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DA.B BROA.DCA.ST NETWORK

Graduation Project
EE-400

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List of Abbrevlatlons

A/D	Audio/Data
ACS.	Access Control System
ADF	Absolute Doppler Shift Flag
AES	Audio Engineering Society
MC	Auxiliary Information Channe]
AppTy	Application .Type
CA	Conditional Access
CAid	Conditional Access Identifıer
CCIR	Comıtç Ç2t1.~ı.ıltatiflntemationaldes Radiocommunications
CI	Cont~~tsJpqip~tor
CIF	Commçn··!t1.t~deave4.Frame
CRC	Cy'1lip !çqu.t1.daııçyCheck
CU	CapfctWlJt1.it
D-QPSK	Diff'er~t1.tial QPSK
DAB	Digital A.u.qio Broadcasting
DGCA	:I:)8:ta Group Conditional Access
DRC	Dyn.amic Range Control
EBU	European Broadcasting Unipn
EE,p	Equal Error Protection
EId	Ensemble Identifier
ETS	European 'I'elecomrnunication Standard
F-·PAD	Fixed Program Associated Data
FFT	Fast Fourier Transform
FI	Frequency .Infçr11ation
FIB	Fast Information Block
FIC	Fast Information.Channel
FIDCId	Fast Information Data Channel Identifier
FIG	FastJıifür111atiçnGroup
HEO	f.I.i~lı!~ilt1.cJim~di:E:lfipticalOtbit
ID	Iqçt1.ti~çrofAudiq Çodi11g Algorihm
IEC	International Electrçtechnical Commission
ISO	International Organization for Standardization
LFN	Logical Frame Number
LSb	Least .Significant Bit
LSB	Least Significant Byte

LS,F.	Lower Sampling Frequency
LTO	Local Time Offset
MiS	Music/Speech
Mainld	Main Identifier ofa Transmift~f
MCI	Multiplex Configuration Informatiôri
MPEG	Moving Pictures Expert Group
MSb	Most. Significant .Bit
MSB	Most Significant Byte
MSC	Main Service Channel
OE	Other Ensemble
OFDM	Orth9~9~alir~equenfy Division Multiplex
P/D	Program DataService Flag
PAD	Pr9gr.~.A.~şoctat~fi. I>ata
PCM	Pulse Côded Modulatfon
PTy	Program'Type
QPSK	Quadratut'~.Phase Shift Keying
S/D	Static/Dynamic
SC	Service Component
SCId	Service Component Identifier
SCTy	Service Component Type
SFN	Single Frequency Network
SI	Service Information
SId	Service Identifier
SubCh	Sub-channel öf the MairService Channel
SubChId	Sub-channel Identifier
SubId	Sub-Identifier ofa transmitter
TCId	Type Component Identifier
TD	TimeDelay
TII	Transmitter Identification Information
TMId	,Transport Mechanism Identifier
UEP	Unequal Error Protection
UHF	Ultra High Frequency
UTC	Co-ordinated Universal Time
VHF	Very High Frequency
X-PAD	Extended Program Associated Data

Abstract

When FM was established in the middle of the past century it was specified for reception with stationary receivers equipped with directional antennas mounted in a height.of approximately 10 meters. Now a days this specification is no Ionger up to date since more than 85% of the radio receivers are portable or mobile systems connected to a simple non-directional antennas. Also the expectations in the quality of the received signal have changed.

By this way, the Broadcastin.ğisystem DAB was developed by the EUREKA 147 project and fully standardized by the Eu:ropean Telecommunications Standards Institute (ETSI).

The DAB system is a transparenı digital transmission channel that allows to transport any information that can be expressed in bits and bytes to stationary, portable and mobile receivers. The capacity of this transmission channel can be split into a number of sub-channels which can carry independehtaudio or <lata programs with different <lata rates and protection levels. For the transmissfön of these digital programs the innovative modulation technology Coded Orthogonal Frequency Division Multiplex (COFDM) is used

To maintain an efficient use-of.the provided transmission channel it was also decided to use a <lata compression system.

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Chapter Two
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Introduction

The section 1.1 of the project gives an overview of the principles which should be considered and applied when planning the implementation of a DAB Broadcast Network. In this project, the DAB Broac:lçast. Network is taken to encompass ali of equipment between the audio coders (or <lata source equipment in the case of a <lata service) located at the studio centre (or <lata <>rfgination point) and the input to the DAB receiver.

A conceptual picture of the DAB Broadcast Network·from source coders tô ttaris:ı:ıiitters is iatroduced in figure 1.2>1 Each ofthe elements ôfthe cônceptıial rietWôrkis analysed and some of the stfateğfesiwhich could be. emplöyed fôr siğ:rıal dist:ributiôn in the different parts of the Netwörk are introduced. This section concludes with some illustrative examples of Broadcast Network implementation.

Section 1.2 proposes a oonceptual 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.

In section 1.3 we have a look at the elements of tbe cotıceptual Broadcast Network. This includes some aspects of the use of the Service and EriseınbleTrarispô::ftl:rıteıfaces. The concluding of section Iooks at some rnore general networking aspects including timing and synchronisation as well as some of the considerations which apply when reconfiguring the Ensemble Multiplex.

The section 1.4 considers how the factors presented in the previous sections should be applied when considering distribution of service and ensemble information.

Section 1.5 looks at some DAB Network implementations which are in operation. The examples serve as an illustration of some of the aspects to be mentioned.

The chapter two starts with a detailed look at the modulation and channel coding used by the DAB system. It continues with a look at some particular aspects including synchronisation rnethods and the > technique employed to permit transmitter identification;The chapter concludesby considering some RF aspects and broadcast network planning techniques. In partictilafithe concept ofa Single Frequency Network (SFN) is examined in some detail.

Chapter three concludes our project

DAB BROADCAST NETWORK

Chapter One

Implementatien and Operation of the DAB Broadcast Network

1.1 The coneeptual DAB Broadc~ş,~~twork

Figure 1.2.1 shows the coneeptüal netwot:k/.in.<diagrammaticform: >'fhe·rn.etwork..is envisaged as a three stage process whete eachisfüğeiis managed<byatdifferen.t entity. The three managementseütities are: the *Service provider,* the *Efisemble .provider* and 'the *Transmitter Network provider.*

The Service provider is conoemed with building a part of the multi-service Ensemble multiplex. Typically; this would be an individual service (or service component), thou.ğh it could extend to a number ofservices, 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 tums this into the transmitted signal at one or, more typically, many transmitter sites. In this final stage, t~e data which identifies uniqely each transmitter in the network (Transmitter IdentiflcationInfonnation) must be added if required.

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

Figure 1.2.1 also shows the flow of control information in the Network. Since the Ensemble provider Iooks after the constnıetion 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 Transmissicfü:<Networkprovider. There will also be a requirement for control information to flôwrfrom ithe Service providers back to the Ensemble provider and between different En.seınbleproviders (to exchange information about other transmitted·rensembles.förinsta:rice)Yi'.'J1hese;principalcontrol data flows are also illustrated in Figure 1.2.1.

The lists at the bottom .ofdhe 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 thatthey are not intended to be definitive or exhaustive.

1.1.1 Network Interfaces

It is not necessary that the three stağes>irı building the DAB signal are physically separate. In fact, life is prôbably alotieasier ifthey can be kept close to one another. However, in the typical situation; there willbe many separate Service providers feeding their signals to the Ensemble provider aridthe Ensemble provider will be required to feed the aggregate <lata signal to fua.ıiytrarismitters.

The interface between service and enseın.blegeneration is shown in the diagram as the Service Transport Interface(STI). Its main function is to carry <lata relating to a particular service, or service compöfietit:

The interface between the Enseınblerprovider and the modulation process in the COFDM Generator belonging to the TranSfuissionNetwork provider is shown as the Ensemble Transport Interfaeeffi'I'I). Its main function is to carry <lata which relates to

a full Ensemble Multiplex. The principal characteristics of both interfaces are explored in later sections.

The fundamental difference between these two interfaces is that the STI carries service information in a raw form (i.e. not formatıed'ürto the structure defined for the DAB FIC channel). The ETI carries theservice 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 formattirig the FIC(Fast Information Channel) data.

Both the STI and the ETI li.ave been standardiSed as has athird interface, the Baseband Digital I/Q (DIQ) iriferfac~. Althoŭgh not a distributiôn interface, the DIQ provides a convenient break-pôi.ıt in the traiısmitter between baseband digital processing and radio-frequency ınôdülation equipment,

1.2 Building .tb.e'DAB Signal

1.2.1 The Service Previder

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, tog:ether with supporting information, for onward routing to the Ensemble provider. Some examples of service components'are:

* An aiidio data flow (including the associated PAD);. the audio data flow will generally be the main' componerıt 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 primaryj oomponent, or secondary component Iinked to one otmore ofthe main DAB··se:rvices,
* 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.

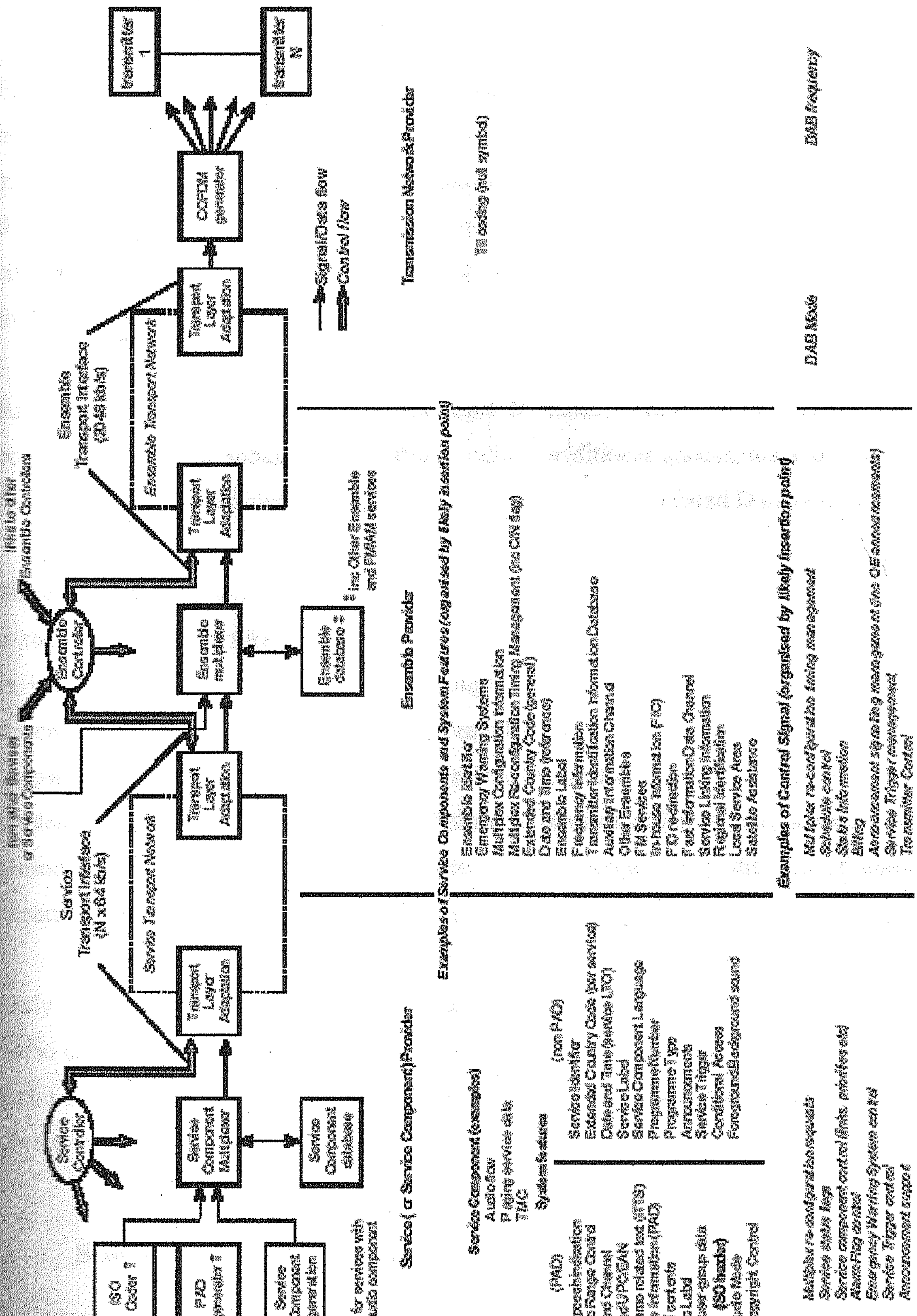

Figure: 1.2.1 The Coneeptual DAB Network

- Source Coding for Audio Flows

For an audio service, source coding takes the form of an ISO/MPEG Layer ll 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))tı:isframes. The input to the coder could be either an analogue audio signal or a digitalsôonnection which would usıially .take the form of an AES/EBU serial interface.

Although based closely on ISO/MPEG-yTuayer.t[L standard frames, DAB .aıidio frames contain a number of enhanCements. These include a.dditiôn.al ehecksums and provision for the inclusion of additiorial<lata, known as Programme AssociatedData (PAD).

Since PAD informationas 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. üne example of the implementation is an RS-232 connection on the audio encoder which prôyides an ISO-frame locked synchronising output to trigger data input from an externa.ltP:ADfermaıter, PAD formatters.have been implemented using PC interface cards. Côritrôlof theforniattingisthen possibJ~using custom software running on the PC. Altemative strategies (e.g. the use of unused capacity in an AES/EBU input) may alsohe possible.

Early audio coders for DAB were equipped with a WG1/WG2 output which requires the audio coders to be in close physical proximity to the DAB Multiplexer. More recently, audio coders have been prodiıced with atı. StI output to permit the buildiııgof more diverse networks.

- Source ceding for Data F'I0\'.VS

For data services, the source codinğ caritake many different forms .dependirıg on the naıure of the particular service. In additiôn, 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 Mıltimedia

Object Transfer protocol (MOT) is one example ofa particular method for dealing with data services which may be employed for DAB.

- 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.

- The Service Component Database

The SCMux also accepts the output ofthe 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 st.atic or dynamic depending on the nature of the data and services. Dynamic data could çljclng€:l under schedule control (i.e. changes take place under the control ofa system çfüçl<:)otçQµldpe trigg~red by external events. An example ofthe latter could be Pty c9g~şyıhich varyinçqnjµnction with prpgramme item changes.

- The Service Centroller

All of the elements of the Service prqvider operate under control of the Service Coetroller which also inserts control-information into the STI (and accepts control information from the Ensemble Contrôllefvia the STI). The Cotı.trôllet deals with the normal scheduling of data (such as Prtuıri. and Pty for example) but could also be responsible for more fundamental chartğes such as those of'the auô.iocoding rate. Some of these changes will have an effect<onôtfıe:rservices, e.g. a•reconfiguration in which a number of services are interchangiıg ca~a.city. In such a situa.tiôhthe Service Controller of any particular service will need to operate in conjunction with other Service Controllers under control ofthe Ensemble Controller.

1.2.2 The Service Transport Interface

The STI 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. In addition, a control channel is also provided which may be used to manage, or monitor, the service compoiients.

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

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 strucfure. Network A.dapted versions are more complex physical manifestations using more complex framing and complete with a degree of error protection. They are designed to cope with partieular network structutes (e.g. G.704).

1.2.3 Cascading of Service Provlsion

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 ra:ther than the Ensemble Multiplexer. In this situation, the STI is used as an input interface to an SCMux as well as an output interface.

1.2.4 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 1.2.1 lists some of the currently defined System Features which coüld' be reqtıifedto be inserted at the level of the EMux. Note, however, that the list could differilldiffefenfirri.pleınentatföns,

* managing the Ensemble Multiplex capacity including the generation öf the MCI. This includes the management of the Service Controllers associated with each service.

- 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 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.

The input to the EMux is characterised 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.

- 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).

- The Ensemble Contreller

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!

1.2.5 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 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, ETl(NI, G703). In this.formit is only suitable for use on simple local connections or data links with relatively<~traightforward characteristics. A Network Adapted versien, ETl(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 oflink errors and contains infotm.ationto control Network delay variations. This becomes important, for example, whellifeeding a Single Frequency Netwörkusing a switched terrestrial transport network.

The following sectfönsgivesom.e ğeıiefal guidanceon'the use ôftheET[

- Using the ETI

ETI(NI, G703) is a simple form of'the ETi which may be used fora direct connection or connection via a relatively simple network. Its electrical characteristics conform to those defined in ITU-T Recommendation,G.703: it contains rııdimentary 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 at24 kHz).is rather weakinthe presence of errors. Nevertheless, the ETI(NI, G703) could.be.usedıon a satellite connection where protection agairıst errors is provided within the modulation and demodulation equipment. The time delays in such a Network are known' with sufficient precision so thatdynamic delay correction is not required.

ETI(NA, G704), is an adaptation of'theinterface 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 Iarter case, it is of course necessary that the send and receiving uriits 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.

- ETI Capacity

The capacity required for the ETL 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. Altemative versions of the ETI must be used in this case.

- Ensemble Transport Network Performance

This sub-section attempts to set performance targets for the behaviour of the Ensemble TransportNetwork.

The performance is defined in terms of the behaviour of the network frorn 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 ofthe ETI. Fora 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 charaoteristics, some preliminary definitions are necessary. The ETi comprises 24ms frames. Each frame is assumed to consist of 24 blocks (giving 1000 blocks per second) with 1920 bits in each block.

We define:

* A delay Slip as a change in Network-Transit Time from one frame to the next of more than 50% ofthe DAB Guard Interval for the DAB Transmission mode in use.

* An Errored Block (EB)to be ablockm'\,'yithatleastone errored bit,

* A Severely Errored Block (SEB) to be a block with at least 8 errored bits,

* AnErrored Frame (EF) to be a frame with at least one EB,

* A Severely Errofed 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)tobe'aföfö:ıevııith atleast ı·.·sEF(or atleastl UF).

The Network is co1 1sideredUnavailableifframe synchronisation is Iost, or more than 10 SEF were received in the last 40. The channel becomes Available as soon as frame synchronisation is achieved for more thari.40 consecutive frames.

Perf.ormanceobjectives can now be otitlirted:

* Network Transit Time (Mean): the mean Network Transit Tirne shöuld be fixed and known with an accuracy of± 1 μs. 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.

* Network Transit Time (Variance): the variance in the Network Transit Time must not cause the jitter and wander on the received 2Mbit/s signal to exceed the limits.

* 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 L3.J.

Table 1.S.1. Error Performance Objectives

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

* Network Unavailability: The Network should be Unavailable less than once per year. Each frame thus has 5760 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.

* 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.

* Differential Transit Time (Variance): Performance target to be defined,

• Signalltng 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. 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 <Section "Cascading of Ensemble Provision" [1.2.6]).

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 fiinctions to be implemented to permit tailored systems to be built. üne example 8:faUser defined function could be the control of COFDM generator parameters (sueli' as time delay or TII code) from a remote terminal. Other transmitter control furictiôns Côuld also be implemented.

In addition to the MNSC, since the ETI(NA, 0704) 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 (ITU-T Recotnmendation G.704).

- 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 thai 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 ofCR.C-4.

- Use of time-stamps

In order that the ETI receiver can restore a cônsisterıt network transit time, information about signal timing must be included in the transmitted ETI. For this reason, timestamps are included mit.hin. tRP. H:ET.

1.2.6 Cascading of Ensemble Provision

Although the conceptual model shows the Emux (and associated equipment) asa single entity it may be necessary in some instances for the Ensemble provider to operate in a distributed fashion. For instance, at« ther,.firsf.level a partial Ensemble Mııltiplex consisting of. a common sub..set of nation.alservic:es could\he>built This would be distributed to a.second level ofEnsem.bleMultiplexingwhicha.ddslocal variants ofthe remaining services. Such an architecturetğquires the use ofa miılti-frequency network, MFN

In this case the output of one level o:fiErısemble Provision (the ETI) is. followed by anceher Ievel of Ensemble Multiplexitığ\rather than the COFDM generator. In such circumstances, the ETI must be capable ofoperating as an input interface to an Emux as weU as its output interface.

Timestamps are included in the basicaiefiıiition of the ETiand a further timestamp is included at the Network Adapted layer.Ina·network using.cascaded multiplexers, the latter may be used to control transit.delayin a section of.thenetwork, ensuringseamless 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 asa 'ıime-reference mııltiplexer' 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 andthe timestamp (TIST) field.

1.2.7 The 'I'ransmission Network Provider

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

- Signal **Distribution in the Transmission Network**

The ehoice of a suitable distribution sigual to feed the distant transnıitters will be made largelyon economic.çonsiderations.

For operational networks, by far the best clıgiqe is the use of the ETI either in Network Independent or Network Adapted form. Tlıiş 2MJ?it/s,signal may be carried relatiy~ly easily using standard techniques. Uis thç tp.()$t çfp.qi~IJt W1q .flç0iple.rμet~<>dpfq~y~p.ş the signal, and all known qperatiomıl ıiçtW9f~SusçJhişteq~p.ift~e.

However, use of the ETI has the disadyantage that a COFDM geııerB.t()rs r~q~İfeqiat each transmitter site. If only a small number oftransmitters are required, forexample in experimental networks, then this may not offer the cheapest solution depending on the balance of circuit and equipment costs, Two other techniques are pgssible:

1) the modulated signal may be prQduced at a low interııediaJe freqııency (in the vision band) and distributed to tlıe. transmitters usiııg.vi.sion circuits. This is refesred to as the "pseudo-vidço" method. A number of ensembles could be carri~d by a single vision qirçuit by using a different centre frequency for each. All that is required at the transrnitter 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 synchronise the ftequency conversions at each transınitter;

* 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) modulated signel 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 wherethefrequencies are available. In a SFN, a method pf locking the frequency co1lyert~rs must be d~yiş~g; ſ.l:4~ tr~ş11:üşşie>ı+.:Pf additional tones has usu~lly ./beeıı:; used in experiinental work. The .sarrte limitaıions raised in 1) above applytöthe management oftransmitter timing atıd insertion of TII.

In passing, it is worth noting that the techıiique of off-air relays, commonly used iı.FM networks, is more difficult in a SFN since there is no separation betw~e11 the .transmit and receive frequency for any given tr~smitter site. This ca,11 Jead *tp* difficulties in achieving adequate aerial isolation, particularly at VHF, to preve11t. 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 timiıg of low power transmitters, and would Iead to more than one transmitter radiating same TII code, which could give rise to difficulties in receivers which make use of TII codes.

- COFDM Geueration

The COFDM generator uses the ETI to produce the analogue DAB ensemble. Control information could also be used, and inclµded 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 standardised as a convenient interface between the baseband processing equipment and the radio-frequency modulation equipment. This is the baseband digital 1/Q interface.

1.2.8 Signal Timing and Synclır(lnişıtion

There are a number of issues côtı.cer:rıedwithSignal timiııg and' synchronisation which should be considered when designing aD.AB Network.

The following lists some ofthe issuescôncerned with data rate synchronisation:

* the audio coder samples the audfö ata frequency of 48 kHz (noııinal)or 24 kHz and formats the resulting codeditiformation into fraıneswith alenğthôfl 152 sample periods (nominally 24 ms ôr 48 ms, depending üpôtithe aüdiô sampling frequency). Ifthe input to thecodeFis a digital signalthetı.the coder's sampling frequency and the incoming data. s:irrıple rate must be syrtchronised. The output <lata rate of the coder will be afriiıteger number ôf bits per frame; the exact number is determined by the output data-rate selected for the coder, which includes all control informatföri,'sh.ıffing bits and P.AD 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.

* the SCMux accepts data at the rate suppHed by the audio coder and associated equipment, and may add additionaledata.. The output of the SCMux must be / synchronous, (or plesiochronous, .asy.cietermined by the nature of the Transport Network) to the input ofthe Serviçf.'l/.'l'ransport Network.

* The EMux accepts the data.fı;<>fü.X~ ıı.tıfü,l'.f.'l[qf Service. Transport Networks and produces a single oµtput 1-\\gaiı:ıiıa g4fü.s .fı;afü.f.'l \~pgtl:ı iş us.f.'ld. at the .output oftlıe EMux. A strategy 111-cıst bf.'l ~<iQpt~~l tq.ensute}lµıt/f.'lijçh;i.ı4ms/fr~W-~•HHtput.·PY.tlıe EMux preserves>,the .fi-an:ı~. stnıcture of the data <from i·~~çl:ı.jrıpµt. .. E:itlıer ttl:ıe frames (at outputand 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 capaetty must be large enough to cope with the data-rate differences and to ensure thatbuffer slips, if any.sareemade in integer frame multiples, In other words, frame alignment must be .ı:naintained<by. dropping er stuffi.ng whole frames from a particular input, as appropriate (irı the Iatter, case .this couldrbe achieved by repeating the previôıiS1 frarrı~).

* The DAB enseınble prôdu8ed byeflıe COFDM generator is .Iocked to the .24ıns frame ofthe ETI output b)'theEMux. However, ifan EMux feeds more than 1 COFDM generator in a SFN therıthe 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 ınustbe very close to each other (within about 1% ofthe carrier spacing), implyirığth.ateach transmitter-must maintain a frequency reference. If the delay of Enserrıble Transport networkcis not fixed, then each transmitter also requires atirrıeteference which is also available to the EMux.

In addition, there are related issues cdrıcerned with the handling of time information carried in theDAB 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 reeeived. 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 synchronisation requirements and network transit delays. This delay must be fixed and known to the required accuracy. UK time pips are usııally transmitted with an accuracy of about SOms.

* 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 au.dible<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 ofa DAB service (if available) to.ifill in gaps in the DAB coverage, which iare 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 ofreal time.

1.2.9 Multiplex Recenflguratlon - Netwnrk Issues

The DAB System permits the flexible and dynamic re-configuration of the Mıltiplex. 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. üne of the functions-of the control information included in the STI, is to allow the broadcaster to manage and control these re-configurations.

- A cautionary note

In the interest of simplification, many of the detailed considerations applying to multiplex re-configurations have been s~~e~~at glossed over. For instance, the data interleaving employed within the Ensemble Multiplex, imposes a latency of 15 frames during configuration changes, i.e. data cıpacity 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 OAB versiöl l. of an ISO-frame (scale-factor CRCs and PAD) applies- to other frames. Thisinf<>rrp.~tıqp.<n:ıay.need to be suppressed, or ignored, over the period ofreconfiguration.

1.3 Strategies for SignalDistribution

1.3.1 Local Connectiôns

Most early implementations ofDAB systertıs relied on the local proximity ofthe audio coders to an integrated Service and Ensemble Multiplexer.

Connections between the audio coder:$ al.l.c.f the have been made using the WG1/WG2 Interface. Signal timing and synchtorıisatiôn is straightforward and can rely on a local "Master"generator which is usuallythe rri:ı.ıltiplexer.

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

1.3.2 Terrestrial Dlstributlon

In the longer term, terrestrial data circfüfs offer the most riatufal method of carrying information. about Services and Service Côrııponents betweeıı Service Multiplexing and Ensemble Multiplexing equipmeıif irtdiffetent locations; irideed, some networks have already been implemented using this approach. It is also likely that terrestrial circuits will be the preferred choice for distribution ofthe 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. The general considerations apply.equally to distribution using the ETi or the COFDM signal.

- Terrestrial Distribution, STI

The STI may be carried on many different kinds of physical links. ETS 300 798 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 comrrrunication between the Ensembie Controller and the various Service Controllers may require thci§'ftlink~ 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.

- Terrestrlal Distributton, ETi

The terrestrial distribution of the ETI could be done eithefusfüğ>fixed'links defücated to the purpose, or using 2Mbit/s da.ta circuits provided as part of a 'I'elecommunication Network.

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

It is recommended that one of the Netwôrk Adapted versiöns ôf the ETI is chosen because of their superior robustness compared with the Network Independent versions. In particular, ETI(NA, G704)s316 has been found to offer good performance in most situations including .carriage on ATM ri.etworks.The capacity·available on this variant of the ETI should suit most application.s,tJ:ıÔÜgh may not be ad.equatefor 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 synchronising pulses derived from a satellite TV channel.

The frequency of each transmitter in a SFN also needs to be [illegible] 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, synchronising 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.

1.3.3 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 oftransponder linearity.

1.3.4 Sharlng the Dlstrlbution Netwotk

In some cases, broadcasters may wish to use the distribution network to feed D~ transmitters together with transmitters operating in other frequency bands (e.g. FM). Duplicated services can share the saıne distribution feed, non-duplicated services could be fed using either spare capacity withirı.W,the\E'I'l or, additional capacity on the same circuit.

A detailed analysis of the problems involved with common distribution paths is beyond the scope of this project 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 teclıtdques Onthe DAB Services.

* 'Data' requirements of,other services may be substantially different (e.g. RDS for FM services).

1.4 Some Real Examples

1.4.1 The BBC's DAB Network

Figure 1.5.1 shows an outline ofthe BBC'sDAB network. A network of27 transmitters has been implemented to cover 60% of.the UK population, and' the majority of major motorway routes. This is a Single Ji'requency Network operating in Band III (Block 12B), and the transmitter output pdwets afein the range 1.kW to 10 kW ERP.

Signal distribution is accomplishedsusing 2 Mbit/s telecommunication circuits using ETI(NA, G704)s376, A mixture ofleased SDH and PDH circuits are used to feed?the transmitters and, in most cases, afüllytedundant 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 providea.time-feference (and frequency-reference) at all sites forthe control of delay variatiorıs and .transmitter :frequency.

1.4.2 L band DAB networks in France

Due to the difficulty of obtaining adequate VHF spectrum, only the frequency band 1452-1492 MHz,(feferred 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. Fof example, first regular experimental transmissien 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 ofhighways.

Since the beginning of 1997, operational netwofks 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 Of several transmitters. DAB mode II is used.

A transport network feeding the tfansfu.ittefs sites and including the ensemble multiplexer. As the transmitters, this multiplexer is also locally located. This permits to incorporate local progfammes. Betweenthe 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, Of local and sent by microwave links Of digital lines to the multiplexer.

In the beginning of 1997, opefational 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 synchronised 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 4 towns : Lyon, Marseille, Toulouse, Nantes. Other authorisations are expected for the other main French towns.

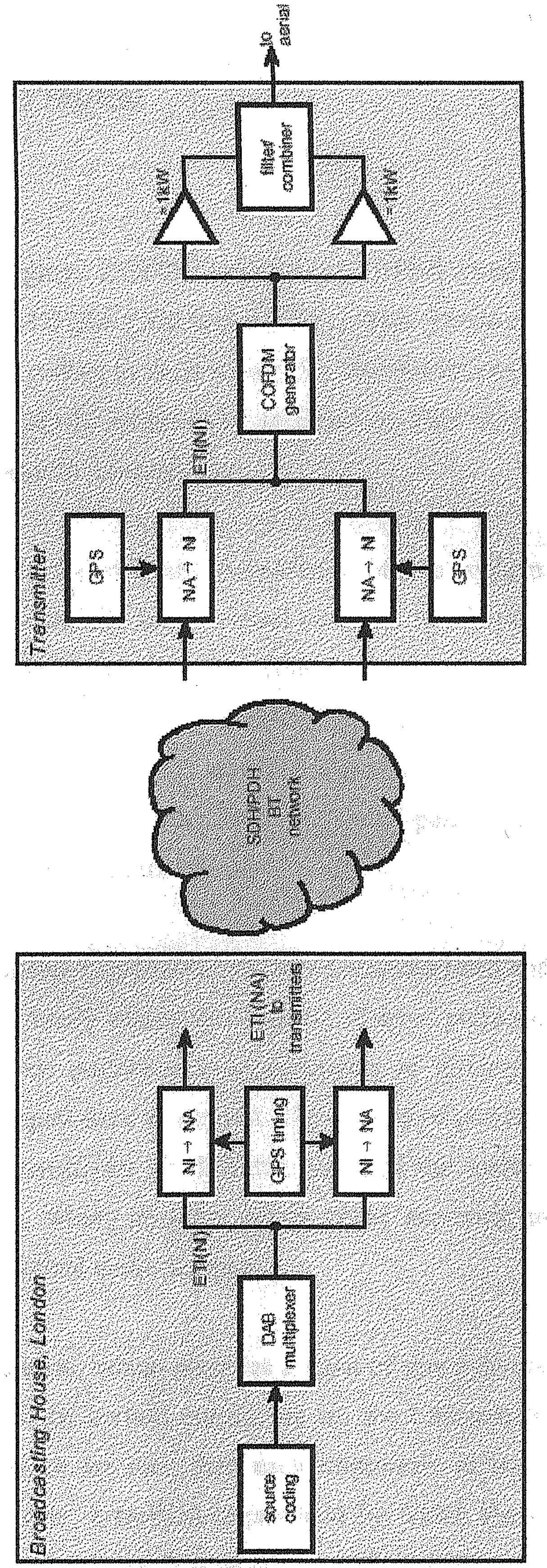

Figure 1.5.1.0utline of BBC DAB Network

Chapter Two

The Transmitted Signal

2.1 Overview

The DAB system employs COFDM modt.ı.lation, which combines the multi-carrier modulation technique OFDM ("Orthogortal Frequency Division Multiplexing") with convolutional channel coding in such a way that the system can exploit both time and frequency diversity. .Thiş.is aclıiey~g liyjfrıtedeaving data symbols, in the time and frequency domains, prior to transfüission.

OFDM contributes to the inherent ruggedness of the systeın ~gainst ınulti~pı:ı,th distortions due to the relatively large symbol duration. In addition, a guard interval is used to help remove interference between consecutive symbols. In order ta 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 fhe syınbol period üµıd guard..interval) and carrier spacing are varied to suit the situation.

The DAB systemuses fl!W~~~!f~i~\rı~ed convolutional Côdi'S for {<l~~..~ correction. This code family allowş the .arn.ount of error protection to be individually chosen according to the performance requirements of different services. For aııdio 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 ofa guard-interval, which provides a form of space-diversity, allows a SFN to be implemented. Provided certain cohstraints 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 pattem, radiated during the synchronising 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 of the 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.

2.2 Channel Coding and Môdulation

2.2.1 OFDM Modulation and Transmission F:rame

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. These 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 propagatiön...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.

Mode I is intended for terrestrial brôadcasting 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.

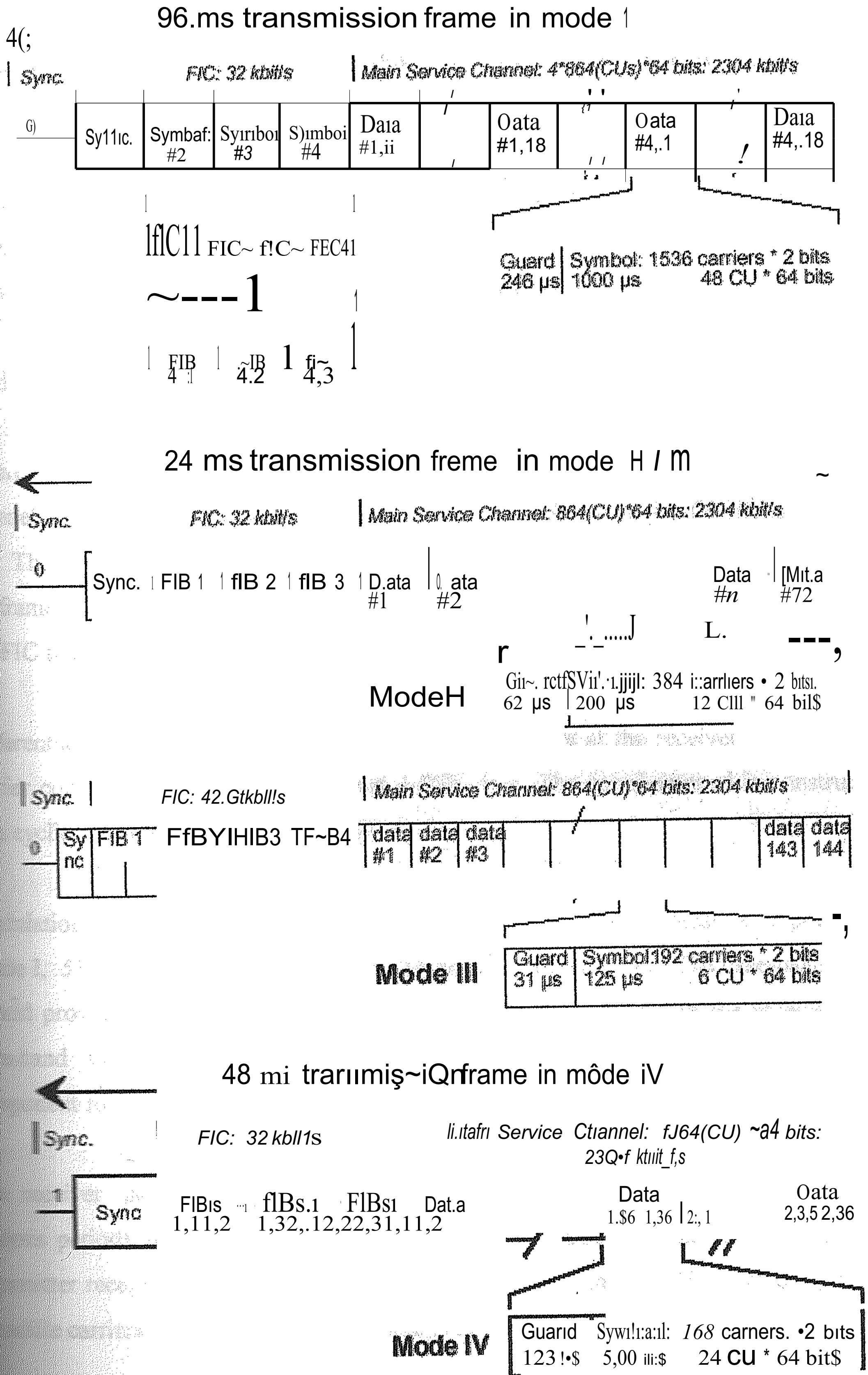

Figure ; 2.1. Strueture of the Traasuiissto» Frame

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 gııard-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 systemsand complementary services at 1.5GHz.

Figure 2.1 shows the basic structtıre of each Tra.nsmissförinıôde.}'I'he main service channel is subdivided into Capacity Units (CUs). Each CU contains 64 e:ı:föoded 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 reeovery at the receiver. Each OFDM carrier contains two bits of Gray-coded 4--PSK data, The >guatcl interval is constnıcted by a cyclic continuation-of eachsyın.bol.

Modulation in the transmitter may be realised with an IFFT of at least 2048 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. The base-band signal sampled at 2.048 MHz for both the in-phase and quadrature component forms the IFFT output block.

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.sy:rı.chronisation but can also carry Transmitter Identification Information. The receivef:côµld also use the null symbol to analyse the transmission channel and take into accôuı:ıtthe level of interference or noise which are present.

The phase reference symboLprôyides\Jine syndhr<>:rıiŞa,tiôn.itıformatjonallowingcthe receiver to extract frequency iriforrtıa.tion (for carrier acquisitiotı.ii{lıtouğh~C}.ancl.a phase reference for di:fferentialdemodulation. The receiver does not need to extract a carrier reference for signal demodulation.

2.2.2 Channel Coding

inside the coverage area of the DAB service, a quasi .error-free reception is generally obtained due to the high perfonrıance /of tlıe applied channel coding schemes. Nevertheless, transmisSıohci.f&r~Feliiiı.()t1iB~b()ınplefely(a'.Vöid~a:ıe§~~bicı.lly it the edge ofa service area. Therefore,.twogöalsfofChannel coding lıave to be considered: firstly, error-free reception within tlıe coverağe area; secondly, some kind of graceful degradation at the edge of it. Botlı ar~ ~9,hieved by applying source ada.pt~d .chaımel eoding. Data services where the bits slı9xyaıequal sensitivity to bit.errors ar~ protected in an equal manner (EEP: equal .efl"()tpr<>tection). Sound services, -where.groupsof bits having different.sensitivities to bit~rı-çfş/.ı:treprotected with anon.,.uniform code (UEP: unequal error protection). Tlıis allows economical use of the available redundancy and therefore a high protection performance.

Error protection in DAB is based on cônvölutional codes witlı a memory of 6 bits, i.e. the number of successive data bits wlıich are used for creating code bits is equal to 7 (or, in the jargon, a constraint length-of "). The basic code rate (mother-code) is of rate R=l/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 ofthe error correction process is required. FIBs are protected by a 16 bit cyclic redundancy code (CRC). For/data in Packefmôde aııd data groups respectively, another optional CRC may be used. For Audiô>services, a CıtC is prôvided for the control infor:mation [Header, BAi, ScFSJ) accôrding to th~ IS91Il'74-$ staııdard. An additional C:RC is provided; one to

error concealment may

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 gracefu.1 degradation at the edge of the service area may be achieved,

2.2.3 Unequal Error Protection (UEP) for Audio (48 kHz sampling)

- Proteetion elasses

A DAB audio stream contains componen.tsofaudio data with different sensitivities'to bit errors (significance) . Every 24 msallthecomponents are transmittedusi:tıgtheDAB audio frame. For those cortıpôı:ıen.tswhicfühave nearly the same sigtıi:fü::aı:ıce a common protection class is applied. The D~<audiôframe uses four different protection classes, applied as follows:

Protection class 1:

ISO~HEADER	l'leader iırıfcmnat:ion	32 bit
CRC	CRC frır ermr dıetectıon withln f.:On!ıro,l inWoorıatt,r,ın: neader, bit allocatiı;mı arrcd sıcaıte faıetoır ::ııelect iııil'oifma,tıorıı	"116 bit
BAl	bit aıiocaıtloıı informatiorı	2'\88 bit
ScFSI	seaıe faotor seıoot infoırmauon	2'54 bit

Protection class 2:

S.6f· '	scale factors	2*486 bit

.Protecllon daşs 3::

SAMPLE	sub-band samples	varıablie
STUFF	stuffing bits	varla,blie
X-PAD	extended programme associated data	vı:ıriabıte

Protection class 4

X~PAD	extended programme associated data	
SoF.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 Informatföıı (CI)fot 'the audio decoding process. AH this informaıion 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 caııse very annoying 'blips' ..But because ofthe a.ppliedfseale•factôr> error>.COncealıneııt/ iri conjunction with the ScF-CRC, the performance requirementfor error coırection is not as great compared to CI.

Protection class 3 is used for sub-band samples covering the largest part ofthe audio frame. Since sample errors are only perceivable when the bit error ratiÔ is aböve 10^{-4}, the error protection can be lower than that required for CI and scale-factors. 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 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 ofthe X-PAD Information.

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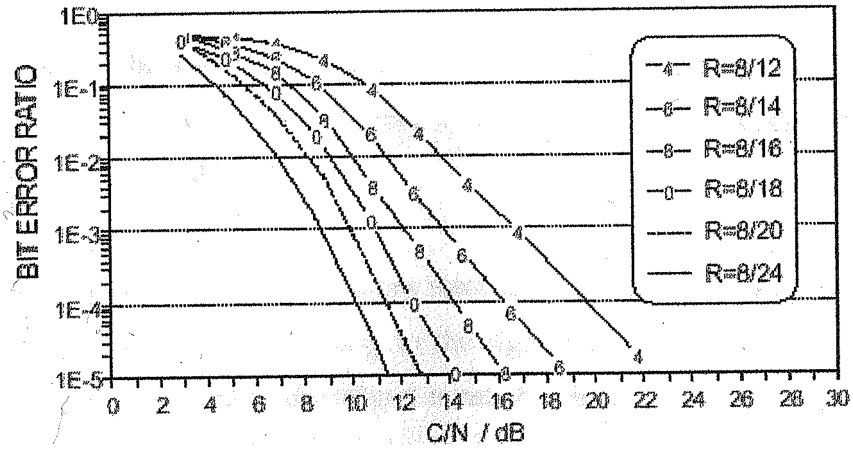

Figure 2.2: Residual bit etror ratio for di:fferentprotectionclasses with code rates (Rayleigh Channel COST 207 Rural Area)

The code rate of a pfötectiôti ·class is set by selecting a rate coıhpa.tible.<puncttıring scheme. Code rates frôfu Rl =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 2.2 shows some curves of the residual bit error ratio using different protection classes with côde rate R_i.

The curves were measured using the "3rd Geııera.tioti'>. experimental equipment, over a simula:ted COST 207 Rural Area channel: vehicle speed 50 km/h at a frequency of 232 MHz).

- Protectlon pfofiles

The number of bits for each protection class depends on the specific audio ,lata rate aiid the audio mode defined by the header information. The error protecti~n ~lasse.s ~it~n one audio frame are defined by a protection profile, which carries information about the length L_j of each protection class, i. and the corresponding index number, PIj, 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. Thelength of the protection class is always chosen to match the worst case for each audio dara rate. For example, an audio frame for .a stereophonic service ata data rate of 128 kbit/s requires twice the BAI and ScFSlôfa monophonic service at 128 kbit/s. The latter will benefit from the extra protection

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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 ınonophonic sound signal.

The protection profiles for audio services were designed by optimising the distributiorı of the available redundancy according to the significance of their components. Since protection class .3 is applied to the largest number ofbits, its code rate, R_i, was chosen to leave sufficient transmission capacity for the higher protection classes. In further ste:ps, 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 encugh 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. Fora given average code rate, the error protection ofa service with high audio <lata 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 redundançy d<:ıpengs\otıh.ow)the 864 capacity units of the Main Service Channel are allocated to süb..chaıınels. Because of the large flexibility in. arranging ;the:, DAB multiplex "gold" numbers were chosen for the amount of capacity u11.itş. tıs(:}9 py the protection profiles. This approach allows certain multiplex re-configurn,timış.-witn.oµt the need to rearrange other sub-channels (e.g. splitting one service a,t t5f~bit(sinto two services, each at 128 kbit/s).

Table 2.1 gives an overview of number ofCUs utilised for each of the audio data rates.

Table 2.1: "Gold" numbers ofCapacity uuits used for pretection profiles

Numbeır of CUs:	16			21	24		29
Audio Data Rates (kbit/s):	32			32	32,48		32,41),, 56
Number of CUs:	32	35	40	42	46	52.	56
Audıo Data Rrıites (kbiVs):	64.	32,.48,		48, 54.},	64,. 96	48, 56, 80	64,.80, 96,. 1'12:
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:						416	
Audio Data Rates (kbit/s):		256,. :~20.					

The various numbers ofcapacity units for each ofthe 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 retum the cenvolutional-encoder-into the zero state, i.e. to clear its memory. This so-called blocked convoluticnal-coding permits closure of the decoder trellis .and .allows independent decoding of the <iifferent sub-channels.

- **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. Prt0foction 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 protectionfor applications with a very high sensitivity to transmission errors. Table 2.2 gives an.>6verview öf the protection levels and .the ccrresporidlng code rates. Figure 2.3 showsJ~e protection profiles for an audio data rate of 192 kbit/s at all protection levels.

For the compilation ofa DAB multiplex, a reasonable ttade-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 ofthe residual bit error ratio versus C/I become steeper. In high speed, mobile, receptiön an error flöor 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.

Protection			Code Rates *R*				Coding
Level		Application	ıge	protection class 1	protection class 2, 4	protection class 3	Gain *
							Cll
1	very high	special	0.34•0.36	8/32	8/2/5m8/28	8/19-8122	+4
2	high	ımohlle hig,rı	0..4-0.43,	8130"8132	8/20,Sl.26	8/16-8/11	+2
3	good	mobile	OJH).51	8/23;.8/24	8l1t.1MV18	8/14,8/15	0
4	mediu:m	mobile weak	0.57~0.6,2	8/17"8/21	8/i 4,,.S/1 /	8112~13	•1.ö ..<t5
5	low	cable	0.72-0·.15	8113"'811€l	811 i-8114	J.\lf1(L	""

* 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

Table 2.2: Overview of the protectiôn.levels and the corresponding eode rates

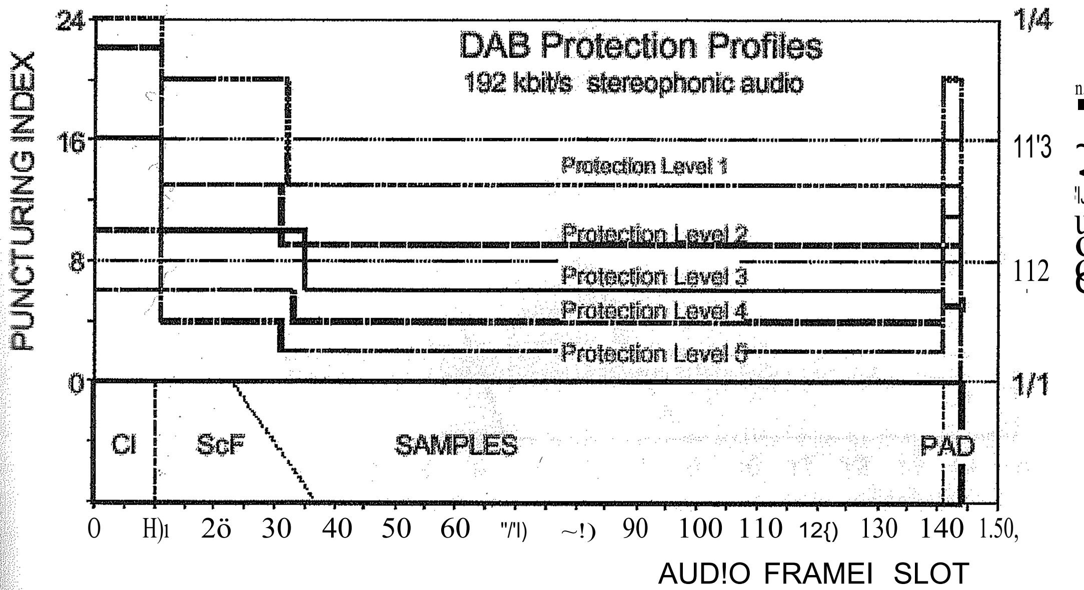


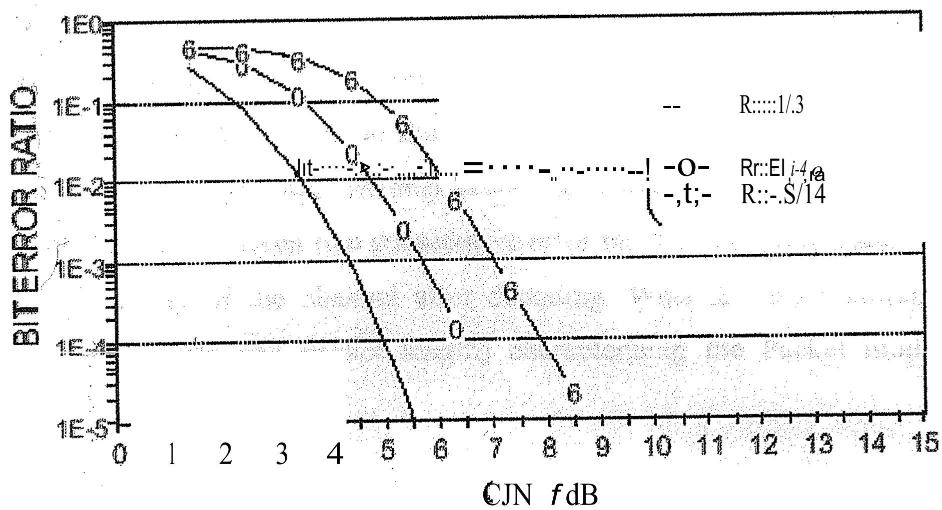
Figure 2.3: Examples of DAB protection profiles för the audio data rate 192 kbit/s (1 Slot = 32 Bit)

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 "3 rd 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 2.4 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.



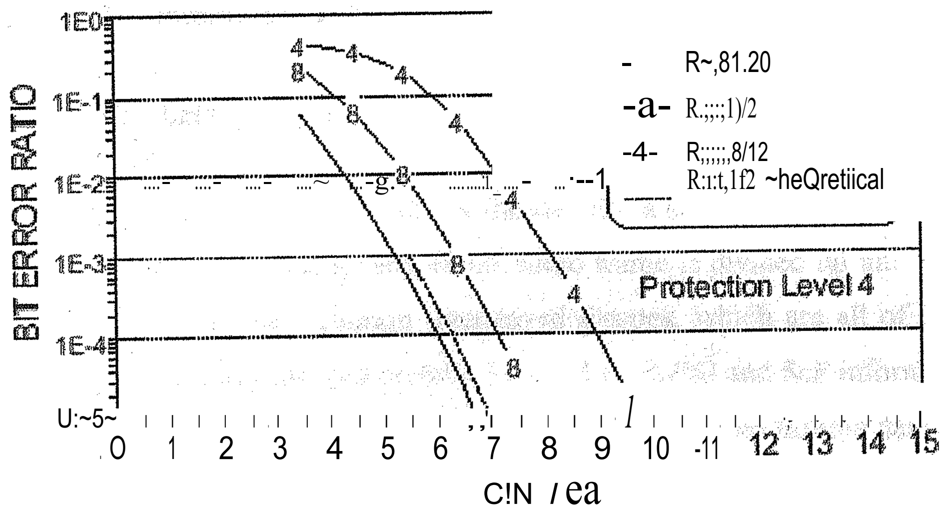


Figure 2.4: Bit error ratie of the 3 proteeüon elasses in a Gaussian clıaıınel

top: a) 192 kbitls, R=0.5, Proteetlon Level3

hottom; b) 224 kbit/s, R=0.6, Protecticn Level 4

2.2.4 Equal Error Protectlon

Within the framework of rate-compatible convolutional codes, provision is made in the DAB system for encoding sub-channels carrying data service components with Eqüal 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 2.2 and 2.4.

If a Sub-channel is organised in Packet möde, however, the (average) bit error rate is not the only important figure. Due to the pröperties 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 t}}'p parameters (data group and packet length) characterising the Packet mode can be chosen.

2.2.5 Error Protecthm for Low Sampling Frequeney (LSF) Audlo (24 kHz Sampllng)

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, ScFSFand ScFiiiformation 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 fiill' sampling frequency audio, not all LSF sub-channels will be able to use the UEP profiles. Such sub-channels will need to use 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.

2.2.6 Error Deteetion in The Fast Informatien Channel

The FIC carries information about the configuration ofa DAB Ensemble multiplex. Decoding this information correctly is vital fot proper receiver operation. Therefore it is important to know to what extent transmission errors can be detected. 'TheFIC 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-distribüted by the dis-interleaving process. As the bit error rate (BER) is not the only paraınetervvlıichinfll.ierices error detection, it is necessary to take into account the "burstiriess't·of the chaiıriel,i.e. itS."m.efüôry".. 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

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

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

Error pattem fields with the length of one FIB (256 bits) were created with the Gilbert model. These error fields were added bitbyhit (modulo-zjjothe ."ref~rence 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 2.4 shows the number of UEFIBs versus BER based on 126 simulatiöns 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.

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.

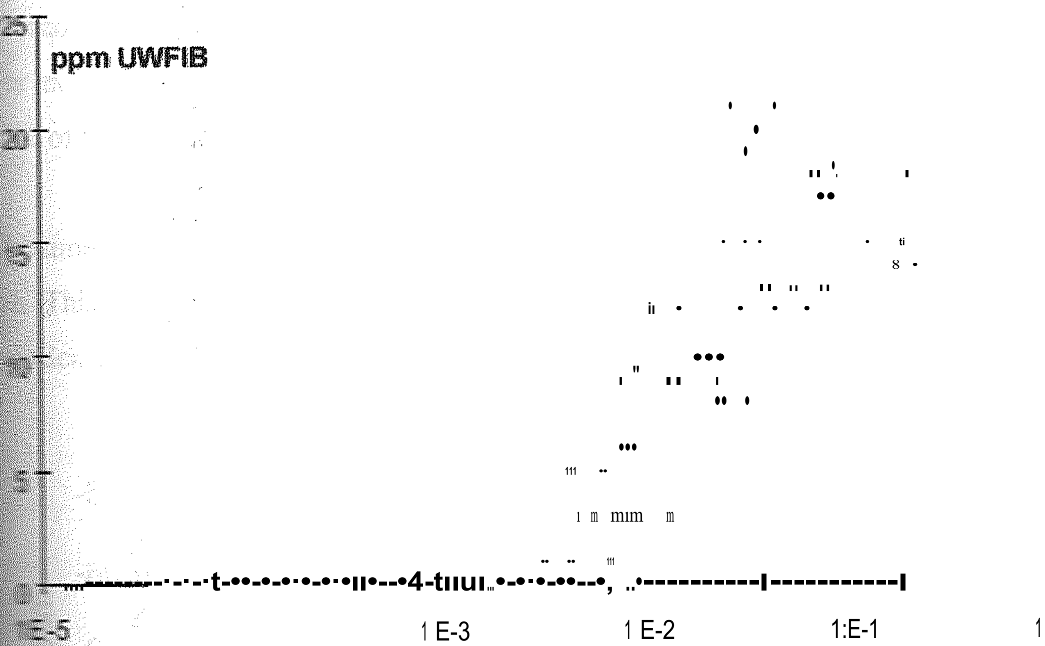


Figure 2.5: Number of UEFIBs out of 10^6 FIBs versus BER

2.2.7 Time and Frequency Interleaving

- **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 oonvolutional codes in the presence of error bursts,

- Time Interleaving

Time interleaving improv'r~t~~ ~~~~9~'$~~ ;gf the ff.ror correction in a time-variant transmission channel. Specifically inthe<case of.mobile reception, even deep fades whieh affect all OFDM cartiers (flaffadiriğ caused by i.shott patlı differences) can be overecme. The longer thetinıe interleaving the better th~.protectfon(against flat fades. For example, when a convolutional code with rate R=l/2 is used, a fade rnay last up to 1/10 ofthe interleaving time with no degradation at high SNR.

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

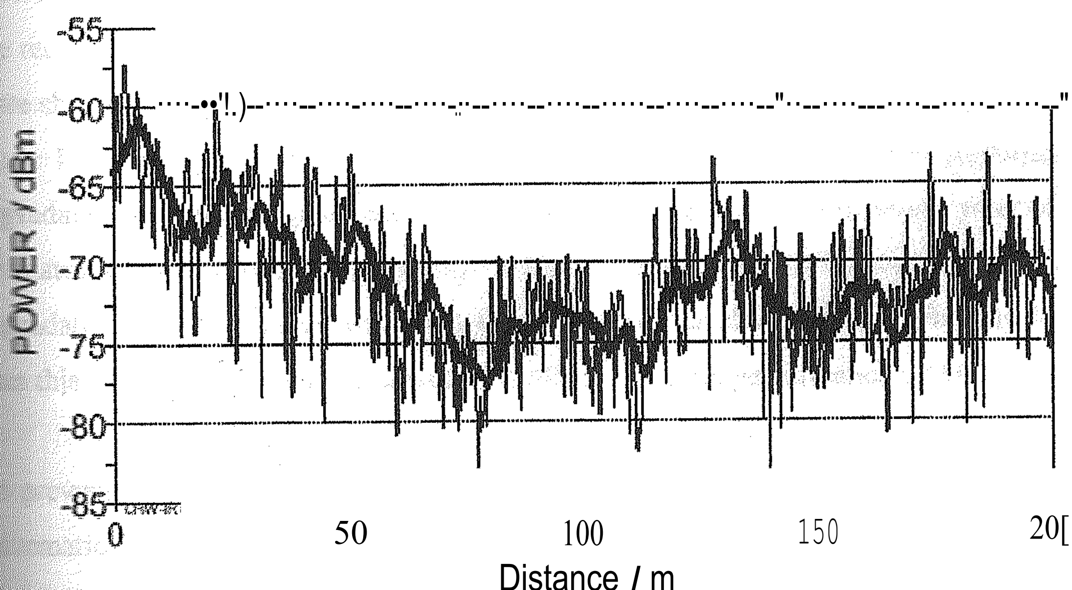


Figure 2.6: Reeeived power in a carmoving at 50 km/h (thin line) shewing the effeet of averaging due to time interleaving (bold Iine),

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Figure 2.6 shows an example of the received power ofa 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 characterised 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 als<> be seen as an averaging over the distance covered during the interleaving time. The average received power is included in Figure 2.6, for a vehicle speed of 50 km/h, and shows the beneficial effect of time interleaving.

2.3 Synchronisation and Transmitter Infermatien

2.3.1 Synchrenlsatlon 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 qııality, a BER of 10^{-4} is needed.

However, frequency deviations ofthe base-band signal, ora corresponding deviation of the receiver clock oscillatoıtsjwill 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 ofthe oscillatorts). Occasionally exoeeding these values will not cause a significant degradation. The frequency deviatiôıı of the transmitter should be. significantly lower than this 1% value. (No measurements are-available on the performanee ofMode IV).

To prevent the use of expensive highiptecision 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 noıse 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 ofthe 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 thesyst.;m cıoc.~ Ofthe Chafulef enco~er ~:tı~ d~codermay be any multiple m (m = 1,2,3, ...) of 2.048 MHz. It is recommend.ed that the encoder clock have an accuracy of about 1 ppm. The system clock of the channel decoder should be §ynchronised to the encoder clock. Synchronisation is derived from analysis of the channel impulse response, which may be estimated from the phase reference symbol.

2.3.2 Transmitter Identification lnfürmation

- General Deserfptien

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 localising information which could be used to filter out the relevant <lata.

The Transmitter Identification Information {TII) provides this localising 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

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The identifier comprises two parts; a main and a sub identifier for every transmitter in the SFN. From analysis of'theNull'symbol a receiver can derive the identifiers ofthose transmitters which are currently received.CThereceiver can use these identifiers directly for service information selection based on geograpnical criteria. For a more precise localisation, the geographical <lata of the transmittersmay also be conveyed in the Sl With the help of this information the receiver can estimate its location inside the coverage area ofan SFN.

- 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.iltı ôrdertô allow the receiver to perform channel state analysis, the TII signal is oılyttansm.itted in every other frame. The synchrofüsatfoıı 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 organised 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 mateh the 768 available carriers; and'in Mode I, the structure is repeated four times in the frequency domain, to match the 1536 available carriers.

In Transmission Mode III, the TII signa.Lis based on a block of 192 carriers, again organised as 24 combs of carrier pairs, ea.ch comb comprising 4 carrier pairs.

In all Modes,the 24 combs, which correspond to the set of possible Sub Ids of the transmissions, allow the conventional network planning of the TII signal inside the SFN. The allocation of Sub Id 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 combina.tions of 4 from a set of 8 is 70, this results in 70 unique "pattems" ofcarrier

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pairs per comb, which correspond to the set of possible Main Ids of the transmissions. The allocation ofMainid to a transmitterdetermines which ofthe pattems (i.e. which 4 out of the 8 carrier pairs in 'the comb) itwill fransmit.

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 5 unique "pattems" of carrier pairs per comb, and hence 6 possible Main Ids.

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 Ih Mode II four pairs of carriers (8 in total) are used, and in Mode Ill two pairs of carriers (4 in total) are used. The ratio of carriers in a TII symbol to anormal 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 synchronisation from the null symbol containing TII is stili possible,

For example in Figure 2.7, the TII symbol for the BBC's experimental transmission at Crystal Palace is illustrated. The Main Id is OB (hex) which corresponds to a pattern of "00110110". The Sub Id is Ol (hex) which uses the second of the 24 possible pairs of carriers.

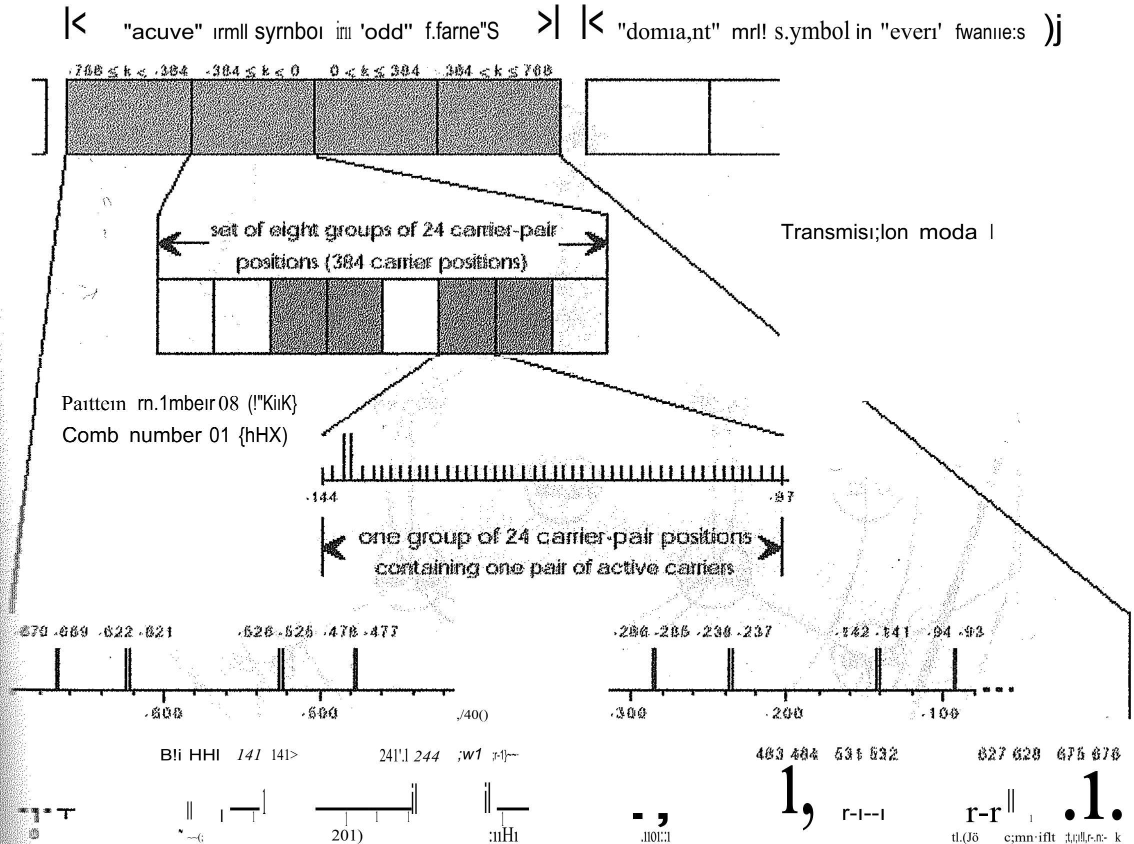


Figure 2.7: Comb strueture (Transmlssion mode I)

Figure 2.8 shows an idealised 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. Sub Id 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 eoverage area of one SFN is larger, the hexagons of Figure 2.8 can be arranged with diflerent main identifiers. Every hexagon of 21 transmitters has its own main identifier. An example is shown in Figure 2.9.

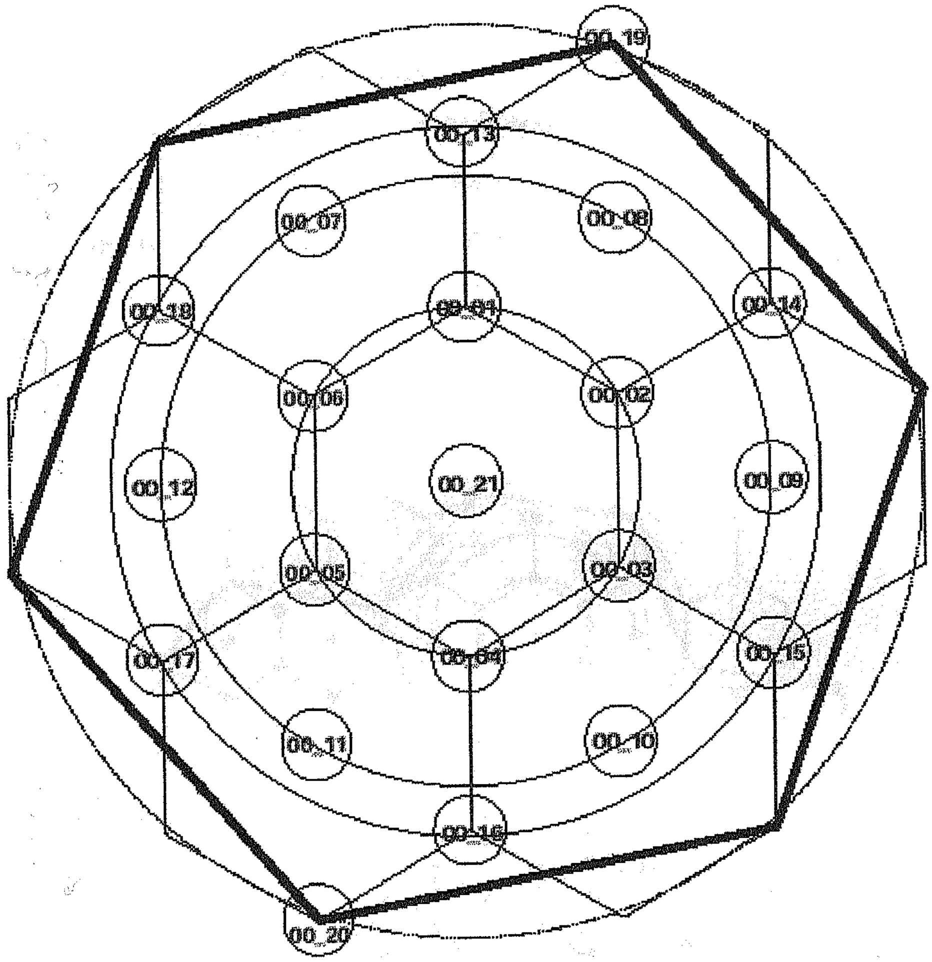


Figure 2.8: Sub-Identifier 01 to 21with one Main Identifier {00)

2.4 RF Aspects

2.4.1 Time demahı representatlen

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 Iarge amount of headroom to prevent non-linear effects from occurring.

A DAB-like symbol was simulated with 1536 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 2.10. 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.

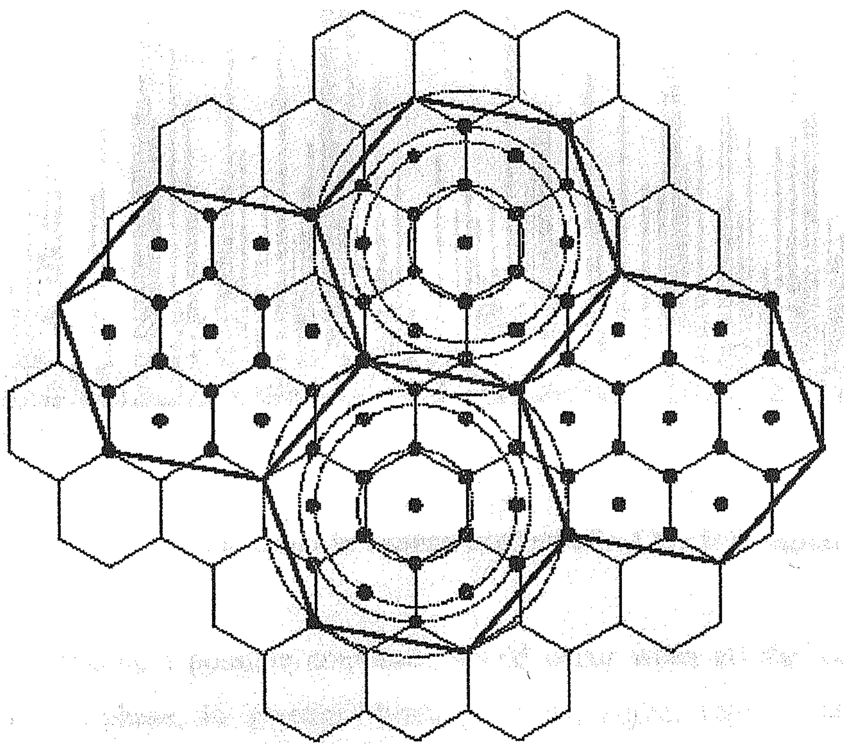
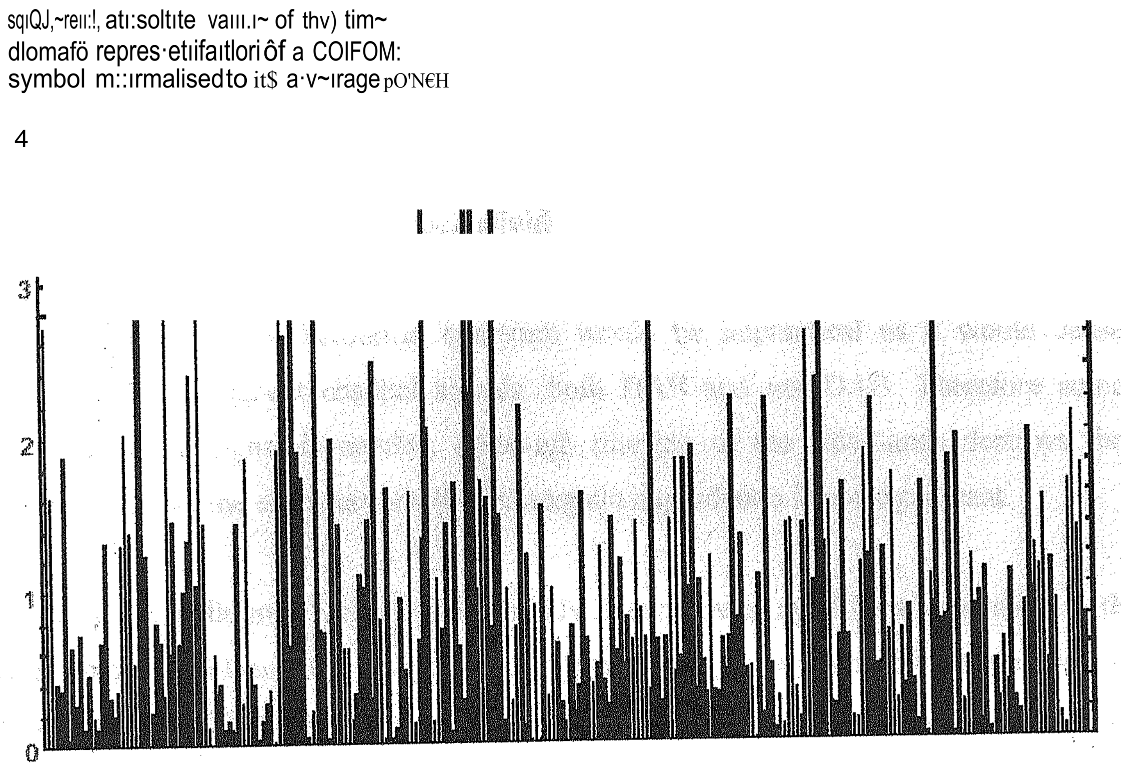


Figure 2.9: SFN with 21 Sub-Identifiers for 4 different Main Identifiers



Ftgure 2.10: Time-domain,represeııtat,oıı ofa.ÇQ!DM signal

In theory the maximum possible /a:ınplitude would occur when all the carriers were simultaneously in-phase, In upfactice,/:lıımiting of the digital representation of the COFDM signal never .allowsthis tôrhappen: .The result of the inevitable signal clipping and other non-linearity isthatout .'of-bandcomponents are generated at the output ofthe digital to analogue converter in DA.B 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 aınplitudewill only occur veryinfrequently.

Care should be taken that clippifig•ittıdiôther non-linear effects within the transmitted signal do'not degrade the overall perfôrnıance to a significant Ievel.

2.4.2 Frequeney domain representation

In The theoretical spectrum ofa COFDM signal for the four Transmission modes, the levels ofthe side-bands beyond the last active carrier frequency are the sums ofthe 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 additipn,al p:roblenı, that highly-linear power anıplifierş ./. opepiting yYith Iarge 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 Iikely to restrict the useof adjacent DAllchannels.in certain cases. This decreases the efficiency of spectrum utilisation and increases the problems of intemational co-ordination of DAB frequency allpçati<:>ns.

An important issue is thereforethedefiri.ition ofa suitable spectrum mask. This must be a compromise between the needs ôffrequency planners for efficient use of the spectrum and the needs of broadcasters forcôst-effective transmitters.

- VHF speetrum mask

As a result, a dual mask has beeri. specified for VHF. The first mask would be used for transmitters in critical 'situatious,' Where the adjacent frequency regiorı needs specific protection. The second, less stringeht, mask may be used for transmitters in situations with more relaxed requirements. Both masks are shown in Figure 2.11. The vertical scale reflects the permitted out-of-baııd 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 lessstringent mask might be used for the other transmitters in the middle of the network, or in locations where a highefJevel of radiation into the adjacent channel could be tolerated (e.g. <X>-sitedeıriisSiöniiôfa.djacentc:hannels). In cases where adjacent channel broadcasting ~~tjff~()~ côLsiteffr~s~tters~ COtlltll0n ~mplification may be possible. This could pe;mit ~ ,c:onsiderable relaxation oftherequirements.

The less stringent spectn.ıınfüask 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.ouı-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)~:<,t~~seused for aeronautical services. As the more stringent mask would l)e used in critical situations, it should be used as the hasis for deriving protection ratios.

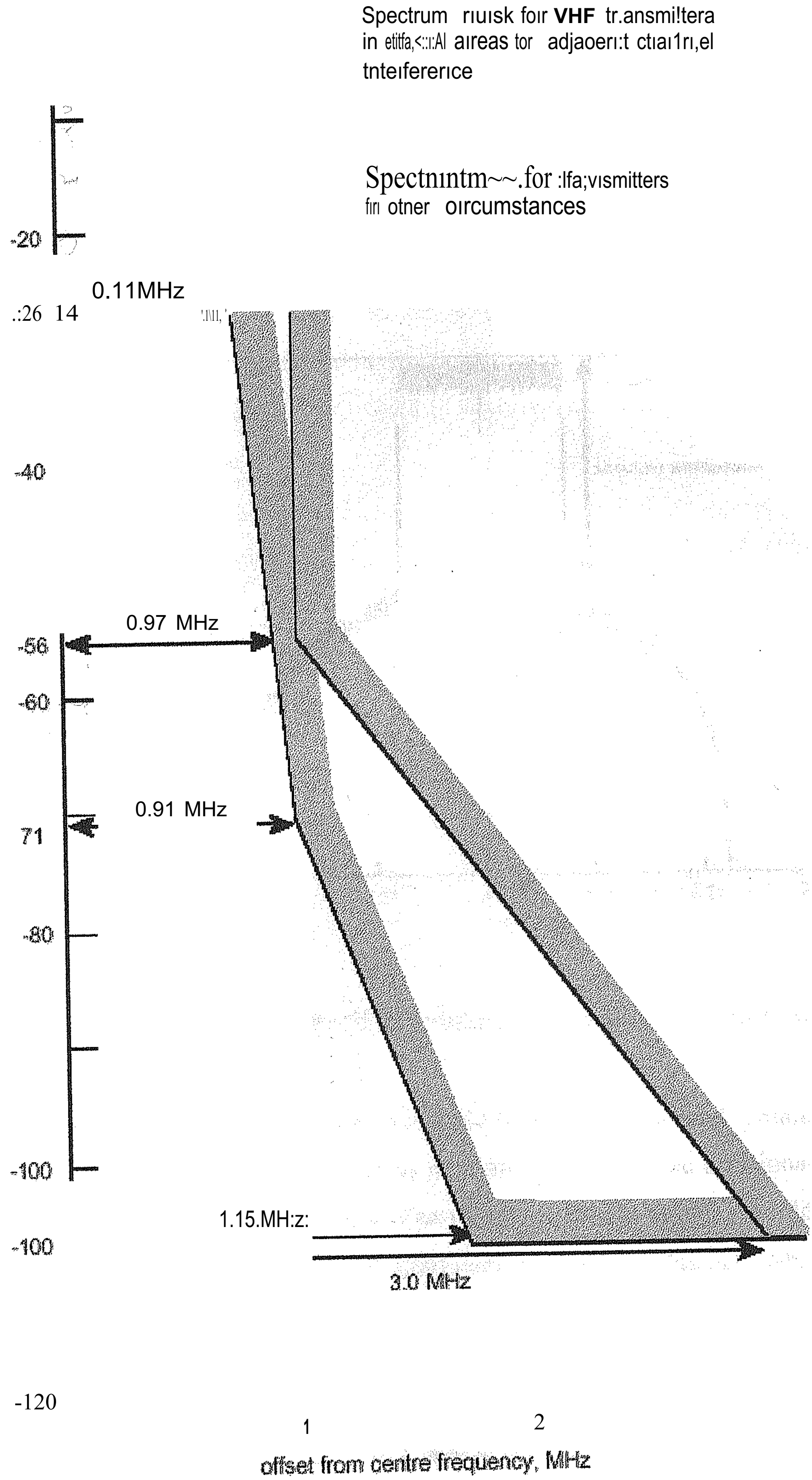
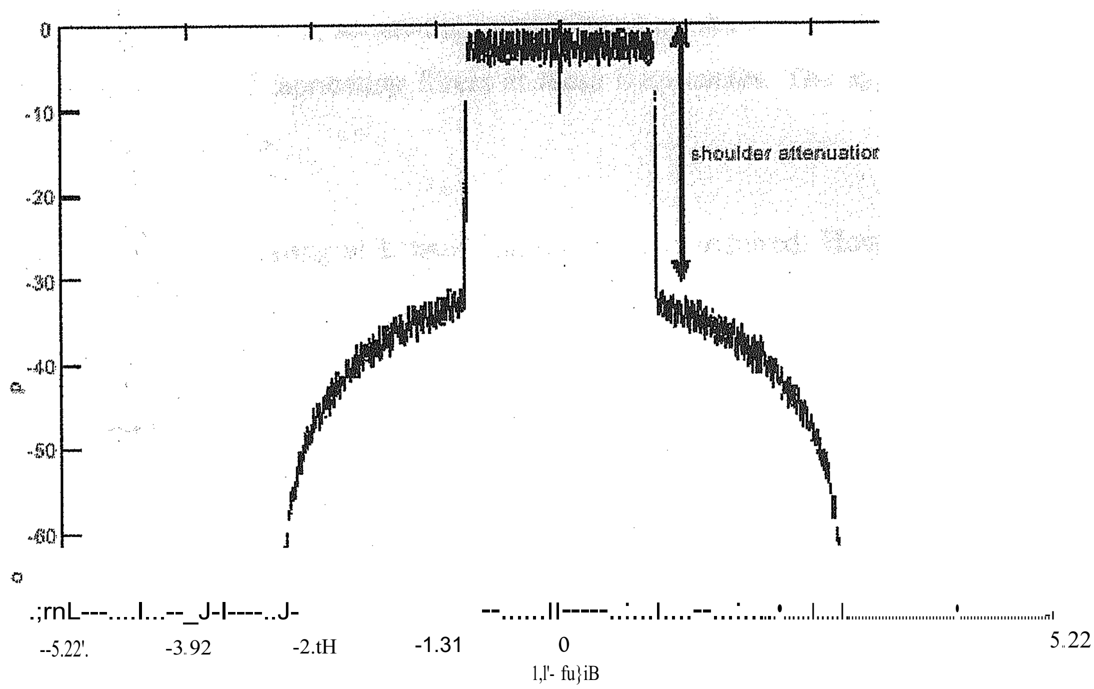


Figure 2.11: VBF speetrum mask

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 entput 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 introducinga higher-order filter, which may have a relatively high insertion loss and reduce the total power level of the wanted signal.



Flgure 2.12: OFDM spectrum showbıg shoulder attenuatien

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 2.12. Conventionally, its level has been measured as the ratiô 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.

- 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 2.13.

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

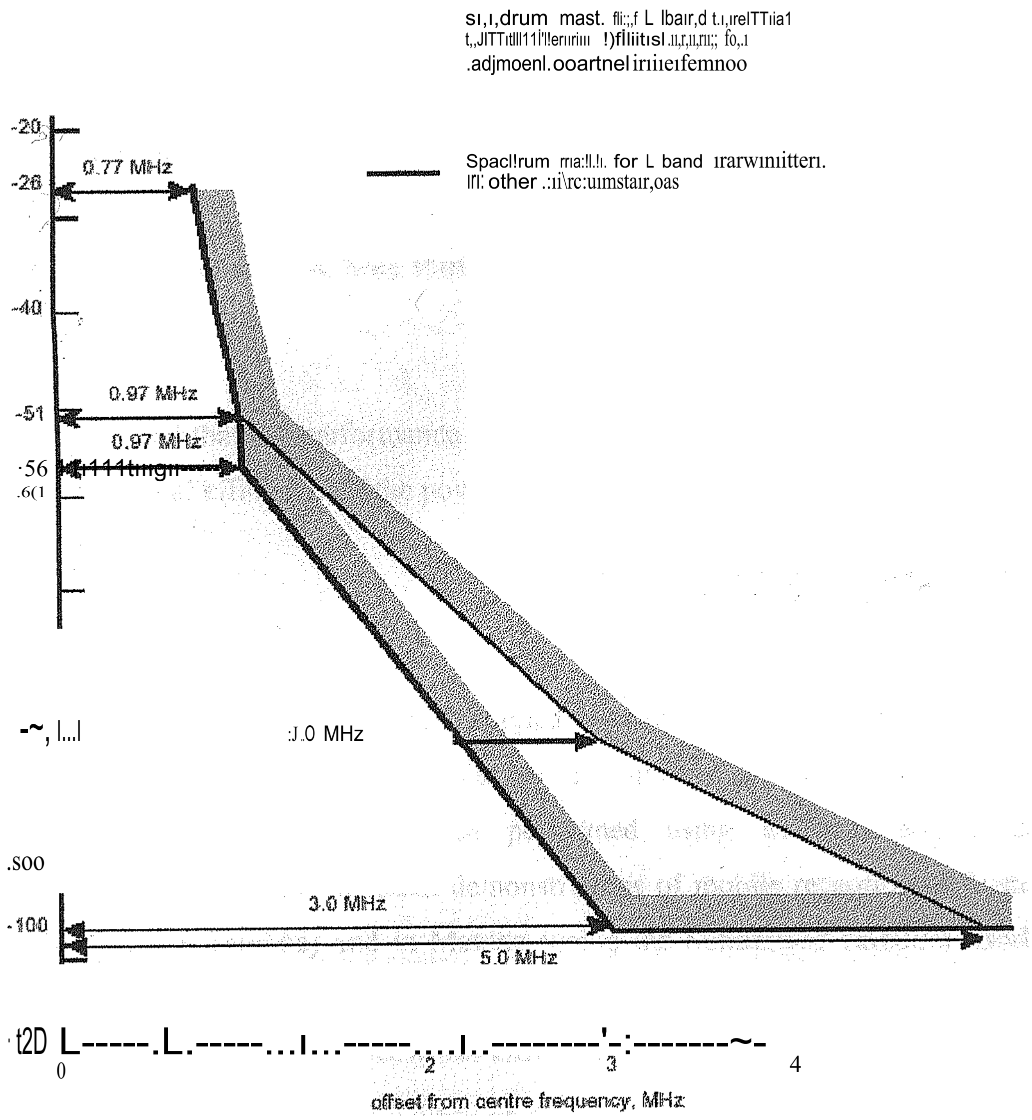


Figure 2.13: L-Band spectrum mask

2.4.3 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 DA.13 blôcks>areclosely 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-correctioıı techniques have been shown to improvethe powefi~f'fiçietiçyofpractical 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.

2.4.4 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 Mexicô (using the Solidaridad satellite). Both of these satellites used frequencies i111~jdi~telyadjacent to the L-barid allocation for DAB, although they had somewhatlower EIRP than might be expected from a genuine broadcast satellite. The success öf.thesei.experiments is sufficient to demonstrate that provision ofDAB services via satellit~c:lôes uot preserıt major technical difficulties.

Both Optus and Solidaridad were geôstationary satellites. Another satellite system that has been studied is the HEO (HighEIHptic Orbit) type constellation. ln Europe, the active HEO satellite would be received at elevation angles which are significantly higher than for geostationary satellites,uand.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 a.cceptablemobile reception at high latitudes, which is hard to achieve with a geostatioııary satellite..

A satellite link is usually power limited and therefore the target is to maximise the efficiency of the SSPA (Solid State PôWel" Arnplifier).>Asatellite output stage is likely to be implemented asa phase shift network driving several power amplifiers which feed a matrix network. This matrix finally fe~ds 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 fllterts) pass(es) the whole band. This combination causes a wide spectruın with relatively high necause ınany 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.~tage. It includes one DAB ensemble going through one SSPA stage. This exercise gives an order of magnitude estimate far the additional margin to the link budget which is required by the non-linear component using a COFDM-like signal.

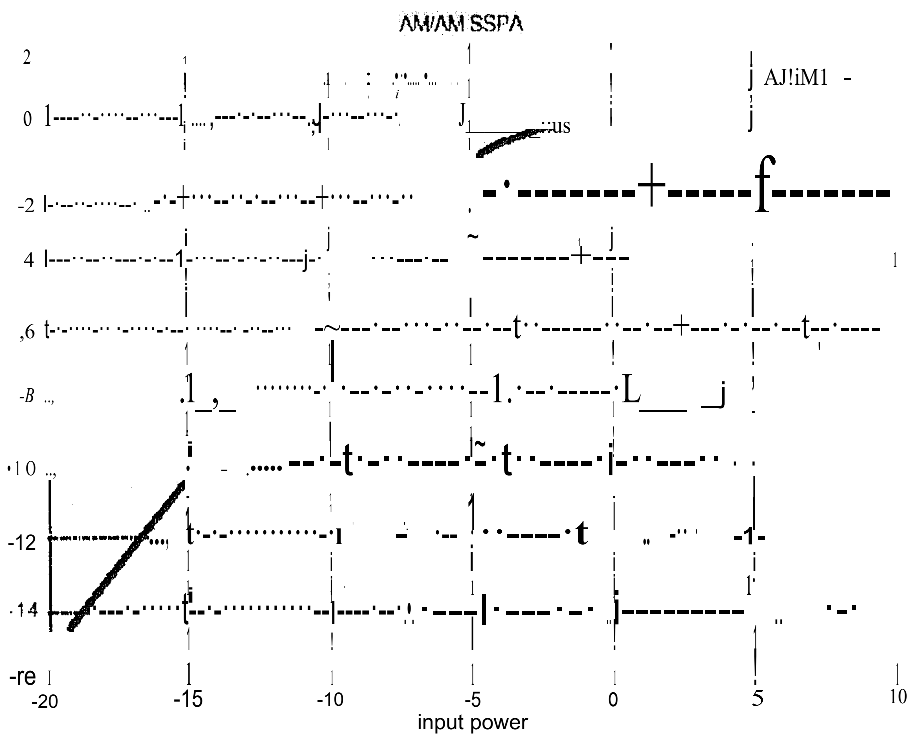
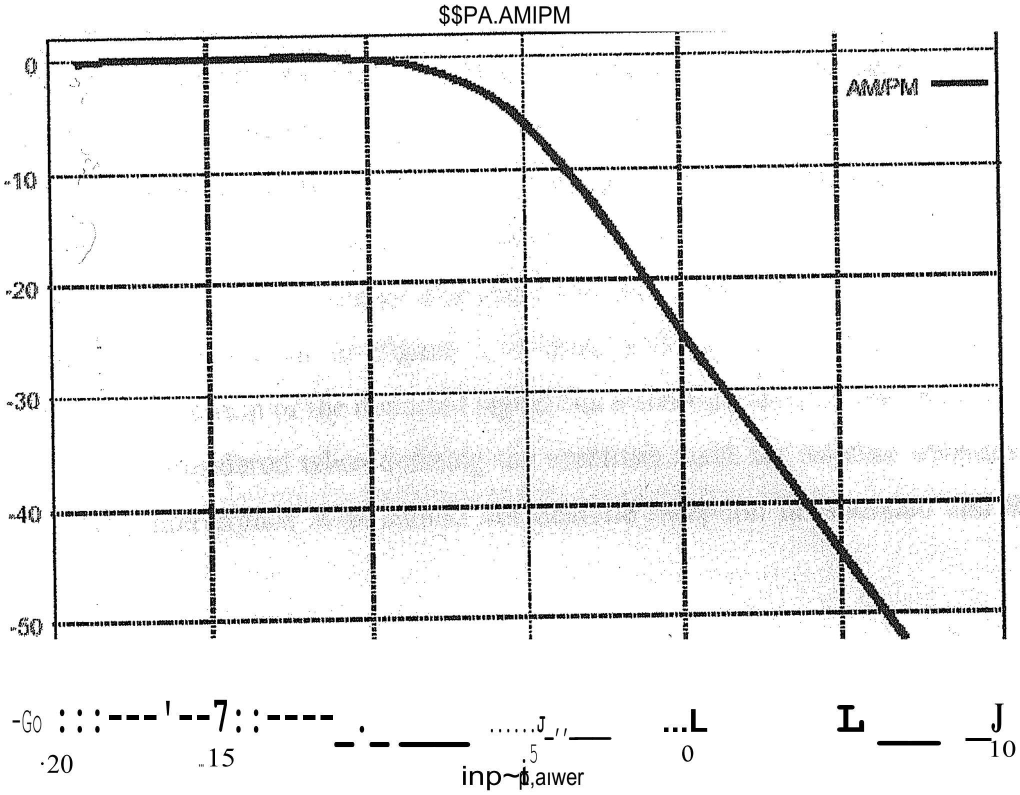


Figure 2.14: .AM/A.M eurve ofa tjpical SSPA.



Figare 2.15: AM/PM curveofaty.pical SSPA

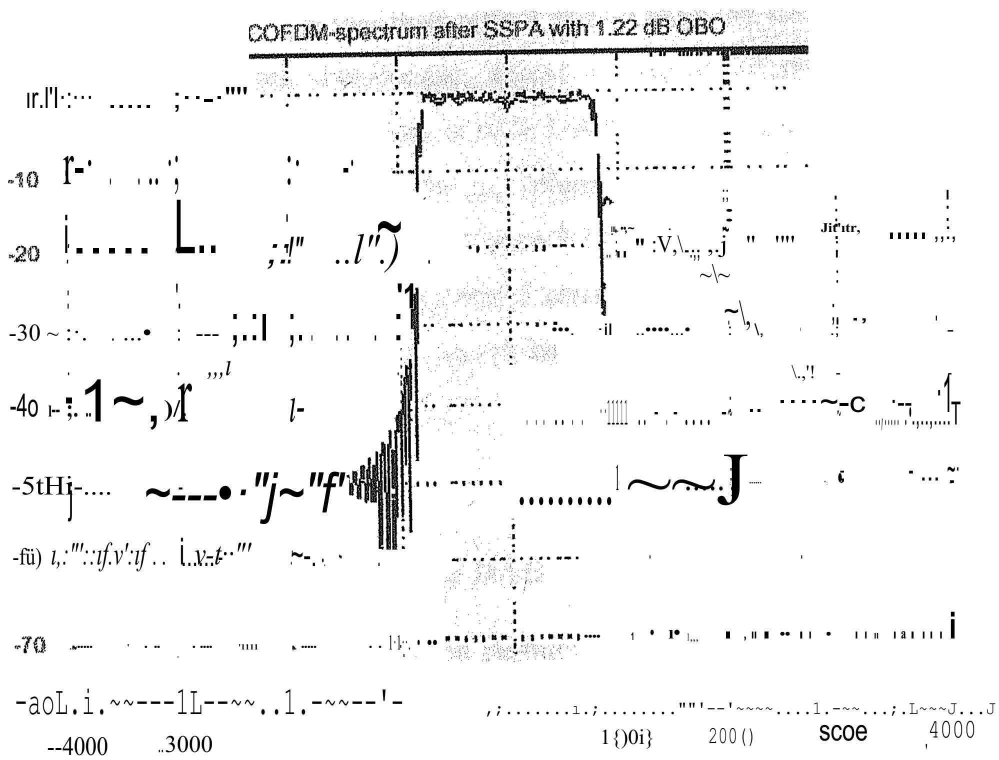


Figure 2.16: Spectrum ofa CQFDM signal before and after SSPA (see.text)

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The AMJAMand·AM!PMvaluesfö:fatypfüalSSPA(Figures 2.14 and 2.15) have been used in. the simulatioif to firid an' optimum bperatiôn 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 Ôperating in Transmission mode III. The optimum operating point was found to be an ()BO (Output Back Oft) 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 i:h Figüre 2.16 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, eertain considerations apply to such gap-fillers.

The first consideration is tlıattlı~i{ö~tJü{$iğnaı; ~~y;J~; ş~~p9wer spectrum as the output of the satellite .transponder, unless additional fı.ltering 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 2.13), there is a danger that aterrestrial 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.

2.4.5 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.

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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 speetrum efficiently under a range ofsharing scenarios,

The recommended frequencies **are shown in Tables** 2.5.1, 2.5.2 and 2.5.3.

Table 2.5.1 Baml **1 frequencies given priority in Eureka** 147 reeeivers

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 2.5.2 Band 3 freqmmcies given priority in Eureka 147 reeefvers

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
11'1 C	220.352
i1"D	222.064
~	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 2.5.3 L-Band freque!lc:ies ghıen priority in Eureka 147 receivers

Channel	Frequency (MHz)
L A	1452.960
L B	
L C	
L D	
L E	
iL F	
L G	
L H	
L I	1466.656
J	1468.368
	1470.080
i. L	1471.792
L M	1473.504
L N	1475.216
L O	1476.928
t p	1478.640
	1480.352
t R	1482.064
	1483.776
	1485.488
	1~7.2100
I, il n	
1452.816	
I	
&.	
"t' 41	
iL 5,	
il 6	1461.536
t 7	1463.280
I a	1465.024
1. g,	1466.768
li. Hr	1468.512
iL 11	1470.256
t. 1:;q	1472.000
I, 13	1473.744
I. M,	1475.488
I t!.i,	1477.232
iL SS	1478.976
i. rr	141'...0.,J!IJ
t Hl.	1482.464
1. H1	1484.208
iL 2'.(J,	1485.952
f. ?1	1487.696
i. 2:2	1489.440
li. 2:3	

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2.4.6 Expected Receiver Performance

- General

The European NomıEN50248, "Characteristics ofDAB Receivers", gives methods of measurement ofthe characteristics ofD.ABfeceivers:
DAB signal strengths and receive:rpara:ı:rieters-- targets for typical operation.

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

- Amplifier Linearity and Selectivity

Non-Iinearities of amplifiers, mixer stages ete. produce a distortion öfthe!signalitself (amplitude and phase errors) and subsequently interfering products inside and outside th.e transmitted band. These effects may cause additional bit errors and limit the maximum input power ofa receiver (and hence its dynamic range). Further, the out-of-band products may influence the.selectivityoftheteceiveı\ifthe.signalinthe adjacetıt channel is of the same power or ströri.gerthan the wan.tedisignalJintliis<case<the interference, generated by norı-lineaarities within the: receiver, lies iri.side the wafüed channel and cannot he removed by.Subsequeıtfiltering.

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

1. The wanted signal is embedded-insrblock ofDAB signals, each 1.54 MHz wide, and the complete block is receivedfrom a single transmitter,

2. The wanted signal is received:together.with an .adjacent DAB signal (or signals) which is (are) derived from otherttarısmitter(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 leastfrom another transmitter antenna.

Despite fading effects ofthe 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 .receiyets, a.chieved by applying one SAW filter of 1/2 jnch chip size and base-band filters of degree 5, may be taken as a guideline. The large-signal behaviour ofthe front-end was identical to that of present TV tuners.

The measurement conditions were: both .wanted an<.tinterfering signals are)JAB-Hke with 1.54 MHz bandwidth; centre fr~qµen.cy separation 9f J.7 ~ çe>rresp9ndingto about 0.17 MHz guard-band. Noise leveladjusted to giveaHERqfl0"5;·in.terferersetto a level which increases the BER to 10 ,

Wa!flıted~s~g~etı inpuıt ~em	seıec1i'ılil:y (BER = 10:41)
~ '90 dBm	46. ea
.,OOi/JBm	3.f3.d8

The selectivity at higher inputlevels 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.

- Dynamic Range

The input signal range is limited at lower l~vels 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 fora DAB receiver·in Band Ill. 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 of6 dB may be appropriate but this may be the composite some or ali ·of the following contributors: an active antenna's noise figure, cable attenuation and the receiver noise figure. Care has to be tak:en 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 ôerived rroıri 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 BandIIl will be at least 1 0 dB .less .than in FM networks due to the better behaviour <>füth~ digital (DAB)systefü. Thus, the feceiver input levels to be expected should alsôbelOdB less compated with the hiğhest·valtıes known from FM, which results in valu.es afoıind 0 dBm. L-band systeıtı.s may be characterised by physically smallef feceiving antennas, and possibly also . by transmittingantennas employing shapinğôfthe vertical radiation pattem. These factors may allow a further reduction in the requirenient for maximum input power, to around -15 dBm, but agaitfboth active anteıfü.a<arı.d receiver, as appropriate, will need to handle such power levels.

The dynamic ranges given in Table 2:6<wereachieved by prototype receivers which, in terms oftheir handling oflarge siğnalsiwereidentical to present TVtuners:

	Average value	Poorest value	
Modle t			
Ba.rııı:11	-98 ... -3	-95 ... -10	dBm
Band m	-96 ... -1	-93 ... -10	dBm
Mode II	-95 ... -1	-93 ... -10	dBm
Mode III	-95 ... -1	-92 ... -10	dBm

Mode IV: No figures available

Table 2.6: Inputdyna.mic>range of seme protetype DAB reeeivers

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 maximıım 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 inflnenced by noise and füterfereııce from the digital part of the receiver itself which is picked up by tbe 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.

- Miscellaneous

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

2. Laboratory measurements on prototype receivers (see Table 2.6) sho:w typical input sensitivities of about -96 dBm and worst case vahıes of about -93 qBmina

Gaussian channel for the VHFIUHFittarlsnıission<bands.>An average sensitivity of-93 dBm has been measutedinBa.ri.ffIII;fıeldtrial~in areaaffee.from man-made noise. These receivers were>füısedon television .tuners and receivers designed for reception of DAB c~tı.,be<expectedto offer superion performance. This would be offset to sonı.e.;~xtent by a margin to encompass receiver production tolerances. The.inöis~.':fiğuresnoted above (- 6 dB) are consistent with areceiver sensitivityôf'tli~törderof- 97 dBm.

3. Future receivers may ha:veCtlieiabilityto detect and suppress the effect of CW and narrow band •inter:fer~rsiExperimentalwork has demonstrated ·a substantial improvement may·beôbtainedinthe.case ofa single interferer.

2.5 Broatiç~~t~~fıvotPlanning 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 alk>cation of frequencies and co-ordination of' transmitters.. However, additional consideratıons 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 section.

All broadcast networks can be noise or interference limited. In practice, both apply in different areas of the network. How.ever, 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 arid1adjace:ntchannel interference in conjunction with propagation data for the radio waves are the hasis for transmitter frequency co-ordination. One of the key figures füiconventional planning is the reuse distance fota given frequency. This is thedistancebetweeritransmitters operating at thesarne fr~quency, that is necessary to reducethe C()ichatıtıel 'interference to the minimum level indicated by the protectfon.ratios.iProtectiofü:ratiosindicate the level of interference that is ,permissible in order to maintain .a Jceftaitı .minimum service quality. As radio propagation is time variant, worst case ·Sifüatiôns have to be considered. Normally, the planning techniques aim to provide the nıitıinum service quality for 99 % of time at the edge of.the-coverage area. ThereföreB'~tt~rr~ception.conditions are achieved at lower percentages of time, especially irisidetlı.e:•seı-yiceborders.

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. Cotıventional 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 Iocation 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.

2.5.1 Planning of Conventional Networks

Conventionally planned broadcasting rietwôrks consist of transn:ıittefs(within.depetıdent programme signals and with individuaFradio frequencies, (In contrasr to a SFN, the transmitters do not have to obey strictfüles of synchronous emission.) The allocation of the radio frequency for each transrnitter needs thorough calculation of the mıtual interference of all transmitters inside ari.d outside the network.

In analogue services a small violation ofthe protection ratio only results itı a smalF degradation of'the service quality. Asthedegradation is also limited in tiıne(see<aHôve) it is considered to be an economical compromise. For example, the FM se~iceöffersa.

margin of about 30 dB between a smalllôss ofquality 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 twith digital transmission, a.irel~ti~~ly.a.brupt break occurs when the RF signal conditions do not fulfil the protectiônıs~~i~/|~~ireıneıts.Therefore, within the service area.the RF signal conditions mustbeyşatiş:fi~dforahighpercentage oflocations; say 99%, as well as for a high percerttağe:.~rtiinet~ayalsô:99'%.

These'considerations leadto ·theitnpôrtant result••t1ıa~;,;~:itiô;at/cq11:yeııtiônalplartninğ methods cannot be used' ditectly for planning digital serviöes. Modi:fications are necessary to account for the differences in behaviour of analogue and digital systems, üne 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-channelinterference determines the reuse distance>forthe RF channel. The lower the protection ratio the smaller the reuse distance. As thE, prôfoction.ratio ofDAB is about 25 dB lower than the protection ratio ofFM a straight•fotward conclusion could be that the reuse distance in the DAB case is considerably smallerthan inthe FM case. Asa consequence, the number of RF channels needed to cover largeiareas with aı-least one RF signal would seem to be considerably smaller than in the FM casef'However, using this planning method, with its 50 % coverage criterion at the edgeofthe.service.area, in conjunction with the "brick-wall behaviour" ofa digital system,wôuldileave aIarge part of the envisaged service area unsetved, In conventional-planning, the,only way to avoid such gaps in. the service area is tö enlarge the distance -for: frequency re-use. Adding a margin tothe protection ratio isapragmatic wayto achleve 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 eonsideration are as large as 20 dB. In this casesthe 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

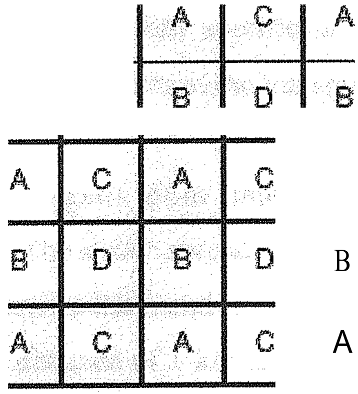
for conventional planning of DAB may norbe so much less than for planning analogue (FM) services.

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 utilisation 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 föt iı:ırnmatiôı:ıal. frequeı:ıcy<all61:trıents.The detailed planning of the network, i.e, the exact transmitter locations, ERP etc., is then performed using more sophisticated, terrain based techniques.

2.5.2 SingleFreiju.ericyNetwork

In an SFN, all transmitters are synchrortqı.:ısly ınç,qulaJ~d with the same signal and radiate on the same frequency. This network CQµç~pt.()ffer§1 111ll2h. 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 ()fSFN§ S~eıns to be very high compared to conventionally planned networks... Hovv~y~f, taki11.g into account the presence of similar networks offering other ens.eınbles in\agja2e11.t areas, further DAB channels are required fer international frequency iCO-Qrdination. If all service areas are. lafge enough, lrıtheqı-y four different channe.lsat'~ şµfficient to provide any of the areas withits individual ensen:ıble(see Figure 2.17'\VithDAB channels A, B, C and D). Each DAB channel can bere..used in the next but one area if the respective re-use distance is not less than aboiıt 100 km. However, in almost all practical situations, the location of transmitt~f}şites, local terrain, lower re-use distances and number of other factors may combinfto require the use of occur a fifth, sixth or even a seventh channel,



Figu:re 1.17: Assiğnınent of SFN blocks to regiens

Inside large areas the frequency channels of adjacent areas can be re-used if the rule' öf 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 efficientbııt also power efficient. This can be explained by considering the strong local variatiôns ôffield .str~ngth of any given transmitter. In conventionally planned networks, cöın~ö~!~!r t~ ac~:.:~ ~e~ice contimııty 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 componeııts froın 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. A.ccording 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 eften referred to as Network Gain.

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

When a demodulator8teç~ives sigı:ıals\from multiple transmitters, they appear like echoes of one original signal. The detay spread of such a "virtual" channel depends upon the distance betweeırthetrarisfüitters and the free space attenuation, which is itself a function of the frequeı:ıcyA.Cdista.nce of 1 km is equivalent to a propagation delay of about 3.3 μs.

OFDM systems cambe adapted to different multi-path environments by changing their three main.parameters.rThese are the inter-carrier distance F, the guard Interval A, and the symbol duration T. *i*In order to obtain the desired oıthogonality of the OFDM carriers, these parameters must fulfil the relation

('J.''' f!!:<< i>'lı.lıf' -

DAB provides a bandwidth anda data ratewhich areiı:ıdepen.derttoftheiselection ofthe OFDM parameters. This requires the ratio *A!Tto* be constant.Hence, only onedegree of freedom remains, e.g. the guard interval can be fixed according to tlı~ delı:ıy spı:ead. of the radio channel, and the other parameters depend upon that selection, .Alter:nativeIy, the inter-carrier spacing, F, can be fixed according to the Doppler spr~ad.yfor.agiven vehicle speed and frequency, in which case the other parameters wiltciepend on .that selection. This latter approach is described in depth in the following section,

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

The differential delayeef two siğrtals frôüFadjacent transmitters musr be, at most, smaller than the guard interval. Additionahheadroom for synchtonisation and normal multi-path propagation should be prôyidedYi This results in recommendations for maximum transmitter distances,

It should be .noted that these vahıes ate exaınples :from studies of ideal, regular lattices of transmitters which are syrtchrôriously modulated and have uniform- pôwers. Practical transın~)s~~~ ~i,f\ııtjable and depend upon topograplıiC~ considerations. However,.there are also some situations where transmitters shoutd be purposely delayed or advanced, .côınpared to ideal synchronicity, in order to improve the coverağe.·.This is· partföularly tıııe if the SFN. inchıdes both high and low pôwer transmitters. Because öf"its hiğh power, the signal of a strong trarisınitter cari be received at a large distarice 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 synchronised. Tpis differential delay can be reduced by advancing the strong transmitter relative to the SF.J'.!•F ıneaıı\delay, or conversely by delaying the weak transmitter.

Gaps in the coverage area of an SFN are easily filled by addirıg one rteW tra:tısmitter without the need for additional :frequencies. This technique offers ai.very>efficient spectrum utilisation, especially in large area networks for national or reğiôtıaLservice coverage. This is true as long as the whole DAB ensemble is filled by services with the same required coverage area.

2.5.3 Calculatlon ofThe VehicleSpeed at Which DAB Reception Beeomes Degraded

Whenever a DAB signal is received in .a moving vehicle, especially when there is multipath propagation, there is like1y to be some degradation of 'perfonnance('I'he COFDM signal has been optimised 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 perforınance çan be :related to a parari:ıeter p. This carı. be interpreted as the repre$eij.ti9µ 9f~lı.~{qişfpJijçeınent öf the wehicle expressed in number of wavelengths during one Syı:ri.bôfduta.tion Ts when the vehicle has a speed of v (metres-per second).

Let β = fmax.Ts = (v.fo/c).Ts = v.Ts/λ (2.1)

The reference value föf Pis o.o&ifor a 4 dB degradation at approx. 10 -3 BER in the most difficult ınultip~th>corıditions (dispersive Doppler effect, constant probability density of the received power over the 21t range of reception in the horizontal plane, as opposed to a simple Doppler shift). Putting these figures into equation 2.1:

β = Ts . fo . v/c = 0.08

(lu + .1) . fo . v.IC ;;; (U)B

(Tu + Tu/4} fo . vIC .:ı 0.0a

{5lfuf4)ifo . v/c ;;; 0.08

Tü.ro.vJ:C = (U)64 (2.2)

Equation (2.2) represents the speed versus frequency curves with the symbol duration Tu as a parameter. 'This is a function ofthe Mode.

For c = 3 x 10^8 m/s equation (2.2)

we obtain,

TuJo.v ;;: o.064 x 3 :ıı: 1oı.ı = 1·9.2 x 101$ (2.3)

with

Tu = useful symbol duration in seconds

fo = frequency in Hz and v = vehicle speed in m/s

When in equation (2.3), fo is expressed in MHz and v in km/h then

$$Tu.fQ,\_V \;=\; 0.064 \times 3 \times \frac{10^8 \times 3600}{-l,,rf \times 10^3} \sim 10 \text{ (approx)} \quad (2.4)$$

Sothat;

70/Tu.fo and (2.5)

iOfru (2.6)

By means of equation (2.5) the maximıım speed can be calculated that is possible ata certain frequency. By means of equation (2.6) the maximum frequency can be calculated that is possible at a certain vehicle speed.

Examples:

1) Calcufatfön of the maximum speedvthat is possible in the 4 modes I,Il,ID, IV for a nominal frequency fo of 375 MHz, 1.5 GHz, 1.5 MHz and 3 GHz respeenvelr.

Môde I : Tu = 1 ms = 0.001 s and fo = 375MHz
From equation 2.5: the maximum speed isii7.0/0.00lx375= 186 km/h(point A in Figure2.18)

Mode IV: Tu = 500 ms = 0.0005 s and fo = 1.5 GHz = 1500 MHz
From equation 2.5: the maximum speed is 93 km/h (point Bin Figure 2.18)

Mode Il : Tu = 250 ms = 0.00,025s and fo = 1.5 GHz = 1500 MHz
From equation 2.5: the maximum speed is 186.km/h (point Cin Figure 2.18)

/Mode fil: Tu = 125 ms = 0.000125.s and fo = 3 GHz =3000MHz

Fromequatfon•2.5: the maximum speed is 186 km/h (point D in Figure 2.18)

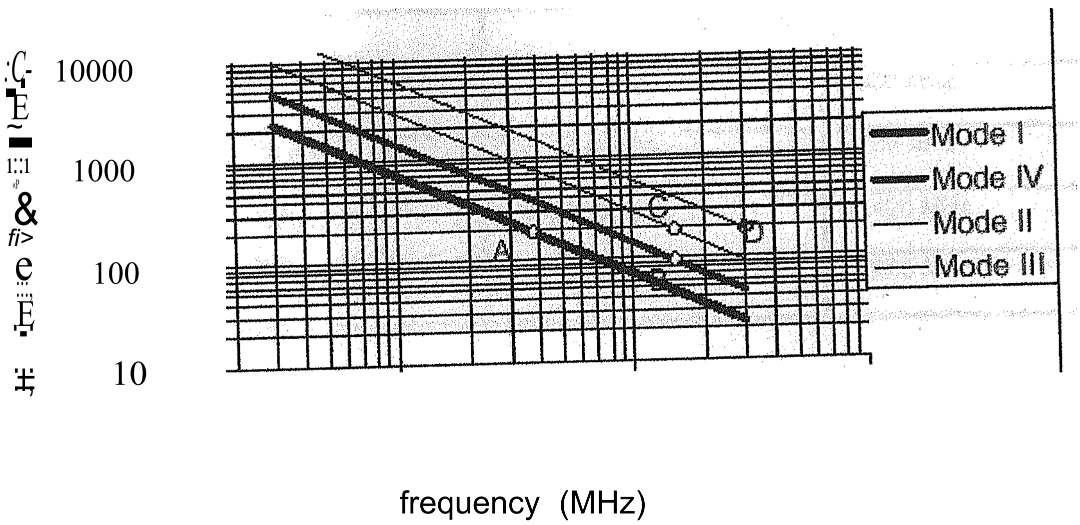


Fig 2.18 - Maximum Speed versus Freijie1icf fôr Beta:::: ö1ôs

2) Calcalatien of the max, speed at 100 MBz

Using equation 2.5:

			maximum speed
Mocte I :	0.001 El:	70/{t001x100	= 700 km/h
Mode !IV:	0.0005 s;	70/0.0005x100	= 1400 km/h
Mode ll .	o.0002ss:	70/Q.00025x100	.:: 2.800 km/h
Mode lil :	0.00125 s;	70/0.000125x100	= 5600 km/h

3) Calculatıon of the max, usable **frequency per mode when** moving at a speed of 200km/h

Using equation 2.6:

			maximum frequency
Modıe 1:	0,.001 s	70/0.001x200	= 350 MHz
Mod!e IV:	0·.0005 s	70/0.0005x200	= 700 MHz
Mocte, H:	0.00025 ıs	701(f.00025ı<200	= 1400 MHz
Modem:	0.000125 s	70/0.000125x200	= 2800 MHz

) Calculatlon of the max, vehiele speed at L-Band at 1.5 GHz

Only applicape to Mode II,·III and IV)

	Tı.ı	maximum speed	
Mode II :	0.00025 s	70/0.00025x1500	= 186km/h
Mooe m:	0·.000125 s	70/0.000125x1500	= 373 km/h
Mode IV :	0.0005 s	7'()/{),,Qı(J(Jfü\1500	= 93 km/h

5) Otber values of β :

In some papers such as in the Montreux proceedings and the speed/frequency curves are drawn in such a way that for factor β a vahıe of 0.0625 is taken rather than 0.08.

Tu.fcuı/ıo = 0.0625 x 4/5

Tu.fo.v = $0.00625 \times 4 \times 3 \times 10^{8} / 5 \times 10^{6}$

When fo is expressed in MHz and v in km/h then Tu.fo.v = 55

so that:

v = 55/Tu.fo

fo;:;;; 55/Tuuı

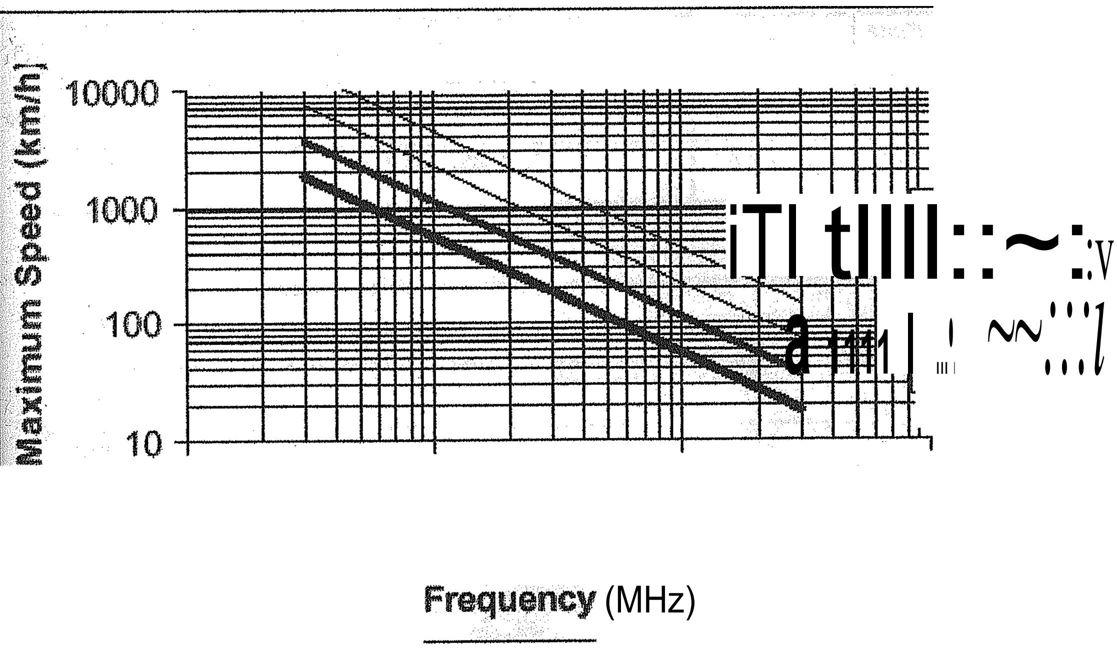


Fig 2.19 - Maximum Speecf~~tsüs Frequeney for Beta = 0.0625

Calculated examples for the case that β = 0.0625:

Calculation of the maximum speed v that is possible in the 4 mode, I, II, III, IV for nominal frequency fo of resp. 375 MHz, 1.5 GHz and 3 GHz.

		Speed	
I:	55/0.001*375	=147	km/h
Mode IV:	55/0.0005x1500	= 73	km/h
Mode II:	55/0.00025x1500	=147	km/h
Mode III:	55/0.000125x3000	= 141	km/h

Calculation of the max. speed at 100 MHz

		Speed	
Mode I:	55/0.001x100	= 550	
Mode IV:	55/0.0005x100	=1100	km/h
Mode II:	55/0.00025x 100		km/h
Mode III:	55/0.000125x100	= 4400	km/h

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

		Max Frequency (fo)	
Mode I:	5/0.001x200	= 275	MHz
Mode IV:	55/0.0005x200	= 550	MHz
Mode II:	55/0.00025X200	= 110	MHz
Mode III:	55/0.000125x200	=2200	MHz

Calculation of the max. vehicle speed at L-Band at 1.5 GHz

		Speed	
Mode II:	55/0.00025x1500	= 147	
Mode III:	55/0.000125:x 1500		km/h
Mode IV:	55/0.0005x1500	= 13	

nensional frequeney-delay domam

le-off between RF frequency and the maximum delay)

car is driving at high speed it is picking up signals coming from the front ing to a positive Doppler shift) and signals corning from the rear (negative Doppler . The worst-case condition occurs when the two signals have nearly the tude. Satisfactory recepti on is assumed when under this worst-case coridition num equivalent noise degradation is less than 4 dB at 1o-4:BER.

speed constraint of 200 km/h, the maximum frequeıı~y that caıı cöpe. with the num multipath delay can be calculated.

equation 2.1:

= Ts . fo . v/c=

Tu + &), fo.w/c= OJJ8

(5 Δ). fo .v/c= (1.08

edo.<ıp.• QJJ8 x ş x 1t;t
5

(2.7)

in equation (2.7) fo is expressed in MH.zand v in km/h equation (2.7) then :

Δfo.v = 0.0§ x :3 ;ır; 10~:x 3600 = 17.28
5 :::: 10'3:X 103

(2.8)

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

C/I = 10 dB and Δ = Tu/4 then:

$$\tau_m = 1.2 \times \Delta$$

or

$$\Delta = \tau_m / 1.2$$

$$\tau_m . fo . v = 17.28 \times 1.2 = 20.736 \quad (2.9)$$

a vehicle speed v = 200 **km/h equation (2.9) becomes :**

$$\tau_m . fo = 20.736/200 = 0.10368 \quad (2.10)$$

For a given vehicle speed of 200 km/h, equation (2.10) allows us to derive the maximum usable frequency fo in function of the maximum delay τ_m beyond which addition of delayed signals causes degradation. In table 2.7 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 oflight.

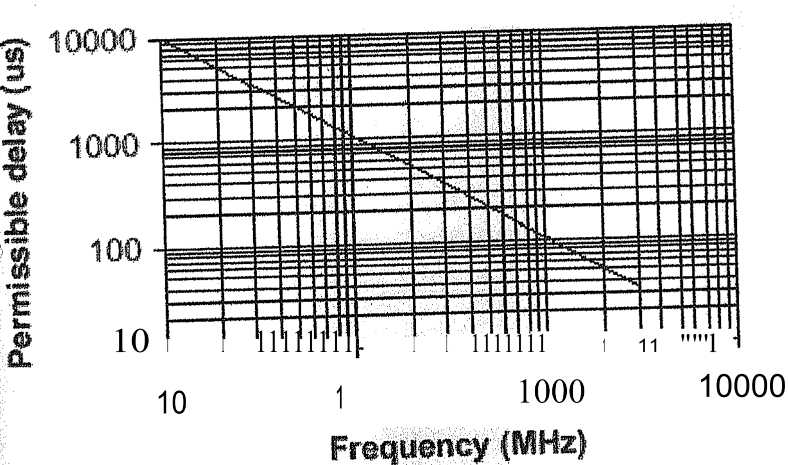


Fig 2.20 - Permissible **delay as a function of** frequency fer a vehiele movhıg at 200 **km/h**

2.7: Yariation oflimits of **frequency and transmission distance as a function of Mode**

	Mode I	Mode IV	Mode II	Mode III
	246.ms	123 ms	61.5 ms	30.7 ms
$= \Delta \times 1.2$	300 ms	150 ms	75 ms	37 ms
$o = 0.10368/ \tau_m$ $v = 200$ km/h)	345 MHz	690 MHz	1280 MHz	2800 MHz
Maximum transmission distance $d = c \times \tau_m$		45 km	22 km	11 km

2.5.4. 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 minimise the re-use distance, and hence optimise 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 localised 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.

Chapter Three

Concluding the Project

Conclusions :

Since the beghıning of the radio broadcast era, frequency planning aims to avoid the interference caused.by the overlapping of the transmitters service areas. Unfortunately, transmitters overlap is not the unique source of interference; the terrestrial channel has a complex propagatioll model which produces echoes (multi-path propagation) and when addressing mobile receivers, Doppler frequency shift. As a consequence, in each point of a service area, the signal captured by the receivers results of the sum of several elementary signals including the original signal, some delayed replicas and channel noise.

To bypass this physical degradation, the traditional method was to increase the power of the original signal (e.g.: the transmitting power). Asa direct consequence, this method enlarges the lilllitôfthe•channel re-usability and accordingly,.contributes the artificially in.creasethe radıôfreqtıency spectrum occupancy.

A modulation system have been studied which is sufficiently robust and efficient to carry digital signals and to save radio frequency spectrum; the Coded Orthogonal Frequency Division Multiplex (COFDM).

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