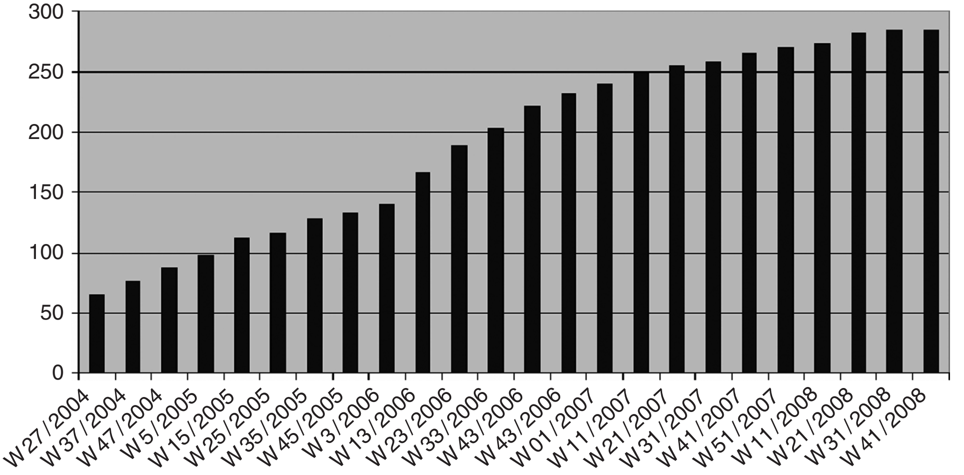

Figure 8.26 DAB product manufacturers worldwide. Reproduced by permission of Swiss Satellite Radio

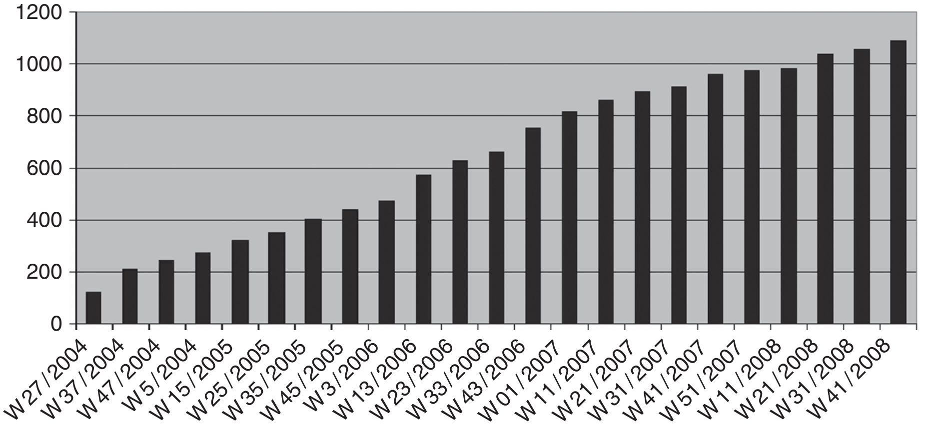

Figure 8.27 DAB receiver products worldwide. Reproduced by permission of Swiss Satellite Radio

Euros. The choice of different models also became impressive. From the source [www.dab-digitalradio.ch] one can extract that in 2008 more than 250 manufacturers offered almost 900 different receiver products worldwide as depicted in Figure 8.27.

8.8 Receiver Features

8.8.1 General Aspects

The possibilities of data provision with digital broadcast systems are almost infinite. As a consequence, the number of different types of data services increased over the years. But this also showed that customers do not accept all services, especially when a user friendly operation is not guaranteed.

The different receiver categories concentrate on different services and features. The benefit of the features can be derived from the use of the receivers. In the car for example, traffic information is of particular interest. Traffic information is necessary to ensure the safety of the car driver. So control instruments of the receiver which reduce the driver's attention to the traffic (for instance when the driver has to look at the receiver while setting it) must be minimised and other control instruments such as voice recognition must become more important.

Often portable receivers only have very small displays, so that the possibilities to present data services also are limited. In contrast, PC-based receivers have larger displays which much enhance possibilities. Receiver manufacturers have to decide carefully about the features their receivers should support considering the HMI (Human Machine Interface).

To obtain a better overview, a classification into three feature categories is appropriate. The first category needs a signalling in the DAB data stream to allow the use in the receiver. Examples of this are programme type, announcements and service following as described below. Often the FIC is used to transport the signalling bits. The features of the second category are independent of the DAB standard. But the implementation became much easier with the use of digital signal processing in the receiver which in turn was forced by the use of a digital transmission system. A typical example is the pause/rewind feature. The third feature category consists of the whole bunch of different data services. The utilisation is only possible if the broadcasters decide to spend some data capacity for these services. Examples are TOPNews, Journaline and electronic programme guide (EPG).

8.8.2 Examples of Receiver Features

8.8.2.1 Programme Type (PTy)

The assignment of one or more programme types (PTys) by the service provider allows selection of a programme, matching the interests of the user (see also section 2.5.5.2). PTy is a feature that is known from RDS, but for DAB use the capabilities have been largely extended. DAB offers the possibility to assign more than one PTy to a programme. Another novelty is dynamic PTy codes that can be used to describe, for example, the song that is currently on-air, whereas the static PTy reflects the flavour of the service itself. Therefore, the receiver HMI has in principle to offer three modes:

Service search. Search for services with a specific flavour. That is, the static PTy of the service matches the user's choice.

Programme search. Search for services with a specific programme. That is, the dynamic PTy of the service matches the user's choice.

Background scan (watch mode). In principle, one of the above, when there is no service available that offers the programme, according to the user's choice. The programmes are watched and if a programme according to the scan-list is turned on, the receiver automatically switches to that programme.

8.8.2.2 Announcements

Announcements (see also section 2.5.6) in DAB are similar to traffic announcements (TAs) in RDS, although offering a wider range of categories such as news, weather information, events, etc. In total, 32 different categories have been defined. Announcements are an interesting feature for all types of receivers. Traffic announcements are a must for car receivers which furthermore are simple to use with only one button.

8.8.2.3 Alternative Frequencies and Service Following

DAB provides frequency information about other DAB ensembles as well as FM or AM broadcast (see section 2.5.8). This serves as a tuning aid to the receiver control and is similar to the alternative frequency (AF) feature provided by RDS receivers. It offers the possibility to follow a service when the reception conditions of the actual frequency degrade too much. When the receiver leaves the coverage area of the DAB signal, it can automatically switch to the same programme being broadcast on an FM or AM frequency. It can also be used to switch (back) to DAB as soon as the receiver enters the coverage area. This feature is of particular interest in the transition phase from analogue to digital broadcasting for car receivers. The simplicity for the user is given because the user does not have to adjust the HMI.

8.8.2.4 Pause/Rewind and Record Function

With the Pause/Rewind feature the user of the receiver can pause and rewind the information received when he has missed something. With the single touch of a button the reception of the favourite station can be paused. Later on, the user can continue listening from the point at which he left the programme. Depending on the internal memory capacity and the data rate of the service some ten minutes can be recorded. When using an additional flash card memory the recording time can be increased considerably.

8.8.2.5 Audio-Based Information Service TOPNews

Screen-based data services with graphics are an attractive way to offer a wide range of information to the user. However, the information provided by this type of service cannot be used by the driver while driving. That's why an audio-based information service called TOPNews was developed and standardised [TS 101498-3].

TOPNews can be regarded as a kind of 'audio text' and is similar to the well-known TV video-text. A bouquet of different categories of information like sport, weather, traffic or news is provided to the user. Each category carries a dozen or so latest news clips. These clips are carried either in the programme associated data field of an audio service or in a dedicated data service and are stored in the receiver. A further option is that the user preselects his favourite categories (user profile) in order to reserve the whole memory only to those categories he or she is most interested in. In combination with a timer, the user will always get the latest news after entering the car.

The user can browse with simple navigation means through all the information. One concept used in the field test of TOPNews is similar to the operation of a CD changer. The next or previous news flash within a category is selected in the same way as the next or previous track on a CD. In the same way, the next or previous category is selected like the next or previous CD on a CD changer. Due to this concept this service can even be used by the driver while driving.

8.8.2.6 Electronic Programme Guide (EPG)

The EPG is designed to offer the customer an easy way to inform himself about current and future programmes (see section 5.5). Access to additional information for each programme is possible. For simple use the receiver needs a display with several lines to provide a good presentation. The EPG can be easily combined with the record function described above. So the user never misses his favourite programme. To achieve an accurate start and stop of the recording, the receiver has to evaluate the signaling linked with the EPG.

8.8.3 Future Trends

The last sections showed some impressive examples of receiver features and already existing data services. But combination of fast signal processing in receivers with high data rate delivery of DAB will further increase receiver functions for customers. For this the larger displays more often available nowadays are the basis for a good presentation. Other new functions will not depend on displays. The example in the next section will work for receivers with and without larger display.

8.8.3.1 The mobile.Info Project

The full name of this trendsetting project is *Mobile Platform for Efficient Traffic Information Services* [www.mobile-info.org]. The reason to launch the project was the steadily increasing traffic on Europe's roads which results in more traffic jams. DAB can be one important part to defuse this situation. It offers suitable geographical coverage and sufficient capacity for the transmission of traffic services. Therefore the basic services are from the TPEG Automotive Profile (TAP), a family of TPEG (see section 4.6.2) application protocols dedicated to automotive information and adopted for the use in navigation systems. It allows comprehensive traffic information which in turn is the basis for dynamic routeing of cars. The project covers the whole distribution chain consisting of data collection and processing, broadcasting as well as the processing in the receiver in combination with a navigation system. For the field trials different types of terminals were implemented. The tests included live traffic data, road works, local hazard warnings, speed limits, synthetic traffic messages and Journaline text messages. The successful field trials showed the potential of future

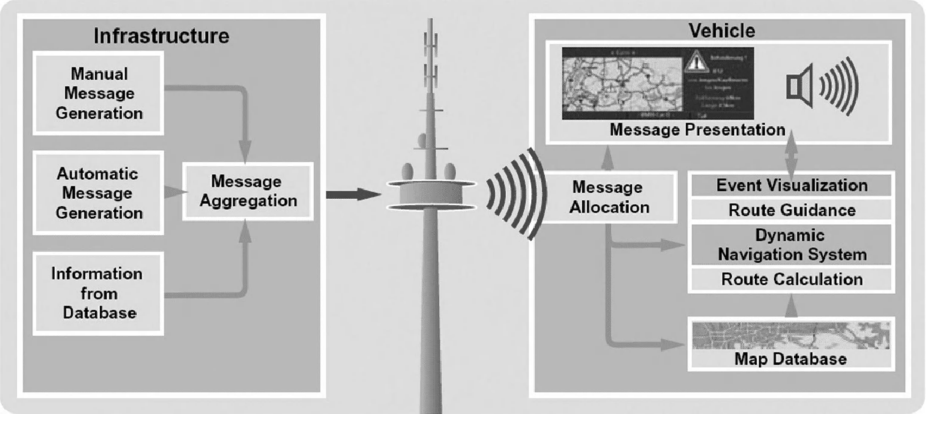

Figure 8.28 Content processing in the mobile.info project

intelligent traffic information. Figure 8.28 shows an overview of the content processing.

The increasing interest of several countries to apply mobile.info platforms resulted in the decision to join forces with the TMC Forum and the TPEG Forum to form the Traveler Information Services Association (TISA).

9

Mobile Television and Multimedia

9.1 Overview

9.1.1 History and Requirements

DAB was originally developed in the late 1980s and early 1990s. At that point in time, radio was the gravity point of this system. It became equipped with features well known from FM-RDS. Targets to be reached were high quality audio, homogeneous RF coverage leading to an uninterrupted and undisturbed sound reproduction also in a mobile environment as well as a more efficient use of spectrum – nowadays called the digital dividend. The only multimedia element at that point of time was the Dynamic Label, an advanced version of what was called Radiotext for RDS.

In the late phase of the standardisation process – the first edition of the core DAB standard was published in September 1995 – multimedia entered the stage [Hallier, 1994b], [Kozamernik, 1995], [Schneeberger, 1996], [Lauterbach, 1996]. The first applications were the Slide Show (SlS) and the Broadcast WebSite (BWS) (see sections 4.4.1 and 4.4.2). Early versions of these applications were already implemented for the first DAB pilot projects addressing mainly car receivers like the Grundig DRC1000, the Blaupunkt Hannover 106 or the Panasonic CN-MX0620/0621L.

In the middle of that decade the EU-supported Eureka Project 147 that brought DAB to life was not over, but accepted the challenge to move towards a mobile TV application for DAB. Addressing mobile receivers with multimedia content – especially AV streaming or 'Mobile TV' – relevant for larger user groups is a promising market evolution, but at the same time a challenge provoking appropriate answers to the following issues:

- reliable reception up to vehicle speeds of 300 km/h;
- sufficient reproduction quality – quasi error-free;
- limited power consumption of the terminals;
- high flexibility regarding employment of new applications;
- low costs.

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

It took another two years before in 1997/1998 an initial draft for a DAB-based audio/video streaming application was developed by the corresponding task force. This first DAB mobile TV specification made use of the MPEG-1 standard for source coding and was already called DAB-DMB [Amor, 1996], [Siegle, 1996], [Lauterbach, 1997c].

Later one migrated to MPEG-2 with a much better coding efficiency. It was especially the German company Bosch that pushed a correspondingly adjusted standard towards ratification in WorldDAB, but the Forum was afraid to mix radio with television and thus compete with the DVB Project. Hence it did not approve this still early version of DAB mobile TV for delivery to ETSI. Nevertheless closed user-group applications of this early 'DMB' system were implemented, in particular for television systems on trains [Kowalski, 1999] and underground lines, e.g. in Berlin.

It took some more years until the idea born in the EU-147 project was combined with cutting edge video coding technology, namely H.264/MPEG-4 AVC, first proposed in Germany [Bauer, 1997], [Grube, 2001] and actually implemented later in South Korea [Lee, 2005], [Cho, 2007], [Lee, 2007]. Another key feature was the improvement of DAB reception performance by additionally using a Reed-Solomon code [Chang, 2003], see sections 2.2.3.2 and 2.2.7.2.

The discussion about the right way to mobile TV started again, when this new proposal entered the WorldDMB Forum Technical Committee (TC) in September 2004. The situation at that point of time was a bit different from the one a few years before. Competition had arisen. In the first place it was DVB-H that targeted the same market. And the support for this development was wide. But DAB-DMB was one to two years ahead in terms of its development status.

Another challenge came from the USA with MediaFLO and from Japan with ISDB-T (1 segment). Both systems target at similar applications and markets as DMB.

The advantage of DAB-DMB seen by WorldDAB was the envisaged availability of terminals in the near future and the advanced status of the draft specification under development. As a background to this fortunate situation was the intention of a number of key players from South Korea to implement DAB-DMB (or T-DMB for a better distinction in Korea from S-DMB, which is technically different) there very soon. DMB obtained official approval as a European ETSI (European Telecommunication Standard Institute) standard in July 2005 [TS 102427], [TS 102428].

In early 2007 another DAB mobile TV variant appeared that was based on DAB IP Tunnelling. An example for the latter path is known in the UK as BT Movio. This application appears on top of IP algorithms that were not fully specified with open standards. In the meantime BT Movio disappeared from the market again.

During the standardisation of DMB several WorldDMB members recognised that this approach would lead away from convergence and alignment to other standards, especially DVB-H. Hence in spring 2005 the way towards alignment was paved with the inauguration of a corresponding TC Task Force. It was obvious from the beginning that convergence towards DVB, 3G-MBMS and OMA would mean a technical approach that is based on the Internet Protocol (IP). An extensive comparison between the remaining two substantial proposals – one acronymed as EDG (Enhanced Data Groups) and the other as DXB (or more precisely eDAB/DXB) –

lead to mainly one disadvantage for each of the two proposals. EDG foresaw a special Transport Stream variant (DAB-TS) slightly deviating from the MPEG-2 TS, accompanied by a new DAB Data Group variant. Also the signaling was different from eDAB/DXB. In summary, EDG came with lower overheads and simpler signaling structures. On the other hand, eDAB/DXB had a higher overhead, but was protocol-stack- and signaling-wise identical with IP DataCast over DVB-H. The increased overhead sat in the PSI/SI and its TS packet structure, often transporting a substantial amount of padding for a limited amount of PSI/SI bytes. Finally this drawback was eliminated by allowing – in a backwards-compatible way – the insertion of different table sections into the same TS packet.

Based on this proposal, the DAB-IPDC (Internet Protocol Datacasting) standard was created and was published by ETSI in 2008 [TS 102978].

9.1.2 Situation of Mobile TV in Different Parts of the World

At the time of writing (spring 2009) DAB-DMB is regularly implemented in South Korea and was on air in Germany for a period of almost two years, but switched off in early 2008. DVB-H was implemented in Italy, the Netherlands, Austria and Switzerland and was foreseen for implementation in France. The media license awarded in Germany was returned to the authorities and it is still unclear if it will be renewed. Japan enjoys mobile TV via ISDB-T. Especially in Europe mobile TV is still waiting for its commercial breakthrough. One commercial project even failed already in the UK with BT Movio based on an older DAB standard called IP Tunnelling.

A new approach towards mobile TV via classical broadcasting networks is the use of DVB-T. This is feasible in the light of the fact that a state of the art DVB-T tuner typically consumes 200 mW and an OLED 2.2’ display 100 mW. Although this power consumption is still significantly higher than that required for decoding DVB-H, but still a mobile device can reproduce mobile TV with good image quality for a few hours. The advantage of this approach is that more or less all services are free of charge and the DVB-T coverage is convincingly wide in Europe, but it is necessary that the multiplex configuration and the network implementation allow for handheld portable (or even mobile) reception. This is not the case throughout Europe.

DAB-IPDC as a new standard has not been implemented regularly so far.

9.1.2.1 Europe: EU List of Standards

In March 2008 the European Commission decided to add the DVB-H standard to the EU List of Standards [EC 2008]. Although member states will be required to encourage the use of DVB-H, this decision hasn’t any binding or legislative character regarding mobile TV implementations in the member states.

9.1.3 Spectrum Resources

With the agreements found at the end of the RRC04/06 process (now referred to as Geneva 2006 or GE06, see section 7.8.2), frequency resources in Europe for DVB-T and

DAB in VHF and for DVB-T in UHF were increased. Herewith spectrum will also become available for DVB-H.

An opportunity for an introduction of MediaFLO in Europe could occur in the UK with the result of the L-Band spectrum auction in 2008 (40 MHz won by an organisation mainly known before as a chip manufacturer). This might become easier through the ETSI standardisation process for FLO, which is already in progress.

As far as spectral efficiency is concerned, all systems in question are of similar performance when (D)QPSK is being used – and this is the case for all implementations so far.

On the one hand, DVB-H suffers from quite a high amount of overhead, but on the other hand, with DVB-H statistically multiplexed video coding can be applied leading to a reduced data rate at the same subjective quality.

9.1.4 Integration of Mobile TV Services in DAB/DAB$^+$ Radio Ensembles

A special advantage of DAB was always the individual character of its sub-channels. The total capacity of 864 so called capacity units is shared beneath these physical layer pipes. The inner code rates can be applied individually to each sub-channel and the size of stream mode channels can be chosen freely with a granularity of 8 kbps. Also the structure of the data travelling in those sub-channels can be chosen individually (see section 2.3). Therefore different protocols can be applied on top of the physical layer. This in the end leads to a simple to organise co-existence of DAB radio with DAB+ radio, DAB-DMB and DAB-IPDC – all dependent on the special requirements of the related providers. This freedom of multiplexing is illustrated in Figure 9.1 below, where a DAB Ensemble carries 8 DAB+ and 3 DAB radio services as well as 2 DAB-DMB

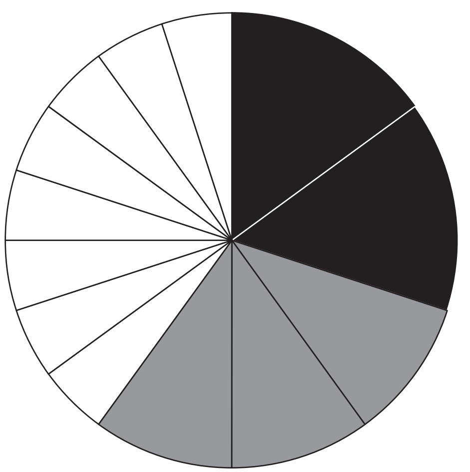

Figure 9.1 DAB Ensemble with DAB (light grey) and DAB+ (white) radio services as well as DAB-DMB (dark grey) television services

services. In this example, each service consists of one service component and occupies a single sub-channel. Thirteen of these pipes are set up with the following attributes (code rate 1/2 for all, data rate is sub-channel rate):

a) Each DAB + services	40 kbps	Equal Error Protection Level 3-A	30 capacity units.
b) Each DAB service	128 kbps	Unequal Error Protection Level 3	96 capacity units.
c) Each DAB-DMB service	224 kbps	Equal Error Protection Level 3-A	168 capacity units.

9.1.5 Transport Mechanisms with Improved Robustness

The more efficient state-of-the-art source coding algorithms are naturally more sensitive to transmission errors. Here the original algorithms based on MPEG-1 and MPEG-2 were significantly more robust and error-tolerant. H.264 video coding requires an average BER as low as 10^{-8} of the stream at the input of the decoder. DAB was originally designed for a BER of 10^{-4} being sufficient at the input of the MPEG-1/-2 Layer II audio decoder. Therefore, new transport mechanisms have been introduced into DAB using a concatenation of the original convolutional code with an additional Reed-Solomon code. These are used for DAB+, DMB and enhanced packet mode, see sections 2.3.3.2 and 2.3.3.5.

9.1.6 The MPEG-2 Transport Stream

The MPEG-2 TS is defined within the standard [ISO/IEC 13818-1] and provides a structure composed of packets of fixed length. In addition, the above mentioned systems part of the MPEG-2 standard defines a limited set of service information with four corresponding tables. For all DVB systems this was the common denominator from the start. The first DVB generation also adopted Reed-Solomon error control coding with a fixed number of parity bytes accompanying each TS packet. Also a convolutional byte-wise interleaver was employed.

The adoption of this transport structure, the error control coding and the interleaving scheme for DAB means that for the first time this layer is not aligned anymore to logical frames on the physical layer as was the case for the MSC stream audio transport mechanism for MPEG-1/-2 Layer II audio and for the MSC packet data transport mechanism. Hence the MPEG-2 TS reflects a new asynchronous packet mode for DAB with fixed packet size. As far as the provision of metadata is concerned, it is available from a number of layers – DAB (MCI/SI), MPEG-2 TS (PSI/SI), MPEG-4 (object, scene descriptors) and – for DAB-IPDC – IP (ESG).

Generally a Transport Stream can carry a single or multiple service. As far as DMB is concerned, each single TS transports a single service in order to stay aligned to traditional DAB service structures.

Figure 9.2 shows the structure of the MPEG transport stream packets. The meaning of the header fields is as follows:

- Sync: Synchronisation byte, always set to 0x47, allows – together with the fixed packet size of 188 Bytes (error control coding excluded) – for easy synchronisation and packet access on receiver side.
- TEI: Transport Error Indicator, indicates packet errors already present before transmission.
- PUSI: Packet Unit Start Indicator, indicates the presence of an additional Byte after the 4-Byte long standard header. If present, the additional Byte indicates the start position of a new datagram inside the TS packet payload.
- TP: Transport Priority, classifies the priority of the TS packet payload in two categories.
- PID: Packet IDentifier, identifies elementary streams (e.g. video 1, audio 1, . . .).
- TSC: Transport Scrambling Control, indicates the encryption of the payload.
- AFC: Adaptation Field Control, indicates the presence of an adaptation field after the header.
- CC: Continuity Counter, allows for tracking sequence of packets belonging to the same elementary stream.

The TS multiplexes elementary streams carrying a single service component (video, audio, service information, data applications) each, as shown in Figure 9.3. It furthermore provides a receiver clock reference (PCR – program clock reference, 27 MHz) for the accurate reproduction of the aforementioned service components on the receiver side.

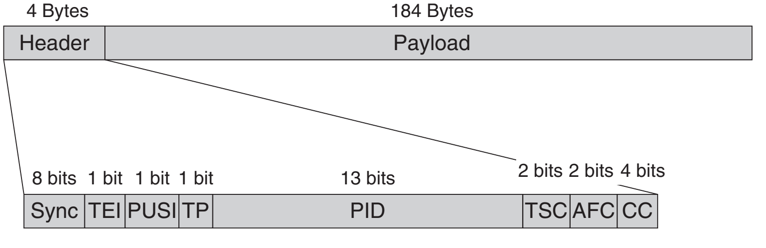

Figure 9.2 MPEG-2 Transport Stream Packet

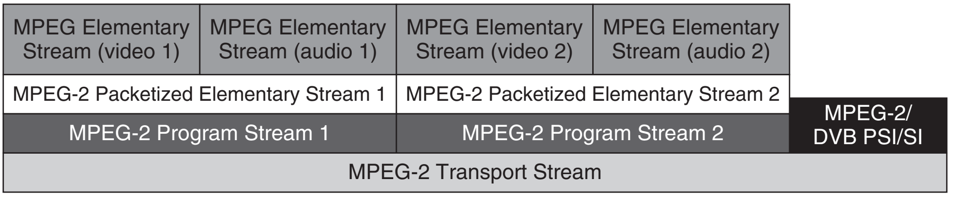

Figure 9.3 MPEG protocol stack

The above mentioned systems part of the MPEG-2 standard [ISO/IEC 13818-1] also defines the basic programme service information (PSI) tables, namely the Program Association Table PAT, the Program Map Table PMT, the Conditional Access Table (CAT) and the Transport Stream Description Table (TSDT). It also refers to the NIT, but does not specify that table.

The DVB service information (SI) standard [EN 300 468] defines further tables required for DVB (and DAB) operation. The most important ones are the Network Information Table NIT, the Service Description Table (SDT), the Event Information Table (EIT) as well as the IP/MAC Notification Table (INT).

The PAT indicates for each service in the multiplex the PID (Packet Identifier) of the corresponding PMT and also of the NIT common to the multiplex.

The PMT provides the PIDs for all components making up a service as well as the PID of the relevant Program Clock Reference (PCR).

The NIT conveys information relating to the physical organisation of the multiplexes/TSs carried via a given network, and the characteristics of the network itself.

The SDT gives further human-readable information about the services like service and provider names.

The EIT contains information about events for each service in chronological order. Parameters are the event name, start time and duration. An event can be a programme item.

Each table – apart from the PMT – carries a pre-defined PID for table identification purposes.

9.1.7 Commonalities of DAB-DMB and DAB-IPDC

Starting from the top of the protocol stack, there is quite a high degree of commonalities of DAB-DMB and DAB-IPDC (assuming a mobile TV application according to [TS 102978] and the corresponding DVB standards). Source coding will be the same in most cases, i.e. video is encoded with H.264/MPEG-4 AVC and audio with HE-AAC v2. Below, the MPEG-2 Transport Stream as well as its error control coding and the convolutional interleaving are the same for both. Also all layers below are fully identical – apart from the fact that DMB uses one TS and one sub-channel per service and IPDC can insert several services into one TS and distribute this single TS via several sub-channels.

9.2 DAB-DMB

9.2.1 Overview

Although the acronym 'DMB' sounds like a new broadcasting system, it is in fact an application building on the same physical layer DAB as all other applications.

With the MPEG-2 Transport Stream, DAB has adopted for the first time an asynchronous transport mechanism. From the beginning, DAB structures were built around the MPEG-1 Layer II audio frames of 24 ms duration. This length is adopted for the Logical Frames of DAB. Also traditional DAB packets – the MPEG-2 TS is nothing else than a new packet structure – were aligned to this time period in a way that with all sub-channel rates always an integer number of packets would be delivered.

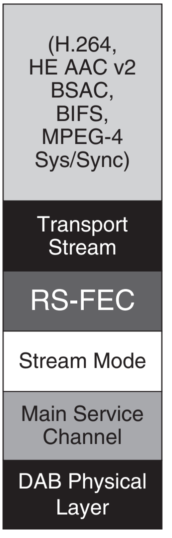

Figure 9.4 DAB-DMB Stack

The MPEG-2 TS is accompanied by Reed-Solomon parity bytes and builds a structure of constant length packets of 204 bytes, see Figure 9.4. These packets are convolutionally interleaved. The whole mechanism is the same as the one used for DVB.

The above mentioned outer coding and interleaving is necessary for assuring a sufficiently low BER for quasi error-free video reproduction in all DAB reception environments. Scalability is still given with the different inner code rates provided by the physical layer DAB.

The adaptation of the MPEG-4 system and synchronisation layer to the MPEG-2 system layer is described in the following sub-clause. For source coding H.264/MPEG-4 AVC (video) and HE-AAC v2 (audio) or alternatively MPEG-4 BSAC can be applied, i.e. two different profiles are defined for the application DMB.

9.2.2 Transport

The source-coded video and audio content is structured in so called Access Units. These are wrapped independently into Synchronisation Layer (SL) packets. In the next step MPEG-2 structures take over with separate Packetised Elementary Streams for both, audio and video. These as well as the PES packets carrying Object and Scene Description (OD, SD) and also BIFS animation data are multiplexed together into the MPEG-2 Transport Stream.

In addition the Initial Object Description (IOD) becomes part of the Programme Map Table PMT in the form of a corresponding descriptor (IOD_descriptor) in the first descriptor loop of the PMT. The corresponding stack illustrating these protocol layers is provided with Figure 9.5 below. It further details the figure in Appendix 3 regarding the layers above the MPEG-2 TS.

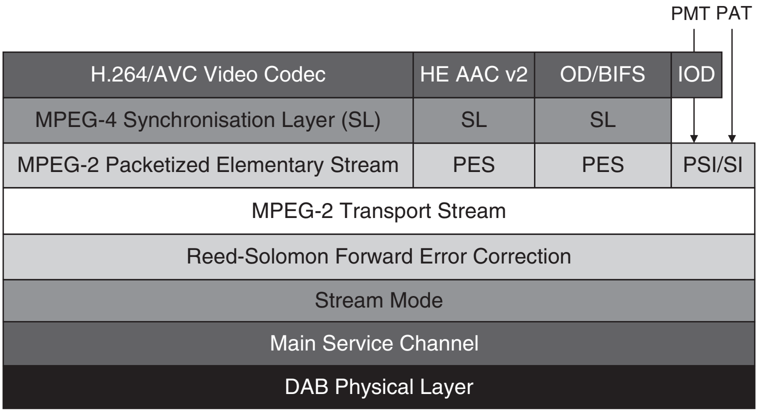

Figure 9.5 DAB-DMB protocol stack in detail

The Object Clock Reference (OCR) for synchronisation of MPEG-4 objects and the Composition Time Stamp (CTS) are inserted into SL/PES packets at least every 100 ms.

9.2.3 PSI/SI Signalling

For DAB-DMB a limited set of PSI/SI tables [TS 102 428] is employed, mainly describing the configuration of the single service content within the related Transport Stream (for details see sub-clause 9.1.8. above). These are the tables – together with their minimum repetition rates:

- PAT: Due to the specific DMB structure, i.e. one service per TS, the Programme Association Table describes only that one programme service – with a minimum repetition rate of two per second.
- PMT: The Programme Map Table obeys a number of restrictions for DMB purposes as follows:
  - A group of descriptors with Restriction 'A' in the table shall include an IOD_descriptor.
  - A group of descriptors with Restriction 'B' in the table shall include an SL_descriptor for an ES_ID.

The minimum repetition rate is the same as for the PAT above, i.e. at least twice per second.

- PCR: Minimum repetition rate is 10 times per second.
- SD/OD: For both, the scene and the object description information the minimum repetition rate is twice per second.
- OCR/CTS: For both, the object clock reference and the complementary time stamp, the minimum repetition rate is 10 times per 7 seconds.

9.2.4 DMB-Radio

The DAB-DMB standard clearly identifies this application as a video one and the whole standard does not cover its use as a radio application. Nevertheless, on the first view it seems to be quite a trivial exercise to let DAB-DMB be audio- instead of video-centric. But this first impression is not a very accurate one.

Precedence for the use of DAB-DMB for radio purposes provided the German 'watcha' DAB-DMB project that started in spring 2006. The simple reason for that approach was that all remaining services consisted of mobile TV content and one didn't accept that for MPEG-2 audio the capacity investment would have been in the same order as for the TV services in question. This resistance is certainly justified by the standardisation reality at that point of time, but in the meantime DAB+ (see section 3.4.2) has been developed for the purpose of audio transmission using MPEG 4.

Also the big French radio broadcasters announced that they will employ DAB-DMB for radio purposes. Reasons named were the rich multimedia options and the availability of smart devices. Whereas there could hardly be doubts regarding the latter remark – well-known reference is the iRiver B20 – the multimedia options of DAB-DMB are not richer than those based on IP.

From the sheer technical point of view DAB-DMB misses many traditional radio functions, among them:

- no announcements enabled;
- no Pty signaling;
- no Service Component language;
- no Cross-referencing to other delivery systems.

9.2.5 Overhead of Low Rate Applications

Every broadcasting system requires the addition of a particular overhead on top of the sheer content to be transported. Typical examples for such an overhead are synchronisation signaling, error control coding as well as service parameter signaling and metadata. In particular, when narrowband applications are transported with DAB-DMB, the figures given range from 'a few percent' to half of a stream. In order to give such discussion a reliable basis, an example case is discussed here in detail – a DMB streaming application.

Let us consider a time slot of a narrowband DAB sub-channel used for DAB-DMB and consisting of a single stream. Because it is a common denominator of the entities we want to discuss, let it be 7 seconds long. The sub-channel bitrate for this example (other examples can be derived from this exercise) is 40 kbps. In the binary case 'k' for 'kilo' is equivalent to 1024. An MPEG-2 Transport Stream fills the sub-channel completely, which means that within the 7 seconds 175.69 MPEG-2 TS packets – all with 16 Reed-Solomon parity bytes attached – can be transported.

Starting with the PSI/SI signalling required for this use case, we need to transport the two tables PAT (Program Association Table) and PMT (Program Mapping Table) every 500 ms. Hence within 7 seconds 28 of those tables occur and 147.69 TS packets are left for other purposes – assuming that each of the tables can be accommodated by one single TS packet. The MPEG-4 Initial Object Descriptor IOD is part of the PMT.

For a complete description of the object transported – i.e. the application data stream – the MPEG-4 system layer entity OD (Object Description) gets its PID and corresponding TS packets that shall be repeated every 500 ms again. So 14 packets are assigned to OD and the remaining amount sums up to 133.69 TS packets. These packets offer 184 bytes each for the payload. So altogether 24,598.96 bytes are available for payload to be transmitted.

For controlling the 27 MHz receiver clock accurately, the 'Program Clock Reference' PCR parameter is required every 100 ms. PCR is provided within the so called adaptation field being part of a TS packet and located right after the 4-byte header. The PCR carries the same PID as the accompanied stream and can herewith be transported in the TS packets carrying the payload. PCR travels in the Adaptation field and occupies 8 bytes per occurrence. In total 560 bytes will be consumed by PCR in seven seconds. With this, 24,038.96 Bytes are left.

MPEG-4 Access Units (AU) carry the MPEG-4-encoded content. Each AU is embedded in an SL packet and the SL packet in a PES packet. Insertion of PES packets into TS packets can be done in a fragmented way. Assuming a length in time of 60 ms for each MPEG-4 AU, 116.67 of them need to be transported within seven seconds. This is equivalent to the number of SL and PES packets employed for the transport. The PES packet overhead is 5 bytes per packet and the SL packet overhead is 1 byte. Hence the complete overhead for 7 seconds is 700 bytes.

Due to the fact that every 700 ms the Object Clock Reference (OCR) for synchronisation of MPEG-4 objects and the Composition Time Stamp (CTS), each of them 33 bytes long, are required, every 11th PES/SL packet carries 66 bytes of this overhead in addition, which sums up to – once again – 700 bytes.

Subtracting these 1400 bytes from the 24,038.96 above, there are 22,638.96 Bytes available for the transport of naked Access Units. This value can be converted to 25.27 kbps remaining bitrate for the Access Units. It is equivalent to 63.17% of the sub-channel bitrate of 40 kbps. So the overhead for the discussed example is 36.83%.

With this calculation and the related assumptions applied to several more sub-channel bitrates we get the results presented in Figure 9.6. In summary, it is recognised that for

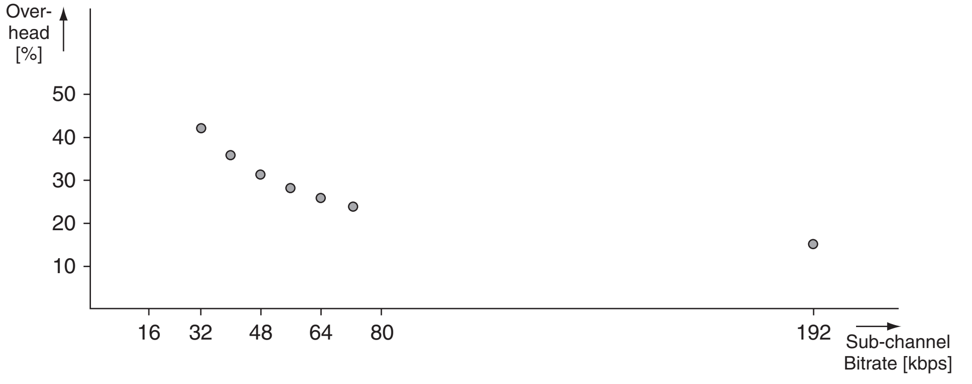

Figure 9.6 Overhead (minimum) as function of application data rate

applications requiring low sub-channel bit rates the combination of MPEG-2 and MPEG-4 system layers leads to quite a significant overhead. For higher data rates the overhead is less significant.

9.2.6 DMB Implementations

The first country that regularly implemented DMB was the one that also initialised the standardisation process within the WorldDAB Forum – South Korea. The introduction there dates back to late 2005 and 'T-DMB' – as DAB-based DMB is identified in that country – co-exists with a system called 'S-DMB', but which is technically completely different from the broadcasting system described here.

In Germany DMB was introduced as a regular service in May 2006, i.e. in time for the football world cup 2006 taking place in this country. It remained on air until early 2008. The two devices attracting the highest interest were the mobile phones Samsung SGH-P900 and LG V9000 with integrated DAB/DMB functionality.

The portable multimedia device receiver B20 shall be named here as well since it is probably the most advanced receiver for many of the applications described in this book at this point of time – including DAB+.

Also the Peoples Republic of China put DMB on air in several areas of the country in preparation for the 2008 Olympics. Initially, half a million devices were imported, but later on the production of DMB devices started in China. In this large country several different systems were tried out and currently there seems to be a preference for systems developed in China, e.g. CMMB/STiMi.

Tests with DMB were executed in many European countries in 2008, among these Czech Republic, Italy, Luxemburg, Malta, the Netherlands, Norway, Portugal Spain, Sweden, Switzerland and the UK. Outside Europe trials are undertaken in Australia, Ghana, Indonesia, Kuwait, Malaysia, New Zealand, Singapore and South Africa.

A special case is France where decisions were taken that DMB shall become a regular service for multimedia radio purposes soon.

9.3 DAB-IPDC

The movement towards this standard was initialised in March 2005 by kicking off WorldDAB TC Task Force Alignment. The basis for this was the presence of first hybrid consumer equipment and the expectation for more. Hybrid devices employ more than a single broadcasting and/or mobile communication standard.

Although a number of proposals were made – most promising were certainly the EDG/DAB-TS and the eDAB/DXB ones – the WorldDMB Forum in the end decided in favour of the one explained in detail below, because a family approach within the broadcast society with hooks towards the mobile communication world was seen as most future-proof.

9.3.1 Overview

Similar to DAB-DMB described above, DAB-IPDC makes use of the MPEG-2 Transport Stream as an underlying layer, but opposite to DAB-DMB, DAB-IPDC is able to provide room for several services in the same TS. This directly leads to the significant advantage

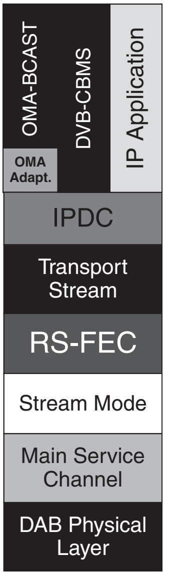

Figure 9.7 DAB-IPDC Stack

that statistical multiplexing can be employed and the corresponding gain can be used for transporting more services. The protocol stack of DAB-IPDC is shown in Figure 9.7.

In the same way as IPDC via DVB-H, DAB-IPDC encapsulates content in IP datagrams and those datagrams in multi-protocol encapsulation (MPE) sections. Opposite to DVB-H, DAB-IPDC does not require a third layer of error protection. It is sufficiently robust even for mobile reception with the DAB-immanent convolutional coding cascaded with the Reed-Solomon error control coding as used for DVB-T. Here DAB-DMB and DAB-IPDC are equivalent.

As far as overhead is concerned, DAB-IPDC naturally comes with IP protocol overhead that is not present for the application DAB-DMB.

Although both mechanisms might serve for mobile TV applications in the first place, DAB-IPDC can make direct use of a range of existing CBMS applications defined by the DVB project or – via the OMA-BCAST adaptation specification for DVB-H – can even feature quite a wide range of applications defined by the Open Mobile Alliance.

Last but not least, DAB-IPDC is in line with the current EU-driven movements in Europe favouring DVB-H as the preferred technology for mobile TV on that continent.

In order to enable power saving best, the above mentioned single TS can be distributed service-wise over several smaller DAB sub-channels in order to maintain the option to decode just those sub-channels that are carrying the currently reproduced service.

9.3.2 Transport

As illustrated with Figures 9.7 and 9.8 below, DAB-IPDC builds a basis for mobile TV applications as well. In the same way as DMB, it makes use of the MPEG-2 Transport

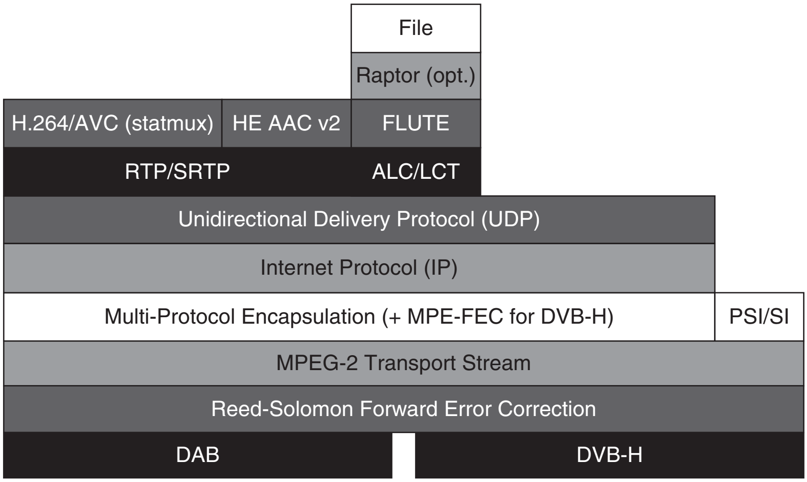

Figure 9.8 Protocol stacks *for* IPDC on *top of* DAB and DVB-H physical layers

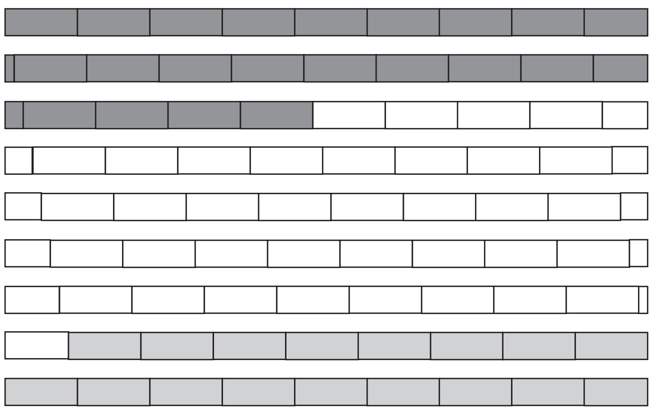

Figure 9.9 Single big sub-channel (600 kbps) transporting TS packets of 3 services in macro-sliced manner (1800 bytes per 24 ms logical frame, equivalent to 8.82 TS packets incl. RS-FEC)

Stream and applies outer error control coding as well as convolutional interleaving to the TS packets.

The main difference to DMB is that several services can travel in the same TS, which in turn enables the application of statistical multiplexing to the video streams of those services. This smart approach results in one to two more TV services that can be transported at the same quality level with a single DAB Ensemble!

In addition, the TS packets belonging to the same service are assigned to a certain group of DAB sub-channels. Therefore the receivers need to decode only those sub-channels that carry the desired service. Due to the statistical multiplexing a part of those sub-channels will also carry TS packets of other services. Hence after sub-channel selection still packet stripping needs to be applied by the terminals.

Furthermore, DAB-IPDC is closely aligned to DVB-H and hence applies IP protocols above the TS. Also all signaling and IP structures are identical with DVB-H. So once implemented, the same middleware can process DVB-H and DAB-IPDC. Opposite to the DAB, DVB-H requires an additional error control coding layer on MPE level (MPE-FEC) accompanied by virtual time interleaving.

The difference between the classical transport of a 'big' Transport Streams via a single sub-channel and its transport via several smaller sub-channels shall be illustrated with Figure 9.10 below. The gain in terms of micro time slicing is described above.

9.3.3 PSI/SI Signalling

A few rules are applied to the provision of PSI/SI in order to enable receivers to easily find that information especially for the sake of a quick start of content reproduction after device switch-on:

- All PSI/SI shall travel in the primary sub-channel of the DAB-IPDC service.
- The Programme Association Table (PAT) describes only one programme service and shall be repeated at least twice a second.
- One single Program Map Table (PMT) shall describe the whole IP Platform with its transported MPEG Services. Its minimum repetition rate is the same as for the PAT's, i.e. twice a second.

9.3.3.1 Compact PSI/SI/SAT

As illustrated with Figure 9.11, the signalling overhead for DAB-IPDC was significantly reduced by a simple measure. If PAT-carrying TS packets provide enough unused space,

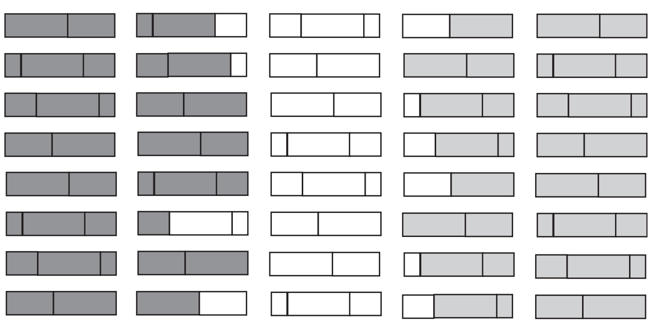

Figure 9.10 Five sub-channels (120 kbps each) transporting TS packets of 3 services in micro-sliced manner (360 bytes per sub-channel per 24 ms logical frame, equivalent to 1.76 TS packets incl. RS-FEC)

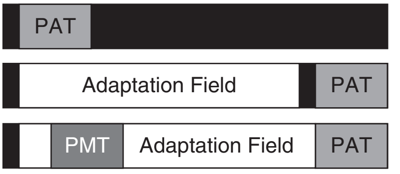

Figure 9.11 Compact PSI/SI/SAT (here: PMT inserted into PAT-carrying TS packet)

other PSI/SI/SAT table sections (e.g. PMT) might be inserted into the adaptation field. It is not permitted to move all tables to the adaptation fields. A particular portion shall travel in the conventional way. For details see [TS 102978].

This way, even legacy TS processing would not be confused, i.e. this is a backwards-compatible solution. Of course, existing middleware would not find the inserted tables and hence for new devices the corresponding amendment should be applied to the existing PSI/SI middleware.

9.3.4.2 Sub-channel Assignment Table (SAT)

The Sub-channel Assignment Table (SAT) maps the services by means of their elementary_PIDs to the sub-channels (physical layer pipes) that are identified with their service_component_index. With this information, the receivers are enabled to grab those and only those sub-channels that are carrying the desired service. This way, micro-time slicing can still be applied – in the presence of a 'big TS' permitting for statistically multiplexed video coding of the different video streams.

9.3.4 Application 'Mobile TV'

Source content is encoded with e.g. H.264/AVC (video) and HE AAC v2 (audio) algorithms. As far as video is concerned, the encoding process shall be realised in a statistically multiplexed manner. For this approach a maximum constant bitrate is fixed for the sum of the output of all correspondingly grouped video codecs. This available bandwidth is assigned partially to the single video streams depending on what they require for particular time intervals that the coding process is structured with – e.g. one second. With this, the streams will differ over time in their capacity requirements, but altogether will never exceed the fixed total bandwidth. This process leads to significant capacity gains in the order of 20 to 30%, depending on image quality requirements and the number of streams encoded in parallel. The processing of the different streams is shown in Figure 9.12.

The access units output by the video and audio codecs are wrapped into RTP datagrams first. This datagram is extended to a UDP one suited for unidirectional transport means, i.e. broadcasting. After that the IP datagram is completed with the IP header consisting of source and destination address. During this transport protocol encoding process content of services and service components is kept together. Synchronisation of content belonging together – with video and audio as the most critical components in

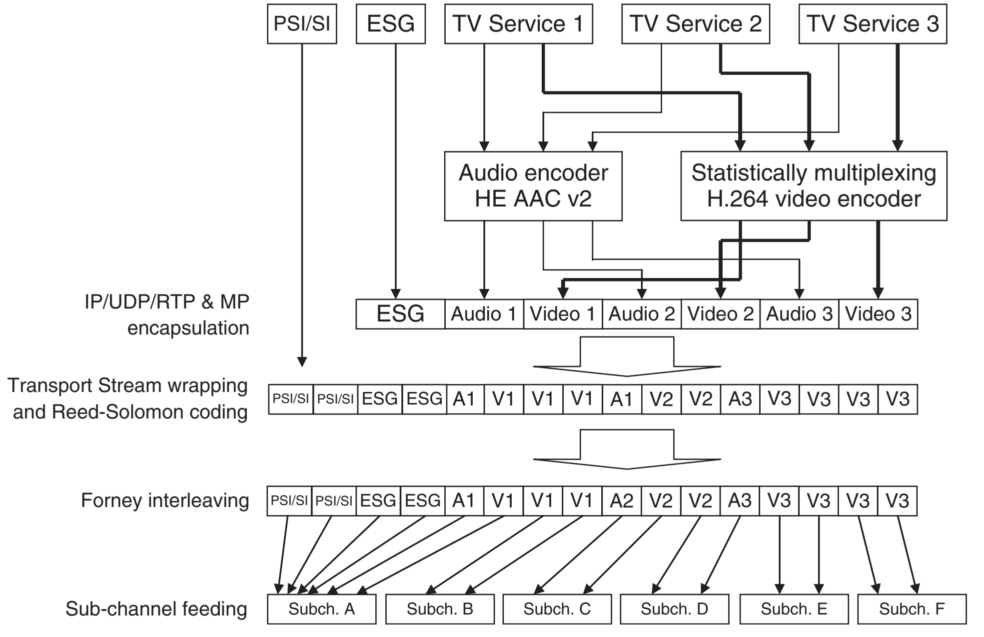

Figure 9.12 Assigning service content to multiple sub-channels

terms of lip sync accuracy in the first place – is realised by means of RTCP, a time stamp mechanism allowing for accurately synchronised reproduction of the different streams at the receiving end of the broadcasting chain.

IP datagrams are then encapsulated in multi-protocol encapsulation (MPE) sections. MPE is also able to encapsulate other higher layer protocol datagrams (e.g. Ethernet). Opposite to DVB-H, no MPE-FEC sections are built since DAB doesn't require error control coding layer for fulfilling the mobile TV service quality requirements. That also means lower overhead (here DVB-H typically adds an overhead of 25%).

The MPE sections are inserted into TS packets of 188 bytes length each and those are accompanied by 16 Reed-Solomon parity bytes resulting in 204 byte long packets. In addition the packets are convolutionally interleaved in a byte-wise manner. These are all standard DVB procedures.

After that stage, the RS-FEC'ed TS packets are filled in order into a limited number of pre-defined sub-channels (→ SAT, Sub-channel Assignment Table) in a deterministic way enabling their extraction in the same order on the receiver side. Due to the statistical multiplexing process, there will always be enough space for all TS packets, but the packets of single service will swap into neighbouring sub-channels whenever their capacity requirement is higher. All the sub-channels a single service is allowed to occupy are described in the sub-channel assignment table (SAT). So the receiver will only strip the content of this limited number of sub-channels and will ignore the rest. This way power saving mechanisms based on micro time slicing like power cycling can be applied. Of course, receivers still need to separate wanted from unwanted content, because one or more of the decoded sub-channels are consisting of content of more than the desired service.

9.4 Application Standardisation

Historically the bearer-related fora overtook the responsibility of drafting the norms for the applications they intended to apply individually. Nowadays this approach would be too inflexible on its own. Therefore the adoption of externally developed application and transport protocol standards will become a common exercise. Corresponding liaisons are built already with the Joint Technical Group (JTG) of the DVB Project and the WorldDMB Forum.

Due to the more challenging competition with wireless and mobile communication standards and implementations, but also for enabling seamless handover between different bearer technologies, the collaboration within the broadcast camp and with the groups producing the wireless and mobile communication as well as bearer-agnostic application standards shall be strengthened.

Convergence towards a ubiquitous IP environment for communication purposes might be the mid to long term target.

9.5 Conclusions

Although structurally DMB and DAB-IPDC represent different layers of the DAB protocol stack, they are also used as acronyms for two different ways DAB offers for transporting television and multimedia content, i.e. these acronyms cover the whole stack above the physical layer.

Both are requiring an outer FEC layer realised with a Reed-Solomon code of same length and error correction capability. Also on the top end of the stack, both routes are equivalent in terms of the mobile TV application regarding source coding – video with H.264/AVC and audio with HE-AAC v2 (plus MPEG-4 BSAC for DMB). The main difference sits in the middle and affects how access units are transported. Also the Layer II signaling differs consistently.

DMB sticks to the original service structure of DAB and employs a single service per (stream mode) sub-channel (in fact, it differs in so far as DAB foresaw a single service component per sub-channel and not a service) transported by means of the standard DVB protocol – the MPEG-2 Transport Stream. MCI/SI signaling and service access structures of DAB could be retained and were accompanied by MPEG-2 (PSI/SI) and MPEG-4 system layer signaling elements where required. Herewith existing DVB middleware can be re-used with limited adjustment to the application in question.

The drawback of this solution is that statistical multiplexing in conjunction with the video source coding is not enabled. Hence the significant gain of this approach cannot be realised.

DMB does not wrap source-coded content into IP/UDP/RTP datagrams, which on the one hand side saves quite some overhead, but on the other hand does not lead to alignment in a more and more IPised communication environment.

The lower the bitrate of the DMB service the higher the overhead portion will be. For typical audio stream bitrates with state-of-the-art source coding, DMB comes with quite a significant overhead. Therefore DMB was not specified for radio purposes.

Although seldomly used, BIFS is a powerful tool and enhances DMB in an attractive way. It would even be possible to use DMB-BIFS as an application accompanying DAB and DAB+ radio services.

DAB-IPDC uses a different approach and puts emphasis on making use of the statistical multiplexing gain while enabling micro time slicing through a dedicated distribution of service content over a limited number of sub-channels. The receiver – as with radio services – only needs to grab the capacity units that belong to the desired sub-channels and can power down in between. Also macro-time slicing can be used in a similar way as for DVB-H, but would require corresponding memory on the receiver side. The efficiency of macro time slicing for DAB in general is influenced by the fixed time interleaving of 384 ms.

The statistical multiplexing gain leads to one or two more TV services per multiplex, depending on the settings.

The use of IP datagrams led, leads and will also in the future lead to discussions in the broadcasting environment. It is obvious that for free-to-air services the overhead might be unnecessarily high. On the other hand addressing of devices for e.g. content requiring subscription is well eased. Also the distribution of content received via DAB in a home network requires IPisation. In a communication environment more and more based on IP, broadcasting cannot ignore this trend. For the mentioned reasons, other systems have chosen an 'all IP approach', i.e. transporting content in any other form than IP datagrams is not enabled at all.

The main goal of DAB-IPDC was to achieve the maximum degree of convergence with DVB-H. This was achieved and as shown above, both IPDC mechanisms are equivalent above the bearer system's physical layer. Herewith the implementation in hybrid devices able to receive DAB and DVB-H in parallel is eased dramatically.

This equivalence also opens the door towards OMA-BCAST applications in parallel to DVB-H.

Last but not least DAB-IPDC fulfills the requirement for a unique standard for mobile television in Europe.

Appendix 1

DAB Parameters for Modes I, II, III and IV

A1.1 System Parameters

Parameter	Mode I	Mode IV	Mode II	Mode III
Sub-carriers				
Number of sub-carriers: K	1536	768	384	192
Sub-carrier spacing: Δf	1 kHz	2 kHz	4 kHz	8 kHz
Time relations				
Transmission frame duration: T_{Frame}	96 ms	48 ms	24 ms	24 ms
	196608T*	98304T	49152T	49152T
Symbol duration: $T_{symOFDM} = T_{guard} + T_u$	1246 µs	623 µs	312 µs	156 µs
	2552T*	1276T	638T	319T
Guard interval duration: T_{guard}	246 µs	123 µs	62 µs	31 µs
	504T*	252T	126T	63T
Symbol duration without T_{guard}: $T_u = 1/\Delta f$	1000 µs	500 µs	250 µs	125 µs
	2048T*	1024T	512T	256T
Null-symbol duration: T_{null}	1297 µs	648 µs	324 µs	168 µs
	2656T*	1328T	664T	345T
OFDM symbols				
OFDM symbols per transmission frame (without null symbol): L	76	76	76	153
OFDM symbols with PR data	1	1	1	1

(*continued overleaf*)

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

(*continued*)

Parameter	Mode I	Mode IV	Mode II	Mode III
OFDM symbols with FIC data	3	3	3	8
OFDM symbols with MSC data	72	72	72	144
FIC/MSC				
FIC: FIBs per transmission frame	12	6	3	4
FIBs per 24 ms frame	3	3	3	4
MSC: CIFs per transmission frame	4	2	1	1
CIFs per 24 ms frame	1	1	1	1
FIBs/CIF	3	3	3	4
Transmission frame				
Bit per OFDM symbol	3.072 kbit	1.536 kbit	0.768 kbit	0.384 kbit
Bit per transmission frame (without PR symbol)	230.4 kbit	115.2 kbit	57.6 kbit	58.368 kbit
Transmission frames per second	10.416	20.832	41.666	41.666
Data rates				
FIC data rate (gross, code rate always 1/3)	96 kbps	96 kbps	96 kbps	128 kbps
MSC data rate (gross)	2.304 Mbps	2.304 Mbps	2.304 Mbps	2.304 Mbps
Max. MSC net data rate for a single sub-channel**	1.824 Mbps	1.824 Mbps	1.824 Mbps	1.824 Mbps
Total data rate (with PR Symbol)	2.432 Mbps	2.432 Mbps	2.432 Mbps	2.448 Mbps
Network specific parameters				
Maximum echo delay ($\approx 1.2 \times T_{guard}$)	300 μs	150 μs	75 μs	37.5 μs
Maximum propagation path difference	≈100 km (90 km)	≈50 km (45 km)	≈25 km (22.5 km)	≈12.5 km (11.25 km)
Maximum f_{RF}***	340 MHz		1.38 GHz	2.76 GHz
****	375 MHz		1.5 GHz	3.0 GHz

* System clock: 2.048 MHz with a period T of 0.48828 μs.

** The multiplex configuration for maximum data rate is as follows: one sub-channel with 1.824 Mbps and code rate 4/5 and a second sub-channel with 16 kbps and code rate 3/4. The remaining multiplex capacity of 64 bit per 24 ms frame is equivalent to an uncoded data rate of 8 kbps.

*** @ max f_{RF}: maximum S/N-degradation of 4 dB for a BER of 10^{-4} at a speed of 200 km/h [Le Floch, 1992].

**** @ max f_{RF}: maximum S/N-degradation of 4 dB for a BER of 10^{-4} at a speed of 180 km/h and 1 dB at 90 km/h [Kozamernik, 1992].

A1.2 Important Relations

1 CU = 64 bits = 8 bytes
1 CIF = 864 CU = 55.296 kbits = 6.912 kbytes
1 FIB = 256 bits = 32 bytes
Where: CU = Capacity Unit

A1.3 Coarse Structure of the Transmission Frame

|Null-symbol | PR | FIC (FIBs) | MSC (CIFs)|

Where: PR = Phase Reference symbol
FIC = Fast Information Channel
FIB = Fast Information Block
MSC = Main Service Channel
CIF = Common Interleaved Frame

Appendix 2

Frequencies for Terrestrial and Satellite DAB Transmission

The frequencies for terrestrial DAB transmission (T-DAB) were co-ordinated at a CEPT planning conference [CEPT, 1995] for Band I (47 MHz to 68 MHz), Band III (174 MHz to 240 MHz) and L-Band (1452 MHz to 1467,5 MHz), valid in particular for European terrestrial DAB implementations (T-DAB), see Table A2.1. A further international planning conference for the VHF and UHF broadcasting bands in Europe, Africa, Middle East and the Islamic Republic of Iran held its final session in Geneva from May to June 2006 [RRC06, 2006]. This resulted for T-DAB in increased spectrum in Band III (VHF), which is partially already available. After a transition period it will become fully available in 2015 (except for a few countries, where this will not happen until 2020). The actual T-DAB frequency blocks remained unchanged in this process, only a new assignment of which block may be used in which area.

Slightly different centre frequencies for Band III were specified for DAB and DRM services in Korea, see Table A2.4 [WorldDMB Website]. The DAB block numbers in Band I and Band III correspond to the naming convention for TV channels in Europe.

CENELEC made a subsequent Industry standard [EN 50248] which does not recommend frequencies in Band I (printed in *italics*), due to the high man-made noise within this frequency range. Therefore, there will be no receiver equipment available which does support Band I.

NOTE 1: CENELEC introduced three additional DAB blocks named 10N, 11N, 12N which occupy some of the broader guard intervals, see Table A2.1. Offsets to Blocks 10A, 11A and 12A will allow these blocks to be used in areas also covered by B/PAL/NICAM TV transmitters operating in the lower adjacent channels. The TV transmitters also need to be offset in frequency by the maximum allowable amount (approx. 200 kHz).

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

Table A2.1 Frequencies for T-DAB in Europe

T-DAB Block number	T-DAB Block label	Centre frequency (MHz)	Block corner frequencies (MHz)	Lower/upper guard distance (kHz)
Band I: 47,0 to 68,0 MHz				
(1)	2A	47,936	47,168 to 48,704	168/176
(2)	2B	49,648	48,880 to 50,416	176/176
(3)	2C	51,360	50,592 to 52,128	176/176
(4)	2D	53,072	52,304 to 53,840	176/320
(5)	3A	54,928	54,160 to 55,696	320/176
(6)	3B	56,640	55,872 to 57,408	176/176
(7)	3C	58,352	57,584 to 59,120	176/176
(8)	3D	60,064	59,296 to 60,832	176/336
(9)	4A	61,936	61,168 to 62,704	336/176
(10)	4B	63,648	62,880 to 64,416	176/176
(11)	4C	65,360	64,592 to 66,128	176/176
(12)	4D	67,072	66,304 to 67,840	176/160
Band III: 174,0 to 240,0 MHz				
13	5A	174,928	174,160 to 175,696	160/176
14	5B	176,640	175,872 to 177,408	176/176
15	5C	178,352	177,584 to 179,120	176/176
16	5D	180,064	179,296 to 180,832	176/336
17	6A	181,936	181,168 to 182,704	336/176
18	6B	183,648	182,880 to 184,416	176/176
19	6C	185,360	184,592 to 186,128	176/176
20	6D	187,072	186,304 to 187,840	176/320
21	7A	188,928	188,160 to 189,696	320/176
22	7B	190,640	189,872 to 191,408	176/176
23	7C	192,352	191,584 to 193,120	176/176
24	7D	194,064	193,296 to 194,832	176/336
25	8A	195,936	195,168 to 196,704	336/176
26	8B	197,648	196,880 to 198,416	176/176
27	8C	199,360	198,592 to 200,128	176/176
28	8D	201,072	200,304 to 201,840	176/320
29	9A	202,928	202,160 to 203,696	320/176
30	9B	204,640	203,872 to 205,408	176/176
31	9C	206,352	205,584 to 207,120	176/176
32	9D	208,064	207,296 to 208,832	176/336
33	10A	209,936	209,168 to 210,704	336/(176)
NOTE 1	10N	210,096	209,328 to 210,864	
34	10B	211,648	210,880 to 212,416	(176)/176
35	10C	213,360	212,592 to 214,128	176/176
36	10D	215,072	214,304 to 215,840	176/320
37	11A	216,928	216,160 to 217,696	320/(176)
NOTE 1	11N	217,088	216,320 to 217,856	
38	11B	218,640	217,872 to 219,408	(176)/176

Table A2.1 (*continued*)

T-DAB Block number	T-DAB Block label	Centre frequency (MHz)	Block corner frequencies (MHz)	Lower/upper guard distance (kHz)
39	11C	220,352	219,584 to 221,120	176/176
40	11D	222,064	221,296 to 222,832	176/336
41	12A	223,936	223,168 to 224,704	336/(176)
NOTE 1	12N	224,096	223,328 to 224,864	
42	12B	225,648	224,880 to 226,416	(176)/176
43	12C	227,360	226,592 to 228,128	176/176
44	12D	229,072	228,304 to 229,840	176/176
45	13A	230,784	230,016 to 231,552	176/176
46	13B	232,496	231,728 to 233,264	176/176
47	13C	234,208	233,440 to 234,976	176/32
48	13D	235,776	235,008 to 236,544	32/176
49	13E	237,488	236,720 to 238,256	176/176
50	13F	239,200	238,432 to 239,968	176/32
L-Band: 1452,0 to 1467,5 MHz				
51	LA	1452,960	1452,192 to 1453,728	192/176
52	LB	1454,672	1453,904 to 1455,440	176/176
53	LC	1456,384	1455,616 to 1457,152	176/176
54	LD	1458,096	1457,328 to 1458,864	176/176
55	LE	1459,808	1459,040 to 1460,576	176/176
56	LF	1461,520	1460,752 to 1462,288	176/176
57	LG	1463,232	1462,464 to 1464,000	176/176
58	LH	1464,944	1464,176 to 1465,712	176/176
59	LI	1466,656	1465,888 to 1467,424	176/–

The remaining frequency range in L-Band (1467,5 MHz to 1492 MHz) was now recommended for DAB satellite applications (S-DAB), see Table A2.2.

A slightly different frequency scheme for the L-Band was recommended for use in Canada (see Table A2.3), which is now also foreseen for use in Australia. These channels have been numbered separately from 1 to 23.

Table A2.2 Recommended centre frequencies for S-DAB in Europe according to [EN 50248]

T-DAB Block number	T-DAB Block label	Centre frequency (MHz)	Block corner frequencies (MHz)
L-Band: 1452,0 to 1467,5 MHz			
60	LJ	1468,368	1476,600 to 1469,136
61	LK	1470,080	1469,312 to 1470,848

(*continued overleaf*)

Table A2.2 (*continued*)

T-DAB Block number	T-DAB Block label	Centre frequency (MHz)	Block corner frequencies (MHz)
62	LL	1471,792	1471,024 to 1472,560
63	LM	1473,504	1472,736 to 1474,272
64	LN	1475,216	1474,448 to 1475,984
65	LO	1476,928	1476,160 to 1477,696
66	LP	1478,640	1477,872 to 1479,408
67	LQ	1480,352	1479,584 to 1481,120
68	LR	1482,064	1481,296 to 1482,832
69	LS	1483,776	1483,008 to 1484,544
70	LT	1485,488	1484,720 to 1486,256
71	LU	1487,200	1486,432 to 1487,968
72	LV	1488,912	1488,144 to 1489,680
73	LW	1490,624	1489,856 to 1491,392

Table A2.3 Recommended centre frequencies for DRB in Canada (the Eureka 147 DAB system in Canada is officially designed as DRB = Digital Radio Broadcasting) according to [EN 50248]

Canadian DRB Channel number	Centre frequency (MHz)
L-Band:	1452,0 to 1467,5 MHz
1	1452,816
2	1454,560
3	1456,304
4	1458,048
5	1459,792
6	1461,536
7	1463,280
8	1465,024
9	1466,768
10	1468,512
11	1470,256
12	1472,000
13	1473,744
14	1475,488
15	1477,232
16	1478,976
17	1480,720
18	1482,464
19	1484,208
20	1485,952
21	1487,696
22	1489,440
23	1491,184

(NOTE 2: These frequencies are also valid for Australia)

Table A2.4 Recommended centre frequencies for DAB / DMB in Korea according to [WorldDMB Website]

Block Label	Centre frequency (MHz)
ROK 7A	175.280
ROK 7B	177.008
ROK 7C	178.736
ROK 8A	181.280
ROK 8B	183.008
ROK 8C	184.736
ROK 9A	187.280
ROK 9B	189.008
ROK 9C	190.736
ROK 10A	193.280
ROK 10B	195.008
ROK 10C	196.736
ROK 11A	199.280
ROK 11B	201.008
ROK 11C	202.736
ROK 12A	205.280
ROK 12B	207.008
ROK 12C	208.736
ROK 13A	211.280
ROK 13B	213.008
ROK 13C	214.736

Appendix 3

DAB System Protocol Stack

The following figure called DAB System Protocol Stack (taken from [Herrmann, 2007]) shows all relevant audio, video and data transmission protocols and applications for the DAB system family (DAB, DAB+, DMB) services in a compressed overview. The detailed functions and requirements are described in the related chapters as indicated in the figure.

Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7

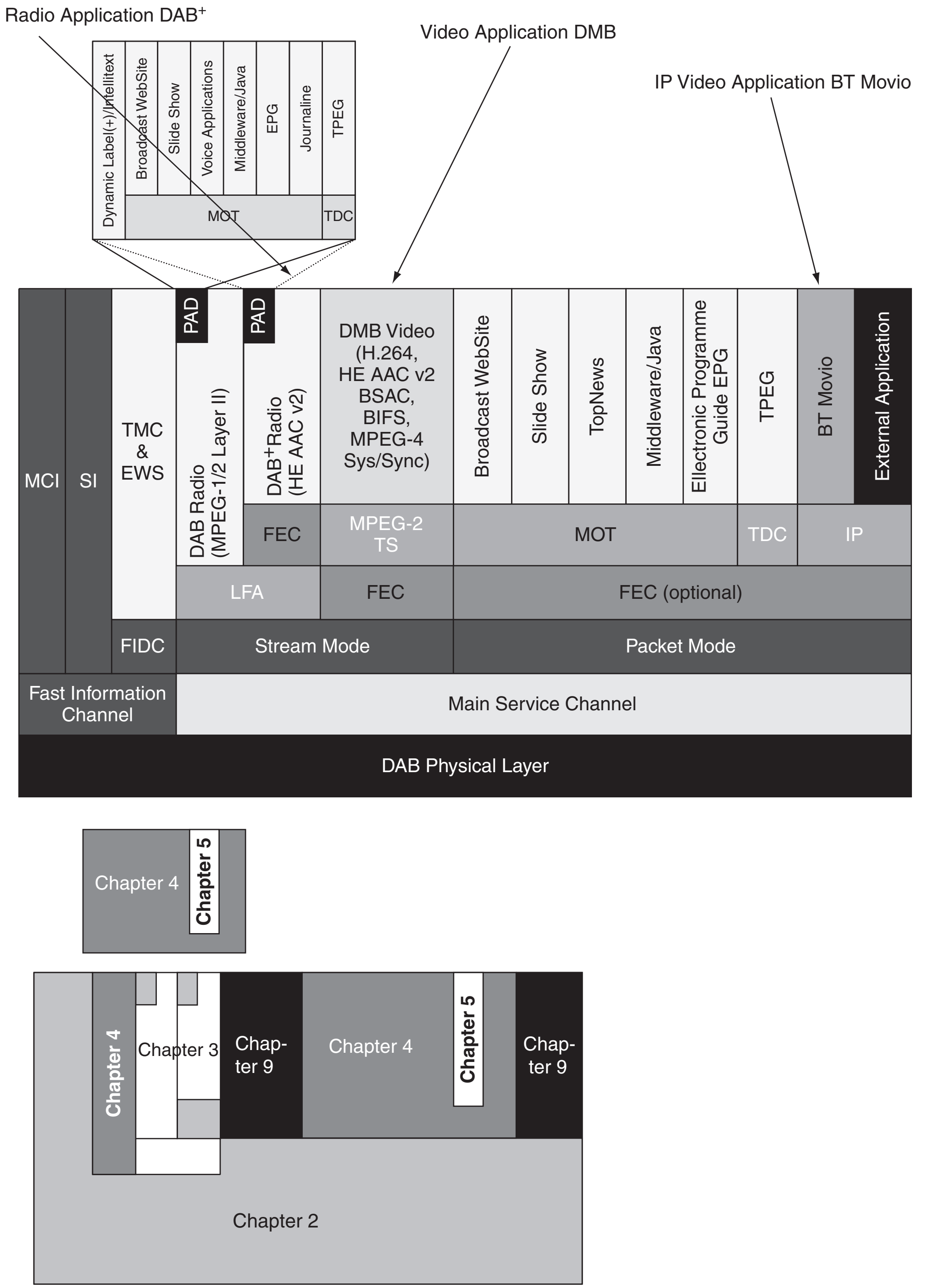

Figure 1 DAB Protocol Stack

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Key

BO.	ITU-R Recommendation (Satellite sound and television broadcasting)
BPN	EBU: Internal Technical Document (informative)
BR.	ITU-R Recommendation (Sound and television recording)
BS.	ITU-R: Recommendation (Sound broadcasting)
EN	ETSI or CENELEC European Telecommunication Standard (normative)
ES	ETSI: European Standard (normative)
IS	ISO/IEC: International Standard (normative)
P.	ITU-R Recommendation (Radiowave propagation)
R	EBU: Technical Recommendation
Tech	EBU: Technical Document (informative)
TR	ETSI: Technical Report (informative)
TS	ETSI: Technical Specification (normative)

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Internet Links

Note: *The reader should be aware that any reference to an Internet web-page will be of informal character only. The content of the cited web-page and/or the URL itself may be subject to change.*

[www.ASTRA] Web-site of SES-ASTRA Digital Satellite Radio. URL:http://www.ses-astra.com

[www.atdi] Web-site of ATDI – Software solutions in radiocommunications. URL:http://www.atdi.com

[www.dibeg] Web-site of DiBEG, the Japanese Digital Broadcast Experts Group URL:http://www.dibeg.org

[www.DRM] Web-site of Digital Radio Mondiale. URL:http://www.drm.org

[www.DVB] Web-site of the Digital Video Broadcasting (DVB) Project URL:http://www.dvb.org

[www.ETSI] Web-site of ETSI (European Telecommunications Standards Institute) URL:http://www.etsi.org

[www.HECA] Web-Site of Fraunhofer Institut für Integrierte Schaltungen IIS, with details of the conditional access system HECA. URL:http://www.iis.fraunhofer.de/bf/db/proj/heca.jsp

[www.iBiquity] Web-site of iBiquity Digital Corporation URL:http://www.iBiquity.org

[www.irt] Web-site of IRT (Institut für Rundfunktechnik) GmbH, who offer the network planning tool FRANSY. URL:http://www.irt.de

[www.lstelcom] Web-site of LS telcom AG, who offer the broadcast planning tool CHIR+. URL:http://www.lstelcom.de

[www.Mobil-info] Web-site of Mobil-Info service. URL:http://www.mobil-info.org

[www.NIST] Web-site of the National Institute of Standards and Technology, Computer Security Resource Center. URL:http://csrc.nist.gov

[www.Siriusradio] Web-site of Sirius Satellite Radio, Inc. URL:http://www.siriusradio.com

[www.Wiley] Web-site of the Wiley InterScience Home Page – the gateway to the online edition of the book 'Digital Audio Broadcasting: Principles and Applications. URL:http://www3.interscience.wiley.com

[www.WorldDAB] Web-site of the WorldDAB Organisation, London, U.K. URL:http://www.worlddab.org

[www.WorldDMB] Web-site of WorldDMB Forum, URL:http://www.WorldDAB.org

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Digital Audio Broadcasting: Principles and Applications of DAB, DAB+ and DMB, Third Edition Edited by W. Hoeg and T. Lauterbach
 ISBN: 978-0-470-51037-7