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Foreword

This Technical Report (TR) has been produced by Joint Technical Committee (JTC) Broadcast of the European Broadcasting Union (EBU), Comité Européen de Normalisation ELECtrotechnique (CENELEC) and the European Telecommunications Standards Institute (ETSI).

NOTE 1: The EBU/ETSI JTC Broadcast was established in 1990 to co-ordinate the drafting of standards in the specific field of broadcasting and related fields. Since 1995 the JTC Broadcast became a tripartite body by including in the Memorandum of Understanding also CENELEC, which is responsible for the standardization of radio and television receivers. The EBU is a professional association of broadcasting organizations whose work includes the co-ordination of its members' activities in the technical, legal, programme-making and programme-exchange domains. The EBU has active members in about 60 countries in the European broadcasting area; its headquarters is in Geneva.

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The Eureka Project 147 was established in 1987, with funding from the European Commission, to develop a system for the broadcasting of audio and data to fixed, portable or mobile receivers. Their work resulted in the publication of European Standard, EN 300 401 [1], for DAB (see note 2) which now has worldwide acceptance. The members of the Eureka Project 147 are drawn from broadcasting organizations and telecommunication providers together with companies from the professional and consumer electronics industry.

NOTE 2: DAB is a registered trademark owned by one of the Eureka Project 147 partners.

The present document is part 3 of a multi-part deliverable covering Guidelines and rules for implementation and operation for Digital Audio Broadcasting (DAB), as identified below:

Part 1: "System outline";

Part 2: "System features";

Part 3: "Broadcast network".

1 Scope

The present document is Part 3 of the Guidelines and Rules of Implementation and Operation for the Digital Audio Broadcasting (DAB) system. It focuses on the broadcast network. The guidelines have been developed by the Eureka Project 147 as the major companion document to the DAB system specification given in EN 300 401 [1] . They are intended to provide additional information to aid interpretation of the on-air signal and to assist broadcasters and manufacturers to implement systems using the specification features as intended. TR 101 496-1 [8] focuses on the system outline and TR 101 496-2 [9] gives a detailed description of the system features.

2 References

For the purposes of this Technical Report (TR), the following references apply:

[1] ETSI EN 300 401: "Radio Broadcasting Systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers".

[2] ETSI EN 300 797: "Digital Audio Broadcasting (DAB); Distribution interfaces; Service Transport Interface (STI)".

[3] ETSI ETS 300 799: "Digital Audio Broadcasting (DAB); Distribution interfaces; Ensemble Transport Interface (ETI)".

[4] ETSI EN 300 798: "Digital Audio Broadcasting (DAB); Distribution interfaces; Digital baseband In-phase and Quadrature (DIQ) Interface".

[5] ETSI EN 301 234: "Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) protocol".

[6] ETSI TR 101 758: "Digital Audio Broadcasting (DAB); Signal strengths and receiver parameters; Targets for typical operation".

[7] ITU-T Recommendation G.704: "Synchronous frame structures used at 1544, 6312, 2048, 8448 and 44 736 kbit/s hierarchical levels".

[8] ETSI TR 101 496-1: "Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 1: System outline".

[9] ETSI TR 101 496-2: "Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 2: System features".

[10] ITU-T Recommendation G.703: "Physical/electrical characteristics of hierarchical digital interfaces".

[11] ISO/IEC 11172-3: "Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s - Part 3: Audio".

[12] EN 50248: "Characteristics of DAB receivers".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

Access Control System (ACS): particular set of rules for managing entitlement checking and conditional access messages

audio bit stream: sequence of consecutive audio frames

audio mode: audio coding system provides single channel, dual channel, stereo and joint stereo audio modes
In each mode, the complete audio signal is encoded as one audio bit stream.

Capacity Unit (CU): smallest addressable unit (64 bits) of the Common Interleaved Frame (CIF)

Common Interleaved Frame (CIF): serial digital output from the main service multiplexer which is contained in the Main Service Channel part of the transmission frame
It is common to all transmission modes and contains 55 296 bits (i.e. 864 CUs).

convolutional coding: coding procedure which generates redundancy in the transmitted data stream in order to provide ruggedness against transmission distortions

DAB audio frame: same as audio frame, but includes all specific DAB audio-related information

DAB transmission signal: transmitted radio frequency signal

ensemble: transmitted signal, comprising a set of regularly and closely-spaced orthogonal carriers
The ensemble is the entity which is received and processed. In general, it contains programme and data services.

Equal Error Protection (EEP): error protection procedure which ensure a constant protection of the bit stream

Extended Programme Associated Data (X-PAD): extended part of the PAD carried towards the end of the DAB audio frame, immediately before the Scale Factor Cyclic Redundancy Check (CRC)
Its length is variable.

Fast Information Block (FIB): data burst of 256 bits
The sequence of FIBs is carried by the Fast Information Channel. The structure of the FIB is common to all transmission modes.

Fast Information Channel (FIC): part of the transmission frame, comprising the Fast Information Blocks, which contains the multiplex configuration information together with optional service Information and data service components

Fast Information Group (FIG): package of data used for one application in the Fast Information Channel
Eight different types are available to provide a classification of the applications.

Fixed Programme Associated Data (F-PAD): fixed part of the PAD contained in the last two bytes of the DAB audio frame

logical frame: data burst, contributing to the contents of a sub-channel, during a time interval of 24 ms
For example, data bursts at the output of an audio encoder, a Conditional Access scrambler and a convolutional encoder are referred to as logical frames. The number of bits contained in a specific logical frame depends on the stage in the encoding process and the bit rate associated with the sub-channel.

Main Service Channel (MSC): channel which occupies the major part of the transmission frame and which carries all the digital audio service components, together with possible supporting and additional data service components

Multiplex Configuration Information (MCI): information defining the configuration of the multiplex
It contains the current (and in the case of an imminent re-configuration, the forthcoming) details about the services, service components and sub-channels and the linking between these objects. It is carried in the FIC in order that a receiver may interpret this information in advance of the service components carried in the Main Service Channel. It also includes identification of the ensemble itself and a date and time marker.

null symbol: first Orthogonal Frequency Division Multiplex (OFDM) symbol of the transmission frame

OFDM symbol: transmitted signal for that portion of time when the modulating phase state is held constant on each of the equi-spaced, equal amplitude carriers in the ensemble
Each carrier is four-phase differentially modulated from one symbol to another, giving a gross bit rate of two bits per carrier per symbol.

Programme Associated Data (PAD): information which is related to the audio data in terms of contents and synchronization
The PAD field is located at the end of the DAB audio frame.

protection level: level specifying the degree of protection, provided by the convolutional coding, against transmission errors

protection profile: defines the scheme of convolutional coding applied

service: user-selectable output which can be either a programme service or a data service

service component: part of a service which carries either audio (including PAD) or data
The service components of a given service are linked together by the Multiplex Configuration Information. Each service component is carried either in a sub-channel or in the Fast Information Data Channel.

Service Identifier (SId): 16- or 32-bit code used to identify a particular service

Service Information (SI): auxiliary information about services, such as service labels and programme type codes

Single Frequency Network (SFN): network of DAB transmitters sharing the same radio frequency to achieve a large area coverage

synchronization channel: part of the transmission frame providing a phase reference

transmission frame: actual transmitted frame, specific to the four transmission modes, conveying the Synchronization channel, the Fast Information Channel and the Main Service Channel

transmission mode: specific set of transmission parameters (e.g. number of carriers, OFDM symbol duration)
Four transmission modes (i.e. I, II, III and IV) are defined to allow the system to be used for different network configurations and a range of operating frequencies.

Unequal Error Protection (UEP): error protection procedure which allows the bit error characteristics to be matched with the bit error sensitivity of the different parts of the bit stream

X-PAD data group: package of data used for one application in the Extended Programme Associated Data (X-PAD)

3.2 Symbols

For the purposes of the present document, the following symbols apply:

c	velocity of light
C/N	ratio of power of carrier to that of the noise
F	inter-carrier distance
$f$max	maximum recommended frequency for a given speed of vehicle (for good reception of DAB)
f_o	frequency in Hz
G(x)	polynomial generator of the cyclic redundancy code
L	length of the data coded with a given error protection profile
R	code Rate for convolutional code
S/N	ratio of the power in the signal to that of the noise in the same bandwidth
T	symbol duration
T_s	complete symbol period
T_u	duration of the useful part of the symbol
v	vehicle speed in m/s
β	as the representation of the displacement of the vehicle expressed in number of wavelengths during one symbol duration
Δ	length of the guard interval
λ	wavelength of the signal
τ_m	maximum delay beyond which the addition of delayed signals causes degradation

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AES/EBU	audio interface created by the Audio Engineering Society and the European Broadcasting Union
AFC	Automatic Frequency Control
AM	Amplitude Modulation
BAI	Bit Allocation Information
BBC	British Broadcasting Corporation
BER	Bit Error Ratio

C/N	Carrier-To-Noise Ratio
CCETT	Centre Commun d'Études de Télédiffusion et Télécommunications
CI	Contents Indicator
COFDM	Coded Orthogonal Frequency Division Multiplex
CRC	Cyclic Redundancy Code
CU	Capacity Unit
DAB	Digital Audio Broadcasting
DIQ	Digital In-phase and Quadrature - the baseband representation of a DAB signal
EB	Errored Block
EF	Errored Frame
EMux	Ensemble Multiplexer
ETI	Ensemble Transport Interface
FIB	Fast Information Block
FIC	Fast Information Channel
FM	Frequency Modulation
GPS	Global Positioning System
HEO	Highly inclined Elliptical Orbit
IFFT	Inverse Fast Fourier Transform
ISO	International Organization for Standardization
ITTS	Interactive Text Transmission System
ITU	International Telecommunications Union
LSF	Lower Sampling Frequency
LTO	Local Time Offset
MCI	Multiplex Configuration Information
MFN	Multi Frequency Network
MNSC	Multiplex Network Sub Channel
MPEG	Moving Picture Experts Groups
MSC	Main Service Channel
NA	Network Adapted
NI	Network Independent
OBO	Output Back Off
OFDM	Orthogonal Frequency Division Multiplex
PAD	Programme Associated Data
PDH	Plesiochronous Digital Hierarchy
PM	Phase Modulation
PTy	Programme Type
ScF	Scale Factor
SDH	Synchronous Digital Hierarchy
SEF	Severely errored Frame
SFN	Single Frequency Network
SI	Service Information
SSPA	Solid State Power Amplifier
TII	Transmitter Identification Information
TMC	Traffic Message Channel
TPEG	Transport Protocol Experts Group
UEP	Unequal Error Protection
UF	Unavailable Frame
US	Unavailable Second
VHF	Very High Frequency
X-PAD	Extended Programme Associated Data

4 Implementation and Operation of the DAB Broadcast Network

4.1 Introduction

This clause of the document gives an overview of the principles which should be considered and applied when planning the implementation of a DAB [1] Broadcast Network. In the present document, the DAB Broadcast Network is taken to encompass all of equipment between the audio coders (or data source equipment in the case of a data service) located at the studio centre (or data origination point) and the input to the DAB receiver.

A conceptual picture of the DAB Broadcast Network from source coders to transmitters is introduced. Each of the elements of the conceptual network is analysed and some of the strategies which could be employed for signal distribution in the different parts of the Network are introduced. The clause concludes with some illustrative examples of Broadcast Network implementation.

4.2 The conceptual DAB Broadcast Network

4.2.1 Introduction

This clause proposes a conceptual DAB Broadcast Network. This extends from the source coders (associated with each individual service) to the transmitted COFDM signal, the Ensemble. The Ensemble carries a multiplex of services, known as the Ensemble multiplex.

4.2.2 The Conceptual Network

Figure 4.2.1 shows the conceptual network in diagrammatic form. The network is envisaged as a three stage process where each stage is managed by a different entity. The three management entities are: the Service provider, the Ensemble provider and the Transmitter Network provider.

The Service provider is concerned with building a part of the multi-service Ensemble multiplex. Typically, this would be an individual service (or service component), though it could extend to a number of services. In a typical DAB network there will be many Service providers, each associated with a set of one or more of the service components. Each service is itself a multiplex of data. For example, an audio service consists of coded audio data, Programme Associated Data, and additional Service Information supporting that particular component.

The Ensemble provider collects together all of the data sets describing the individual service components. Additional, ensemble related Service Information (such as the Multiplex Configuration Information) is added and a data set representing a complete Ensemble Multiplex is built. In general there will only be one Ensemble provider for each transmitted ensemble.

The Transmission Network provider takes the data representing a full Ensemble Multiplex and turns this into the transmitted signal at one or, more typically, many transmitter sites. In this final stage, the data which identifies uniquely each transmitter in the network (Transmitter Identification Information) must be added if required.

As may be seen from the above description, the building of the Ensemble Multiplex is a multi-stage process where data is originated at many points in the network and added to the full Multiplex in stages. Nevertheless, data flow is unidirectional from Service provider, through Ensemble provider and on to the Transmitter Network provider.

Figure 4.2.1 also shows the flow of control information in the Network. Since the Ensemble provider looks after the construction of the complete Ensemble Multiplex, then control information is likely to be required to flow from the Ensemble provider to all Service providers and to the Transmission Network provider. There will also be a requirement for control information to flow from the Service providers back to the Ensemble provider and between different Ensemble providers (to exchange information about other transmitted ensembles for instance). These principal control data flows are also illustrated in figure 4.2.1.

The lists at the bottom of the figure give examples of the type of data which is inserted at different points in the network and of control information which could flow in the network. The entries in the lists are located below the principal originator of the named data type but note that they are not intended to be definitive or exhaustive.

4.2.3 Network Interfaces

It is not necessary that the three stages in building the DAB signal are physically separate. In fact, life is probably a lot easier if they can be kept close to one another. However, in the typical situation, there will be many separate Service providers feeding their signals to the Ensemble provider and the Ensemble provider will be required to feed the aggregate data signal to many transmitters.

The interface between service and ensemble generation is shown in the diagram as the Service Transport Interface (STI). Its main function is to carry data relating to a particular service, or service component.

The interface between the Ensemble provider and the modulation process in the COFDM Generator belonging to the Transmission Network provider is shown as the Ensemble Transport Interface (ETI). Its main function is to carry data which relates to a full Ensemble Multiplex. The principal characteristics of both interfaces are explored in later clauses.

The fundamental difference between these two interfaces is that the STI carries service information in a raw form (i.e. not formatted into the structure defined for the DAB FIC channel). The ETI carries the service information in a formatted form (the form required for the FIC). At its simplest level, the conversion between STI and ETI could be seen merely as the process of formatting the FIC data.

Both the STI and the ETI have been standardized [2], [3] as has a third interface, the Baseband Digital I/Q (DIQ) interface [4]. Although not a distribution interface, the DIQ provides a convenient break-point in the transmitter between baseband digital processing and radio-frequency modulation equipment.

4.3 Building the DAB Signal

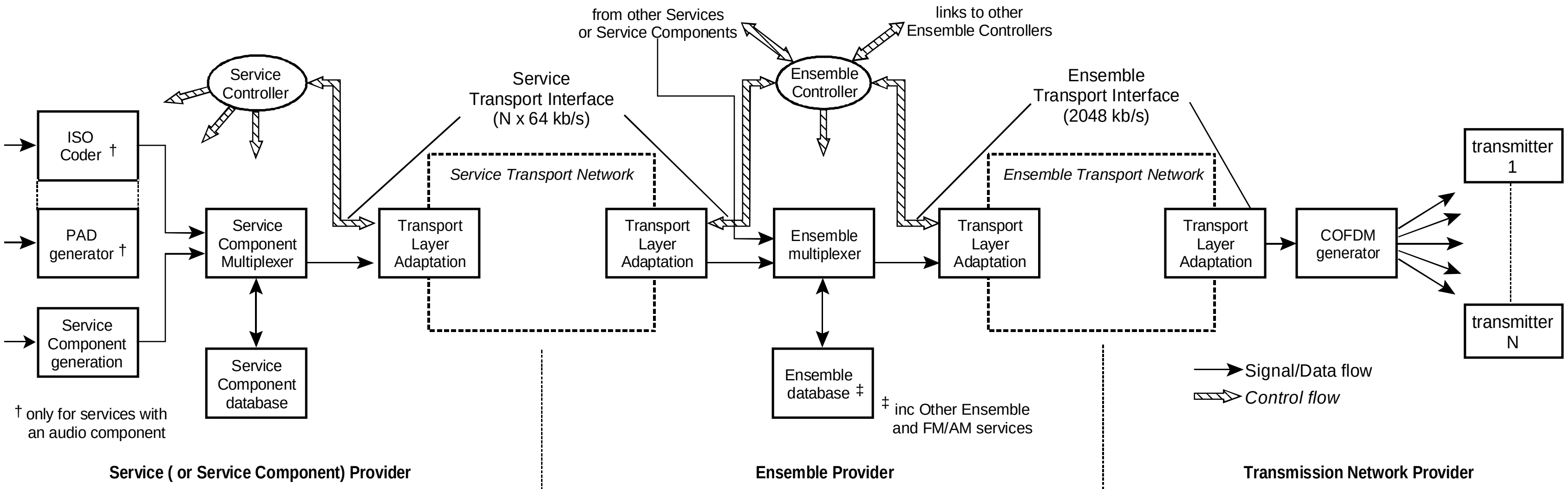
4.3.1 Introduction

This clause starts by taking a more detailed look at the elements of the conceptual Broadcast Network. This includes some aspects of the use of the Service and Ensemble Transport Interfaces. The concluding clause looks at some more general networking aspects including timing and synchronization as well as some of the considerations which apply when reconfiguring the Ensemble Multiplex.

4.3.2 The Service Provider

The basic building blocks of a DAB Ensemble Multiplex are service components. The role of the Service provider is to assemble a set of one or more service components, together with supporting information, for onward routing to the Ensemble provider. Some examples of service components are:

- an audio data flow (including the associated PAD); the audio data flow will generally be the main component of an audio service but could also be a secondary component;
- a text data flow;
- a TMC or TPEG data flow; this could be a primary component, or secondary component linked to one or more of the main DAB services;
- a packet data flow; DAB data services can be configured as a packet data channel which could itself be configured as a number of data service components.



Examples of Service Components and System Features (organised by likely insertion point)

Service Component (examples)

Audio flow
Paging service data
TMC

System features

(PAD)
Music/Speech indication
Dynamic Range Control
Command Channel
ISRC and UPC/EAN
Programme related text (ITTS)
In-house Information (PAD)
Table of contents
Dynamic Label
Closed user-group data

(ISO header)

Audio Mode
Copyright Control

(non PAD)
Service Identifier
Extended Country Code (per service)
Date and Time (service LTO)
Service Label
Service Component Language
Programme Number
Programme Type
Announcements
Service Trigger
Conditional Access
Foreground/Background sound

Ensemble Identifier
Emergency Warning Systems
Multiplex Configuration Information
Multiplex Re-configuration Timing Management (inc C/N flag)
Extended Country Code (general)
Date and Time (reference)
Ensemble Label
Frequency Information
Transmitter Identification Information Database
Auxiliary Information Channel
Other Ensembles
FM Services
In-house Information (FIC)
FIC re-direction
Fast Information Data Channel
Service Linking Information
Regional Identification
Local Service Area
Satellite Assistance

TII coding (null symbol)

Examples of Control Signal (organised by likely insertion point)

Multiplex re-configuration requests
Service status flags
Service component control (links, priorities etc)
Alarm Flag control
Emergency Warning System control
Service Trigger control
Announcement support

Multiplex re-configuration timing management
Schedule control
Status Information
Billing
Announcement signalling management (inc OE announcements)
Service Trigger management
Transmitter Control

DAB Mode *DAB frequency*

Figure 4.2.1: The conceptual DAB network

4.3.2.1 Source Coding for Audio Flows

For an audio service, source coding takes the form of an ISO/MPEG Layer II audio encoder in which the audio data is sampled at a frequency of 48 kHz for full-bandwidth audio or 24 kHz for audio with reduced bandwidth. The output of the encoder is data at the defined rate formatted into 24 (or 48) ms frames (see note). The input to the coder could be either an analogue audio signal or a digital connection which would usually take the form of an AES/EBU serial interface .

NOTE: In the ISO/MPEG standard, audio data sampled at 48 kHz results in 24 ms frames whereas audio signals sampled at 24 kHz result in 48 ms frames. In the DAB system, the 48 ms frames are treated as a pair of 24 ms frames.

Although based closely on ISO/MPEG Layer II standard frames, DAB audio frames contain a number of enhancements. These include additional checksums and provision for the inclusion of additional data, known as Programme Associated Data (PAD).

Since PAD information is intimately related to the audio signal and needs to be included in the associated audio frame then PAD insertion will take place in the audio encoder or in intimate association with it. One example of the implementation is an RS-232 connection on the audio encoder which provides an ISO-frame locked synchronizing output to trigger data input from an external PAD formatter. PAD formatters have been implemented using PC interface cards. Control of the formatting is then possible using custom software running on the PC. Alternative strategies (e.g. the use of unused capacity in an AES/EBU input) may also be possible.

Early audio coders for DAB were equipped with a WG1/WG2 output2 which requires the audio coders to be in close physical proximity to the DAB Multiplexer. More recently, audio coders have been produced with an STI output to permit the building of more diverse networks.

4.3.2.2 Source coding for Data Flows

For data services, the source coding can take many different forms depending on the nature of the particular service. In addition, appropriate transport protocols will need to be used for carriage of data services within a DAB ensemble. The most appropriate transport protocol will be determined by the nature of the application. The Multimedia Object Transfer protocol (MOT) is one example of a particular method for dealing with data services which may be employed for DAB [5].

4.3.2.3 Service Component Multiplexing

The Service Component Multiplexer (SCMux) is the heart of the Service provider's system. It accepts the output of the source coders (which could take the form of one or more audio coders or data formatters depending on the nature of the service) and multiplexes them, along with other data, to form the Service Transport Interface. The simplest form of an SCMux is, of course, an audio encoder with an STI output.

4.3.2.4 The Service Component Database

The SCMux also accepts the output of the Service Component Database which holds information about the DAB System Features which apply to this particular set of services. The data in the database may be static or dynamic depending on the nature of the data and services. Dynamic data could change under schedule control (i.e. changes take place under the control of a system clock) or could be triggered by external events. An example of the latter could be PTy codes which vary in conjunction with programme item changes.

4.3.2.5 The Service Controller

All of the elements of the Service provider operate under control of the Service Controller which also inserts control information into the STI (and accepts control information from the Ensemble Controller via the STI). The Controller deals with the normal scheduling of data (such as PNum and PTy for example) but could also be responsible for more fundamental changes such as those of the audio coding rate. Some of these changes will have an effect on other services, e.g. a reconfiguration in which a number of services are interchanging capacity. In such a situation the Service Controller of any particular service will need to operate in conjunction with other Service Controllers under control of the Ensemble Controller.

4.3.3 The Service Transport Interface

The STI [2] provides a convenient interface for carrying DAB service components, for example between an audio encoder and Service Component multiplexer or between the Service and Ensemble multiplexers. It could also be used as the interface between two Service Component multiplexers to allow services to be built up in a distributed fashion. The STI provides a transport mechanism for all DAB service components and service information as defined in EN 300 401. In addition, a control channel is also provided which may be used to manage, or monitor, the service components.

The STI uses a layered structure, comprising a Logical Interface and several physical implementations which may be Network Independent or Network Adapted.

The Logical Interface is the basic definition of the interface and defines the structures used to carry data and control information but has no physical manifestation. The Network Independent interfaces are the simplest physical manifestations of the STI and provide a simple transport framing structure. Network Adapted versions are more complex physical manifestations using more complex framing and complete with a degree of error protection. They are designed to cope with particular network structures (e.g. G.704).

A full description of the STI may be found in EN 300 797 [2].

4.3.4 Cascading of Service Provision

Although the conceptual model shows the SCMux (and associated equipment) as a single entity, it could be necessary in some instances for the Service provider to operate in a distributed fashion. In this case the output of one level of Service Provision (the STI) is followed by another level of Service Multiplexing rather than the Ensemble Multiplexer. In this situation, the STI is used as an input interface to an SCMux as well as an output interface.

4.3.5 The Ensemble Provider

The Ensemble provider manages the full capacity of at least one DAB Ensemble multiplex. A single Multiplex can have up to 64 sub-channels which could each carry a service or service component. The role of the Ensemble provider includes:

- accepting sub-channel information, and associated control information, from the Service providers and re-formatting these inputs to build the Ensemble Transport Interface,
- accepting service-related System Feature data from the Service provider and formatting these to make appropriate FIC information for inclusion within the ETI,
- adding ensemble-related System Feature data (for this and other ensembles or transmissions) to the FIC information. Figure 4.2.1 lists some of the currently defined System Features which could be required to be inserted at the level of the EMux. Note, however, that the list could differ in different implementations,
- managing the Ensemble Multiplex capacity including the generation of the MCI. This includes the management of the Service Controllers associated with each service.

4.3.5.1 Ensemble Multiplexing

The heart of the DAB network is the Ensemble Multiplexer (EMux). It accepts the service data from one or more SCMux and uses it to generate all of the common (see note) component parts of the DAB Ensemble Multiplex. The output of the EMux is a data signal which describes, uniquely, a DAB ensemble and this may then be connected to a COFDM generator which produces the modulated signal.

NOTE: In this context, the term "common" is used to mean the various parts of an Ensemble Multiplex which are common to a number of transmitters. Usually, all the component parts of a DAB signal are common - with the sole exception of the TII.

The input to the EMux is characterized by many data links whose main task is to carry information about services, or service components, to the EMux.

The output of the EMux is an interface signal which contains all the information necessary to generate the radiated COFDM signal at a given transmitter, or set of transmitters. In general, the output of the EMux is a single interface which is fed, in parallel, to many destinations.

4.3.5.2 The Ensemble Database

The EMux also accepts the output of the Ensemble Database which holds the DAB System Feature information which applies to this particular ensemble and related information. The data in the database may be static or dynamic depending on the nature of the data and the status of service components etc. Dynamic data could change under schedule control (i.e. changes take place under the control of a system clock) or could be triggered by external events (for example, a service changes from one having an FM alternative to one without).

4.3.5.3 The Ensemble Controller

The Ensemble Controller is responsible for controlling the action of the EMux, including the control of scheduled configuration changes for instance. It is also responsible for the overall management of the ensemble's configuration and for co-ordinating any changes in service status - and resolving any conflicting demands!

4.3.6 The Ensemble Transport Interface

The ETI is used to carry information about a full, or partial, ensemble between Ensemble multiplexers, or (in the case of a full ensemble) from Ensemble multiplexer to COFDM Generator. It is distinguished from the STI by the fact that it carries the service information formatted in the DAB FIC format and the control requirements are much simpler.

The ETI is defined in a European standard [3] which gives full details of the interface and describes its use.

In a similar manner to the STI, the ETI is defined in a number of layers: a Logical layer and Network Independent and Network Adapted forms. The most commonly used form of the ETI is a 2 Mbit/s G.703 interface, ETI(NI, G703) [3]. In this form it is only suitable for use on simple local connections or data links with relatively straightforward characteristics. A Network Adapted version, ETI(NA, G704), suitable for 2 Mbit/s G.704 connections, is also defined. This is generally more useful as it is more robust in the presence of link errors and contains information to control Network delay variations. This becomes important, for example, when feeding a Single Frequency Network using a switched terrestrial transport network.

Detailed information on the structure of the ETI can be found in ETS 300 799 [3]. The following clauses give some general guidance on the use of the ETI.

4.3.6.1 Using the ETI

ETI(NI, G703) is a simple form of the ETI which may be used for a direct connection or connection via a relatively simple network. Its electrical characteristics conform to those defined in ITU-T Recommendation G.703 [10]. It contains rudimentary error checks which permit integrity checking but does not allow for any error correction. In addition, there is no mechanism for coping with changing Network delays and the long frame structure (24 ms for audio samples at 48 kHz, or 48 ms for audio sampled at 24 kHz) is rather weak in the presence of errors. Nevertheless, the ETI(NI, G703) could be used on a satellite connection where protection against errors is provided within the modulation and demodulation equipment. The time delays in such a Network are known with sufficient precision so that dynamic delay correction is not required.

ETI(NA, G704), is an adaptation of the interface for use on terrestrial switched G.704, 2 Mbit/s networks. An error correcting mechanism is included together with a much shorter frame structure. In addition, provision is made for time stamping of data so that the timing variations on the network can be corrected. In this latter case, it is of course necessary that the send and receiving units maintain "a sense of time", i.e. a common time reference must be available at both ends of the Ensemble Transport Network. Some current implementations use GPS-derived clocks for this purpose.

The time-stamps carried in the Network Adapted ETI also allow for "seamless-switching" between multiple feeds of the ETI to a transmitter. This would typically be done to improve the reliability of the DAB network. The separate feeds can be time-aligned independently, using the time-stamps. Switching between the separate feeds can then be accomplished without any loss of data.

4.3.6.2 ETI Capacity

The capacity required for the ETI is a function of the number of services and the capacity of each service before coding is applied. In general, a 2 Mbit/s circuit provides ample capacity even allowing for the overheads required for framing, error correction etc. Note, however, that in some circumstances a capacity greater than that allowed by a 2 Mbit/s circuit is required. Alternative versions of the ETI must be used in this case.

ETS 300 799 [3] gives a detailed treatment of how to calculate the ETI capacity requirement.

4.3.6.3 Ensemble Transport Network Performance

This clause attempts to set performance targets for the behaviour of the Ensemble Transport Network. The text of this clause is provisional.

The performance is defined in terms of the behaviour of the network from the output of the Ensemble Multiplexer (*before* any network adaptation) to the input of the relevant COFDM generator (*after* any relevant network adaptation). In other words, the performance is assessed by reference to Network Independent versions of the ETI. For a simple point-to-point connection, the characteristics to be considered are the Network Transit Time (mean and variances) and the Error Performance. Additionally, for a point-to-multi-point connection (as used to feed a SFN) the Differential Transit Time (mean and variances) must also be considered.

In order to assist with the definition of these characteristics, some preliminary definitions are necessary. The ETI comprises 24 ms frames. Each frame is assumed to consist of 24 blocks (giving 1 000 blocks per second) with 1 920 bits in each block (see note).

NOTE: Each frame thus has 5 760 bytes which are made up of data plus framing overhead etc. These are the bytes which are mapped into one of the Network Adapted versions of the ETI.

We define:

- a Delay Slip as a change in Network Transit Time from one frame to the next of more than 50 % of the DAB Guard Interval for the DAB Transmission mode in use,
- an Errored Block (EB) to be a block with at least one errored bit,
- a Severely Errored Block (SEB) to be a block with at least 8 errored bits,
- an Errored Frame (EF) to be a frame with at least one EB,
- a Severely Errored Frame (SEF) to be a frame with at least 5 SEB,
- an Unavailable Frame (UF) to be a frame with at least 9 SEB,
- an Unavailable Second (US) to be a frame with at least 1 SEF (or at least 1 UF).

The Network is considered Unavailable if frame synchronization is lost, or more than 10 SEF were received in the last 40. The channel becomes Available as soon as frame synchronization is achieved for more than 40 consecutive frames. Note that reference [2] defines the method to be adopted for frame synchronization.

Performance objectives can now be outlined:

1) Network Transit Time (Mean): the mean Network Transit Time should be fixed and known with an accuracy of ±1µs. The mean Transit Time is measured over a period of 1 month, neglecting the effect of Delay Slips caused by Network effects. The target performance for Delay Slips is fewer than 1 Delay Slip per month.
2) Network Transit Time (Variance): the variance in the Network Transit Time must not cause the jitter and wander on the received 2 Mbit/s signal to exceed the limits given in [2].
3) Error Objectives: the Error Objectives are set on the assumption that an error of a few bits in the transmission of the ETI, although giving rise to an incorrectly modulated signal, does not give rise to significant degradation of the received signal. Badly corrupted frames, however, are likely to have severe consequences. The targets are presented in table 4.3.1.

Table 4.3.1: Error Performance Objectives

Classification	Target
EF	< 1/minute
SEF	< 1/hour
UF	< 1/day
US	< 1/month

4) Network Unavailability: The Network should be Unavailable less than once per year.

5) Differential Transit Time (Mean): The Differential Transit Time between the ETI signals received at any two COFDM generators should be substantially less than 10 % of the DAB Guard Interval of the DAB Transmission mode in use.

6) Differential Transit Time (Variance): Performance target to be defined.

4.3.6.4 Signalling in the ETI

The ETI(NI) layer contains a signalling channel which may be used for signalling information between the EMux (or the Ensemble Controller) and the COFDM generator, or between cascaded EMuxes. This is referred to as the Multiplex Network Service Channel (MNSC).

The MNSC carries 16 bits per frame, corresponding to a data rate of 666,7 bits/sec. The structure of this channel is defined in ETS 300 799 [3]. Signalling is possible in two different modes; Frame Synchronous or Asynchronous.

Frame Synchronous signalling carries information which is relevant to the containing frame (or frames). It is used, for instance, to carry time information between the different levels of Ensemble Multiplexing (see clause 4.3.7).

Asynchronous signalling carries information which is not linked to particular frames of the interface and could carry, as an example, information about forthcoming changes to the configuration of an Ensemble Multiplex. Again, this could be useful with cascaded Ensemble Multiplexers.

Both signalling protocols allow user defined functions to be implemented to permit tailored systems to be built. One example of a user defined function could be the control of COFDM generator parameters (such as time delay or TII code) from a remote terminal, see ETS 300 799 [3]. Other transmitter control functions could also be implemented.

In addition to the MNSC, since the ETI(NA, G704) corresponds to the G.704 framing structure, time slot 16 in every frame is available for signalling information. This time slot is free for user applications, see ITU-T Recommendation G.704 [7].

4.3.6.5 Monitoring in the ETI

The ETI carries CRC checksums which allow for data integrity checking. Separate CRC checks are used for header and data fields. This allows different strategies to be used when errors occur in the separate parts of the ETI. For instance, errors in the header field could be mitigated by assuming that the header information is unlikely to change from one frame to the next. Data errors could be ignored in isolated frames but some action may be required if data errors occur frequently.

The ETI(NA, G704) corresponds to the G.704 framing rules and standard G.704 monitoring techniques may be used in addition to the monitoring provided at the NI interface. This could include the use of CRC-4 [7].

4.3.6.6 Use of time-stamps

In order that the ETI receiver can restore a consistent network transit time, information about signal timing must be included in the transmitted ETI. For this reason, timestamps are included within the ETI. Detailed information on the coding and use of the timestamps can be found in ETS 300 799 [3].

4.3.7 Cascading of Ensemble Provision

Although the conceptual model shows the EMux (and associated equipment) as a single entity it may be necessary in some instances for the Ensemble provider to operate in a distributed fashion. For instance, at the first level a partial Ensemble Multiplex consisting of a common sub-set of national services could be built. This would be distributed to a second level of Ensemble Multiplexing which adds local variants of the remaining services. Such an architecture requires the use of a multi-frequency network, MFN.

In this case the output of one level of Ensemble Provision (the ETI) is followed by another level of Ensemble Multiplexing rather than the COFDM generator. In such circumstances, the ETI must be capable of operating as an input interface to an EMux as well as its output interface.

Signalling between the cascaded layers of Ensemble Multiplexing can use the MNSC field defined in the ETS 300 799 [3].

Timestamps are included in the basic definition of the ETI and a further timestamp is included at the Network Adapted layer. In a network using cascaded multiplexers, the latter may be used to control transit delay in a clause of the network, ensuring seamless switching between a main and reserve feed for instance. The former may be used to manage the overall delay of the cascaded network. This is particularly relevant where, as noted above, cascaded multiplexers are used to provide a mixture of national and local services in a MFN, where it is desirable to ensure co-timing of the national components. The first multiplexer acts as a "time-reference multiplexer" and generates the basic timestamp which may be used by the final multiplexers in the cascade to ensure that the delay through the complete multiplex structure can be controlled. In this case, all the multiplexers must maintain the relationship between the Frame Count (FCT) field (see [3]) and the timestamp (TIST) field.

4.3.8 The Transmission Network Provider

The Transmission Network provider is responsible for building the COFDM signal and for the transmission of this signal from a single transmitter or a network of transmitters.

4.3.8.1 Signal Distribution in the Transmission Network

The choice of a suitable distribution signal to feed the distant transmitters will be made largely on economic considerations.

For operational networks, by far the best choice is the use of the ETI either in Network Independent or Network Adapted form. This 2 Mbit/s signal may be carried relatively easily using standard techniques. It is the most efficient and flexible method of carrying the signal, and all known operational networks use this technique.

However, use of the ETI has the disadvantage that a COFDM generator is required at each transmitter site. If only a small number of transmitters are required, for example in experimental networks, then this may not offer the cheapest solution depending on the balance of circuit and equipment costs. Two other techniques are possible:

1) the modulated signal may be produced at a low intermediate frequency (in the vision band) and distributed to the transmitters using vision circuits. This is referred to as the "pseudo-video" method. A number of ensembles could be carried by a single vision circuit by using a different centre frequency for each. All that is required at the transmitter is a frequency converter, which leads to minimum transmitter cost.

   Disadvantages of this method include:

   - high circuit costs; this method cannot be recommended for anything other than feeding a very small numbers of transmitters;
   - in a single frequency network a pilot-tone is usually required, again located within the vision pass-band, to synchronize the frequency conversions at each transmitter;
   - the relative timing of transmitters is dictated by the circuit delays;
   - TII information must also be keyed into the signal generated at each transmitter and no practical method has been demonstrated for achieving this.

2) the modulated signal could be produced at any other frequency which is available for distribution (in the UHF or SHF bands for instance) and frequency converted at the transmitter sites. This is the technique employed for many of the experimental transmissions but is usually prohibitively expensive when serving many transmitters, even where the frequencies are available. In a SFN, a method of locking the frequency converters must be devised. The transmission of additional tones has usually been used in experimental work. The same limitations raised in 1) above apply to the management of transmitter timing and insertion of TII.

In passing, it is worth noting that the technique of off-air relays, commonly used in FM networks, is more difficult in a SFN since there is no separation between the transmit and receive frequency for any given transmitter site. This can lead to difficulties in achieving adequate aerial isolation, particularly at VHF, to prevent instability or keep signal impairment to an acceptable level. However, this technique could still be valuable in the case of L Band Networks or low-power "fill-in" transmitters. In either case, a mixture of ETI feeds to the main stations and off-air feeds to the low-power stations could be envisaged. Note however, that this imposes limitations on the timing of low power transmitters, and would lead to more than one transmitter radiating the same TII code, which could give rise to difficulties in receivers which make use of TII codes.

4.3.8.2 COFDM Generation

The COFDM generator uses the ETI to produce the analogue DAB ensemble. Control information could also be used, and included in the ETI, for transmitter control purposes. The COFDM generator also inserts TII information into the appropriate null symbols under control of information carried in the ETI. This is necessary because the TII is unique to each transmitter location. Note that in the case where the COFDM signal is re-radiated by an off-air relay then the relay will have the same TII code unless the null-symbol information is over-written as mentioned above.

An intermediate interface has also been standardized as a convenient interface between the baseband processing equipment and the radio-frequency modulation equipment. This is the baseband digital I/Q interface which is described in EN 300 798 [4].

4.3.9 Signal Timing and Synchronization

There are a number of issues concerned with signal timing and synchronization which should be considered when designing a DAB Network.

The following lists some of the issues concerned with data rate synchronization:

- The audio coder samples the audio at a frequency of 48 kHz (nominal) or 24 kHz and formats the resulting coded information into frames with a length of 1 152 sample periods (nominally 24 ms or 48 ms, depending upon the audio sampling frequency). If the input to the coder is a digital signal then the coder's sampling frequency and the incoming data sample rate must be synchronized (see note). The output data rate of the coder will be an integer number of bits per frame; the exact number is determined by the output data-rate selected for the coder, which includes all control information, stuffing bits and PAD as well as the encoded audio. The audio coding algorithm may also sample the input at a rate of 24 kHz (nominal). This gives rise to a 48 ms audio frame which is split into two halves (of 24 ms each) for carriage by DAB.

NOTE: This may involve sample rate conversion if the incoming sample rate is not 48 ksamples/sec.

- The SCMux accepts data at the rate supplied by the audio coder and associated equipment, and may add additional data. The output of the SCMux must be synchronous, (or plesiochronous, as determined by the nature of the Transport Network) to the input of the Service Transport Network.

- The EMux accepts the data from a number of Service Transport Networks and produces a single output. Again a 24 ms frame length is used at the output of the EMux. A strategy must be adopted to ensure that each 24 ms frame output by the EMux preserves the frame structure of the data from each input. Either the frames (at output and all inputs of the EMux) must be synchronous, or buffering must be employed to even out the differences. Where buffers are used, then the buffer capacity must be large enough to cope with the data-rate differences and to ensure that buffer slips, if any, are made in integer frame multiples. In other words, frame alignment must be maintained by dropping or stuffing whole frames from a particular input, as appropriate (in the latter case this could be achieved by repeating the previous frame).

- The DAB ensemble produced by the COFDM generator is locked to the 24 ms frame of the ETI output by the EMux. However, if an EMux feeds more than 1 COFDM generator in a SFN then the timing of each ensemble generator in the Network should be kept very close to that of the others (within at most 10 % of the guard interval, unless timing offsets are employed). Additionally, all the transmitter centre frequencies must be very close to each other (within about 1 % of the carrier spacing), implying that each transmitter must maintain a frequency reference. If the delay of Ensemble Transport network is not fixed, then each transmitter also requires a time reference which is also available to the EMux.

In addition, there are related issues concerned with the handling of time information carried in the DAB signal:

- Audible time marks (such as the time pips broadcast in the UK) must bear some resemblance to the time at which the pips are received. The delay through the entire Network is likely to approach 1 second or more when account is taken of processing delays, time interleaving in the DAB signal, buffer delays to take care of synchronization requirements and network transit delays. This delay must be fixed and known to the required accuracy. UK time pips are usually transmitted with an accuracy of about 50 ms.

- Time information carried in the FIC is inserted at the EMux. The precision with which this time is received is not specified but could be expected to be at least an order of magnitude more accurate than the audible pips mentioned above. Again this requires that the delays in the Ensemble Transport Network are accurately controlled.

- DAB Services may also be radiated on FM channels. In this case, account must be taken of the relative delays which will occur in the distribution of signals to both networks. Typically, the delays involved in FM distribution will be considerably shorter than those involved in DAB. Ideally, the received DAB and FM signals should be co-timed. This allows the receiver to use the FM version of a DAB service (if available) to fill in gaps in the DAB coverage, which are inevitable in the early days of any DAB network. However, inserting the implied delay in the FM Network may not be trivial, as broadcast centres would need to run ahead of real time.

4.3.10 Multiplex Reconfiguration - Network Issues

The DAB System permits the flexible and dynamic re-configuration of the Multiplex. In principle, the mix can be changed every 6 seconds. In a diverse network, where Service providers and Ensemble providers are physically separate, a strategy for managing configuration changes must be put in place. Achieving synchronous coding rate changes, which would normally take place at frame boundaries, will require some considerable care. One of the functions of the control information included in the STI, defined in EN 300 797 [2], is to allow the broadcaster to manage and control these re-configurations.

4.3.10.1 A cautionary note

In the interest of simplification, many of the detailed considerations applying to multiplex re-configurations have been somewhat glossed over. For instance, the data interleaving employed within the Ensemble Multiplex, imposes a latency of 15 frames during configuration changes, i.e. data capacity which is changing hands must be cleared 15 frames prior to its re-use by another Service provider. Some of the information carried within the DAB version of an ISO-frame (scale-factor CRCs and PAD) applies- to other frames. This information may need to be suppressed, or ignored, over the period of reconfiguration.

More information on re-configuration issues can be found in the relevant interface specifications [2] to [4].

4.4 Strategies for Signal Distribution

4.4.1 Introduction

The following clause considers how the factors presented in the previous clauses should be applied when considering distribution of service and ensemble information.

4.4.2 Local Connections

Most early implementations of DAB systems relied on the local proximity of the audio coders to an integrated Service and Ensemble Multiplexer.

Connections between the audio coders and the have been made using the WG1/WG2 Interface [2]. Signal timing and synchronization is straightforward and can rely on a local "Master" generator which is usually the multiplexer.

This mode of operation presents no particular difficulty other than the need for all equipment to be in close physical proximity.

4.4.3 Terrestrial Distribution

In the longer term, terrestrial data circuits offer the most natural method of carrying information about Services and Service Components between Service Multiplexing and Ensemble Multiplexing equipment in different locations; indeed, some networks have already been implemented using this approach. It is also likely that terrestrial circuits will be the preferred choice for distribution of the ETI where a small number of transmitters are involved. Large numbers of transmitters are likely to be more economically fed by satellite circuits.

In some cases, distribution using the COFDM signal itself, generated at a vision frequency or at some other suitable distribution frequency, may provide an acceptable alternative (see clause 4.3.8.1). The general considerations apply equally to distribution using the ETI or the COFDM signal.

4.4.3.1 Terrestrial Distribution, STI

The STI may be carried on many different kinds of physical links. EN 300 797 [2] defines STI structures which may be used on G.703, V.11 or AES/EBU-like links.

It should also be noted that the need for communication between the Ensemble Controller and the various Service Controllers may require the STI links to be bi-directional. The capacity requirement of the return circuit is likely to be considerably less than that of the forward circuit carrying the Service data. This is considered in more detail in the relevant standard [2].

4.4.3.2 Terrestrial Distribution, ETI

The terrestrial distribution of the ETI could be done either using fixed links dedicated to the purpose, or using 2 Mbit/s data circuits provided as part of a Telecommunication Network.

In general, there is no need for a return circuit to be provided unless there is a special requirement in a particular case.

It is recommended that one of the Network Adapted versions of the ETI is chosen because of their superior robustness compared with the Network Independent versions. In particular, ETI(NA, G704)$_{5376}$ (see [3]) has been found to offer good performance in most situations including carriage on ATM networks. The capacity available on this variant of the ETI should suit most applications, though may not be adequate for users requiring a large number of data services.

The use of a Network Adapted version of the ETI is recommended on distribution networks feeding a SFN if the delay variation over the distribution network exceeds a small fraction of the guard interval. This includes most, if not all, telecommunication networks. In this case there will also be a need for a timing reference to be provided at each network destination node so that the timing of the incoming data can be corrected. The timing reference should also be available at the ETI origination point so that data can be generated with the correct, and known, timing. The accuracy of the timing reference needs to be of the order of a few µs. Examples of suitable references are the Global Positioning System (GPS) or frame synchronizing pulses derived from a satellite TV channel.

The frequency of each transmitter in a SFN also needs to be accurate to a small fraction of the intended COFDM carrier frequency spacing. This implies an accuracy of a few parts in 10^8 for a Transmission mode I, Band III transmission. It is likely that each transmitter will need a stable frequency reference. Examples of suitable references are; the incoming data clock, synchronizing pulses from a satellite TV channel, or GPS. Sufficient smoothing of the incoming reference should be provided so that random fluctuations of the derived reference do not cause excessive phase noise to be introduced onto the carrier frequency.

4.4.4 Satellite Distribution

Satellite distribution is likely to be the most economic solution where the requirement is for a single point to feed many destinations. This is exactly the situation for national SFNs where the output of one EMux is required to feed, typically, several hundred transmitters.

In other cases, terrestrial distribution is likely to be more economic, unless the satellite capacity can be shared with other uses or is available for some other reason.

In some cases, the COFDM signal may itself be transmitted via satellite. This should be in the "pseudo-video" mode described earlier. Direct use of the COFDM signal on satellites in the FSS, or DBS, bands is not recommended because of the difficulty of achieving adequate performance either in terms of phase noise at the SHF frequencies employed or of transponder linearity.

4.4.5 Sharing the Distribution Network

In some cases, broadcasters may wish to use the distribution network to feed DAB transmitters together with transmitters operating in other frequency bands (e.g. FM). Duplicated services can share the same distribution feed, non-duplicated services could be fed using either spare capacity within the ETI or additional capacity on the same circuit.

A detailed analysis of the problems involved with common distribution paths is beyond the scope of the present document but some of the issues which should be considered are:

- relative system delays of the different feeds due to the processing delays in the DAB interleaving process;
- the use of data rate reduction techniques on the DAB Services;
- "data" requirements of other services may be substantially different (e.g. RDS for FM services).

4.5 Some Real Examples

4.5.1 Introduction

This clause looks at some DAB Network implementations which are operational at the time of writing. These examples serve as an illustration of some of the aspects mentioned in this clause, though the earlier cautionary words about the relative infancy of DAB Broadcast Network techniques should be noted.

4.5.2 The BBC's DAB Network

Figure 4.5.1 shows an outline of the BBC's DAB network. A network of 27 transmitters has been implemented to cover 60 % of the UK population, and the majority of major motorway routes. This is a Single Frequency Network operating in Band III (Block 12B), and the transmitter output powers are in the range 1 kW to 10 kW ERP.

Signal distribution is accomplished using 2 Mbit/s telecommunication circuits using ETI(NA, G704)$_{5376}$, see [3]. A mixture of leased SDH and PDH circuits are used to feed the transmitters and, in most cases, a fully redundant network is used where each transmitter receives two feeds via diverse routes. The preferred feed is selected on the basis of the error statistics of the links using a seamless-switching technique described earlier. GPS receivers are used to provide a time-reference (and frequency-reference) at all sites for the control of delay variations and transmitter frequency.

4.5.3 L band DAB networks in France:

Due to the difficulty of obtaining adequate VHF spectrum, only the frequency band 1 452-1 492 MHz (referred to as L band) is used in France.

Before 1995, several field trials have been done either in Paris or in Rennes (Brittany) in this band. For example, first regular experimental transmission was started in Rennes in 1993 by CCETT. From these experiments, it appeared that L band could be used for urban coverages and also for the coverage of highways.

Since the beginning of 1997, operational networks are open in France by TDF. There are all based on the same scheme:

A broadcast network covering a town and its suburb and using one or several transmitters. DAB mode II is used.

A transport network feeding the transmitters sites and including the ensemble multiplexer. As the transmitters, this multiplexer is also locally located. This permits to incorporate local programmes. Between the multiplexer and the transmitters, The ETI transport interface is used.

A gathering network, collecting the audio programmes and data channels. The programmes can be national and sent by satellite to the multiplexer, or local and sent by microwave links or digital lines to the multiplexer.

In the beginning of 1997, operational networks have been opened in the Paris area. The administration gave licenses for the broadcasting of three blocks in this region. This represents a capacity of 18 programmes.

The networks installed by TDF in Paris are based on Single Frequency Network. 3 sites located in the suburb of Paris are used. The maximum distance between each site is lower than 20 km. All sites are synchronized and have an omnidirectional antenna pattern. With these three sites, Paris and a main part of its suburb is covered. Since 1999, an extension has been launched with three new sites covering the outside of the previous network. The new sites have directional antennas radiating toward the outside of the network.

In 1998, new networks were open in four towns: Lyon, Marseille, Toulouse, Nantes. Other authorizations are expected for the other main French towns.

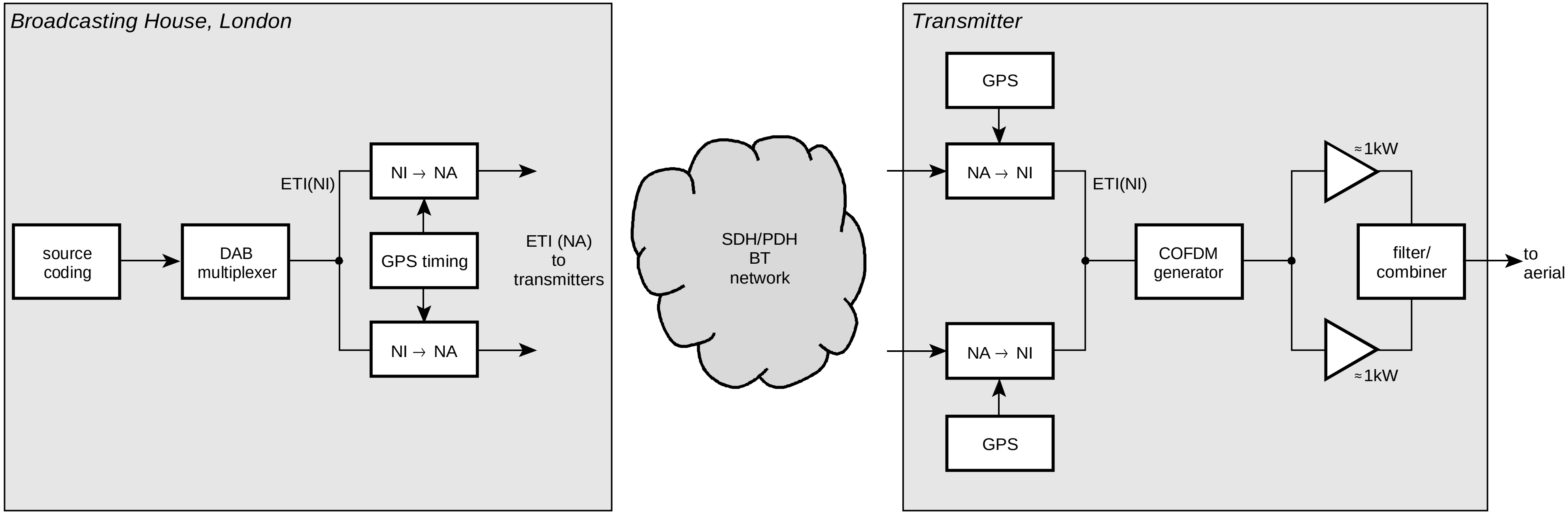


Figure 4.5.1 Outline of BBC DAB Network

5 The Transmitted Signal

5.1 Introduction

This clause starts with a detailed look at the modulation and channel coding used by the DAB system [1]. It continues with a look at some particular aspects including synchronization methods and the technique employed to permit transmitter identification. The clause concludes by considering some RF aspects and broadcast network planning techniques. In particular the concept of a Single Frequency Network (SFN) is examined in some detail.

5.2 Overview

The DAB system employs COFDM modulation, which combines the multi-carrier modulation technique OFDM ("Orthogonal Frequency Division Multiplexing") with convolutional channel coding in such a way that the system can exploit both time and frequency diversity. This is achieved by interleaving data symbols, in the time and frequency domains, prior to transmission.

OFDM contributes to the inherent ruggedness of the system against multi-path distortions due to the relatively large symbol duration. In addition, a guard interval (see note) is used to help remove interference between consecutive symbols. In order to achieve an optimum DAB performance over as wide a frequency range as possible, and with different types of networks, the DAB standard uses four different Transmission modes. The overall capacity remains the same, but the symbol period (and guard-interval) and carrier spacing are varied to suit the situation.

NOTE: The transmitted symbol period exceeds the receiver's analysis window by an amount known as the "guard-interval".

The DAB system uses rate-compatible punctured convolutional codes for forward error correction. This code family allows the amount of error protection to be individually chosen according to the performance requirements of different services. For audio signals, DAB uses unequal error protection. The amount of protection is adjusted to suit the subjective error sensitivity of different parts of the audio bit stream, e.g. bit allocation information, where an error would cause annoying interference, is much better protected than normal audio samples.

The use of a guard-interval, which provides a form of space-diversity, allows a SFN to be implemented. Provided certain constraints on the transmitted symbol timing and centre frequency variance are met, then each transmitter in the network can use the same frequency. However, a method is provided by which the receiver may identify which transmitter (or transmitters) it is receiving. This is achieved by allocating to every transmitter a signal pattern, radiated during the synchronizing period, which is unique.

One consequence of the multi-carrier technique, with the statistical nature of the carrier phases, is a relatively high peak-to-mean ratio the of signal amplitude in the time domain. This leads to a requirement for "linear" signal amplification. Further the power spectral density of an OFDM signal requires filtering in order to keep out-of-band radiation within defined spectrum masks. This restriction is needed to achieve the required channel spacing.

5.3 Channel coding and modulation

5.3.1 OFDM modulation & transmission frame

OFDM is a multi-carrier system. Data is transmitted at a low symbol rate using many narrow-band carriers rather than at a high rate using a single wide-band carrier. The numbers of carriers used in each of the DAB Transmission modes are given in TR 101 496-1 [8]. Carriers are arranged to be mutually orthogonal, so each carrier has its peak amplitude, in the frequency domain, where all others have a zero-crossing.

The bit rate for each carrier is inversely proportional to the OFDM symbol duration. A lower bit rate means that received data suffers less from Inter-Symbol Interference (ISI) in the presence of multipath propagation. Consequently, OFDM is less sensitive to this type of propagation than a wide-band single carrier system. By adding a guard interval between successive symbols, the effect of ISI can be completely eliminated, as long as the delay spread of the received multi-path signal does not exceed the duration of the guard interval. Detailed parameters for each of the Transmission modes are given in TR 101 496-1 [8].

Mode I is intended for terrestrial broadcasting and permits the use of a regional SFN. The required transmitter separation is similar to that for conventional VHF/FM networks. Using the same transmission frequency for the same range of services, the broadcaster can gain a bandwidth saving for national and regional services. The consumer has the advantage that the same service is available anywhere without having to retune his receiver.

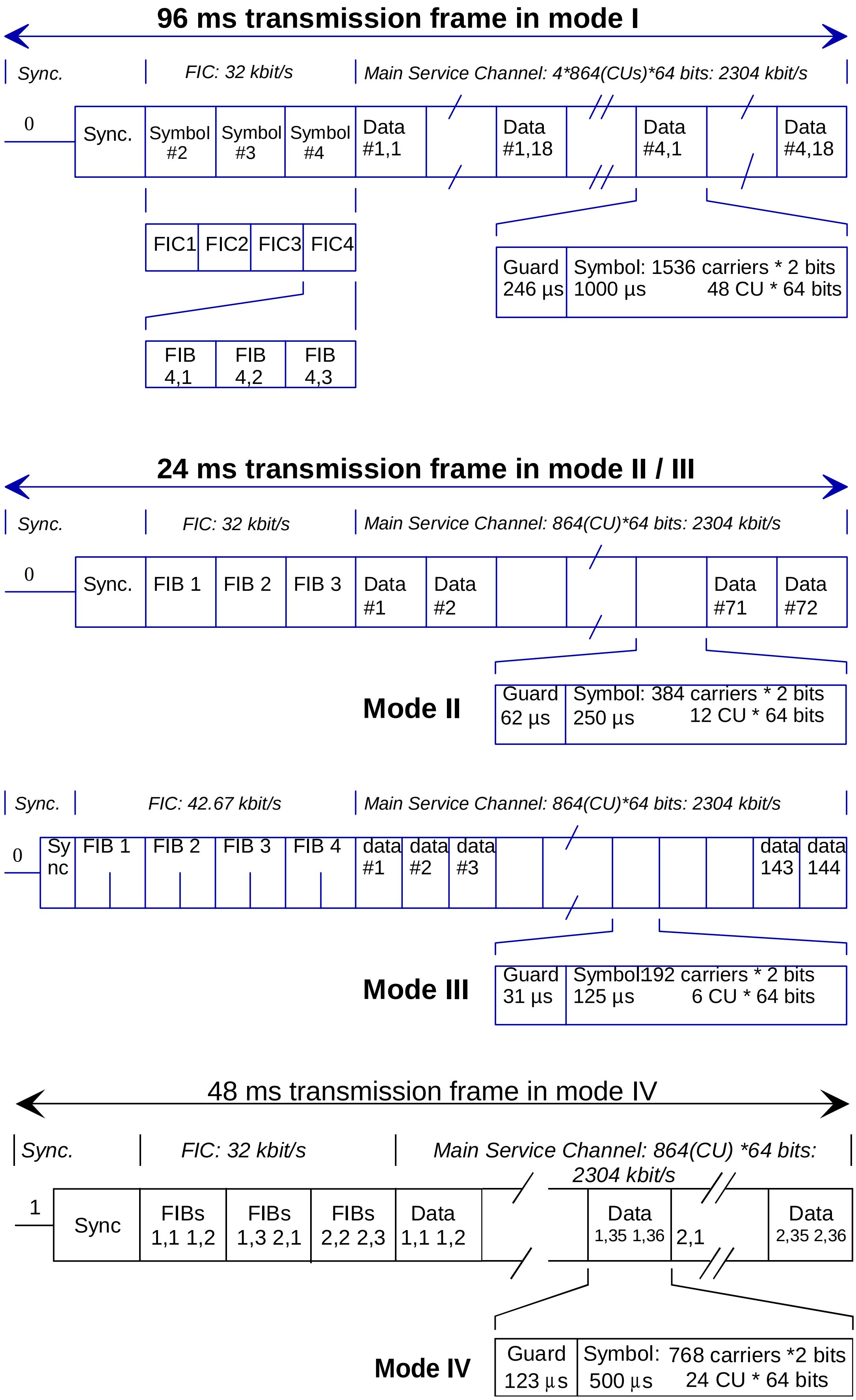


Figure 5.1: Structure of the transmission frame

Mode II may be used at transmission frequencies up to 1,5 GHz, primarily for local terrestrial or satellite broadcasting. A SFN is still possible, but only by implementing a denser transmitter area network to counteract the shorter guard-interval.

Mode III is the most robust against Doppler spread and is useful for transmission frequencies up to 3 GHz. Its primary application is in satellite systems or cable networks. Mode I or II could also be used for the latter.

Mode IV is used for hybrid satellite systems and complementary services at 1,5 GHz.

Figure 5.1 shows the basic structure of each Transmission mode. The main service channel is subdivided into Capacity Units (CUs). Each CU contains 64 encoded source bits. The sub-frame structure chosen for Mode I is such as to allow partitioning into 24 ms frames after demodulation and decoding. Note that Mode III offers one more FIB in the FIC than Modes I, II and IV.

Differential modulation is applied to facilitate bit recovery at the receiver. Each OFDM carrier contains two bits of Gray-coded 4-PSK data. The guard interval is constructed by a cyclic continuation of each symbol.

Modulation in the transmitter may be realized with an IFFT of at least 2 048 points for Mode I, 512 for Mode II, 256 for Mode III, and 512 for Mode IV. The base-band signal should provide enough resolution to prevent an increase of noise in the receiver (see clause 5.5.1). The base-band signal sampled at 2,048 MHz for both the in-phase and quadrature component forms the IFFT output block. Clause 5.5.1 contains information about the required headroom for digital modulation.

The receiver should carefully position its symbol window (equivalent to the FFT analysis period) so that any ISI due to multi-path reception (or, in an SFN, multiple transmitter reception) is kept within the guard-interval. From the FFT resultant, only the N middle carriers contain useful data, where N is a function of the Transmission mode.

Repositioning of the symbol window, from frame to frame, will only result in a phase shift of each carrier. This does not affect differential demodulation between adjacent symbols.

The null symbol provides coarse receiver synchronization but can also carry Transmitter Identification Information (see clause 5.4.2). The receiver could also use the null symbol to analyse the transmission channel and take into account the level of interference or noise which are present.

The phase reference symbol provides fine synchronization information allowing the receiver to extract frequency information (for carrier acquisition through AFC) and a phase reference for differential demodulation. The receiver does not need to extract a carrier reference for signal demodulation.

5.3.2 Channel Coding

Inside the coverage area of the DAB service, a quasi error-free reception is generally obtained due to the high performance of the applied channel coding schemes. Nevertheless, transmission errors cannot be completely avoided, especially at the edge of a service area. Therefore, two goals for channel coding have to be considered: firstly, error-free reception within the coverage area; secondly, some kind of graceful degradation at the edge of it. Both are achieved by applying source adapted channel coding. Data services where the bits show an equal sensitivity to bit errors are protected in an equal manner (EEP: equal error protection). Sound services, where groups of bits having different sensitivities to bit errors, are protected with a non-uniform code (UEP: unequal error protection). This allows economical use of the available redundancy and therefore a high protection performance.

Error protection in DAB is based on convolutional codes with a memory of 6 bits, i.e. the number of successive data bits which are used for creating code bits is equal to 7 (or, in the jargon, a constraint length of 7). The basic code rate (mother-code) is of rate $R = 1/4$ which uses 4 code bits to protect each data bit. The fourth code bit is in fact a repetition of the first; only 3 different generator polynomials are used. Weaker codes, with rates up to $R = 8/9$ are obtained by puncturing the code bits of the mother code. Puncturing means that certain code bits, which are selected by a puncturing vector, are not transmitted.

To cope with poor reception conditions, additional provision to detect any failure of the error correction process is required. FIBs are protected by a 16 bit cyclic redundancy code (CRC) (see clause 5.3.5 and [1]). For data in Packet mode and data groups respectively, another optional CRC may be used (see [1]). The method of calculating the CRC words is also given in [1]. For Audio services, a CRC is provided for the control information (Header, BAI, ScFSI) according to the ISO 11172-3 standard [11]. An additional CRC is provided; one to each set of four groups of scale factors. When bit errors are detected, error concealment may be applied to the audio signal (e.g. repetition of non disturbed scale factors, muting of sub-bands or repetition/muting of frames).

By combining UEP and error concealment, the subjective impairment caused by bit errors is significantly reduced and a graceful degradation at the edge of the service area may be achieved.

5.3.3 Unequal error protection (UEP) for audio (48 kHz sampling)

5.3.3.1 Protection classes

A DAB audio stream contains components of audio data with different sensitivities to bit errors (significance). Every 24 ms all the components are transmitted using the DAB audio frame [1]. For those components which have nearly the same significance a common protection class is applied. The DAB audio frame uses four different protection classes, applied as follows:

Protection class 1:

ISO-HEADER	header information	32 bit
CRC	CRC for error detection within control information:	
	header, bit allocation and scale factor select information	16 bit
BAl	bit allocation information	2×88 bit
ScFSI	scale factor select information	2×54 bit

Protection class 2:

ScF	scale factors	2×486 bit

Protection class 3:

SAMPLE	sub-band samples	variable
STUFF	stuffing bits	variable
X-PAD	extended programme associated data	variable

Protection class 4:

X-PAD	extended programme associated data	32 bit
ScF-CRC	CRC for error detection within 4 groups of scale factors	32 bit
F-PAD	fixed programme associated data	16 bit

The first protection class comprises different kinds of Control Information (CI) for the audio decoding process. All this information shows the same, very high, sensitivity to bit errors. Any single bit error in this information would cause a totally disturbed frame.

The second protection class contains the scale factors. Scale factor errors may cause very annoying "blips". But because of the applied scale factor error concealment, in conjunction with the ScF-CRC, the performance requirement for error correction is not as great compared to CI.

Protection class 3 is used for sub-band samples covering the largest part of the audio frame. Since sample errors are only perceivable when the bit error ratio is above 10^{-4}, the error protection can be lower than that required for CI and scale-factors. For convenience, the early X-PAD information is also included under this category and will be less well protected than later X-PAD and the F-PAD.

Protection class 4 is a continuation of protection class 2. To retain compatibility with the ISO/MPEG standard, the ScF-CRC, which is essentially a part of the scale factor information, is transmitted at the end of the frame. Therefore, the same correction performance is needed for this class as for the scale factors. This protection class is also used for F-PAD and the later part of the X-PAD Information.

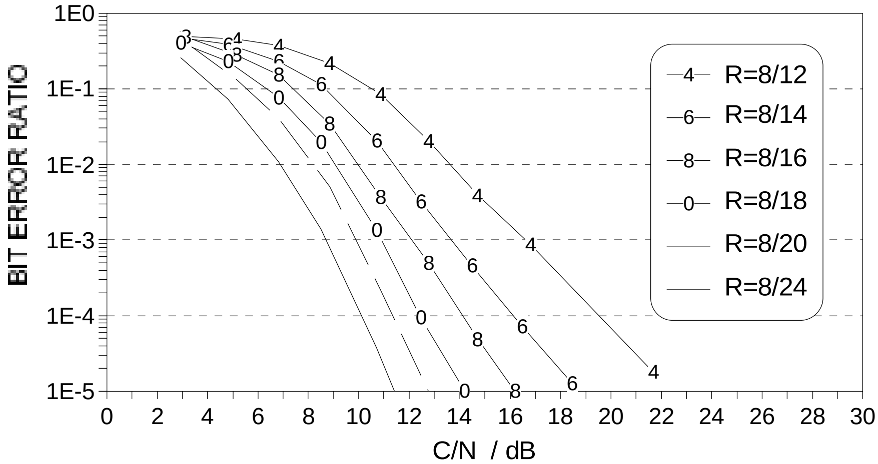


Figure 5.2: Residual bit error ratio for different protection classes with code rates R (Rayleigh Channel COST 207 Rural Area)

The code rate of a protection class is set by selecting a rate compatible puncturing scheme. Code rates from $R_1 = 8/9$, $R_2 = 8/10$, ..., R_i, ..., $R_{24} = 8/32$ are provided and are indicated by the index i of R_i. Figure 5.2 shows some curves of the residual bit error ratio using different protection classes with code rate R_i. (The curves were measured using the "3rd Generation" experimental equipment over a simulated COST 207 Rural Area channel (vehicle speed 50 km/h at a frequency of 232 MHz).)

5.3.3.2 Protection profiles

The number of bits for each protection class depends on the specific audio data rate and the audio mode defined by the header information. The error protection classes within one audio frame are defined by a protection profile, which carries information about the length L_j of each protection class, j, and the corresponding index number, PI_j, of the chosen puncturing vector.

64 protection profiles have been defined, covering all the specified audio data rates. They are designed to be applicable for monophonic, stereophonic, dual channel and joint stereophonic sound coding. The length of the protection class is always chosen to match the worst case for each audio data rate. For example, an audio frame for a stereophonic service at a data rate of 128 kbit/s requires twice the BAI and ScFSI of a monophonic service at 128 kbit/s. The latter will benefit from the extra protection because a part of the scale factors and a small part of the samples are protected to a higher level. This behaviour is consistent with the higher source coding quality of the monophonic sound signal.

The protection profiles for audio services were designed by optimizing the distribution of the available redundancy according to the significance of their components. Since protection class 3 is applied to the largest number of bits, its code rate, R_i was chosen to leave sufficient transmission capacity for the higher protection classes. In further steps, the code rates for protection classes 1, 2 and 4, and the length of protection class 2, were adjusted in order to get an optimal UEP scheme. Wherever there was not enough redundancy available for the ideal protection of class 1, the code rate of protection class 3 had to be increased. In some cases, this led to a higher protection of class 1 and 2. Especially for low audio data rates, the relative length of protection class 1 is large and this results in a weaker protection of the samples. For a given average code rate, the error protection of a service with high audio data rate is increased due to the larger number of sample bits, e.g. 256 kbit/s has a better sample protection than 192 kbit/s for the same protection level.

The available redundancy depends on how the 864 capacity units of the Main Service Channel are allocated to sub-channels. Because of the large flexibility in arranging the DAB multiplex "gold" numbers were chosen for the amount of capacity units used by the protection profiles. This approach allows certain multiplex re-configurations without the need to rearrange other sub-channels (e.g. splitting one service at 256 kbit/s into two services, each at 128 kbit/s). Table 5.1 gives an overview of number of CUs utilized for each of the audio data rates.

Table 5.1: "Gold" numbers of Capacity units used for protection profiles

Number of CUs:	16			21	24		29
Audio Data Rates (kbit/s):	32			32	32, 48		32, 48, 56
Number of CUs:	32	35	40	42	48	52	58
Audio Data Rates (kbit/s):	64	32, 48, 56	80	48, 56, 64	64, 96	48, 56, 80	64, 80, 96, 112
Number of CUs:	64	70	80	84	96	104	116
Audio Data Rates (kbit/s):	128	64, 80, 96, 112	160	80, 96, 112, 128	128, 192	96, 112, 160	128, 160, 192, 224
Number of CUs:	128	140	160	168	192	208	232
Audio Data Rates (kbit/s):	256	128, 160, 192, 224	320	160, 192, 224, 256	256, 384	192, 224, 320	224, 256
Number of CUs:		280				416	
Audio Data Rates (kbit/s):		256, 320, 384				384	

The various numbers of capacity units for each of the protection profiles result from the supported audio data rates. The encoded frame also includes between 12 and 20 encoded termination bits (tailbits, code rate 1/2) to return the convolutional encoder into the zero state, i.e. to clear its memory. This so-called blocked convolutional coding permits closure of the decoder trellis and allows independent decoding of the different sub-channels.

5.3.3.3 Protection Levels

To meet different protection requirements, five protection levels corresponding to five different average code rates, R_{ave}, are provided for nearly every audio data rate. The protection performance can be chosen with regard to the application. Protection level 5 has been designed for cable distribution. It allows a high number of programme services, but does not have the strong error protection which is necessary on multi-path channels. Protection level 3 applies to mobile services. To get more flexibility in accommodating sub-channels, protection levels 4 and 2 have been introduced with weaker and higher protection performance than protection level 3 respectively. Protection level 1 allows a higher protection for applications with a very high sensitivity to transmission errors. Table 5.2 gives an overview of the protection levels and the corresponding code rates. Figure 5.3 shows the protection profiles for an audio data rate of 192 kbit/s at all protection levels.

For the compilation of a DAB multiplex, a reasonable trade-off between the number of programmes, the audio data rate, and the error protection level has to be made. The benefit of the high performance protection levels is that the samples are protected very well and that the curves of the residual bit error ratio versus C/I become steeper. In high speed, mobile, reception an error floor exists for protection class 3 at protection levels 4 and higher. This situation can easily be improved by using the next highest protection level. Protection level 4 (mobile weak) may be used when the service is not addressed to mobile receivers.

Table 5.2: Overview of the protection levels and the corresponding code rates

Protection			Code Rates *R*				Coding
Level		Application	average	protection class 1	protection class 2, 4	protection class 3	Gain * C/I in dB
1	very high	special	0,34-0,36	8/32	8/25-8/28	8/19-8/22	+4
2	high	mobile high	0,4-0,43	8/30-8/32	8/20-8/26	8/16-8/17	+2
3	good	mobile	0,5-0,51	8/23-8/24	8/16-8/18	8/14-8/15	0
4	medium	mobile weak	0,57-0,62	8/17-8/21	8/14-8/17	8/12-8/13	-1,5to-3.5
5	low	cable	0,72-0,75	8/13-8/16	8/11-8/14	8/10	**

* expected coding gain in a Rayleigh channel at BER=10^{-3} with respect to protection level 3

** this channel coding level is not appropriate to a Rayleigh channel

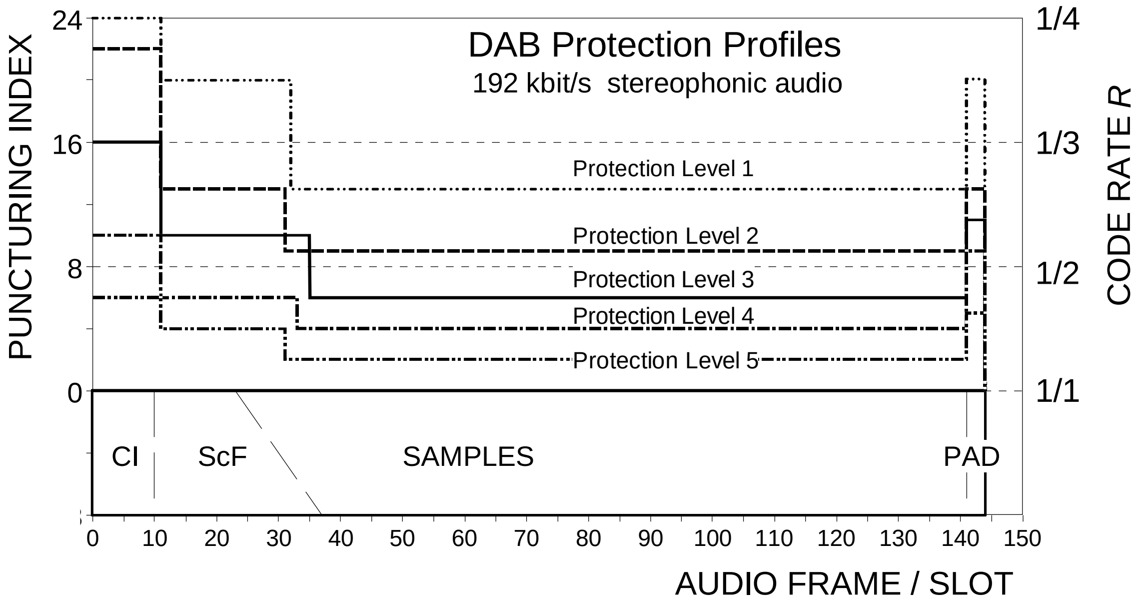


Figure 5.3: Examples of DAB protection profiles for the audio data rate 192 kbit/s (1 Slot = 32 Bit)

To illustrate the performance of the different protection levels, the residual bit error ratios of the protection classes have been measured for two conditions, using the "3rd Generation" experimental equipment. The two conditions are:

a) Protection level 3 at 192 kbit/s,

b) Protection level 4 at 224 kbit/s.

The results for the Gaussian channel are plotted in figure 5.4 as a function of C/N, and show the expected difference in the bit error behaviour. It should be noted that, for protection level 3, two different code rates are specified for protection class 3 (samples) because of the high percentage of control information at lower audio data rates. All data rates below 224 kbit/s use $R_6 = 8/14$ instead of $R_7 = 8/15$, which is used for the data rates of 224 kbit/s and above. Therefore condition a) represents the worst case of protection level 3 with $R_{ave} = 1/2$.

The performance curve of protection class 2 [condition (b)] having a code rate of R=1/2 can be compared with theoretical results from literature. The stroke-dotted line in figure 5.4 includes a 2,3 dB shift due to allow for the effect of differential demodulation. The comparison shows that the implementation margin is lower than 1 dB.

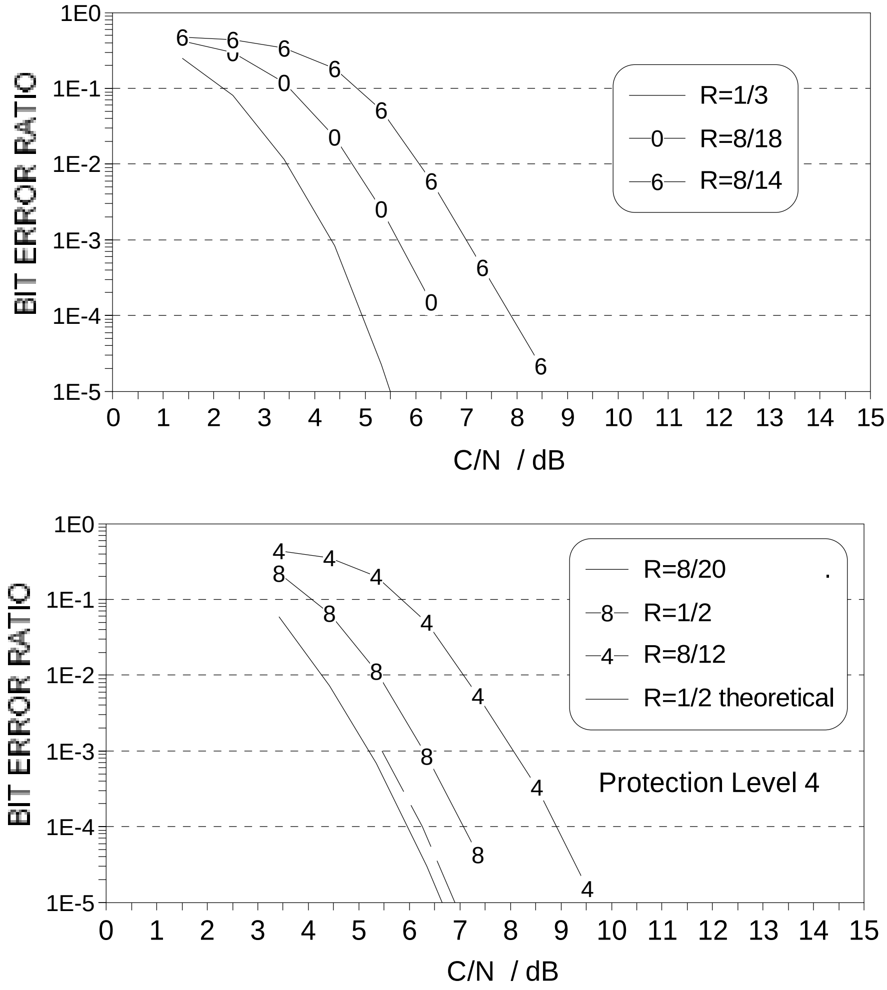


Figure 5.4: Bit error ratio of the 3 protection classes in a Gaussian channel
top: a) 192 kbit/s, R = 0,5, Protection Level 3
bottom: b) 224 kbit/s, R = 0,6, Protection Level 4

5.3.4 Equal Error Protection

Within the framework of rate-compatible convolutional codes, provision is made in the DAB system for encoding sub-channels carrying data service components with Equal Error Protection (EEP). As for audio, a number of protection levels have been defined using code rates between 1/4 and 3/4. The measured bit error rates versus S/N for the different code rates can be deduced from figures 5.2 and 5.4.

If a Sub-channel is organized in Packet mode, however, the (average) bit error rate is not the only important figure. Due to the properties of the code, transmission errors are expected to occur in bursts after channel decoding. Therefore, it is important to know the error free distance between two consecutive error bursts. The appropriate measure is the error gap density, of the channel after decoding. With this information, the two parameters (data group and packet length) characterizing the Packet mode can be chosen.

5.3.5 Error protection for low sampling frequency (LSF) audio (24 kHz sampling)

Low sampling frequency audio uses 48 ms frames. For a sub-channel carrying an LSF audio stream, the data comprising each 48 ms audio frame is divided up into two equal parts for carriage within the Common Interleaved Frames, which are all of length 24 ms. The first part will carry the ISO-Header, CRC, BAI, ScFSI and ScF information and audio sample data. The second part will carry the remaining audio sample data, stuffing bits, X-PAD, ScF-CRC and F-PAD.

Because LSF audio offers some bit-rates that are not accommodated within the "Gold" numbers of CUs assigned to full sampling frequency audio, not all LSF sub-channels will be able to use the UEP profiles. Such sub-channels will need to use EEP profiles. For LSF sub-channels using the bit-rates that are available within the "Gold" numbers of CUs, the use of UEP profiles is possible and may be advantageous because all the data in protection classes 1, 2 and 4 will achieve higher protection than that provided by the use of equivalent rate EEP.

5.3.6 Error Detection in the Fast Information Channel

The FIC carries information about the configuration of a DAB Ensemble multiplex. Decoding this information correctly is vital for proper receiver operation. Therefore it is important to know to what extent transmission errors can be detected. The FIC is convolutionally encoded with a code rate of 1/3 but, in contrast to the MSC, the data is not time-interleaved. Consequently, a "bursty" error characteristic is to be expected since errors are not re-distributed by the dis-interleaving process. As the bit error rate (BER) is not the only parameter which influences error detection, it is necessary to take into account the "burstiness" of the channel, i.e. its "memory". Due to the lack of experimental data a simulation of the error detection in FIBs was performed using the Gilbert model, which is a two-state Markov model able to simulate a channel with burst errors.

FIBs are protected by a cyclic redundancy check (CRC) which is generated by the generator polynomial:

$$G(x) = x^{16} + x^{12} + x^5 + 1$$

For the simulation a "reference FIB" was used. The data field of this FIB was created randomly and the correct CRC was added.

Error pattern fields with the length of one FIB (256 bits) were created with the Gilbert model. These error fields were added bit by bit (modulo-2) to the "reference FIB" and the CRC was performed. This procedure was repeated 1 million times for each choice of the parameters in the Gilbert model. Those erroneous FIBs which were not rejected by the CRC were counted. Therefore, they are referred to as Undetected Erroneous FIBs (UEFIBs).

Figure 5.4 shows the number of UEFIBs versus BER based on 126 simulations performed with a wide range of Gilbert model parameters. As might be expected, the number of UEFIBs increases with the BER. The spread of the data is due to the statistical nature of the errors.

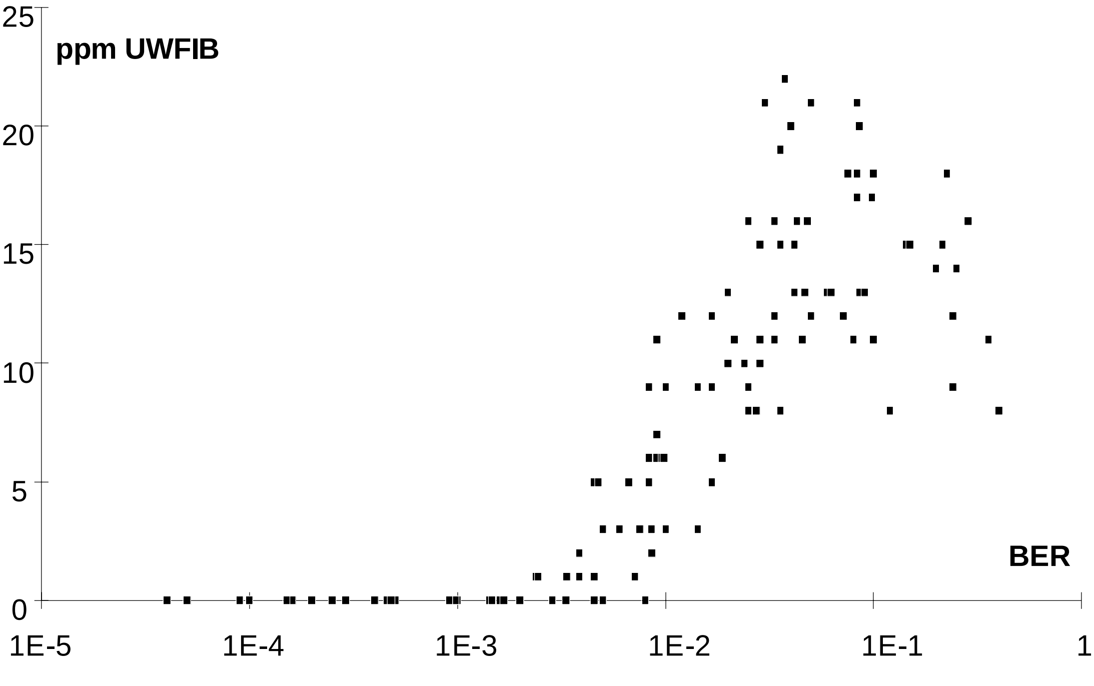


Figure 5.5: Number of UEFIBs out of 10^6 FIBs versus BER

At a BER below 10^{-3}, no UEFIBs were detected in any of the simulations. Although UEFIBs could theoretically occur in such a situation, their probability is very small. From the simulation results, the probability of an UEFIB at a BER of 10^{-4} is estimated as 1 in 10^{10} FIBs. In Transmission modes I and II, this is equivalent to receiving one UEFIB every two and a half years. It should be noted, however, that some worst-case situations may have been missed in the simulations.

5.3.7 Time and Frequency Interleaving

5.3.7.1 Frequency Interleaving

In the frequency domain, multi-path propagation leads to an attenuation, or an amplification, of some of the OFDM carriers. In general, the attenuations of adjacent carriers are strongly correlated. The frequency interleaving procedure ensures that the code bits of any service are shared between the weak and strong carriers. Thus, the performance of the error correction is increased significantly, especially in stationary reception conditions which would otherwise suffer from the relative weakness of convolutional codes in the presence of error bursts.

5.3.7.2 Time Interleaving

Time interleaving improves the ruggedness of the error correction in a time-variant transmission channel. Specifically in the case of mobile reception, even deep fades which affect all OFDM carriers (flat fading caused by short path differences) can be overcome. The longer the time interleaving the better the protection against flat fades. For example, when a convolutional code with rate R=1/2 is used, a fade may last up to 1/10 of the interleaving time with no degradation at high SNR.

The time interleaving covers 16 frames (of 24 ms) resulting in a processing delay of 384 ms. This imposes a significant end-to-end delay compared to conventional analogue broadcasting.

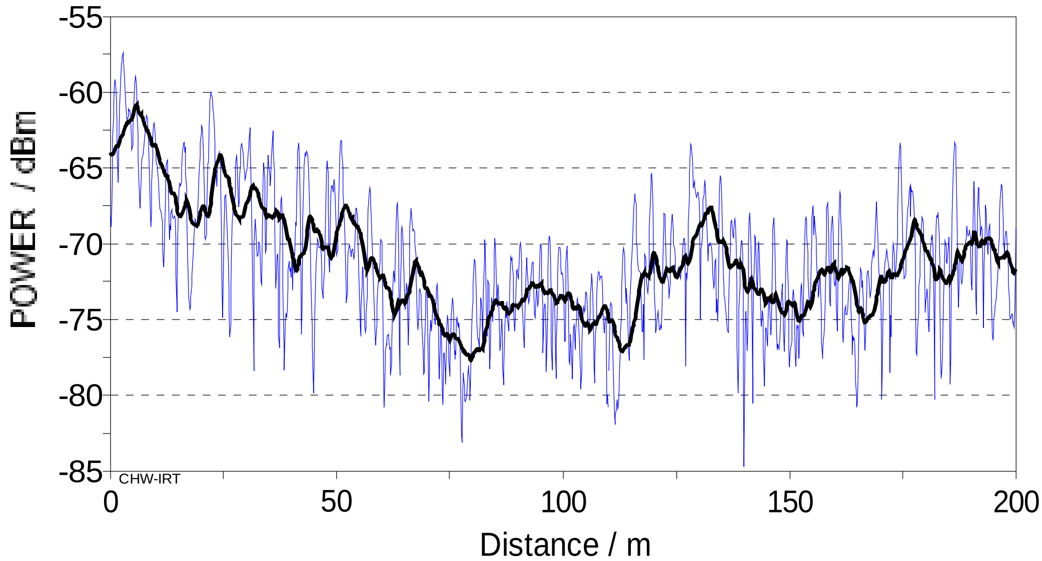


Figure 5.6: Received power in a car moving at 50 km/h (thin line) showing the effect of averaging due to time interleaving (bold line)

Figure 5.6 shows an example of the received power of a DAB signal, plotted against distance traversed, in a mobile reception environment. The power was measured in a 1,5 MHz bandwidth at a centre frequency of 220 MHz. A measurement was made every 12,5 cm. The received power is characterized by many deep fades. To assess the effect of time interleaving, the received power was averaged over 16 successive frames. With a moving receiver, this can also be seen as an averaging over the distance covered during the interleaving time. The average received power is included in figure 5.6, for a vehicle speed of 50 km/h, and shows the beneficial effect of time interleaving.

5.4 Synchronization and Transmitter Information

5.4.1 Synchronization Aspects

During normal reception, the BER is determined by the degree of error protection, the noise of the receiver input stages and the channel characteristics. To obtain good audio quality, a BER of 10^{-4} is needed.

However, frequency deviations of the base-band signal, or a corresponding deviation of the receiver clock oscillator(s) will result in a performance degradation. Measurements have shown that a minimum accuracy of about 1 % of the carrier spacing (e.g. 10 Hz in mode I, 40 Hz in mode II and 80 Hz in mode III) is needed to keep performance degradation within 1 dB (uncoded). These values include any low-frequency jitter of the oscillator(s). Occasionally exceeding these values will not cause a significant degradation. The frequency deviation of the transmitter should be significantly lower than this 1 % value. (No measurements are available on the performance of Mode IV).

To prevent the use of expensive high precision local oscillators, the implementation of Automatic Frequency Control (AFC) is strongly recommended. A frequency domain evaluation of the phase reference symbol can be used to detect frequency deviations. The structure of the phase reference symbol allows a detection range of several carrier-spacings.

The noise side-band, or phase-noise, components of local oscillators must also be considered. For mode I particularly, and with phase-locked oscillators with relatively low reference frequencies (small frequency steps),significant performance degradation can occur. A guideline value for phase-noise components has been set at -60 dBc / Hz at a frequency offset of 25 % of the carrier spacing. The decrease with increasing frequency distance is assumed to be of the order of 6 dB / octave.

The DAB base-band signal is sampled in the receiver at 2,048 MHz for both the I and Q signals. This means that the system clock of the channel encoder and decoder may be any multiple m (m = 1, 2, 3, ...) of 2,048 MHz. It is recommended that the encoder clock have an accuracy of about 1 ppm. The system clock of the channel decoder should be synchronized to the encoder clock. Synchronization is derived from analysis of the channel impulse response, which may be estimated from the phase reference symbol.

5.4.2 Transmitter Identification Information

5.4.2.1 General Description

The coverage area of a SFN with the same ensemble may be very large. The consequence is that some of the information carried in the ensemble may not be relevant for the whole area of the SFN. Therefore there is a need for localizing information which could be used to filter out the relevant data.

The Transmitter Identification Information (TII) provides this localizing feature. The TII signal enables receivers to distinguish the individual transmitters of a network. Every transmitter sends a unique TII signal during the Null symbol of the transmission frame, thus violating the general rules of SFN transmission that requires all transmitters of the network to send identical signals. The potential interference problem is solved by defining TII signals in such a way that only a subset of the OFDM carriers are used by any transmitter. Assignment of TII signals to transmitters is performed so that adjacent transmitters use different carriers. This allocation must follow the rules of conventional network planning.

The identifier comprises two parts; a main and a sub identifier for every transmitter in the SFN. From analysis of the Null symbol a receiver can derive the identifiers of those transmitters which are currently received. The receiver can use these identifiers directly for service information selection based on geographical criteria. For a more precise localization, the geographical data of the transmitters may also be conveyed in the SI, see [1]. With the help of this information the receiver can estimate its location inside the coverage area of an SFN (see TR 101 496-2 [9]).

5.4.2.2 Null Symbol and Network Planning

Every transmitter switches on specific carrier pairs during the Null Symbol. Using carrier pairs instead of single carriers facilitates the determination of the geographical position of a receiver. In order to allow the receiver to perform channel state analysis, the TII signal is only transmitted in every other frame. The synchronization is aligned with the CIF counter.

The structure of the TII signal is based on a block of 384 carriers in Transmission Modes I, II & IV. This block of carriers is organized as 24 "combs" of carrier pairs, each comb comprising 8 carrier pairs. In Mode II, this structure matches the 384 available carriers; in Mode IV, the structure is repeated twice in the frequency domain, to match the 768 available carriers; and in Mode I, the structure is repeated four times in the frequency domain, to match the 1 536 available carriers.

In Transmission Mode III, the TII signal is based on a block of 192 carriers, again organized as 24 combs of carrier pairs, each comb comprising 4 carrier pairs.

In all Modes, the 24 combs, which correspond to the set of possible SubIds of the transmissions, allow the conventional network planning of the TII signal inside the SFN. The allocation of SubId to a transmitter determines which of the combs of carriers it will transmit.

As noted above, in Modes I, II and IV, there are 8 pairs of carriers in each comb. The TII signal for a given transmitter may only use 4 out of these 8 pairs. Since the number of combinations of 4 from a set of 8 is 70, this results in 70 unique "patterns" of carrier pairs per comb, which correspond to the set of possible MainIds of the transmissions. The allocation of MainId to a transmitter determines which of the patterns (i.e. which 4 out of the 8 carrier pairs in the comb) it will transmit.

Mode III is similar but because each comb consists of 4 carrier pairs, the TII signal for a given transmitter may only use 2 out of these 4 pairs. Since the number of combination of 2 from a set of 4 is 6, this results in 6 unique "patterns" of carrier pairs per comb, and hence 6 possible MainIds.

In Transmission mode I, the TII structure is repeated 4 times in the frequency domain, so every transmitter uses four times four pairs of carriers, or 32 carriers in total. In Mode IV, the structure is repeated twice in the frequency domain, so every transmitter uses 16 carriers in total. In Mode II four pairs of carriers (8 in total) are used, and in Mode III two pairs of carriers (4 in total) are used. The ratio of carriers in a TII symbol to a normal DAB symbol is 1:48 for all Modes, so that the signal power in a TII symbol is 16 dB below the signal power of the other symbols. Therefore, coarse receiver synchronization from the null symbol containing TII is still possible.

For example in figure 5.7, the TII symbol for the BBC's experimental transmission at Crystal Palace is illustrated. The MainId is 0B (hex) which corresponds to a pattern of '00110110' [1]. The SubId is 01 (hex) which uses the second of the 24 possible pairs of carriers.

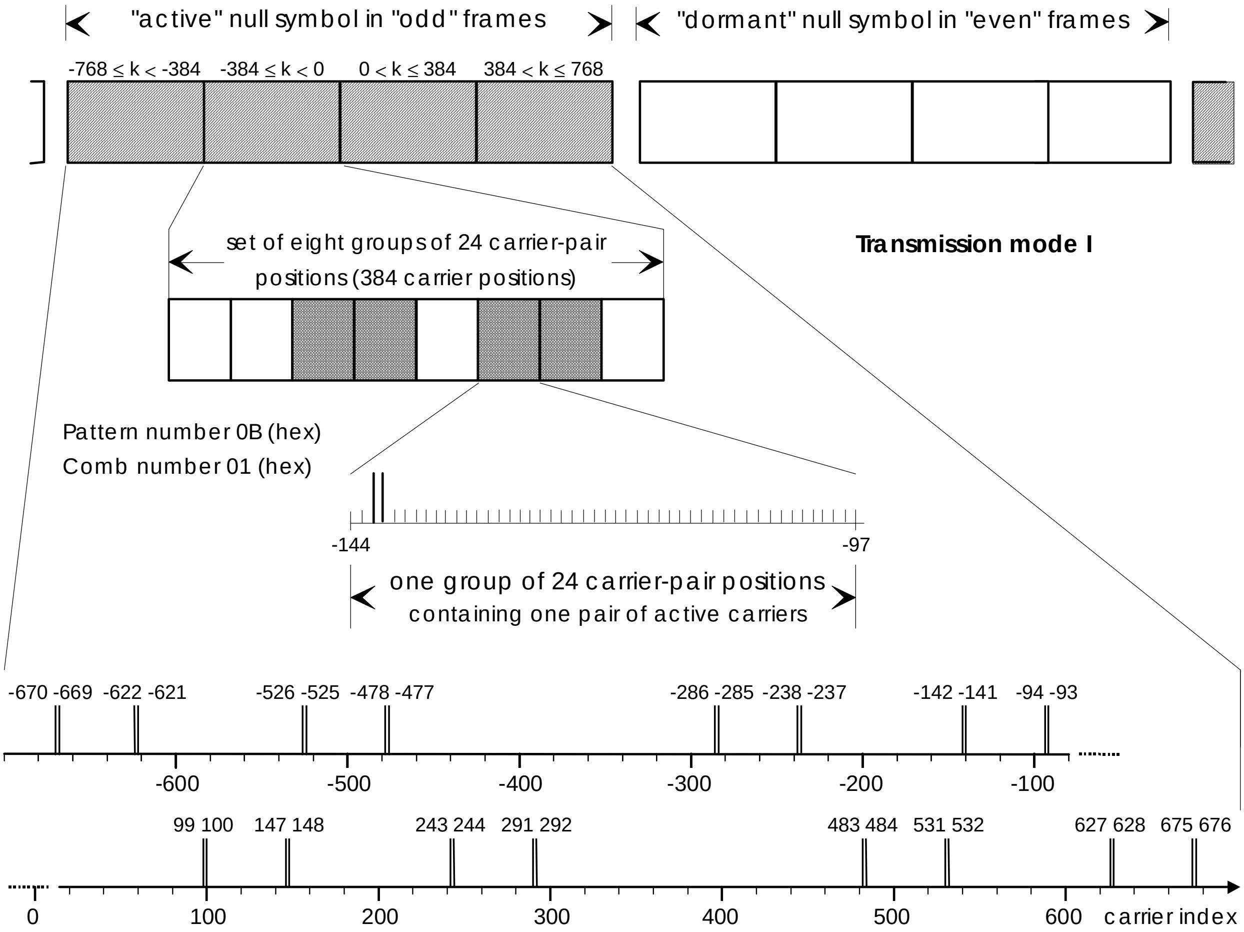


Figure 5.7: Comb structure (Transmission mode I)

Figure 5.8 shows an idealized network planning structure with pattern- and comb numbers corresponding to the main identifier and sub identifier respectively. The first number is the main identifier and the second number is the sub identifier. SubId 0 is reserved for satellite transmission. Assuming that the distances between the transmitters are always the same (e.g. 60 km) and 21 sub identifiers are used, the coverage area is larger than a circle with a diameter of about 240 Km.

If the coverage area of one SFN is larger, the hexagons of figure 5.8 can be arranged with different main identifiers. Every hexagon of 21 transmitters has its own main identifier. An example is shown in figure 5.9.

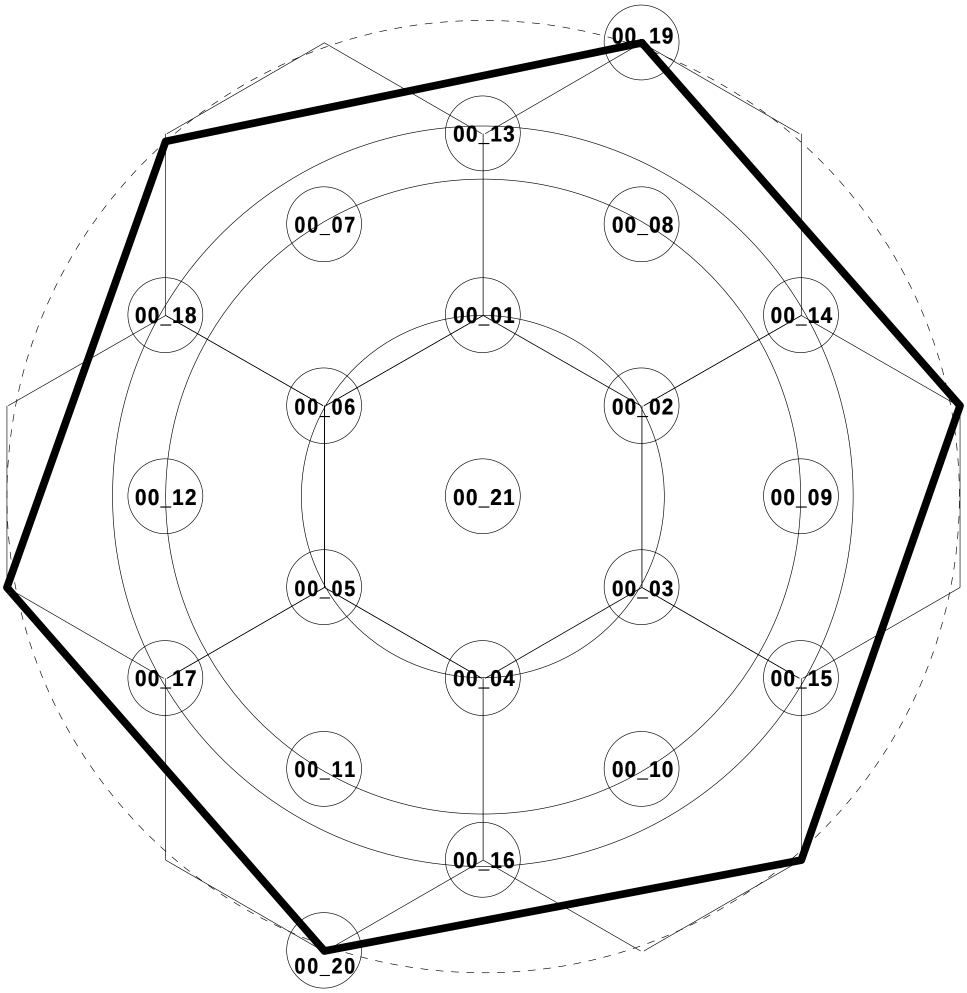


Figure 5.8: Sub-Identifier 01 to 21 with one Main Identifier (00)

5.5 RF Aspects

5.5.1 Time domain representation

In a DAB signal, relatively high amplitude peaks can occur for short periods of the symbol time when the various carriers are in phase. A problem occurs when the signal is transmitted through a practical device (such as an amplifier) as the device must have a linear transfer characteristic with a large amount of headroom to prevent non-linear effects from occurring.

The problem can be simulated mathematically. A DAB-like symbol was simulated with 1 536 equal-power active carriers, each in one of four phase states (chosen randomly). The instantaneous amplitude distribution was then calculated and is shown in figure 5.10. The signal demonstrates a Rayleigh distribution (over the low part of the amplitude range where statistical treatment is applicable) because the signal consists of a large number of carriers each with a randomly chosen phase. The frequency difference between the signals ensures that, from a single reference start point, the phases of the carriers will become entirely random.

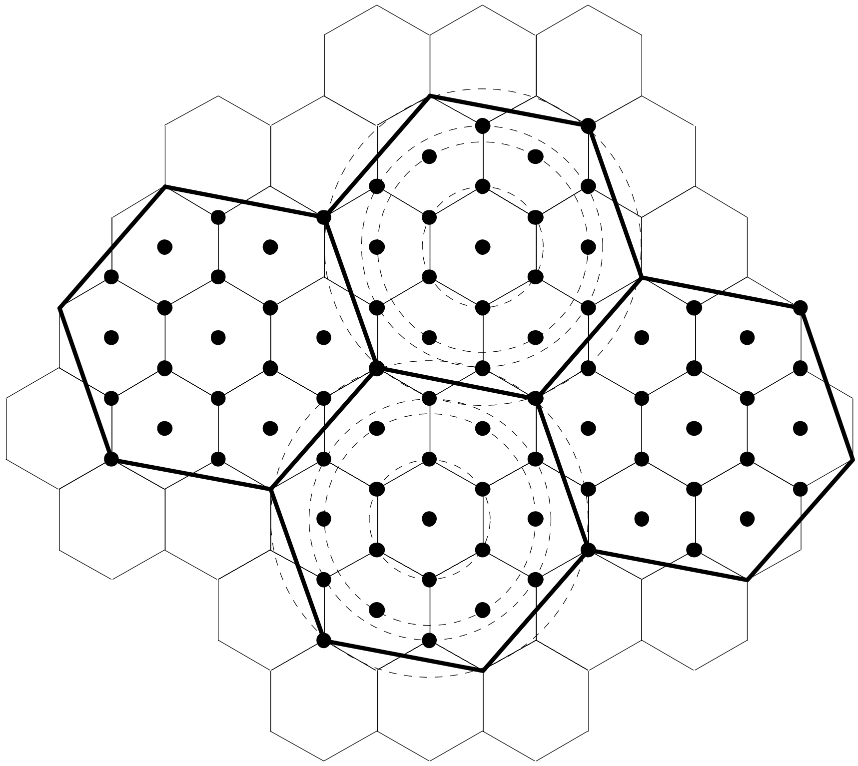


Figure 5.9: SFN with 21 Sub Identifiers for 4 different Main Identifiers

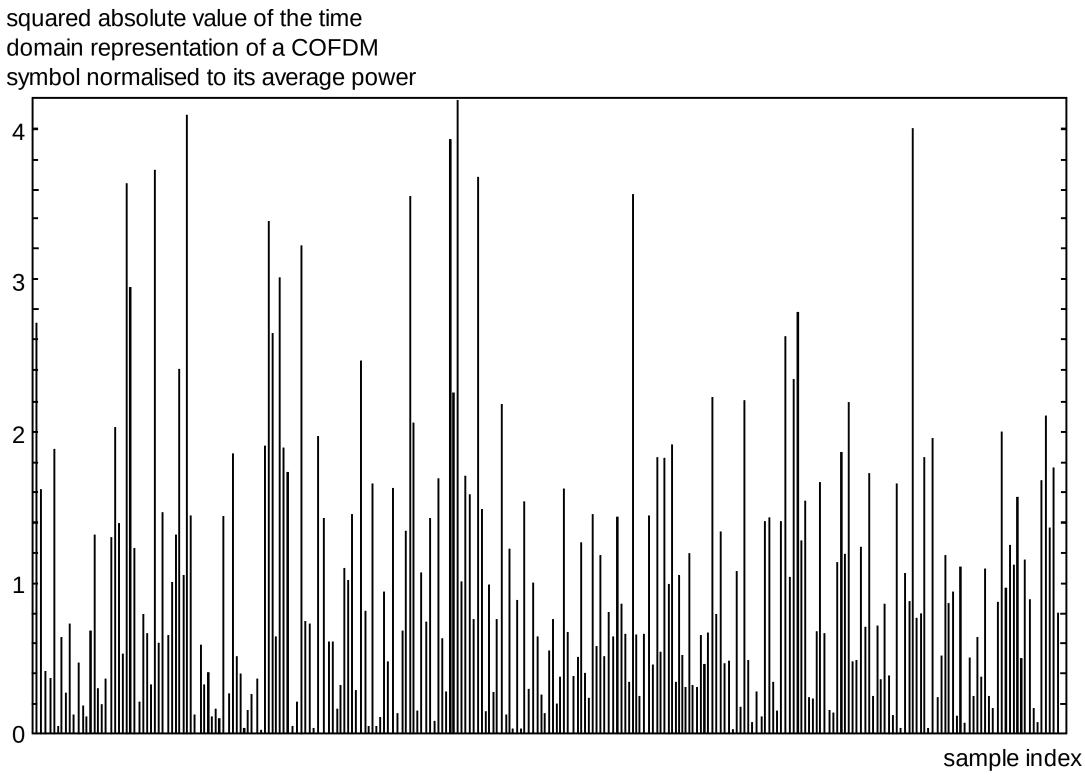


Figure 5.10: Time-domain representation of a COFDM signal

In theory the maximum possible amplitude would occur when all the carriers were simultaneously in-phase. In practice, limiting of the digital representation of the COFDM signal never allows this to happen. The result of the inevitable signal clipping and other non-linearity is that out-of-band components are generated at the output of the digital to analogue converter in DAB modulation equipment. These components are then filtered. The filtering process introduces some over-shoots and increases the peak amplitude. It is this clipping and filtering process which sets the actual peak amplitude which occurs. However, even this amplitude will only occur very infrequently.

Care should be taken that clipping and other non-linear effects within the transmitted signal do not degrade the overall performance to a significant level.

5.5.2 Frequency domain representation

The theoretical spectrum of a COFDM signal is shown in [1] for the four Transmission modes. The levels of the side-bands beyond the last active carrier frequency are the sums of the Sin x / x spectral distributions of the individual carriers.

Broadcasting the full theoretical spectrum would be impractical as it would cause interference to adjacent channel signals, both DAB and non-DAB. Therefore some filtering of the signal is needed. Although filtering of the side-bands destroys the orthogonality of the edge carriers, the consequent degradation is not significant.

There is the additional problem, identified in clause 5.5.3, that highly-linear power amplifiers operating with large amounts of headroom are required to prevent the generation of inter-modulation products (IPs). DAB transmitters engineered in such a way would be very expensive. However, if amplifiers are operated more efficiently the resulting generation of IPs is likely to restrict the use of adjacent DAB channels in certain cases. This decreases the efficiency of spectrum utilization and increases the problems of international co-ordination of DAB frequency allocations.

An important issue is therefore the definition of a suitable spectrum mask. This must be a compromise between the needs of frequency planners for efficient use of the spectrum and the needs of broadcasters for cost-effective transmitters.

5.5.2.1 VHF spectrum mask

As a result, a dual mask has been specified for VHF. The first mask would be used for transmitters in critical situations, where the adjacent frequency region needs specific protection. The second, less stringent, mask may be used for transmitters in situations with more relaxed requirements. Both masks are shown in figure 5.11. The vertical scale reflects the permitted out-of-band radiation levels in a 4 kHz bandwidth relative to the total power in a DAB frequency block.

Examples of the need for this dual-mask approach can be found in many situations. Consider the example of a simple SFN which allows a large number of transmitters in an area to operate on the same frequency. For the purpose of frequency co-ordination with another DAB service on the adjacent channel, the transmitters which are at the edge of the network, near the adjacent service area might require the more critical mask. The less stringent mask might be used for the other transmitters in the middle of the network, or in locations where a higher level of radiation into the adjacent channel could be tolerated (e.g. co-sited emission of adjacent channels). In cases where adjacent channel broadcasting occurs from co-sited transmitters, common amplification may be possible. This could permit a considerable relaxation of the requirements.

The less stringent spectrum mask indicates a potential allowance for additional radiation for 200 kHz on each side of the DAB frequency block (this may be required to allow signal conditioning strategies to be implemented). Beyond this point the mask requires the level of out-of-band radiation to drop very quickly to protect the services in the adjacent channel. The floor of the mask is set to provide an appropriate level of protection for sensitive services such as those used for aeronautical services. As the more stringent mask would be used in critical situations, it should be used as the basis for deriving protection ratios.

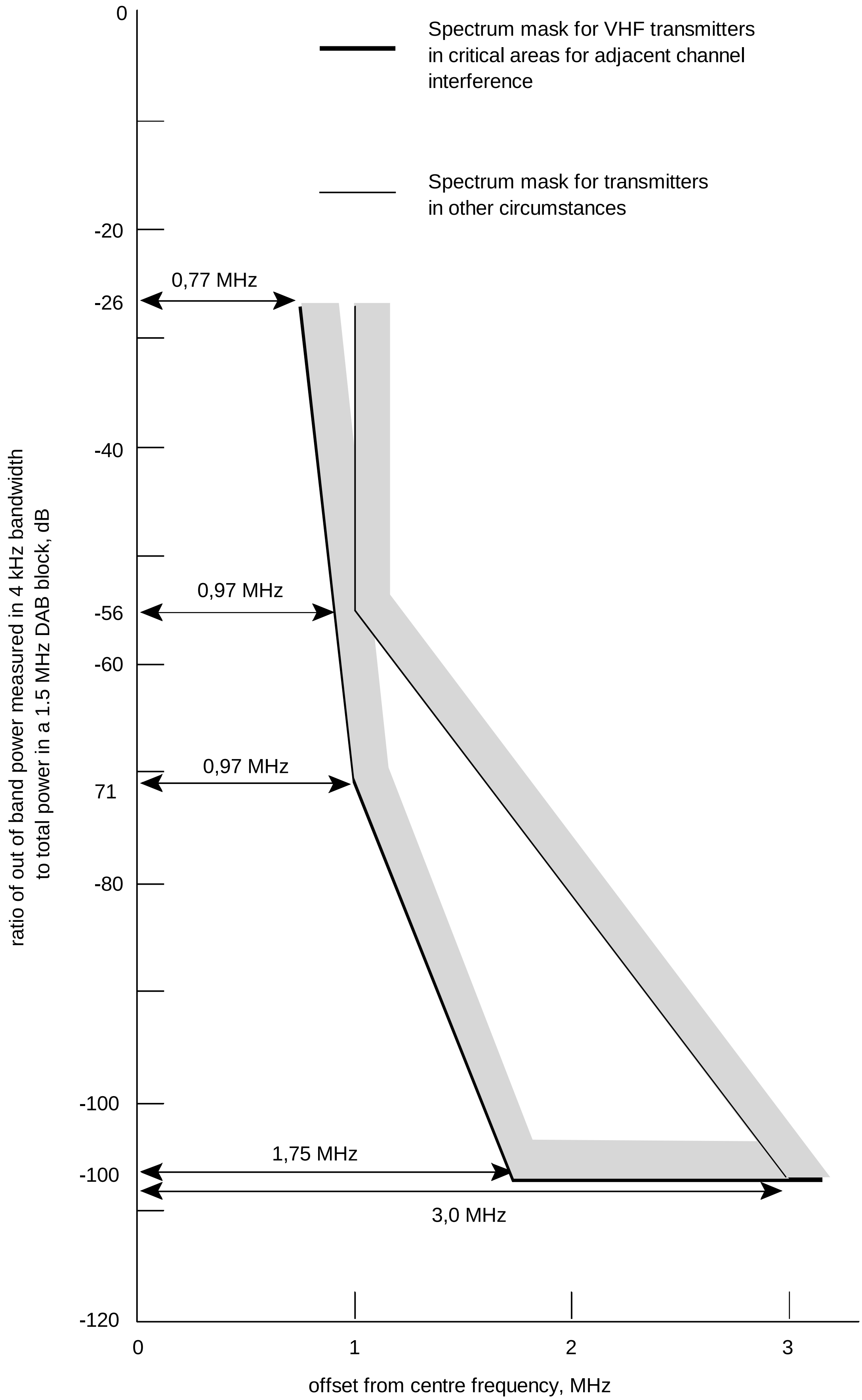


Figure 5.11: VHF spectrum mask

When implementing a transmitter which conforms to the spectrum mask the problem is to control the transmitter cost and the level of power radiated into an adjacent DAB channel, while generating the required output power. As the amplifier headroom is reduced, cheaper amplifiers can be used, but the level of IPs increases. However the amount of power radiated into the adjacent DAB channel can be controlled by introducing a higher-order filter, which may have a relatively high insertion loss and reduce the total power level of the wanted signal.

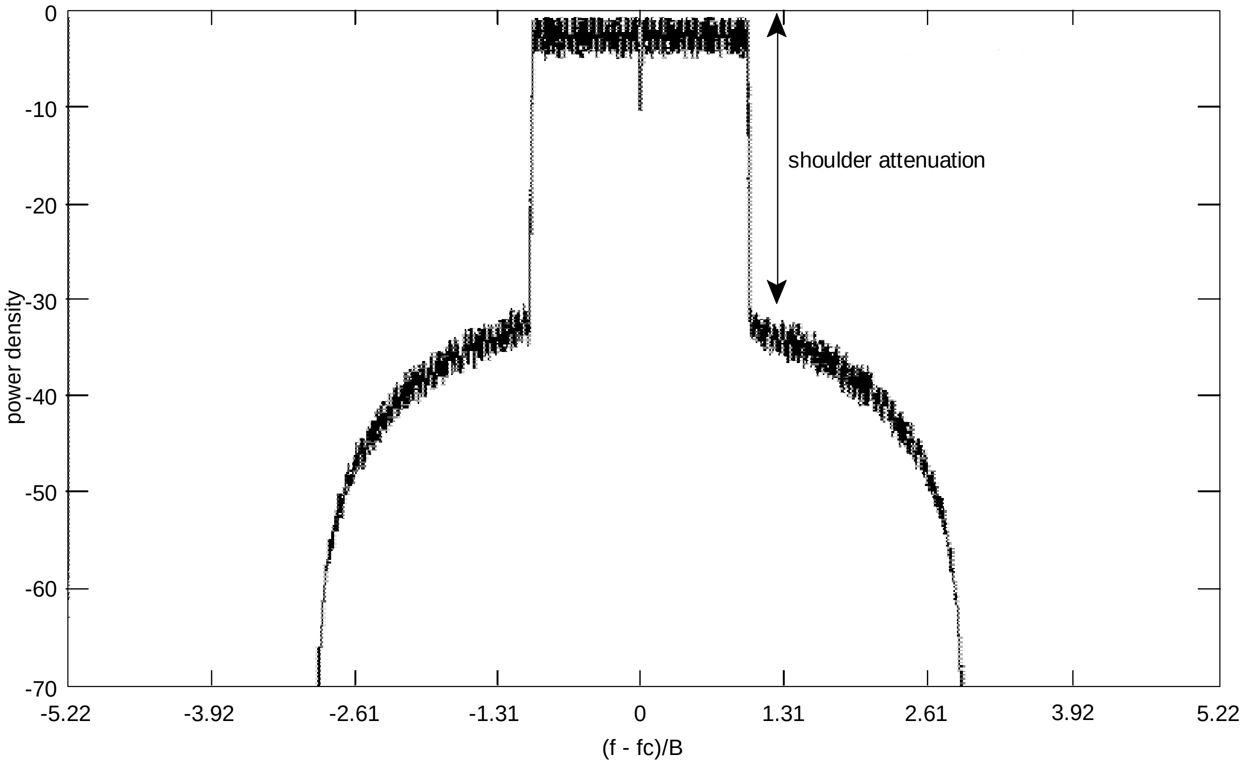


Figure 5.12: OFDM spectrum showing shoulder attenuation

The two key components are therefore the level of out-of-band radiation generated by the transmitter when operating at its nominal power and the additional suppression of this radiation by the output filter. The former is known as the "shoulder level" of the out-of-band radiation. From most amplifiers the shape of this out-of-band radiation has been found to decrease slowly with spacing from the last active carrier. This can be verified by theoretical analysis. The shape is shown in figure 5.12. Conventionally, its level has been measured as the ratio of the in-band to out-of-band power spectral density 200 kHz from the last active carrier in the block.

An initial consideration of amplifier and filter costs suggests that the most economical way of achieving a given level of out-of-band radiation at VHF is to use a high-order (relatively expensive) filter and the minimum possible level of amplifier headroom.

5.5.2.2 L-band spectrum mask

The same considerations apply to the specification of masks for terrestrial L band transmitters. In this case, somewhat more relaxed masks are required which reflect the greater difficulty of fabricating filters at these frequencies. The appropriate masks are shown in figure 5.13.

For satellite broadcasting at L band masks are also required. However, these are still under discussion.

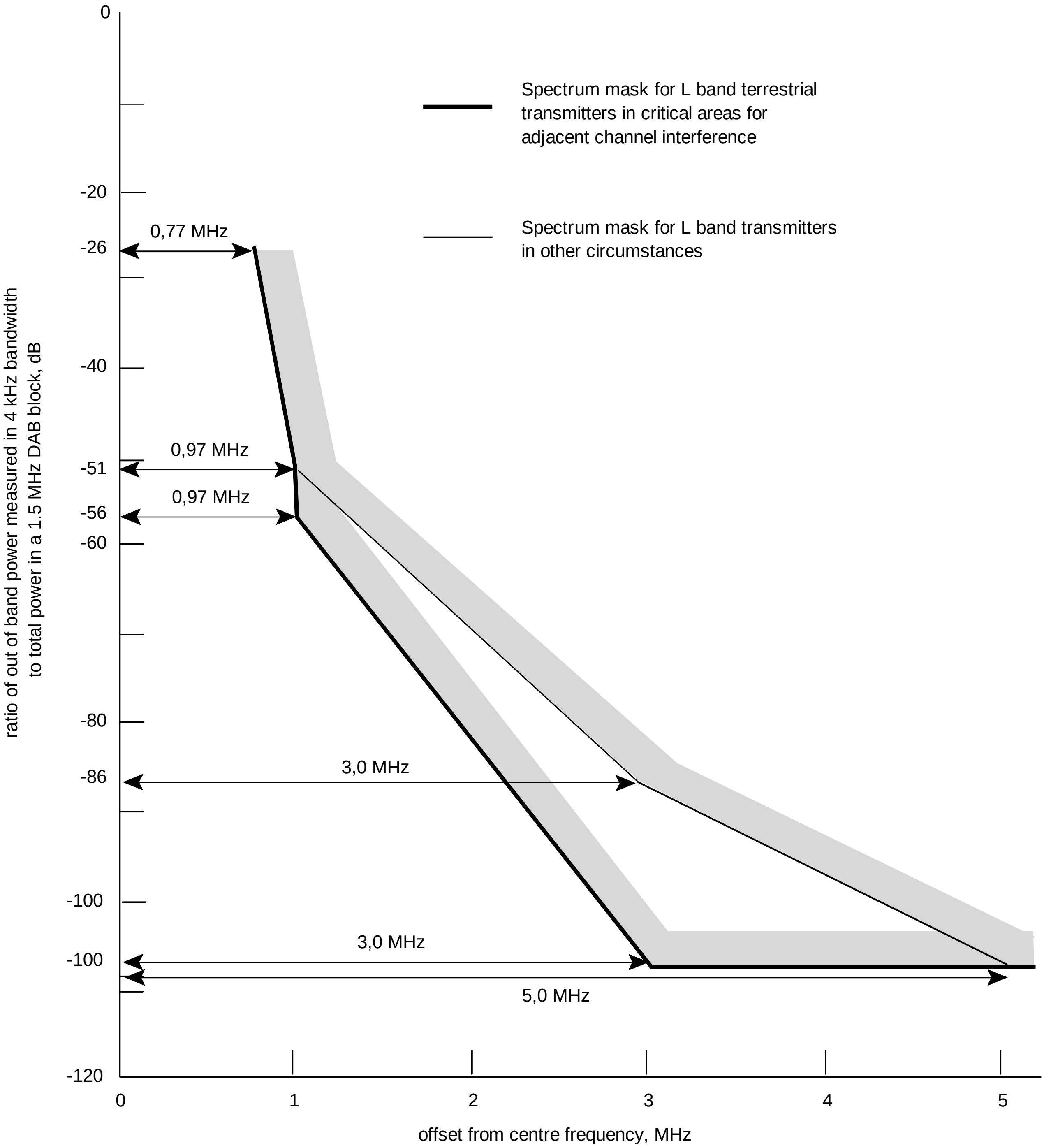


Figure 5.13: L-Band spectrum mask

5.5.3 Amplifier non-linearities

The signal chain can generate non-linearities in many different places. Examples are clipping in the digital representation of the COFDM signal and non-linear responses in analogue components such as the power amplifier in the transmitter.

Non-linearity in the transmitting (or receiving) equipment has two effects on the signal. Firstly, it distorts the wanted signal producing phase and amplitude errors on the individual carriers, thus reducing the noise margin of the system. Secondly, it generates out-of-band intermodulation products which can affect the performance of the adjacent DAB blocks. In a frequency plan in which DAB blocks are closely spaced (e.g. 4 blocks in a band of 7 MHz as in Band III) it is the second effect which is expected to be the dominant problem.

Pre-correction techniques have been shown to improve the power efficiency of practical transmitters.

It is recommended that the performance of terrestrial power amplifiers be measured by noting the electrical efficiency of the power amplifier at a specified IP level.

5.5.4 Satellite Transmission

DAB broadcasting via satellite has been studied in some detail. Although at the time of writing there are no known proposals for a commercial service, a number of experimental transmissions have been performed using existing, non-broadcast satellites. These experiments included demonstrations of mobile reception in Australia (using the Optus B satellite) and in Mexico (using the Solidaridad satellite). Both of these satellites used frequencies immediately adjacent to the L-band allocation for DAB, although they had somewhat lower EIRP than might be expected from a genuine broadcast satellite. The success of these experiments is sufficient to demonstrate that provision of DAB services via satellite does not present major technical difficulties.

Both Optus and Solidaridad were geostationary satellites. Another satellite system that has been studied is the HEO (High Elliptic Orbit) type constellation. In Europe, the active HEO satellite would be received at elevation angles which are significantly higher than for geostationary satellites, and therefore line-of-sight is achieved for a higher percentage of locations. Thus the required link margin for the same service availability is lower, and may allow acceptable mobile reception at high latitudes, which is hard to achieve with a geostationary satellite.

A satellite link is usually power limited and therefore the target is to maximize the efficiency of the SSPA (Solid State Power Amplifier). A satellite output stage is likely to be implemented as a phase shift network driving several power amplifiers which feed a matrix network. This matrix finally feeds a direct radiating antenna. This set-up generates several spot beams with independent ensembles.

The design criterion for the satellite output stage has been a net signal-to-noise ratio of 15 dB, where in this case the noise power is predominantly due to intermodulation in the output stage itself. Each DAB ensemble will go through each power amplifier and the last filter(s) pass(es) the whole band. This combination causes a wide spectrum with relatively high out-of-band levels. However, this may be acceptable because many satellite channels may be transmitted from the same satellite, one ensemble per transponder, and the power flux density (pfd) level on earth is very low in every case, about -113 dBW/m^2.

The optimum operating point for a SSPA is presented below for a simplified satellite output stage. It includes one DAB ensemble going through one SSPA stage. This exercise gives an order of magnitude estimate for the additional margin to the link budget which is required by the non-linear component using a COFDM-like signal.

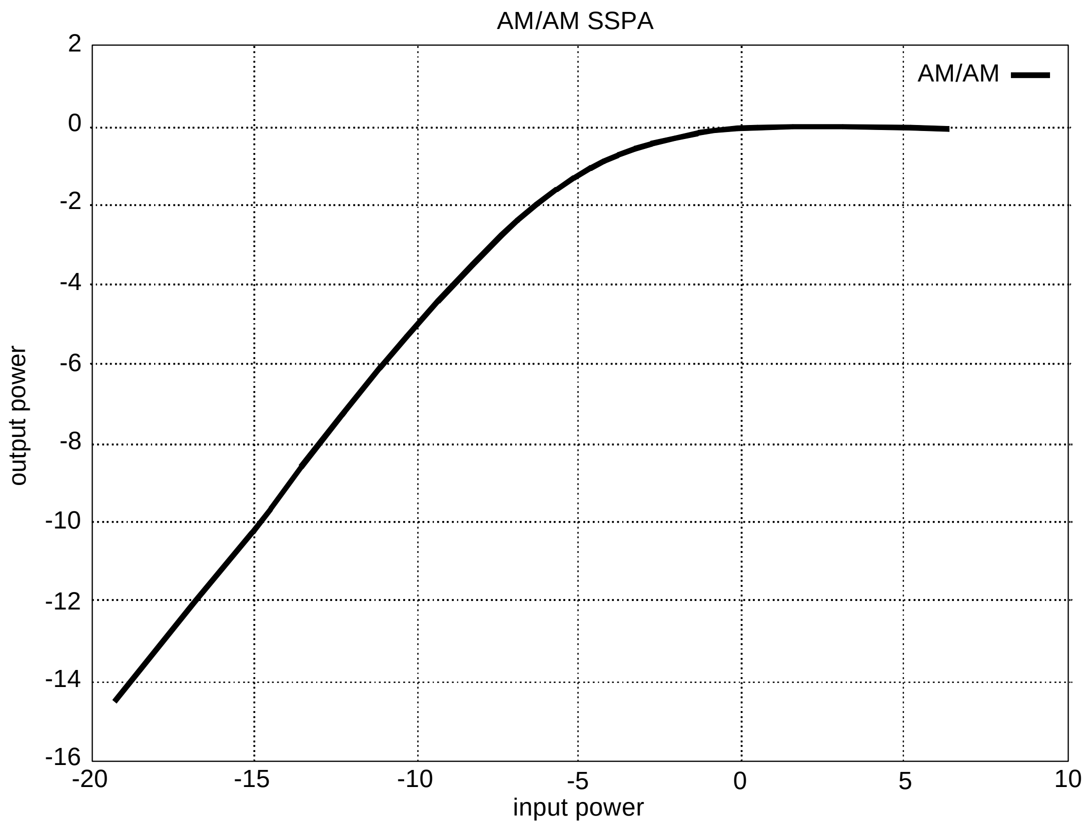


Figure 5.14: AM/AM curve of a typical SSPA

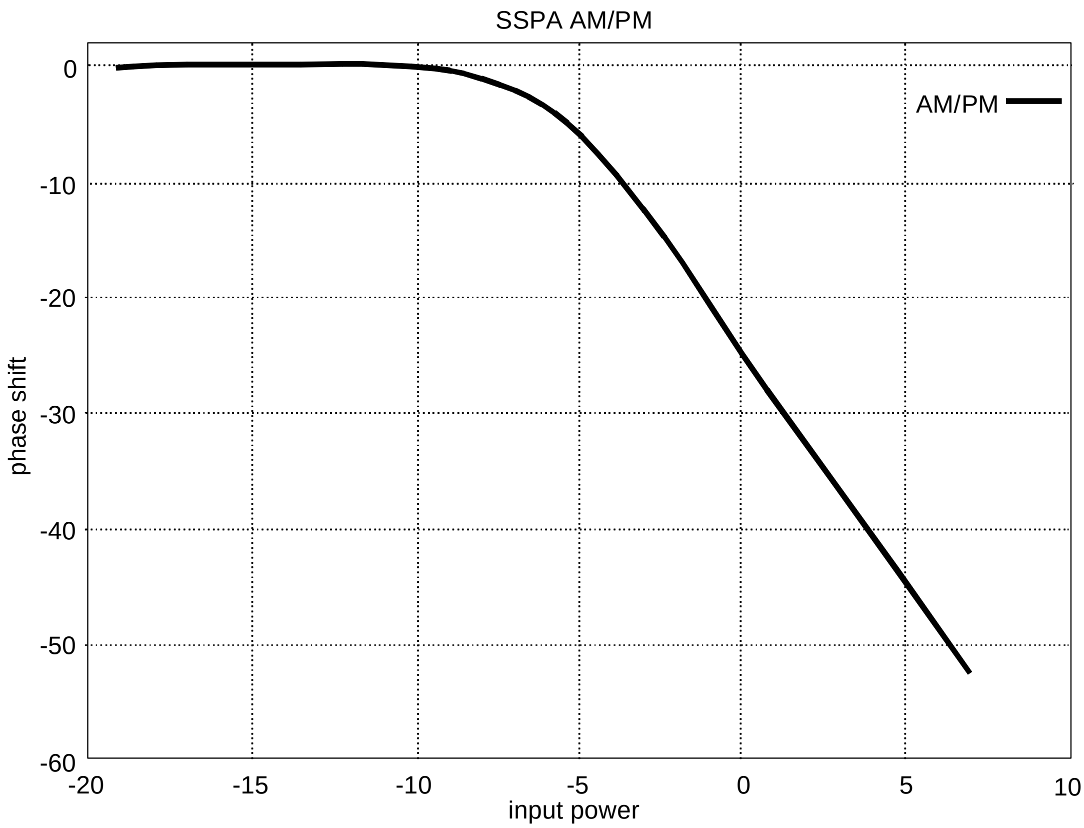


Figure 5.15: AM/PM curve of a typical SSPA

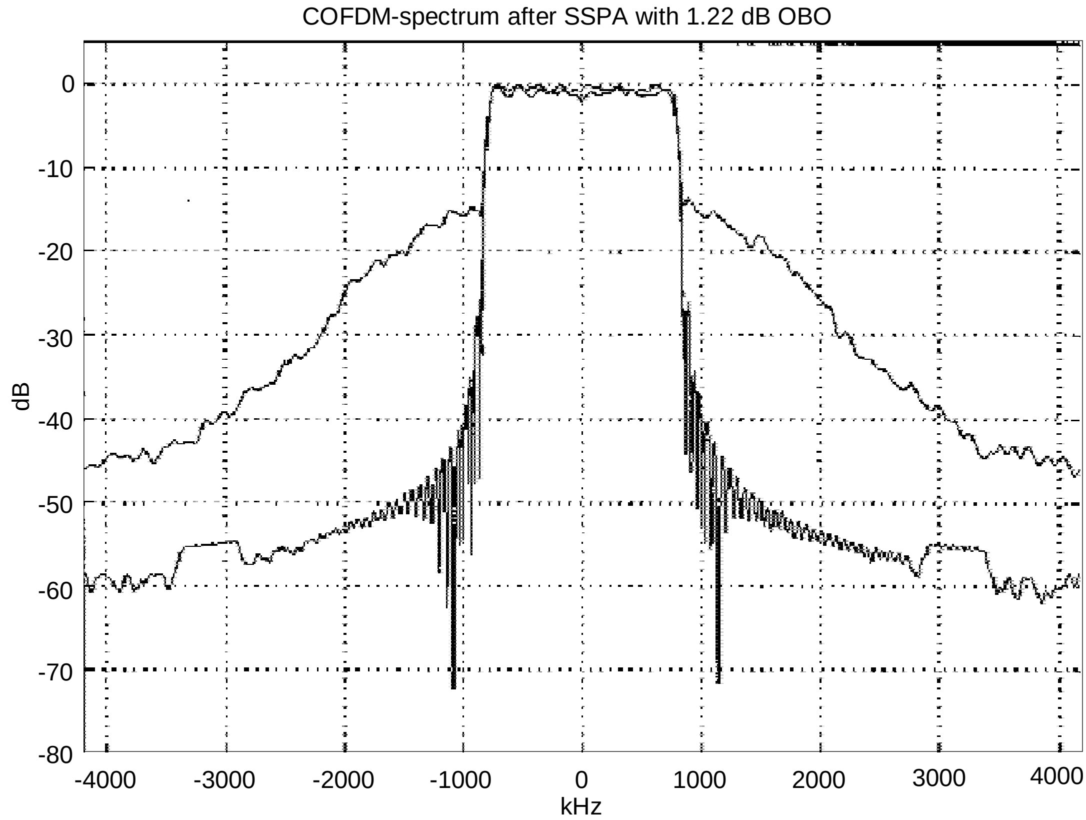


Figure 5.16: Spectrum of a COFDM signal before and after SSPA (see text)

The AM/AM and AM/PM values for a typical SSPA (figures 5.14 and 5.15) have been used in the simulation to find an optimum operation point. The service limit was defined to be a BER value of 10^{-3} with a coding rate of 0,5. The signal is one 1,5 MHz block operating in Transmission mode III. The optimum operating point was found to be an OBO (Output Back Off) value of 1,2 dB at which the SSPA produced a performance loss due to non-linear distortion loss of 1,5 dB. The power spectrum with this OBO value is shown in figure 5.16 with a comparison to the linear power spectrum. The spectrum of the distorted signal has a shoulder attenuation of only 15 dB. This has to be considered when defining the spectrum mask for satellite transmissions. The resulting interference level should not degrade reception in Gaussian and Ricean channels.

Satellite DAB has the advantage that terrestrial gap-fillers can operate on the same frequency, because the properties of COFDM allow the delayed signal from the gap-filler and the direct signal from the satellite to be successfully received. However, certain considerations apply to such gap-fillers.

The first consideration is that their output signals have the same power spectrum as the output of the satellite transponder, unless additional filtering is used. It has to be checked that the nearest services outside DAB satellite transmission band can tolerate the signal levels expected from the gap-fillers. On the other hand, when the non-critical L-band spectrum mask is used (see figure 5.13), there is a danger that a terrestrial DAB signal, even at moderate power, would cause significant interference to an adjacent satellite service. Thus, sufficient guard bands between terrestrial and satellite services are required.

5.5.5 Preferred frequencies for DAB

The Eureka 147/DAB specification permits a large number of centre frequencies to be used. To simplify receiver operations, receivers should scan a sub set of these frequencies as a matter of priority when the receiver is switched on.

It is recommended that these frequencies are preferred to all others in any frequency planning procedure. The options take into account alternatives which may be needed to use spectrum efficiently under a range of sharing scenarios.

The recommended frequencies are shown in tables 5.5.1, 5.5.2 and 5.5.3.

Table 5.5.1: Band 1 frequencies given priority in Eureka 147 receivers

Channel		Frequency (MHz)
2	A	47,936
2	B	49,648
2	C	51,360
2	D	53,072
3	A	54,928
3	B	56,640
3	C	58,352
3	D	60,064
4	A	61,936
4	B	63,648
4	C	65,360
4	D	67,072

Table 5.5.2: Band 3 frequencies given priority in Eureka 147 receivers

Channel		Frequency (MHz)
5	A	174,928
5	B	176,640
5	C	178,352
5	D	180,064
6	A	181,936
6	B	183,648
6	C	185,360
6	D	187,072
7	A	188,928
7	B	190,640
7	C	192,352
7	D	194,064
8	A	195,936
8	B	197,648
8	C	199,360
8	D	201,072
9	A	202,928
9	B	204,640
9	C	206,352
9	D	208,064
10	A	209,936
10	B	211,648
10	C	213,360
10	D	215,072
10	N	210,096
11	A	216,928
11	B	218,640
11	C	220,352
11	D	222,064
11	N	217,088
12	A	223,936
12	B	225,648
12	C	227,360
12	D	229,072
12	N	224,096
13	A	230,784
13	B	232,496
13	C	234,208
13	D	235,776
13	E	237,488
13	F	239,200

Table 5.5.3: L-Band frequencies given priority in Eureka 147 receivers

NOTE: This table was corrected for an error in the channel name entry from Channels LI onwards 11/9/97.

Channel		Frequency (MHz)
L	A	1 452,960
L	B	1 454,672
L	C	1 456,384
L	D	1 458,096
L	E	1 459,808
L	F	1 461,520
L	G	1 463,232
L	H	1 464,944
L	I	1 466,656
L	J	1 468,368
L	K	1 470,080
L	L	1 471,792
L	M	1 473,504
L	N	1 475,216
L	O	1 476,928
L	P	1 478,640
L	Q	1 480,352
L	R	1 482,064
L	S	1 483,776
L	T	1 485,488
L	U	1 487,200
L	V	1 488,912
L	W	1 490,624
L	1	1 452,816
L	2	1 454,560
L	3	1 456,304
L	4	1 458,048
L	5	1 459,792
L	6	1 461,536
L	7	1 463,280
L	8	1 465,024
L	9	1 466,768
L	10	1 468,512
L	11	1 470,256
L	12	1 472,000
L	13	1 473,744
L	14	1 475,488
L	15	1 477,232
L	16	1 478,976
L	17	1 480,720
L	18	1 482,464
L	19	1 484,208
L	20	1 485,952
L	21	1 487,696
L	22	1 489,440
L	23	1 491,184

5.5.6 Expected Receiver Performance

5.5.6.1 General

The European Norm EN 50248 [12], "Characteristics of DAB Receivers", gives methods of measurement of the characteristics of DAB receivers. Targets for sensitivity are shown in TR 101 758: DAB signal strengths and receiver parameters - targets for typical operation.

The following clauses refer to measurements conducted on early prototype receivers in the early 1990s.

5.5.6.2 Amplifier Linearity and Selectivity

Non-linearities of amplifiers, mixer stages etc. produce a distortion of the signal itself (amplitude and phase errors) and subsequently interfering products inside and outside the transmitted band. These effects may cause additional bit errors and limit the maximum input power of a receiver (and hence its dynamic range). Further, the out-of-band products may influence the selectivity of the receiver, if the signal in the adjacent channel is of the same power or stronger than the wanted signal. In this case the interference, generated by non-linearities within the receiver, lies inside the wanted channel and cannot be removed by subsequent filtering.

In order to estimate the required selectivity, three cases of interference may be taken into account:

1) The wanted signal is embedded in a block of DAB signals, each 1,54 MHz wide, and the complete block is received from a single transmitter.

2) The wanted signal is received together with an adjacent DAB signal (or signals) which is (are) derived from other transmitter(s).

3) The wanted signal is received together with an adjacent signal from another service, e.g. TV - normally received from another transmitter or at least from another transmitter antenna.

Despite fading effects of the channel, the DAB signals in case 1 can be assumed to be of comparable power, a case which is less demanding than cases 2 and 3, where higher levels in the adjacent channel have to be taken into account. The difference in level is normally less of a problem if bigger amplitudes from the wanted signal or transmitter are received, because this implies a position nearer to that transmitter, whilst the signal from the adjacent-channel transmitter is assumed to be in the same order as before. Thus, the requirements on selectivity can be reduced at higher wanted-signal levels and a maximum interfering level may be defined. The requirements will further depend to some extent on the conditions under which a DAB service is installed. This may differ from country to country.

The results of measurements using third generation prototype receivers, achieved by applying one SAW filter of 1/2 inch chip size and base-band filters of degree 5, may be taken as a guideline. The large-signal behaviour of the front-end was identical to that of present TV tuners.

The measurement conditions were: both wanted and interfering signals are DAB-like with 1,54 MHz bandwidth; centre frequency separation of 1,7 MHz corresponding to about 0,17 MHz guard-band. Noise level adjusted to give a BER of 10^{-5}; interferer set to a level which increases the BER to 10^{-4}.

- Wanted-signal Selectivity
- input level (BER = 10^{-4})
- -90 dBm 46 dB
- -60 dBm 36 dB

The selectivity at higher input levels is already reduced by non-linearities in the input stages. If possible, a value greater than 40 dB should be provided even at higher levels.

5.5.6.3 Dynamic Range

The input signal range is limited at lower levels by the noise of the receiver input stages and at higher levels by the AGC range and subsequently by non-linearities of the complete chain up to the decoder.

Of course, the lower bound of the dynamic range should be as good as possible. Noise figures of 3 to 6 dB (typical) are standard for FM receivers and TV front-ends, and a noise figure of 6 dB is desirable for a DAB receiver in Band III. This will be primarily due to the receiver front-end alone, because of the relatively low attenuation of the antenna cables. For L Band, a *net* noise figure of 6 dB may be appropriate but this may be the composite some or all of the following contributors: an active antenna's noise figure, cable attenuation and the receiver noise figure. Care has to be taken that these values, under practical circumstances, are not degraded by self interference caused by the digital part of the receiver.

The requirements for the upper bound can be derived from the maximum field strength of transmitters which have to be considered for a DAB service. It can be expected that the maximum transmitter power in Band III will be at least 10 dB less than in FM networks due to the better behaviour of the digital (DAB) system. Thus, the receiver input levels to be expected should also be 10 dB less compared with the highest values known from FM, which results in values around 0 dBm. L-band systems may be characterized by physically smaller receiving antennas, and possibly also by transmitting antennas employing shaping of the vertical radiation pattern. These factors may allow a further reduction in the requirement for maximum input power, to around -15 dBm, but again both active antenna and receiver, as appropriate, will need to handle such power levels.

The dynamic ranges given in table 5.6 were achieved by prototype receivers which, in terms of their handling of large signals, were identical to present TV tuners:

Table 5.6: Input dynamic range of some prototype DAB receivers

	Average value	Poorest value	
Mode I			
Band I	-98 to -3	-95 to -10	dBm
Band III	-96 to -1	-93 to -10	dBm
Mode II	-95 to -1	-93 to -10	dBm
Mode III	-95 to -1	-92 to -10	dBm
Mode IV	No figures available		

For the selection of receivers tested, maximum input level values of about 0 dBm were only achieved by some. Unfavourable values, down to -10 dBm, were tolerable because of the reduced power values used by field test transmitters (in comparison to maximum values in a real service). Investigations have shown that an improvement of the upper values up to 0 dBm is possible.

The final values at lower levels may be influenced by noise and interference from the digital part of the receiver itself which is picked up by the antenna. This has the consequence that the degree of interference depends on the type of the receiver, with portable receivers likely to be most vulnerable. Car radios may also be affected by other in-vehicle electrical systems.

5.5.6.4 Miscellaneous

1) For proper receiver synchronization the delay spread of the received multi-path signal should not exceed the guard interval (e.g. 246 µs in mode I, 62 µs in mode II, 31 µs in mode III, and 123 µs in mode IV). A receiver is assumed to make full use of the guard interval for minimizing ISI.

2) Laboratory measurements on prototype receivers (see table 5.3) show typical input sensitivities of about -96 dBm and worst case values of about -93 dBm in a Gaussian channel for the VHF/UHF transmission bands. An average sensitivity of -93 dBm has been measured in Band III field trials in areas free from man-made noise. These receivers were based on television tuners and receivers designed for reception of DAB can be expected to offer superior performance. This would be offset to some extent by a margin to encompass receiver production tolerances. The noise figures noted above (~6 dB) are consistent with a receiver sensitivity of the order of -97 dBm.

3) Future receivers may have the ability to detect and suppress the effect of CW and narrow band interferers. Experimental work has demonstrated a substantial improvement may be obtained in the case of a single interferer.

5.6 Broadcast Network Planning Techniques

By its nature, broadcasting is a point-to-multi-point service. Techniques have been developed for analogue TV and Radio services to permit the planning of the location and other parameters of transmitters to serve areas of population. A simplified, agreed set of techniques are also used for the international allocation of frequencies and co-ordination of transmitters. However, additional considerations and techniques are important for DAB, because of the digital nature of the signal and the use of Single Frequency Networks. Both conventional techniques and some additional considerations for DAB are described in this clause.

All broadcast networks can be noise or interference limited. In practice, both apply in different areas of the network. However, in areas where the spectrum is intensively used, networks tend to be more interference limited. Therefore, most of the following discussion will concentrate on interference limitations.

Protection ratios for co-channel and adjacent channel interference in conjunction with propagation data for the radio waves are the basis for transmitter frequency co-ordination. One of the key figures in conventional planning is the reuse distance for a given frequency. This is the distance between transmitters operating at the same frequency, that is necessary to reduce the co-channel interference to the minimum level indicated by the protection ratios. Protection ratios indicate the level of interference that is permissible in order to maintain a certain minimum service quality. As radio propagation is time variant, worst case situations have to be considered. Normally, the planning techniques aim to provide the minimum service quality for 99 % of time at the edge of the coverage area. Therefore better reception conditions are achieved at lower percentages of time, especially inside the service borders.

Propagation models do not allow exact prediction of the field strength at a given location. Only the median value, the standard deviation and the shape of a statistical distribution function are derived. Conventional planning techniques only take into account the median values for the wanted signal and use a pragmatic formula for the calculation of the median value of the total of the various interfering signals. Wherever the predicted ratio of the wanted signal to the interfering signal meets the protection ratio, the location is considered to be served.

It is important to note that planning according to such a method cannot avoid situations where the quality requirement is not reached. In principle, at the edge of the coverage area of a single transmitter, only 50 % of the locations offer signal conditions which fulfil the protection ratio requirement. The other half of locations suffers from wanted signals that are too low or from interfering signals that are too high.

5.6.1 Planning of Conventional Networks

Conventionally planned broadcasting networks consist of transmitters with independent programme signals and with individual radio frequencies. (In contrast to a SFN, the transmitters do not have to obey strict rules of synchronous emission.) The allocation of the radio frequency for each transmitter needs thorough calculation of the mutual interference of all transmitters inside and outside the network.

In analogue services a small violation of the protection ratio only results in a small degradation of the service quality. As the degradation is also limited in time (see above) it is considered to be an economical compromise. For example, the FM service offers a margin of about 30 dB between a small loss of quality and total service interruption. Only in rare cases will strong degradation result in a complete break down of the service. Car reception makes use of this robustness of the FM service as far as intelligibility is concerned.

However, with digital transmission, a relatively abrupt break occurs when the RF signal conditions do not fulfil the protection ratio requirements. Therefore, within the service area, the RF signal conditions must be satisfied for a high percentage of locations, say 99 %, as well as for a high percentage of time, say also 99 %.

These considerations lead to the important result that traditional conventional planning methods cannot be used directly for planning digital services. Modifications are necessary to account for the differences in behaviour of analogue and digital systems. One special example of the inappropriate use of the old planning rules may be worth considering. It is well known in conventional planning that the protection ratio for co-channel interference determines the reuse distance for the RF channel. The lower the protection ratio the smaller the reuse distance. As the protection ratio of DAB is about 25 dB lower than the protection ratio of FM a straight forward conclusion could be that the reuse distance in the DAB case is considerably smaller than in the FM case. As a consequence, the number of RF channels needed to cover large areas with at least one RF signal would seem to be considerably smaller than in the FM case. However, using this planning method, with its 50 % coverage criterion at the edge of the service area, in conjunction with the "brick-wall behaviour" of a digital system, would leave a large part of the envisaged service area unserved. In conventional planning, the only way to avoid such gaps in the service area is to enlarge the distance for frequency re-use. Adding a margin to the protection ratio is a pragmatic way to achieve the wanted effect with the conventional planning procedure.

The appropriate value for the margin can be derived from the statistical model of wave propagation; typical values under consideration are as large as 20 dB. In this case the sum of the protection ratio and the margin is in the order of 33 dB. For FM the respective value of the protection ratio is 37 dB. Thus the number of channels needed for conventional planning of DAB may not be so much less than for planning analogue (FM) services.

As a DAB channel is much wider than an FM channel the total bandwidth necessary for a conventionally planned DAB network could lead to substantial frequency bandwidth requirements. However, this bandwidth permits a much larger number of services to be provided in each area, and as a result the utilization of the spectrum remains about the same. Consequently, the principle of conventional network planning is useful for local services which are restricted to a small part of the country. Here the frequency-reuse distance is the ruling figure for international frequency allotments. The detailed planning of the network, i.e. the exact transmitter locations, ERP etc., is then performed using more sophisticated, terrain-based techniques.

5.6.2 Single Frequency Network

In an SFN, all transmitters are synchronously modulated with the same signal and radiate on the same frequency. This network concept offers a much higher spectrum efficiency than a conventionally planned network.

With the SFN technique large areas can be served with a common ensemble at a common radio centre frequency. Therefore the frequency efficiency of SFNs seems to be very high compared to conventionally planned networks. However, taking into account the presence of similar networks offering other ensembles in adjacent areas, further DAB channels are required for international frequency co-ordination. If all service areas are large enough, in theory four different channels are sufficient to provide any of the areas with its individual ensemble (see figure 5.17 with DAB channels A, B, C and D). Each DAB channel can be re-used in the next but one area if the respective re-use distance is not less than about 100 km. However, in almost all practical situations, the location of transmitter sites, local terrain, lower re-use distances and a number of other factors may combine to require the use of occur a fifth, sixth or even a seventh channel.

A	C	A	C	A
B	D	B	D	B
A	C	A	C	A
B	D	B	D	B
A	C	A	C	A

Figure 5.17: Assignment of SFN blocks to regions

Inside large areas the frequency channels of adjacent areas can be re-used if the rule of re-use distance is obeyed. After the frequency co-ordination of large areas a fine co-ordination of frequencies may result in additional allocations for local services.

The SFN technique is not only frequency efficient but also power efficient. This can be explained by considering the strong local variations of field strength of any given transmitter. In conventionally planned networks, a common way to achieve service continuity at a high percentage of locations is to include a relatively large fade margin in the link budget and thus to increase the transmitter power significantly. However in SFNs, where the wanted signal consists of many signal components from different transmitters the variations of which are only weakly correlated, fades in the field strength of one transmitter may be filled by another transmitter. This averaging effect results in smaller variations of the total field strength. According to these considerations, SFNs tend to have relatively low powered transmitters. Typically the e.r.p. is below 10 kW. This power efficiency of an SFN is often referred to as Network Gain.

The price to be paid for this frequency and power efficiency is the need for synchronous operation of all transmitters in a given network. In networks using Transmission mode I, tolerances of ±5 μs should cause little or no performance degradation. This requirement of synchronous transmitter operation has significant impact on the strategies of assembling ensembles and their distribution to the transmitters. The previous clause deals with this in some detail.

When a demodulator receives signals from multiple transmitters, they appear like echoes of one original signal. The delay spread of such a "virtual" channel depends upon the distance between the transmitters and the free space attenuation, which is itself a function of the frequency. A distance of 1 km is equivalent to a propagation delay of about 3,3 μs.

OFDM systems can be adapted to different multi-path environments by changing their three main parameters. These are the inter-carrier distance F, the guard interval Δ, and the symbol duration T. In order to obtain the desired orthogonality of the OFDM carriers, these parameters must fulfil the relation

$$(T - \Delta)F = 1$$

DAB provides a bandwidth and a data rate which are independent of the selection of the OFDM parameters. This requires the ratio Δ/T to be constant. Hence, only one degree of freedom remains, e.g. the guard interval can be fixed according to the delay spread of the radio channel, and the other parameters depend upon that selection. Alternatively, the inter-carrier spacing, F, can be fixed according to the Doppler spread for a given vehicle speed and frequency, in which case the other parameters will depend on that selection. This latter approach is described in depth in the following clause.

OFDM systems may tolerate long echo delays if their parameters are chosen accordingly, i.e. the guard interval is sufficiently large. In contrast to single carrier systems, where the echo rejection capabilities are determined by the length of an equalizer (the complexity of which significantly increases with the delay spread of the channel), the multi-path resistance of OFDM only depends upon the guard interval which does not influence the demodulator complexity.

The differential delay of two signals from adjacent transmitters must be, at most, smaller than the guard interval. Additional headroom for synchronization and normal multi-path propagation should be provided. This results in recommendations for maximum transmitter distances (see TR 101 496-1 [8] for the recommended maximum distances for the three Transmission modes of the DAB System).

It should be noted that these values are examples from studies of theoretical, regular lattices of transmitters which are synchronously modulated and have uniform powers. Practical transmitter spacings are variable and depend upon topographical considerations. Synchronous transmission can be achieved by the methods described in clause 4. However, there are also some situations where transmitters should be purposely delayed or advanced, compared to ideal synchronicity, in order to improve the coverage. This is particularly true if the SFN includes both high and low power transmitters. Because of its high power, the signal of a strong transmitter can be received at a large distance from it. There, it may be superimposed on the signal of a much weaker but closer transmitter. Because of the different propagation distances, the signal of the weak transmitter would arrive much earlier than that of the strong transmitter, if both were synchronized. This differential delay can be reduced by advancing the strong transmitter relative to the SFN mean delay, or conversely by delaying the weak transmitter.

Gaps in the coverage area of an SFN are easily filled by adding one new transmitter without the need for additional frequencies. This technique offers a very efficient spectrum utilization, especially in large area networks for national or regional service coverage. This is true as long as the whole DAB ensemble is filled by services with the same required coverage area.

5.6.3 Calculation of the vehicle speed at which DAB reception becomes degraded

Whenever a DAB signal is received in a moving vehicle, especially when there is multipath propagation, there is likely to be some degradation of performance. The COFDM signal has been optimized to ensure that under normal circumstances, reception is satisfactory. However, it is useful to understand the conditions under which reception starts to be degraded.

Most analysis of the performance can be related to a parameter β (See for example Le Floch et al, IEEE Proceedings CE35, No3 pp 493 et seq.). β can be interpreted as the representation of the displacement of the vehicle expressed in number of wavelengths during one symbol duration T_s when the vehicle has a speed of v (metres per second).

$$\text{Let } \beta = f_{max} \times T_s = (v \times f_o / c) \times T_s = v \times T_s / \lambda \qquad (5.1)$$

The reference value for β is 0,08 for a 4 dB degradation at approx. 10^{-3} BER in the most difficult multipath conditions (dispersive Doppler effect, constant probability density of the received power over the 2π range of reception in the horizontal plane, as opposed to a simple Doppler shift). Putting these figures into equation 5.1:

$$\beta = T_s \times f_o \times v \ / \ c = 0{,}08$$

$$(T_u + \Delta) \times f_o \times v \ / \ c = 0{,}08$$

$$(T_u + T_u / 4) \times f_o \times v \ / \ c = 0{,}08$$

$$(5T_u / 4) \times f_o \times v \ / \ c = 0{,}08$$

$$T_u \times f_o \times v \ / \ c = 0{,}064 \qquad (5.2)$$

Equation (5.2) represents the speed versus frequency curves with the symbol duration *T*u as a parameter. This is a function of the Mode.

For c = 3×10^8 m/s equation (5.2) we obtain:

$$T_u \times f_o \times v = 0{,}064 \times 3 \times 10^8 = 19{,}2 \times 10^6 \qquad (5.3)$$

with:

T_u = useful symbol duration in seconds;

f_o = frequency in Hz and v = vehicle speed in m/s.

When in equation (5.3), f_o is expressed in MHz and v in km/h then:

$$T_u \times f_0 \times v = \frac{0{,}064 \times 3 \times 10^8 \times 3\,600}{10^6 \times 10^3} = 70 \ \text{(approx)} \qquad (5.4)$$

so that:

$$v = 70 \ / \ T_u \times f_o \quad \text{and} \qquad (5.5)$$

$$f_o = 70 \ / \ T_u \times v \qquad (5.6)$$

By means of equation (5.5) the maximum speed can be calculated that is possible at a certain frequency. By means of equation (5.6) the maximum frequency can be calculated that is possible at a certain vehicle speed.

EXAMPLES:

1) **Calculation of the maximum speed v that is possible in the 4 modes I,II,III, IV for a nominal frequency f_o of 375 MHz, 1,5 GHz, 1,5 MHz and 3 GHz respectively.**

-- Mode I: T_u = 1 ms = 0,001 s and f_o = 375 MHz

From equation 5.5: the maximum speed is 70 / 0,001 × 375 = 186 km/h (point A in figure 5.18)

-- Mode IV: T_u = 500 ms = 0,000 5 s and f_o = 1,5 GHz = 1 500 MHz

From equation 5.5: the maximum speed is 93 km/h (point B in figure 5.18)

-- Mode II: T_u = 250 ms = 0,000 25 s and f_o = 1,5 GHz = 1 500 MHz

From equation 5.5: the maximum speed is 186 km/h (point C in figure 5.18)

-- Mode III : T_u = 125 ms = 0,000 125 s and f_o = 3 GHz = 3 000 MHz

From equation 5.5: the maximum speed is 186 km/h (point D in figure 5.18)

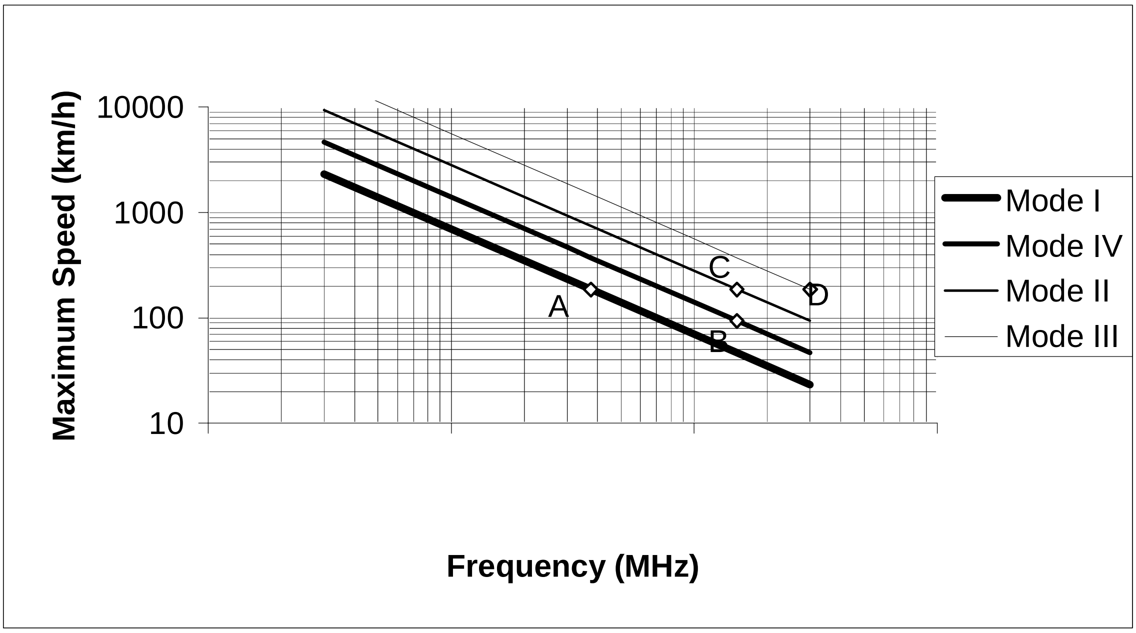


Figure 5.18: Maximum Speed versus Frequency for Beta = 0,08

2) Calculation of the max. speed at 100 MHz

Using equation 5.5:

	T_u	maximum speed	
Mode I	0,001 s	70 / 0,001 x 100	= 700 km/h
Mode IV	0,000 5 s	70 / 0,000 5 x 100	= 1 400 km/h
Mode II	0,000 25 s	70 / 0,000 25 x 100	= 2 800 km/h
Mode III	0,001 25 s	70/0,000 125 x 100	= 5 600 km/h

3) Calculation of the max. usable frequency per mode when moving at a speed of 200km/h

Using equation 5.6:

	T_u	maximum frequency	
Mode I	0,001 s	70 / 0,001 x 200	= 350 MHz
Mode IV	0,000 5 s	70 / 0,000 5 x 200	= 700 MHz
Mode II	0,000 25 s	70 / 0,000 25 x 200	= 1 400 MHz
Mode III	0,000 125 s	70 / 0,000 125 x 200	= 2 800 MHz

4) Calculation of the max. vehicle speed at L-Band at 1,5 GHz

(only applicable to Mode II, III and IV)

	T_u	maximum speed	
Mode II	0,000 25 s	70 / 0,000 25 x 1 500	= 186km/h
Mode III	0,000 125 s	70 / 0,000 125 x 1 500	= 373 km/h
Mode IV	0,000 5 s	70 / 0,000 5 x 1 500	= 93 km/h

5) Other values of β

In some papers such as in the Montreux proceedings and the AES UK conference proceedings the speed/frequency curves are drawn in such a way that for factor β a value of 0,062 5 is taken rather than 0,08.

$T_u \times f_o \times v$ / c = 0,062 5 × 4 / 5

$T_u \times f_o \times v$ = 0,006 25 × 4 × 3 × 10^8 / 5 × 10^6 = 15

When f_o is expressed in MHz and v in km/h then $T_u \times f_o \times v = 55$

so that:

$v = 55 / T_u \times f_o$

$f_o = 55 / T_u \times v$

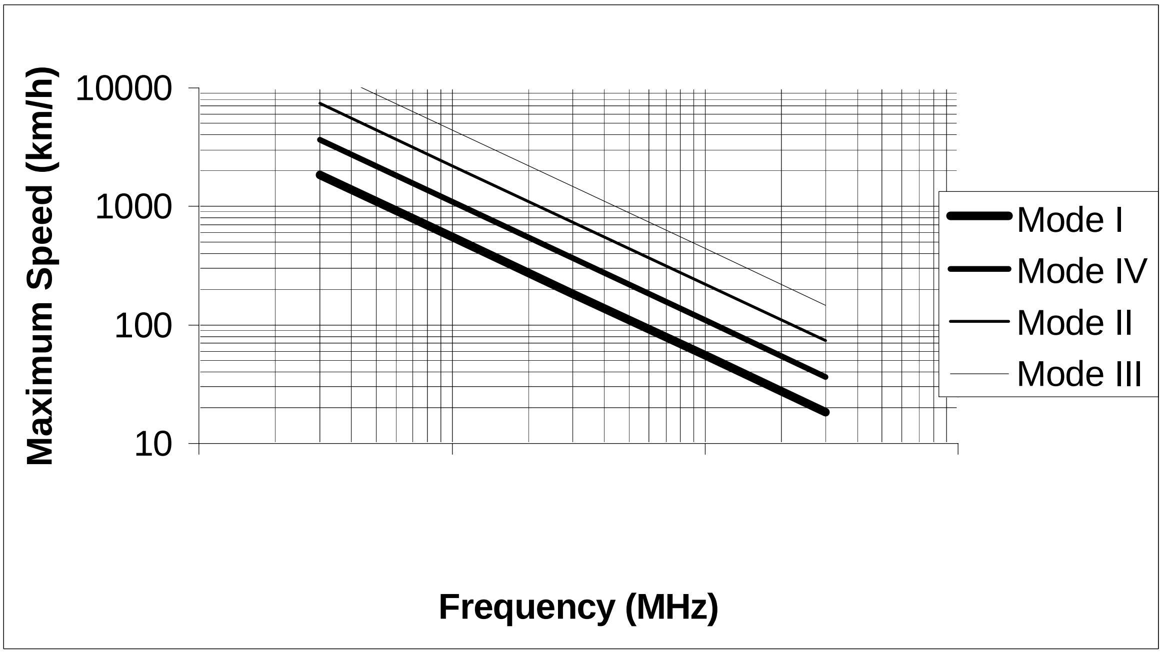


Figure 5.19: Maximum Speed versus Frequency for Beta = 0,062 5

Calculated examples for the case that β = 0,062 5:

- Calculation of the maximum speed v that is possible in the 4 modes I, II, III, IV for a nominal frequency f_o of respectively 375 MHz, 1,5 GHz and 3 GHz

	Speed		
Mode I	55 / 0,001 x 375	= 147	km/h
Mode IV	55 / 0,000 5 x 1 500	= 73	km/h
Mode II	55 / 0,000 25 x 1 500	= 147	km/h
Mode III	55 / 0,000 125 x 3 000	= 147	km/h

- Calculation of the maximum speed at 100 MHz

	Speed		
Mode I	55 / 0,001 x 100	= 550	km/h
Mode IV	55 / 0,000 5 x 100	= 1 100	km/h
Mode II	55 / 0,000 25 x 100	= 2 200	km/h
Mode III	55 / 0,000 125 x 100	= 4 400	km/h

- Calculation of the maximum usable frequency per mode when driving at a speed of 200 km/h

	Max Frequency (*fo*)		
Mode I	55 / 0,001 x 200	= 275	MHz
Mode IV	55 / 0,000 5 x 200	= 550	MHz
Mode II	55 / 0,000 25 x 200	= 1 100	MHz
Mode III	55 / 0,000 125 x 200	= 2 200	MHz

- Calculation of the maximum vehicle speed at L-Band at 1,5 GHz

	Speed		
Mode II	55 / 0,000 25 x 1 500	= 147	km/h
Mode III	55 / 0,000 125 x 1 500	= 293	km/h
Mode IV	55 / 0,000 5 x 1 500	= 73	km/h

2-Dimensional frequency-delay domain

(Trade-off between RF frequency and the maximum delay)

When a car is driving at high speed it is picking up signals coming from the front (leading to a positive Doppler shift) and signals coming from the rear (negative Doppler shift). The worst-case condition occurs when the two signals have nearly the same amplitude. Satisfactory reception is assumed when under this worst-case condition the maximum equivalent noise degradation is less than 4 dB at 10^{-4} BER.

At a speed constraint of 200 km/h, the maximum frequency that can cope with the maximum multipath delay can be calculated.

From equation 5.1:

$$\beta = T_s \times f_o \times v / c = 0{,}08$$

$$(T_u + \Delta) \times f_o \times v / c = 0{,}08$$

$$(5\,\Delta) \times f_o \times v / c = 0{,}08$$

$$\Delta \times f_0 \times v = \frac{0{,}08 \times 3 \times 10^8}{5} \tag{5.7}$$

When in equation (5.7) f_o is expressed in MHz and v in km/h equation (5.7) then:

$$\Delta \times f_0 \times v = \frac{0{,}08 \times 3 \times 10^8}{5 \times 10^6 \times 10^3} = 17{,}28 \tag{5.8}$$

Let τ_m be the maximum delay beyond which the addition of delayed signals causes degradation.

For C/I = 10 dB and $\Delta = T_u / 4$ then:

$$\tau_m = 1{,}2 \times \Delta$$

or

$$\Delta = \tau_m / 1{,}2$$

$$\tau_m \times f_o \times v = 17{,}28 \times 1{,}2 = 20{,}736 \tag{5.9}$$

For a vehicle speed v = 200 km/h equation (5.9) becomes

$$\tau_m \times f_o = 20{,}736/200 = 0{,}103\,68 \tag{5.10}$$

For a given vehicle speed of 200 km/h, equation (5.10) allows us to derive the maximum usable frequency f_o in function of the maximum delay τ_m beyond which addition of delayed signals causes degradation. In table 5.7 this has been done for each of the 4 transmission modes.

The maximum distance between transmitters can be derived directly from the value of τ_m by multiplying it with c, the speed of light.

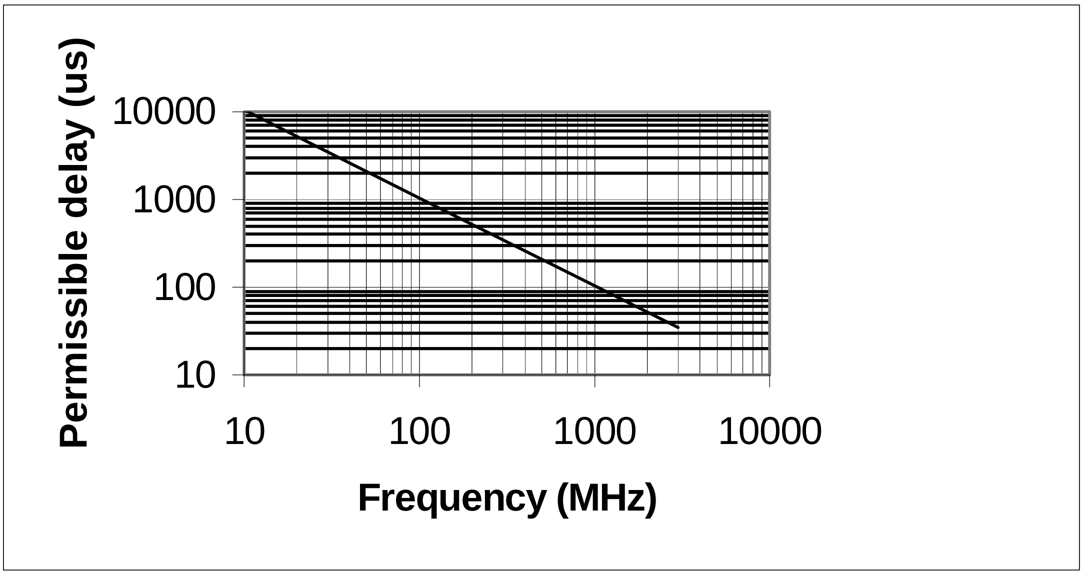


Figure 5.20: Permissible delay as a function of frequency for a vehicle moving at 200 km/h

Table 5.7: Variation of limits of frequency and transmission distance as a function of mode

	Mode I	**Mode IV**	**Mode II**	**Mode III**
Δ	246 µs	123 µs	61,5 µs	30,7 µs
$\tau_m = \Delta \times 1,2$	300 µs	150 µs	75 µs	37 µs
fo = 0,10368 / τ_m (v = 200 km/h)	345 MHz	690 MHz	1 280 MHz	2 800 MHz
Maximum transmission distance d = c x τ_m	89 km	45 km	22 km	11 km

5.6.4 Local Service Options

For local services, a mixture of SFN and conventional techniques can provide the most flexible solution. A few transmitters in a city operated in SFN mode would offer the benefit of the network gain and therefore allow the total power to be reduced when compared to a single transmitter. The interference at a far distance is also reduced.

One way to introduce services with different coverage areas , e.g. local services, is to use another ensemble at a different frequency. Then conventional planning techniques may be used and are being considered in many countries for DAB local radio. In this situation a number of 1,5 MHz blocks are allocated and different ones are used in different geographical areas. These areas may be served by a single transmitter or a number of transmitters operating in a small SFN. An appropriate re-use distance is required before the co-channel block can be allocated to a new area. However, because of the network gain effect, the re-use distance depends on the number and location of transmitters in each network. Terrain-based planning techniques are normally used to minimize the re-use distance, and hence optimize the spectrum efficiency. It must be noted that if the capacity of the DAB ensemble is higher than needed in a certain area the spectrum efficiency is reduced. The possibility to introduce services with different coverage areas in the same ensemble is therefore sometimes required.

One way to fulfil this requirement is to introduce localized services within the ensemble, a feature known as "local service area". This approach makes it possible to use parts of the ensemble in a certain area for local transmission, and is described in Part 2 of the Guidelines (TR 101 496-2 [9]).

History

Document history		
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