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Digital Audio Broadcasting (DAB);
Guidelines and rules for implementation and operation;
Part 1: System outline

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Foreword

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

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

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

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

The present document is part 1 of a multi-part deliverable covering Guidelines and rules for implementation and operation for Digital Audio Broadcasting (DAB), as identified below:

Part 1: "System outline";

Part 2: "System features";

Part 3: "Broadcast network".

1 Scope

The present document gives guidelines for the implementation and operation of the Digital Audio Broadcasting (DAB) system. It forms Part 1 of the guidelines and rules developed by the Eureka Project 147 as the major companion document to the DAB system specification given in EN 300 401 [1]. They are intended to provide additional information to aid interpretation of the on-air signal and to assist broadcasters and manufacturers to implement systems using the specification features as intended. Part 1 focuses on the system outline. TR 101 496-2 [15] considers in detail the implementation and operation of the system features. TR 101 496-3 [16] focuses on the broadcast network.

2 References

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

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

[2] ITU-R Recommendation BS.774-2: "Service requirements for digital sound broadcasting to vehicular, portable and fixed receivers using terrestrial transmitters in the VHF/UHF bands".

[3] ITU-R Recommendation BO.789-2: "Service for digital sound broadcasting to vehicular, portable and fixed receivers for broadcasting-satellite service (sound) in the frequency range 1 4000 - 2 700 MHz".

[4] ITU-R Recommendation BS.1114-1: "System for terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz".

[5] ITU-R Recommendation BO.1130-2: "System selection for digital sound broadcasting to vehicular, portable and fixed receivers for broadcasting-service satellite (sound) bands in the frequency range 1 400 - 2 700 MHz".

[6] ISO/IEC 11172-3 (1993): "Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s - Part 3: Audio".

[7] ISO/IEC JTC-1-SC29-WG11 MPEG 91-101 (1991): "The SR Report on The MPEG/Audio Listening Tests" - Stockholm.

[8] EN ISO 14819-1: "Traffic and Traveller Information (TTI) – TTI messages via Traffic Message Coding – Part 1: Coding protocol for Radio Data System – Traffic Message Channel (RDS-TMC) using ALERT-C".

[9] ETSI ETS 300 174 (1992): "Network Aspects (NA); Digital coding of component television signals for contribution quality applications in the range 34 - 45 Mbit/s".

[10] EN 50094 (1992): "Access control system for the MAC/packet family: EUROCRYPT".

[11] Norwegian Telecom, Issue 2 (20th July 1989): "NR-MSK Access Control System".

[12] ISO/IEC 13818-3 (1998): "Information technology - Generic coding of moving pictures and associated audio information - Part 3: Audio".

[13] ETSI TS 101 500: "Digital Audio Broadcast System (DAB) - Multichannel audio".

[14] ITU-R Recommendation BO.955-3: "Satellite sound broadcasting to vehicular, portable and fixed receivers in the range 500-3 000 MHz".

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

[16] ETSI TR 101 496-3: "Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 3: Broadcast network".

3 Definitions, symbols and abbreviations

3.1 Definitions

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

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

audio bit stream: sequence of consecutive audio frames

audio frame: frame of a duration of 24 ms (at 48 kHz sampling frequency) or of 48 ms (at 24 kHz sampling frequency) which contains a Layer II encoded audio signal ISO/IEC 11172-3 [6], ISO/IEC 13818-3 [12], corresponding to 1 152 consecutive audio samples; the smallest part of the audio bit stream which is decodable on its own

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

Auxiliary Information Channel (AIC): all or part of sub-channel 63, used to carry information redirected from the Fast Information Channel

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

change event indication (CEI): set of FIG fields with particular values to indicate a change of database content for certain service information features

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

Conditional Access (CA): mechanism by which the user access to service components can be restricted

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

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

DAB transmission signal: transmitted radio frequency signal

data service: service which comprises a non-audio primary service component and optionally additional secondary service components

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

Entitlement Checking Messages (ECM): messages containing information about the conditions required for accessing service components, which are intended for restricted access, and for descrambling the data

Entitlement Management Messages (EMM): messages containing information about the conditions required for accessing service components which are intended for restricted access and for descrambling the data

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

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

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

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

Fast Information Data Channel (FIDC): dedicated part of the Fast Information Channel which is available for non-audio related data services, such as paging

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

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

joint stereo mode: audio mode in which two channels forming a stereo pair (left and right) are encoded within one bit stream and for which stereophonic irrelevance or redundancy is exploited for further bit reduction
The method used in the DAB system is Intensity stereo coding.

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

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

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

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

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

packet mode: mode of data transmission in which data are carried in addressable blocks called packets
Packets are used to convey MSC data groups within a sub-channel.

primary service component: first and mandatory component of a service
It can be used as a default selection in the receiver.

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

programme item: time-slice of a programme, for example, a piece of music or a news report

programme service: service which comprises an audio Primary service component and optionally additional Secondary service components

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

protection profile: Defines the scheme of convolutional coding applied.

secondary service component: In case a service contains more than the primary service component, the additional service components are secondary service components.

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

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

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

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

service label: alphanumeric characters associated with a particular service and intended for display in a receiver

single channel mode: audio mode, in which a monophonic audio programme is encoded within one bit stream

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

stereo mode: audio mode in which two channels forming a stereo pair (left and right) are encoded within one bit stream and for which the coding process is the same as for the Dual channel mode

stream mode: mode of data transmission within the Main Service Channel in which data are carried transparently from source to destination
Data are carried in logical frames.

sub-channel: A part of the Main Service Channel which is individually convolutionally encoded and comprises an integral number of Capacity Units per Common Interleaved Frame.

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

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

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

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

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

3.2 Symbols

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

N	number of carriers
p	padding
R_{ave}	mean code rate
Tf	frame duration
T_{null}	null symbol duration
T_s	Total symbol duration
t_s	useful symbol duration
t_Δ	guard interval duration

3.3 Abbreviations

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

AIC	Auxiliary Information Channel
AM	Amplitude Modulation
C/N	Carrier to Noise ratio
CA	Conditional Access
CIF	Common Interleaved Frame
CRC	Cyclic Redundance Check
CU	Capacity Unit
DAB	Digital Audio Broadcasting
DFT	Discrete Fourier Transform
DRC	Dynamic Range Control
EAN	European Article Number
EBU	European Broadcasting Union
ECC	Extended Country Code
ECM	Entitlement Checking Message

EEP	Equal Error Protection
EMM	Entitlement Management Message
EWS	Emergency Warning System
FFT	Fast Fourier Transform
FIC	Fast Information Channel
FIDC	Fast Information Data Channel
FM	Frequency Modulation
F-PAD	Fixed Programme Associated Data
HEO	Highly inclined Elliptical Orbit
IEC	International Electrotechnical Commission
ISO	International Organization for Standardization
ISRC	International Standard Recording Code
ITTS	Interactive Text Transmission System
ITU	International Telecommunications Union
LTO	Local Time Offset
MCI	Multiplex Configuration Information
MJD	Modified Julian Date
MPEG	Moving Picture Experts Groups
MSC	Main Service Channel
MUX	Multiplex
OE	Other Ensemble
OFDM	Orthogonal Frequency Division Multiplex
PAD	Programme Associated Data
PCM	Pulse Code Modulation
PTy	Programme Type
QPSK	Quadrature Phase Shift Keying (4-PSK)
SC	Service Component
SCCA	Service Component Conditional Access
SFN	Single Frequency Network
SI	Service Information
TII	Transmitter Identification Information
TMC	Traffic Message Channel
UEP	Unequal Error Protection
UHF	Ultra High Frequency
UPC	Universal Product Code
VHF	Very High Frequency
X-PAD	Extended Programme Associated Data

4 General Outline

4.1 System overview

The Eureka DAB system is designed to provide reliable, multi-service digital sound broadcasting for reception by mobile, portable and fixed receivers, using a simple, non-directional antenna. It can be operated at any frequency up to 3 GHz for mobile reception (higher for fixed reception) and may be used on terrestrial, satellite, hybrid (satellite with complementary terrestrial), and cable broadcast networks. In addition to supporting a wide range of sound coding rates (and hence qualities), it is also designed to have a flexible, general-purpose digital multiplex which can support a wide range of source and channel coding options, including sound-programme associated data and independent data services. It is, in fact, the only system available in the world which is able to meet all of the demanding requirements drawn up within the International Telecommunications Union (ITU), in order to take a new and revolutionary step in all-digital sound broadcasting, and having a long-term future. These requirements are given in ITU-R Recommendations BS.774-2 [2] and BO.789-2 [3]. The system itself is recommended world-wide by the Inter-Union Technical Committee of the World Conference of Broadcasting Unions and now in ITU-R Recommendations BS.1114-1 [4] and BO.1130-2 [5] and, for terrestrial and satellite broadcasting respectively. The detailed specification of the Eureka DAB System (also known as ITU Digital System A) is given by ETSI in EN 300 401 [1].

The Eureka DAB system is a rugged, yet highly spectrum- and power-efficient sound and data broadcasting system. It uses advanced digital techniques to remove redundancy and perceptually irrelevant information from the audio source signal, it then applies closely controlled redundancy to the signal to be transmitted, to provide strong error protection. The transmitted information is spread in both the frequency and time domains so that the defects of channel distortions and fades can be eliminated from the recovered signal in the receiver, even when working in conditions of severe multi-path propagation, whether stationary or mobile. Efficient spectrum utilization is achieved by interleaving multiple programme signals and, additionally, by a special feature of frequency re-use, which permits broadcasting networks to be extended, virtually without limit, by operating additional transmitters carrying the same multiplexes on the same radiated frequency. The latter feature is known as the Single Frequency Network (SFN). This can also employ the gap filling technique. In this case, a gap filler transmitter receives and re-transmits the signal on the same frequency without demodulation and remodulation. This provides coverage of shadowed areas, which can arise within the overall coverage area provided by the main broadcast network transmitters. Nevertheless, the relatively low co-channel protection ratio of the DAB system also permits adjacent local coverage areas to be planned, on a continuously extending basis, with as few as four different frequency blocks.

4.2 Summary of the major system features

The system provides a signal which carries a multiplex of several digital services simultaneously. The system bandwidth is about 1,5 MHz, providing a total transport bit rate capacity of just over 2,4 Mbit/s in a complete "ensemble". Depending on the requirements of the broadcaster (transmitter coverage, reception quality), the amount of error protection provided is adjustable for each service independently, with a coding overhead ranging from about 33 % to 300 % (200 % for sound). Accordingly, the available bit rate for broadcast services ranges between about 1,7 Mbit/s and 0,6 Mbit/s. The services can contain audio programme data or other data services, and a data service can or can not be related to the audio programme. The number and bit rate of each individual service is flexible and generally receivers are able to decode several service components or services simultaneously. The actual content of the flexible multiplex is described by the so-called Multiplex Configuration Information (MCI). This is transported in a specific reserved part of the multiplex known as the Fast Information Channel (FIC), because it does not suffer the inherent delay of time interleaving which is applied to the Main Service Channel (MSC). In addition, the FIC carries information on the services themselves and the links between the services.

In particular, the following principal features have been specified:

- Audio bit rates from 384 kbit/s down to 32 kbit/s, or even down to 8 kbit/s by applying the half sampling frequency coding technique of MPEG-2 Audio Layer II [12]. This enables the multiplex to be configured to provide typically 6 high-quality stereo audio programme using MPEG-1 Audio at the full sampling frequency or up to 63 mono programmes using MPEG-2 Audio half-sampling frequency coding technique with moderately rugged error protection. An example table of multiplex options for audio services is given in table 4.2.1.

- Program Associated Data (PAD), embedded in the audio bit stream, for data which is directly linked to the audio programme (e.g. dynamic range control data, song lyrics, music/speech flag, etc.). The amount of PAD is adjustable (minimum 667 bit/s with MPEG-1, or 333 bit/s with MPEG-2 Audio at half sampling frequency), at the expense of capacity for the coded audio signal within the chosen audio bit rate.

- Data services, whereby each service can be a separately defined stream or can be divided further by means of a packet structure.

Table 4.2.1: Examples of audio service capacities in a DAB ensemble (Equal Error Protection)

Protection level	3A	4A
mean code rate, R_{ave}	~0,5	~0,6
coded audio rate, kbit/s	No. of audio services	
24	48	63
32	36	41
64	18	20
128	9	10
192	6	7
224	5	6
256	4	5

- Conditional Access (CA), applicable to each individual service and to each individual packet of packet mode data. (Specific subscriber management does not form part of the DAB system specification [1]; DAB provides CA transport and the actual signal scrambling mechanisms.)
- Service Information (SI) for (textual) information on the selected DAB ensemble and selected programme, and also complementary machine code for ease of operation of the receiver. Another important SI-feature is to establish links between different services in the multiplex and links to other (related) services in another DAB multiplex or even to FM/AM broadcasts.

4.3 Outline of system implementation

4.3.1 General

A conceptual block diagram of the DAB system is shown in figures 4.3.1 and 4.3.2; figure 4.3.1 shows a conceptual transmitter drive in which each service signal is coded individually at source level and then error protected and time interleaved. Then it is multiplexed into the Main Service Channel (MSC), with other similarly processed service signals, according to a pre-determined, but changeable, services configuration. The multiplexer output is frequency interleaved and combined with multiplex control and service information which travel in a Fast Information Channel (FIC) in order to avoid the delay of time-interleaving. Finally, very rugged synchronization symbols are added before applying Orthogonal Frequency Division Multiplexing (OFDM) and differential QPSK modulation onto a large number of carriers to form the DAB signal.

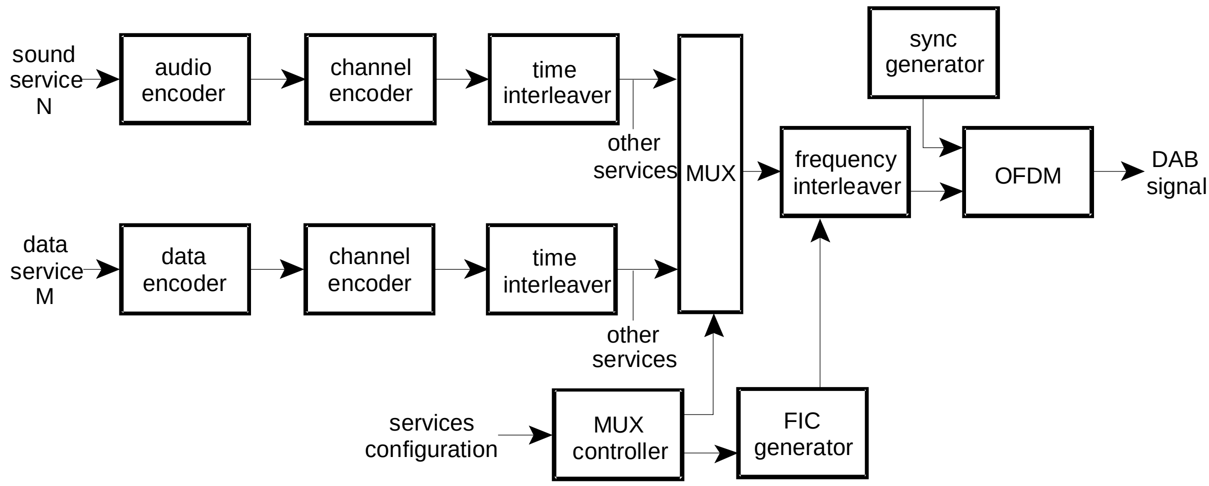

Figure 4.3.1: Conceptual block diagram of the EUREKA DAB system transmitter drive

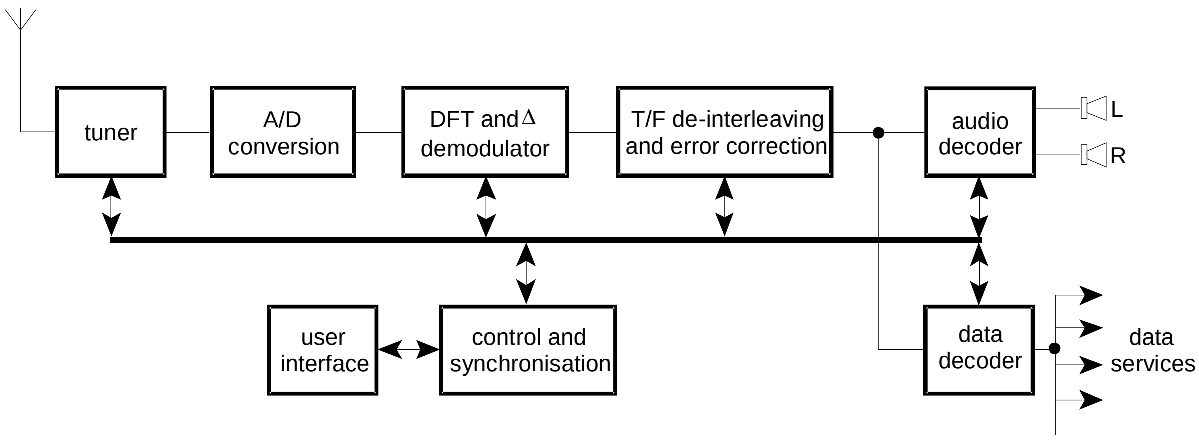

Figure 4.3.2: Conceptual block diagram of the Eureka DAB system receiver

Figure 4.3.2 shows a conceptual receiver, in which the received signal is selected, down-converted and quadrature demodulated before applying it to an analogue-to-digital converter pair. Thereafter, the receiver performs the transmitter operations of Fig. 4.3.1 in reverse order, having selected the wanted DAB ensemble and acquired synchronization. Thus selection is done in the analogue tuner, which performs the tuning and filtering functions. The digitized output of the converter is first fed to the DFT (Discrete Fourier Transform) stage and differentially demodulated. This is followed by time and frequency de-interleaving processes, and error correction to output the original coded service data. That data is further processed in an audio decoder, producing the left and right audio signals, or in a data decoder as appropriate. The decoding of more than one service component from the same ensemble, e.g. an audio programme in parallel with a data service, is practicable and provides interesting possibilities for new receiver features. The system controller is connected to the user interface and processes the user commands, in accordance with the information contained in the FIC.

4.3.2 Audio Services

The audio source coding method is a perceptually based, low bit-rate sub-band coding system for high quality audio signals, standardized by ISO/IEC under the heading ISO/IEC 11172-3 [6] (MPEG-I Audio), and ISO/IEC 13818-3 [12] (MPEG-2 Audio) Layer II [12]. The DAB specification permits use of the flexibility of Layer II except for the fact that for MPEG-1 Audio only the standard studio sampling frequency of 48 kHz , and for MPEG-2 Audio only 24 kHz is used. In the case of half sampling rate coding, a down-sampling filter is used in the encoder to maintain always the standard studio sampling frequency of 48 kHz for the PCM audio input signal. Layer II is capable of processing mono and stereo and different encoded bit-rate options are available (viz: 8, 16, 24, 32, 40, 48, 56, 64, 80, 96, 112, 128, 144, 160 or 192 kbit/s per monophonic channel). Extension to multichannel sound, according to ISO/IEC 13818-3 [12] is also possible [13]. With the exception of 144 kbit/s, in stereophonic mode, the encoder produces twice the bit rate of a mono channel. These options can be exploited by broadcasters depending on the intrinsic quality required and the number of sound programmes to be broadcast. A stereophonic signal can be conveyed in the stereo mode, or, in particular at lower bit-rates, in the joint stereo mode. This exploits the similarity of the two channels of a stereophonic programme to maximize the overall perceived audio quality. The ISO/IEC 13818-3 [12] (MPEG-2 Audio) backwards compatible extension to multichannel audio coding is also possible. This means that the encoded multichannel signal will comprise a conventional stereo signal, decodable by a stereo DAB receiver and additional information which can be used by an extended DAB receiver for the reproduction of the multichannel sound.

Each audio service channel also contains a PAD (programme associated data) channel, having a variable capacity (minimum 0,333 kbit/s for MPEG-2 Audio half sampling frequency coding, or 0,667 kbit/s for full sampling frequency coding), which can be used to convey information which is intimately linked to the sound programme. The PAD is incorporated at the end of a DAB audio frame complying with the ISO standard, and, therefore, cannot be subject to a different transmission delay. Typical examples are dynamic range control information, a dynamic label to convey programme titles or lyrics, and speech/music indication. Additionally, text with graphics features, for example, can be conveyed in the PAD.

4.3.3 Data Services

In addition to the programme associated data which can be carried with the audio, general data may be conveyed as a separate service. This can be in the form of a continuous stream, segmented in 24 ms "logical frames", or arranged as packet data services. The resource allocated to a data service is arranged in multiples of 8 kbit/sec data rate, though individual packet data services may have much lower capacities and be bundled in a packet sub-multiplex. In general, the capacity available for independent data will necessarily be limited by the capacity requirements of the audio programme services making up the DAB multiplex. A Traffic Message Channel (TMC) is an example of a data service which may be carried in the FIC as well as using the packet mode.

4.3.4 Service Information

The following elements of service information (SI) can be made available for display on a receiver:

- basic programme service label (i.e. the name of a programme service)
- time and date
- dynamic programme label (e.g. the programme title, lyrics, names of artists)
- programme language

- programme type label (e.g. news, sport, classical music, etc.)

The following elements of Service Information (SI) can be used for control of a receiver:

- cross-reference to the same service being transmitted in another DAB signal or being simulcast by an AM or FM service

- transmitter identification information (e.g. for geographical selection of information)

Transmitter network data can also be included, for example, for monitoring and control by the broadcasters.

4.3.5 System Organization and Service Control

In order that a receiver can gain access to any or all of the individual services with a minimum overall delay, precise information about the current and future content of the Main Service Multiplex (MUX) is set up and carried by the Fast Information Channel (FIC). This information is the Multiplex Configuration Information (MCI), which is machine-readable data. Data in the FIC are not time-interleaved, so the MCI does not suffer the delay inherent in the time-interleaving process applied to audio and general data services. However, these data are highly protected and repeated frequently to ensure their ruggedness. When the multiplex configuration is about to change, the new information, together with the timing of the change, is sent in advance within the MCI. Essential items of SI, which concern the content of the MSC (i.e. for programme selection), must also be carried in the FIC. More extensive text which is not required immediately on switching on a receiver, such as a list of all the day's programmes, can be carried separately as a general data service. The user of a receiver can select programmes on the basis of the textual information carried in the SI, using the programme service label, the programme type label or the language. The selection is then implemented in the receiver using the corresponding elements of the MCI. Provision is also made for the use of conditional access to services if desired. If alternative sources of a chosen programme service are available and an original digital service becomes untenable, then linking data carried in the SI (i.e. the "cross reference") can be used to identify an alternative (e.g. an FM service) and switch to it. However, in such a case, the DAB/FM receiver will switch back to the DAB service as soon as reception is possible. This is a particularly important feature at the start of DAB services, since not all areas will be served from day one, and the ability to drop back to the same programme on FM, where a simulcast is available, will help maintain service continuity.

4.3.6 Channel Coding and Time Interleaving

The data representing each of the programme services being broadcast (digital audio with some ancillary data, and maybe also general data) are subjected to energy dispersal scrambling, convolutional coding and time interleaving. The convolutional encoding process involves adding redundancy to the service data using a code with a constraint length of 7. In the case of an audio signal, greater protection is given to some source-encoded bits than others, following a pre-selected pattern known as the Unequal Error Protection (UEP) profile. For the bit rates of 8, 16, 24, 40 and 144 kbit/s which can be used by MPEG-2 Audio half-sampling frequency coding, only EEP (Equal Error Protection) can be applied. The average code rate, defined as the ratio between the number of source-encoded bits and the number of encoded bits after convolutional encoding, can take a value from 0,35 (the highest protection level) to 0,75 (the lowest protection level). Different average code rates can be applied to different audio sources, subject to the protection level required and the bit-rate of the source-encoded data. For example, the protection level of audio services carried by cable networks can be lower than that of services transmitted in radio-frequency channels. General data services are convolutionally encoded using one of a selection of uniform rates whilst data in the FIC are encoded at a constant 1/3 rate. Time interleaving improves the ruggedness of data transmission in a changing environment (e.g. reception by a moving receiver) and imposes a 384 ms transmission delay.

4.3.7 Main Service Multiplex

The encoded and interleaved data are fed to the Main Service Multiplexer (MUX) where, each 24 ms, the data are gathered in sequence into the multiplex frame. The combined bit-stream output from the multiplexer is known as the Main Service Channel (MSC) which has a gross capacity of 2,3 Mbit/s. Depending on the chosen convolutional code rate (which can be different from one application to another), this gives a net bit rate ranging from approximately 0,6 to 1,7 Mbit/s, accommodated in a 1,5 MHz bandwidth DAB signal. The Main Service Multiplexer is the point at which synchronized data from all of the programme services using the multiplex are brought together.

4.3.8 Transmission Frame and Modes

The system provides four transmission mode options which allow the use of a wide range of transmitting frequencies, up to 3 GHz for mobile reception. These transmission modes have been designed to cope with Doppler spread and delay spread, for mobile reception in the presence of multi-path echoes. Table 4.3.1 gives the temporal guard interval duration and nominal maximum transmitter separation and frequency range for mobile reception. The reduction in performance at the highest frequency and in the most critical multi-path condition, occurring infrequently in practice, is equivalent to a loss of approximately 1 dB of carrier power at 100 km/h and 4 dB at 200 km/h.

Table 4.3.1: Limiting planning parameter values for each transmission mode

	Transmission Mode			
System Parameter	**I**	**II**	**III**	**IV**
Guard interval duration	~246 µs	~62 µs	~31 µs	~123 µs
Nominal maximum transmitter separation for SFN	96 Km	24 Km	12 Km	48 Km
Nominal frequency range (for mobile reception)	≤ 375 MHz	≤ 1,5 GHz	≤ 3 GHz	≤ 1,5 GHz

From this table, it can be seen that the use of higher frequencies imposes a greater limitation on the guard interval duration and hence on the maximum non-destructive echo delay. Mode I is most suitable for a terrestrial single-frequency network (SFN) in the VHF range because it allows the greatest transmitter separations. Mode II will preferably be used for medium-scale SFN in L-Band and for local radio applications that require one terrestrial transmitter. Larger transmitter spacings can be accommodated by inserting artificial delays at the transmitters and by using directive transmitting antennas. Mode III is appropriate for cable, satellite and complementary terrestrial transmission at all frequencies since it can be operated at all frequencies up to 3 GHz for mobile reception, and has the greatest tolerance of phase-noise. Mode IV is also used in L-band and allows a greater transmitter spacing in SFNs. However, it is less resistant to degradation at higher vehicle speeds. In order to facilitate receiver synchronization, the transmitted signal is built up with a frame structure having a fixed sequence of symbols. Each transmission frame, see figure 4.3.3, begins with a null symbol for coarse synchronization (when no carrier is transmitted), followed by a phase reference symbol for differential demodulation. These comprise the synchronization information. The next symbols are reserved for the FIC, and the remaining symbols provide the MSC. The total frame duration T_F is either 96 ms, 48 ms or 24 ms, depending on the transmission mode as given in table 4.3.2 below. Each audio service within the MSC is allotted a fixed time slot in the frame.

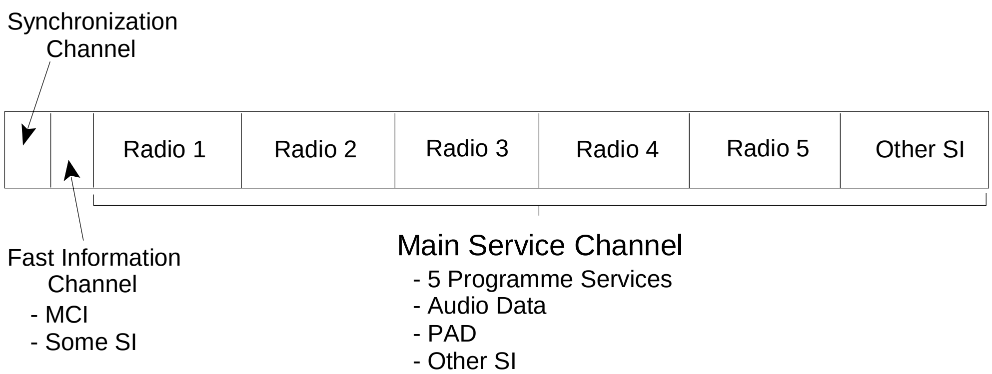

Figure 4.3.3: An example of a DAB multiplex frame

4.3.9 Modulation with OFDM

The DAB system uses differential QPSK modulation coupled with a multi-carrier scheme: known as Orthogonal Frequency Division Multiplexing (OFDM). This scheme meets the exacting requirements of high bit-rate digital broadcasting to mobile, portable and fixed receivers, especially in multi-path environments. The basic principle consists of dividing the information to be transmitted into a large number of bit-streams, having low bit-rates individually, which are then used to modulate individual orthogonal carriers, such that the corresponding symbol duration becomes larger than the delay spread of the transmission channels. By inserting a temporal guard interval between successive symbols, channel selectivity and multi-path propagation will not cause inter-symbol interference. The large number, N, of orthogonal carriers (see table 4.3.2), which can be conveniently generated by a DFT process, is known collectively, as an "ensemble". The spectrum of the signal is approximately rectangular, Gaussian noise-like, and occupies a bandwidth of approximately 1,54 MHz.

Table 4.3.2: DAB Transmission parameters for each Transmission Mode

	Transmission mode			
System Parameter	**I**	**II**	**III**	**IV**
Frame duration T_F	96 ms	24 ms	24 ms	48 ms
Null symbol duration T_{null}	~1297 µs	~324 µs	~168 µs	~648 µs
Guard interval duration t_Δ	~246 µs	~62 µs	~31 µs	~123 µs
Useful symbol duration t_s	1 ms	250 µs	125 µs	500 µs
Total symbol duration T_s	~1246 µs	~312 µs	~156 µs	~623 µs
No. radiated carriers N	1536	384	192	768

Figure 4.3.4 shows an example of the transmitter output spectrum after amplification and filtering. In practice, the peak-to-mean ratio is limited to about 8 dB by digital processing; though this can be further reduced by additional signal conditioning when coupled with non-linear amplification in the transmitter. In the presence of multi-path propagation, some of the carriers are enhanced by constructive signals, while others suffer destructive interference (frequency selective fading). Therefore, the System provides frequency interleaving by a re-arrangement of the digital bit stream amongst the carriers, such that successive source samples are not affected by a selective fade. When the receiver is stationary, the diversity in the frequency domain is the prime means to ensure successful reception; the time diversity provided by time interleaving provides further assistance to a mobile receiver. Consequently, multi-path propagation is a form of diversity and is not considered to be a significant disadvantage for DAB, in stark contrast to conventional FM or narrow-band digital systems where multi-path propagation can completely destroy a service.

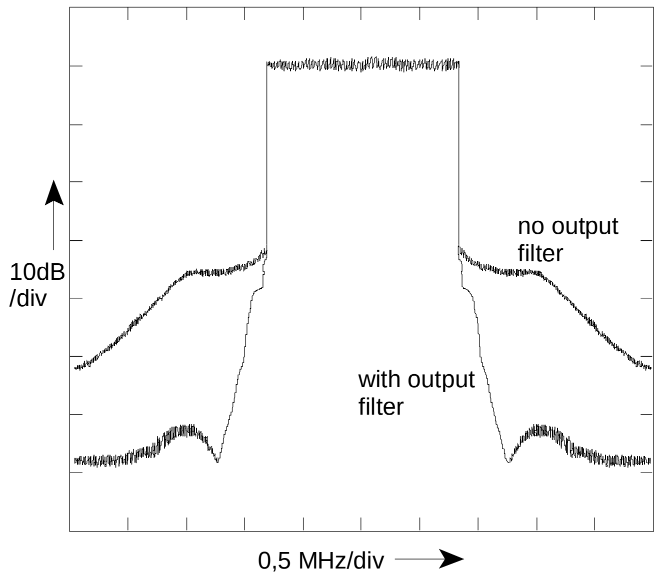

Figure 4.3.4: Example of DAB transmitted signal spectrum (VHF band III)

5 Description of system features

5.1 Introduction

The DAB system features are defined in [1], clauses 5 to 9. In this clause, the features are taken approximately in the same order; data transport mechanisms, the Multiplex Configuration Information (MCI), audio coding, Programme Associated Data (PAD), Service Information, the Fast Information Data Channel (FIDC) and Conditional Access. The descriptions given in this clause are very brief but are intended to give more detailed information than the overview presented in clause 4. A full description will be found in TR 101 496-2 [15] together with detailed implementation and operational guidelines. The DAB system has a very flexible set of system features. In addition, links are provided to permit the easy addition of future features. The procedures for using these links are described here. A summary of the system features is also presented (see 5.10). This includes guidance on the recommended data repetition rates, the preferred transport mechanism and the typical net data capacity. An indication is also given of the inter-dependence of the features. Finally, the capacity of the DAB multiplex is examined in more detail. Some examples of typical ensemble multiplex configurations are given.

5.2 Transport mechanisms

Several different data transport mechanisms are provided within the transmission frame to suit different needs, see 5.1 in [1]. The majority of data is carried in the Main Service Channel (MSC) which enjoys the best protection. This is achieved through the use of time and frequency interleaving as well as channel coding using a powerful convolutional code, see 5.3 in [1]. There are two possible transport mechanisms within the MSC. These are referred to as stream mode and packet mode. In stream mode, data is divided at the source into regular 24 ms bursts. Within the constraint of these 24 ms data bursts, the stream mode can also be used for general data service components. The MSC is divided into sub-channels. Up to 64 sub-channels are available, each of which is treated individually as far as channel coding is concerned. In packet mode, the total data capacity of a sub-channel can be shared by several (up to 1 023) service components, organized in addressable packets. This can increase transport efficiency when several service components have data rates below the minimum sub-channel data rate of 8 kbit/s. The Fast Information Channel (FIC) allows a limited amount of information to be accessed quickly by the receiver - particularly the Multiplex Configuration Information, see 5.2 in [1]. The considerably reduced processing delay is possible because time interleaving is not applied to the FIC. The loss of the ruggedness offered by time interleaving is compensated for by adopting a strategy of repeating the data carried in the FIC at regular intervals. To avoid the need for additional signalling overhead, the FIC uses a fixed, equal (see note) channel coding, with a coding rate of approximately 1/3. Information carried in the FIC is sub-divided and encoded into Fast Information Groups (FIGs).

NOTE: "Fixed" means that it is not alterable; "Equal", since the channel coding is applied uniformly to the FIC data.

One particular sub-channel operating in the packet mode (with sub-channel address '63' and packet address '1023') is reserved for the Auxiliary Information Channel (AIC). The AIC is used to re-direct information, encoded in FIGs, to the MSC, see [1], clause 5.4. This overflow mechanism can be used for certain non-critical information, such as some service information features (see 5.6, 5.7 and [1], clause 8.1.12), which would otherwise reduce the capacity available within the FIC. For the basic audio application, additional non-audio information can be incorporated within the encoded audio frames. This additional information, referred to as Programme Associated Data (PAD), is carried at the end of each DAB frame. It is intended to carry information which needs to be synchronized to the audio programme. One example is Dynamic Range Control (DRC) data (see clause 5.5). PAD can be transported at a basic level (offering a capacity of 667 bit/s) or at an extended level (where a maximum capacity of about 65 kbit/s is available for PAD). In the first case, the PAD is known as "Fixed" (F-PAD) and the capacity is always available without prejudice to the audio data. The use of the extended form (X-PAD) must be exercised carefully since there may be a conflict with the capacity required by the audio application. Service Providers, supplying data to the Ensemble Provider (see clause 4 in TR 101 496-3 [16]) can state a preference for a particular transport mechanism. This statement can be a long-term static requirement of the service provider or the request can be made in real-time (or close to real-time), along with the data supplied to the Ensemble Provider. As outlined in clause 4 in TR 101 496-3 [16], the Ensemble Provider is responsible for managing the transport resources and allocating data to the appropriate route.

5.3 Multiplex Configuration Information

The MSC and FIC carry the components of Services which make up a DAB ensemble multiplex. Each service has one or more Service components. For example, imagine an "Alpha 1 Radio" Service which could contain a stereo audio component, service labels (in the "Alpha 1 Radio" SI) and a Traffic Message Channel (TMC). Several services may be carried in one ensemble multiplex. An example of how a service structure can be organized in an ensemble multiplex is shown in figure 12 in [1]. A listener, or some other user of a DAB receiver, accesses service components by first selecting a service. A distinction is made between the essential Service component of a service which is called the "Primary" component (for an audio service this would normally be the main, or perhaps only, audio stream) and other components of a service, which are considered "Secondary" components. The selection of a single service requires the receiver to decode only a part of the MSC. To benefit from this situation, the MSC is divided into sub-channels, each of which is convolutionally encoded. Each sub-channel can be treated independently, making both the assembly of the multiplex at the encoder, and the decoding operation in the receiver, highly efficient. In stream mode, a sub-channel generally carries a single service component. In packet mode several service components can occupy a sub-channel.

The DAB system allows the ensemble multiplex to be re-configured from time to time. It does not necessarily have a fixed format. Some service providers can require the multiplex to be re-configured frequently (maybe several times a day); others can use this facility rarely, if ever. The Multiplex Configuration Information (MCI) is responsible for defining the organization of the sub-channels, Services and service components in an ensemble multiplex and for managing the effects of a multiplex reconfiguration on the receiver The MCI serves five principal functions:

a) to define the organization of the sub-channels in terms of their position and size in the CIF and their error protection;

b) to list the services available in the ensemble (including service component descriptions);

c) to establish the links between service and service components;

d) to establish the links between service components and sub-channels;

e) to signal an ensemble multiplex reconfiguration.

The MCI is carried in the FIC to avoid the inherent transmission delay associated with the time-interleaving process applied to the MSC.

5.4 Audio coding

The DAB system uses the audio coding algorithm defined in Layer II of the ISO/IEC 11172-3 [6] (MPEG-1 Audio) and ISO/IEC 13818-3 [12] (MPEG-2 Audio) standard [6], [12], with extensions to provide better error ruggedness and flexible transmission of Programme Associated Data. This algorithm has been recommended by ITU-R, after extensive testing [7], for the broadcasting applications of contribution, distribution, and emission. From one mono or stereo input PCM audio signal, sampled at 24 kHz or 48 kHz, a bit rate reduced audio bitstream is produced. Encoded bitrates range from 8 kbit/s to 192 kbit/s for a monophonic programme (or 8 kbit/s to 384 kbit/s for a stereophonic programme) to allow for different balances of audio quality and transmission channel bandwidth [1], table 19. Channel coding is applied to the audio frame, to provide protection against transmission errors. Two types of error protection are possible. Unequal Error Protection (UEP) is preferable to Equal Error Protection (EEP), as it affords more protection to the most critical data.). A number of different profiles, five for UEP and four for EEP are defined, allowing a choice to be made between average coding rate (and hence ruggedness) and utilized channel capacity. For the bitrates 8, 16, 24, 40 and 144 only EEP as defined in table 39 [1] "Set A of Equal Error Protection Profiles" can be used. At all other bit rates the use of UEP is recommended. A detailed description of the UEP is given in clause 5.3.3 of TR 101 496-3 [16].

For audio sampled at half sampling frequency (24 kHz) the encoded bitrates range from 8 to 160 kbit/s but the bitrates 8, 16, 25, 40 and 144 kbit/s use the EEP table. Half sampling frequency coding improves significantly the audio quality at low bit rates (< 64 kbit/s per channel). The lower sampling frequency implies that the frequency range is limited to about 11,5 kHz. Therefore LSF coding is attractive for signals that are inherently band limited, such as speech.

Following audio modes are provided:

- Single channel mode,
- Stereo mode (left and right signals of a stereo pair coded as one bitstream),
- Joint stereo mode (as stereo mode, but with exploitation of the redundancy of stereo material used to improve overall quality).

Reference [6] does not prescribe the encoder. Rather, the format of the coded bit stream and the actions to be taken by the decoder are defined. This has the great advantage that future improvements are possible in the encoder (e.g. by applying improved psycho acoustic models or bit allocation techniques) without the need to change existing decoders. This makes the system future-proof. Annex C of [1] gives an example of a suitable encoder. The inclusion of the ISO/IEC CRC check in the generated audio frame is mandatory. This provides a mechanism to avoid annoying artefacts in the receiver due to transmission channel errors. In addition to the ISO/IEC CRC check, that protects the most error sensitive part of the bit stream, CRC check words protecting the scale factors are defined within the DAB System. These checks allow for effective concealment of errors in the scale factors. These scale factor CRCs are included in the ancillary data field of the ISO/IEC bit stream in a way that is fully compatible with the ISO/IEC standard. Information is included in the bit stream to indicate the copyright and original/copy status of the transmitted material. Channel coding is applied to the audio frame. The protection against transmission errors is applied non-uniformly through the audio frame. Non-uniform coding provides better protection of the more critical data in each frame. This technique is known as Unequal Error Protection (UEP). A number of different profiles (five in all) are defined, allowing a choice to be made between average coding rate (and hence ruggedness) and utilized channel capacity. An Equal Error Profile (EEP) table can also be used and shall be used with certain lower bitrates when using half-sampling frequency.

5.5 Programme Associated Data

Each audio frame contains Programme Associated Data (PAD). In general, this has a strong relation with the audio, both in terms of its content and time-relationship. The PAD is located in the ancillary data field of the ISO/IEC bit stream, in a way that is fully compatible with the ISO/IEC standard. The PAD has two parts, a fixed F-PAD and an optional, extended X-PAD, see [1], clause 7.4 and figure 25). The maximum data rate of the F-PAD is 0,667 kbit/s at 48 kHz audio sampling rate and 0,333 kbit/s at 24 kHz audio half-sampling rate, and the data rate of the X-PAD is in the range 0 to about 65 kbit/s. The F-PAD and some parts of the X-PAD are more strongly protected than most of the other parts of the audio bit stream by the application of UEP.

Some of the F-PAD functions available (see also [1], annex A, clause A.4) include:

- Dynamic Range Control

  With the help of the Dynamic Range Control (DRC) the receiver can compress the dynamic range of the received audio signal in order to improve signal audibility in a noisy environment.

- Music/Speech indication

  The Music/Speech feature indicates whether the transmitted sound consists of music or speech. It also includes provision for sending "no indication". The receiver can use this information to apply different processing to music and speech.

- Commands to a receiver or decoder

  These commands can be used to initiate processes, which need to be synchronized to the audio.

  One example is the provision of a trigger to read out a picture from a buffer memory, which has been written to earlier. The command channel is able to carry commands of a few bytes in length can provide an occasional trigger with a time resolution of 0,2 s to 0,5 s.

- ISRC and UPC/EAN

  The International Standard Recording Code and the Universal Product Code/European Article Number are provided with some pre-recorded software. These could be transmitted by the DAB System.

- Programme-related text

  Programme-related text includes, for example, song titles and programme descriptions. This text can be produced by the programme provider or derived directly from pre-recorded software, or derived by a combination of the two. The data capacity requirements depend on how comprehensive the service provider wishes to make the feature.

- In-house information

  Channels can be provided for both short, synchronous commands or for long strings of asynchronous data. The meaning of these commands and data is not subject to standardization. It is intended for internal use within the broadcast chain and requires special receivers. There are two key advantages of the PAD. Firstly, it is fully synchronized to the audio throughout the transmission chain. Secondly, it remains the preserve of the service provider and the trade-off of PAD capacity and audio quality can be judged by the service provider independently of other multiplex considerations. A disadvantage of the PAD is that it cannot be identified as a separate entity: It is considered to be a part of the audio service component and no part of it may be signalled separately.

5.6 Service Information

Service Information (SI) provides supplementary information about services, both audio programme and data, as well as other information about Ensembles and other miscellaneous features. Some features have general application, such as textual labels for the visual identification of services, Ensemble information, and time and date information. Most of the features are concerned with audio programme services, either for the direct or indirect benefit of the listener and these are listed here:

- The alarm feature signals whether a service carries alarm announcements when available.

- The programme language can be designated and used either for direct display in the receiver or for service selection by the listener.

- The programme number feature allows programme delivery control for off-line recording, for instance.

- A classification of programme types permits another means of service selection. Some programme type codes can be defined over-air using a downloading mechanism. The programme type preview feature allows forthcoming types to be signalled.

- The announcement feature allows the listener to interrupt his chosen programme by verbal announcements carried on some other service. Different kinds of announcements can be selected so that, for example, only travel or weather information is received.

- The service component trigger allows a receiver to respond to a start trigger indicating when the service content is broadcast. This is intended for special, low power-consuming receivers which are dormant for most of the time.

- Frequency information can be provided to signal the centre radio frequency of other DAB ensembles or the frequency of AM/FM services. These can contain alternative sources of the services available in a selected ensemble or further services about which the listener can require information.

- Transmitter Identification Information (TII) allows the geographical location of transmitters to be signalled.

- Services in other Ensembles or FM services can be cross-referenced by signalling their radio frequencies and service identifiers. A limited amount of SI, associated with these alternative services, may be signalled: Programme Number, Programme Type, Announcements and labels for the ensemble and individual services.

- Services carrying the same programme or belonging to the same generic family can be linked together.

- Geographical regions can be identified by a list of transmitter identification codes or a rectangular grid, and assigned a label, to allow the filtering of alternative service sources or announcements, for example.

- Local service areas, within the area covered by a single frequency network, can be defined using the geographical region feature.

- Information may be provided to assist with handover to satellite-based DAB services.

Certain features may be transported in the Auxiliary Information Channel (within the MSC) using a re-direction signalling mechanism (see clause 5.2).

5.7 Fast Information Data Channel

The Fast Information Data Channel (FIDC) is the part of the FIC intended to carry service components with low data rate which are intended to reach dedicated receivers or simple portable and mobile receivers. This applies especially to those receivers for which power consumption is a critical issue. The features defined up to now to be carried in the FIDC are Paging, Traffic Message Channel and Emergency Warning Systems. The Paging feature carries messages to an end-user or a group of end-users equipped with special receivers. Traffic messages may be carried as a "Traffic Message Channel" (TMC) coded according to the "Alert C" protocol [8]. Emergency warnings may be carried to dedicated receivers by "Emergency Warning Systems" (EWS). The details of the coding of EWS remain confidential within the group of users of these systems.

5.8 Conditional Access

The purpose of this feature is to provide a complete access control system, including the following three main functions: the scrambling/descrambling function, the entitlement checking function and the entitlement management function.

The scrambling/descrambling function aims at making the programme incomprehensible to unauthorized users. Scrambling can be applied separately to the different components of a service (for example sound and data) to make these components unintelligible. The scrambling algorithm used for DAB is the algorithm described in 12.2 in [9]. In DAB, it is possible to perform scrambling on data sent in the FIDC or in the MSC for audio and data in stream mode, or in packet mode. Furthermore, for data sent in FIDC or data in packet mode, scrambling can be performed by the Programme Provider, the Service provider or the Ensemble Provider. The entitlement checking function consists of broadcasting the conditions required to access a service, together with the encrypted parameters enabling the descrambling by authorized receivers. These data are sent inside dedicated messages called Entitlement Checking Messages (ECMs). Access conditions may be applied to all components of a service, or to each component separately. The DAB specification provides several transport mechanisms for the ECMs. It is possible to send the ECMs in the FIC (for receivers which would only extract one sub-channel plus the FIC), in the MSC either in sub-channel 63 or in the same sub-channel as the scrambled service component. This last option is possible for data in Packet mode only. The entitlement management function consists of distributing the entitlements to the receivers. There are several kinds of entitlements matching the different means to subscribe to a service. These data are sent inside dedicated messages called Entitlement Management Messages (EMMs). Addressing mechanisms have been implemented, so that the EMMs can be sent to all receivers, to a group of receivers or to a single receiver. The DAB specification [1] provides several transport mechanisms for the EMMs. It is possible to send the EMMs in the FIC (for receivers which would only extract one sub-channel plus the FIC), in the MSC either in sub-channel 63 or in the same sub-channel as the scrambled service component. This last option is possible for data in packet mode only. The mechanisms that have been defined in the specification can be used by most of the access control systems presently on the market. Two of them have already been identified: EUROCRYPT and NR-MSK see [10] and [11] respectively.

5.9 Future Features

To allow the easy accommodation of future features, a number of links are provided within the DAB system. New features can be incorporated into the DAB specification [1] through the appropriate procedures.

5.9.1 Audio

Multi-channel sound, as standardized in ISO/IEC 13818-3 [12], (up to 5 channels with an optional low frequency enhancement channel, plus enhanced multilingual capabilities) greatly enhances the spatial impression of an audio programme. The ISO/IEC 13818-3 [12] standard allows for multichannel audio coding in a backwards compatible way. This means that a stereo DAB receiver decodes properly the basic stereo information from the multichannel bit stream.

5.9.2 Data

If a new feature is to be implemented, the appropriate transport mechanism has to be identified. A new feature could be a Service component (Audio or general data), Programme Associated Data (PAD) or Service Information (SI). The choice of a suitable mechanism is essentially determined by whether the feature is a service component:

a) When the feature is not a Service Component:

- If the feature is closely related to an audio service component and real-time requirements have to be met, the data shall be carried in the PAD. Application type(s) have to be defined (one for the start and possibly one for the continuation of the feature). Note that the PAD field of audio frame (n-1) belongs to the audio samples carried in frame n. If the feature is time critical and decoders are known to need a certain time (e.g. mx24ms) for processing, it would be possible to incorporate the data corresponding to frame n in the PAD field of frame n-m-1. The service provider shall manage the termination of such a service.

- If the feature is related to the ensemble (i.e. common to all services, perhaps with just a few exceptions) or to services in general it shall be carried in the FIC. If there is no strict real time requirement, the feature may be carried in the AIC. A suitable FIG shall be chosen (Type 0 for ensemble and service information, Type 1 for labels, any other type except Type 5 for completely new features).

- For future extensions to service information see [1], clause 6.3.4.

b) When the feature is a service component:

- If the feature can deliver a continuous bit stream with constant data rate it can be carried in stream mode or packet mode.

- If the feature cannot deliver a continuous bit stream with constant data rate, it shall be carried in packet mode.

- If the feature is intended for low power-consuming receivers or even simple receivers without a packet mode decoder, it shall be carried in the FIDC provided that capacity is available. If the FIDC capacity is not sufficient, only a basic part of the feature shall be carried in the FIDC. The full feature shall be carried in packet mode.

In all three cases, the service component shall be characterized by a service component type (see [1], clause 6.3.1).

5.10 System features summary

Table 5.10.1 lists each of the system features and gives details about the applicable transport mechanism, recommended data repetition rate (see key information below the table), typical net data capacity and any dependency on other features. Those features marked by an asterisk are mandatory. Where a FIG type is indicated for the transport mechanism, these features shall be carried in the FIC unless an alternative is given (shown in brackets). The net capacity of the FIC is 32 kbit/s (Modes I, II and IV) or 42,67 kbit/s (Mode III). In the case of system Features carried in the FIC, the typical net capacity figures are also given in terms of the percentage of the FIC (for Modes I and II) required by each feature. Considering a Mode I implementation, with the assumption of note (i) to table 2.10.1, the MCI occupies approximately 30 % of the FIC's total capacity. Adding the ensemble and service labels increases this to about 35 %.

Table 5.10.1: System feature summary

No.	Name of feature	Transport	Rec Rep rate	Typical net capacity {bit/sec} % FIC (see note viii)	Depends on other features	Description
1	Ensemble Identifier	FIG 0/0	A	γ	-	World-wide unique machine-readable code (16 bit)
2	Alarm flag	FIG 0/0	A	Included in 1	13	Response to alarm announcements
3	Multiplex Configuration - Sub-channel organization - Service organization - Data SC type - Global SC	 FIG 0/1 FIG 0/2 FIG 0/3 FIG 0/7 FIG 0/8	 A A B B	(see note i) β α	 - 5 9	 Sub-channel position, size and protection List of services in the ensemble Identifies the type of data Service component Cross-reference of global & ensemble SCs
4	Foreground/ Background sound	FIG 0/3	A	Included in 3	3	Permits a receiver to select a preferred audio component
5	Service Identifier	FIG 0/3	A	Included in 3	3	Machine-readable service code (16- or 32-bit), including 4-bit country Id.
6	Multiplex re-configuration timing management	FIG 0/0	A	extra over 6 secs.	C/N flag	Frame count at re-configuration
7	Extended Country Code (ECC), - ensemble - per service	FIG 0/10	 B C	for basic 7 & 8 δ	 - 5	An 8-bit code to accompany the 4-bit Country Id to create a unique 12-bit country identifier
8	Date and Time, - minute resolution (basic) - milli-second resolution Local Time Offset (LTO), - ensemble - per service	FIG 0/10 FIG 0/9	 B C B C	17 extra for ECC+LTO (see notes i and ii) ε 1,6 extra for milli-sec. resolution ε	 - - - 5	Date in MJD format Time in UTC format LTO in half-hours resolution
9	Labels Ensemble label Programme Service label Data Service label Service Component label	(AIC) FIG 1/0 FIG 1/1 FIG 1/5 FIG 1/4	 B B	 δ γ (see note i)	 - 5 5	16/8 character labels for display in receiver
10	Language	FIG 0/5	A/B	γ	3	Identifies language associated with an audio or data service component
11	Programme Number	FIG 0/16 (AIC)	A/B	δ/service	3, 5	Identifies a programme on a primary audio service component

No.	Name of feature	Transport	Rec Rep rate	Typical net capacity {bit/sec} % FIC (see note viii)	Depends on other features	Description
12	Programme Type - static operation + basic codes - dynamic operation - basic codes + coarse code extension + fine codes - International table identifier - Down-loading - Preview - basic codes + coarse code extension + fine codes	FIG 0/17 FIG 0/17 FIG 0/17 FIG 0/17 FIG 0/9 FIG 1/2 (AIC) FIG 0/12 (AIC)	A/B A/B A/B A/B B C B/C	(see note i) γ γ γ extra δ extra/fine code Included in 7 δ/type δ/type ε extra δ extra/fine code	5	Classification for service selection Signals service flavour Signals programme type currently offered Specifies which international table applies, e.g. EBU table Mechanism for defining new programme types Mechanism for signalling which PTy codes are coming soon
13	Announcements - static -dynamic - without reg id. - with region id.	FIG 0/18 (AIC) FIG 0/19	B A/B	(See notes i and iii) δ γ before message δ extra	3, (5) 33	Identifies whether a service supports any of 16 types of announcements Signals that an announcement is in progress
14	Alternative service sources	FIG 0/21 (AIC)	E	(See note iv) δ	17a, 17b, 33	Alternative frequencies of other DAB ensembles with same service
15	TII coding	Null-symbol				Individual transmitter identification
16	TII database - Main identifier - Sub identifier	FIG 0/22 (AIC)	E	0,5/Tx.group 0,7/Tx.group	15	Geographical area information based on individual transmitter identification codes

No.	Name of feature	Transport	Rec Rep rate	Typical net capacity {bit/sec} % FIC (see note viii)	Depends on other features	Description
17	Other ensembles	(AIC)			1 (OE- flag)	
	- Frequencies					
(a)		FIG 0/21	D	ε/ensemble	17b	Frequencies of other DAB ensembles
	- Services					
(b)		FIG 0/21	D	ε/ensemble (see note i)	17a	List of services in other DAB ensembles
	- Ensemble label					
(c)					17a	
	- Service label	FIG 1/0	D	ε/ensemble		Extends 9 to cover other ensembles
(d)					5, 17a, 17b	
		FIG 1/1	D	ε/service		Extends 9 to cover other ensembles
	- Programme Number				5, 17a, 17b	
(e)		FIG 0/16	B	see 11	5, 17a,	Extends 11 to cover other ensembles
	- Programme Type					
(f)	- PTy preview	FIG 0/17	B	see 12	17b	Extends 12 to cover other ensembles
	- Announcements			δ/ensemble		
(g)	- static	FIG 0/12	C		13,17a	PTy preview of other ensembles
	- dynamic			δ/ensemble		Announcements from other ensembles
		FIG 0/25	B			
				γ before message		
		FIG 0/26 FIG 0/19	A/B			

No.	Name of feature	Transport	Rec Rep rate	Typical net capacity {bit/sec} % FIC (see note viii)	Depends on other features	Description
18	FM and AM Services - Frequencies - Traffic Announcements - static - dynamic - Programme Number - Programme Type - Programme Service labels	(AIC) FIG 0/21 FIG 0/27 FIG 0/28 FIG 0/16 FIG 0/17 FIG 1/1	 D B A/B A/B A/B B	 ε/FM channel δ δ before message		 Frequencies of other FM or AM channels Announcements from FM channels (not applicable to AM) Identifies programmes on FM or AM Classifies programmes on FM or AM Identifies FM or AM services
19	Service Component Trigger - short form - long form + Time, CA, SCCA fields + User Group field	FIG 0/20	 1 per min		3	Informs a group of receivers about the beginning or end of the transmission of a (data) service
20	FIC Overflow indicator	FIG 0/31	B	δ		Allows SI data to be carried in the AIC
21	Conditional Access - Management - EMMs - ECMs - Service overhead	(AIC) FIG 6 FIG 0/3 FIG 0/4	 var A/B A	 Depends on CA system β (see note i)	3	Permits restricted access to services to listeners holding legitimate keys
22	Music/Speech flag	F-PAD	A/B			Distinguishes between music and speech content of an audio programme
23	Dynamic Range Control	F-PAD	24ms			Controls audio compression in a DAB receiver
24	ISRC & UPC/EAN	F-PAD	A/B			Recording code and European article number
25	Dynamic Label segment	X-PAD	B/C			16-character labels for display in receiver (8 form a label)
26	ITTS packets	X-PAD or MSC	var			Texts in 40-character rows for display in receiver
27	In-house information	F-PAD (byte L-1) (byte L) FIG(7)	 A var			Intended for use within a broadcasting organization, using special receivers
28	Closed user group data	X-PAD	Var	(in audio)		Packet & stream channel to carry special services in X-PAD under the control of the service provider
29	Paging - pointer + Times, CA, SCCA fields + User Group field - message	FIG 5/0 or MSC			3 8	Provides radio paging to end-users equipped with special pocket receivers
30	Traffic Message Channel (TMC)	FIG 5/1, or MSC	var			Traffic messages according to the Alert C protocol

No.	Name of feature	Transport	Rec Rep rate	Typical net capacity {bit/sec} % FIC (see note viii)	Depends on other features	Description
31	Emergency Warning Systems (EWS) - Message - Control Info	FIG 5/2, or MSC	var			Provides for the coding of warning messages to be evaluated by special receivers
32	Service linking - short form - long form hard link (note v) soft link	FIG 0/6	 B D	 ε ε		Allows services carrying the same programme or related generically, to be linked together
33	Region identifier - basic with TII list (see note vi) - label	 FIG 0/11 FIG 1/3	 E C	 ε ε/label		Defines a geographical region with a region code and a textual label
34	Local service area	FIG 0/23	B	δ/local service		Area the service is intended to be received in
35	Satellite assistance	FIG 0/30 FIG 0/29	E	n/a		Provides frequency offset & delay data for HEO satellites

NOTES:
i) assume six services, each comprising one audio component and one data service component;
ii) assume all six services have different LTO and ECC;
iii) assume two announcement clusters signalled per service;
iv) assume three alternative sources;
v) assume two services linked together;
vi) assume six transmitters (2@main, 4@sub) per region, and 30 regions.
vii) Key to recommended feature repetition rate codes:

A) 10 times per second;
B) once per second;
C) once every 10 seconds;
D) less frequently than every 10 seconds;
E) all information within 2 minutes.

When a rate A/B, for example, is recommended, this means that a rate between A and B applies.

viii) Key to typical net capacity in terms of % FIC
α) 10 - 30 %, β) 3 - 10 %, γ) 1 - 3 %, δ) 0,1 - 1 %, ε) < 0,1 %

5.11 The Capacity of the DAB Multiplex

5.11.1 Total Capacity of a DAB Ensemble

A DAB ensemble is capable of transmitting 2 432 kbit/s in transmission modes I, II and IV and 2 448 kbit/s in transmission mode III. Some of this total capacity is consumed by the synchronization channel. The remainder is shared by the Main Service Channel (total capacity 2 304 kbit/s in all transmission modes) and the Fast Information Channel (total capacity 96 kbit/s in transmission modes I, II and IV, 128 kbits in transmission mode III). The available net bit rate of the system depends on the Protection level (i.e. channel code rate) used for convolutional encoding. In the FIC this code rate is fixed at 1/3 resulting in a net capacity of 32 kbit/s in Transmission modes I, II and IV and 42,667 kbits in transmission mode III. In the MSC a number of Protection levels are used, corresponding to average code rates between 0,35 to 0,75 (for UEP) or 0,25 and 0,75 (for EEP). Therefore, the available net bit rates in the MSC vary from approximately 800 kbit/s (best protection, level 1) to 1,73 Mbit/s (weakest protection, level 5). At a medium Protection level (level 3 with an average code rate of 0,5) the available net capacity of the MSC is about 1,15 Mbit/s. The quoted figures only apply when the same protection level is used for all sub-channels. This is not necessary, as each sub-channel may use a different level. The following elements of the DAB System, as defined in [1], have the following capacities:

- A Common Interleaved Frame (CIF) is the basic building block of the Main Service Channel. A CIF is transmitted every 24 ms and contains 55 296 bits. A Capacity Unit (CU) is the smallest addressable unit of a CIF and contains 64 bits. A CIF contains 864 CUs.
- A sub-channel of the MSC always has an allocation of an integral number of CUs in each CIF. The number of CUs is called the size of a sub-channel. The data rate of a sub-channel of size *n* is *n*x2,667 kbit/s (this figure includes convolutional encoding).

For sub-channels carrying audio service components, all the possible coding schemes (as defined by the bit rate and Protection level) and the corresponding sub-channel sizes are given in [1], table 7.

The number of CUs allocated to an audio component, when UEP is used, is always one of the set:

{4, 6, 8, 12, 16, 18 , 20, 21, 24, 29, 30, 32, 35, 36, 40, 42, 48, 52, 58, 64, 70, 80, 84, 96, 104, 116, 128, 140, 160, 168, 192, 208, 232, 280, 416}

The corresponding number of CUs, allocated to an audio component, when EEP is used, is:

{15, 18, 27, 30, 36, 40, 42, 45, 48, 52, 58, 60, 63, 64, 70, 72, 76, 80, 81, 84, 90, 96, 104, 105, 108, 116, 126, 128, 140, 147, 168, 189, 192, 208, 232, 280, 416 }

Sub-channels carrying data service components must have a net data rate that is an integral multiple of 8 kbit/or 32 kbit/s. A sub-channel with a net capacity of *m*x8 kbit/s has a size of 12x*m*, 8x*m*, 6x*m* or 4x*m* for Protection levels 1A, 2A, 3A or 4A, respectively. In this case the number of CUs allocated to each data stream is always an integer multiple of 12, 8, 6 or 4 as determined by the protection level. A sub-channel with a net capacity of *n*x32 kbit/s has a size of 27x*n*, 21x*n*, 18x*n* or 15x*mn* for Protection levels 1B, 2B, 3B or 4B, respectively. In this case the number of CUs allocated to each data stream is always an integer multiple of 27, 21, 18 or 15 as determined by the protection level The figures quoted for data service components are independent of the organization of the data sub-channel (stream or packet mode). However, in packet mode, part of the capacity will be required for the organization of the packet structure (headers, error checks etc.). Typically, this could amount to between 5 % to 20 % of the net capacity, depending on the packet length in use.

5.11.2 Examples of Multiplex Configurations

To give some idea of the available numbers of audio sub-channels and the remaining capacity available for additional data services, a number of typical examples have been compiled. In table 5.11.1, for each of the possible audio bit rates, the maximum number of audio sub-channels which can be accommodated in the MSC is given for each of the different protection levels. The remaining gross data capacity, available for data sub-channels is also shown. It has been assumed that all audio service components are encoded using the same protection level. To calculate the net capacity available for data, the information presented in the previous clause can be used. In general, the protection applied to a data sub-channel could be expected to use a code-rate roughly equivalent to the average code-rate employed for the audio sub-channels.

Table 5.11.1: Maximum number of similarly coded audio Sub-channels and residual number of Capacity Units in the MSC

Audio Prot level	**5**		**4**		**3**		**2**		**1**	
Coded Audio bit rate (kbit/s)	**Number of audio Sub-channels**	**Residual capacity units**	**Number of audio Sub-channels**	**Residual capacity units**	**Number of audio Sub-channels**	**Residual Capacity Units**	**Number of audio Sub-channels**	**Residual Capacity Units**	**Number of audio Sub-channels**	**Residual Capacity Units**
8 (note 2)	n/a	n/a	216 215	0 4	144 143	0 6	108 107	0 8	72 71	0 12
16 (note 2)	n/a	n/a	108 107	0 8	72 71	0 12	54 53	0 16	36 35	0 24
24 (note 2)	n/a	n/a	72 71	0 12	48 47	0 18	36 35	0 24	24 23	0 36
32	54 53	0 16	41 40	3 24	36 35	0 24	29	23	24	24
40 (note 2)	n/a	n/a	43 42	4 44	28	24	21	24	14	24
48	36 35	0 24	29	23	24	24	20	24	16	32
56	29	23	24	24	20	24	16	32	n/a	n/a
64	27 26	0 32	20	24	18 17	0 48	14	52	12	24
80	21	24	16	32	14	52	12	24	10	24
96	18 17	0 48	14	52	12	24	10	24	8	32
112	14	52	12	24	10	24	8	32	n/a	n/a
128	13	32	10	24	9 8	0 96	7	52	6	24
144 (note 2)	n/a	n/a	12 11	0 72	8 7	0 108	6 5	0 144	4 3	0 216
160	10	64	8	32	7	52	6	24	5	24
192	9 8	0 96	7	52	6	24	5	24	4	32
224	7	52	6	24	5	24	4	32	3	168
256	6	96	5	24	4	96	3	168	3	24
320	5	64	4	32	n/a	n/a	3	24	n/a	n/a
384	4	96	n/a	n/a	3	24	n/a	n/a	2	32

NOTE 1: In some configurations, padding (amounting to 8 kbit/s or less) can be required to fill the multiplex.

NOTE 2: Audio Sub-channels at this bit rate as well as Data Sub-channels, employ equal error protection. Consequently, these Sub-channels use CUs in integer multiples of 12, 8, 6 or 4 per 8 kbit/s of net capacity for Protection levels 1, 2, 3, and 4 respectively. These audio Sub-channels, as well as Data Sub-channels are always allocated in multiples of 8 kbit/s. For example, 24 CUs used with protection level 3 yields a net rate of 24 kbit/s.

NOTE 3: n/a = not applicable.

Table 5.11.2: Example configurations of MSC using Protection Level 3A (Rate = 0,5)

Minimum Bit rate for Data service	32 kbit/s (840 CUs available)	64 kbit/s (816 CUs available)	128 kbits (768 CUs available)	256 kbit/s (672 CUs available)
Number of Audio Sub-channels	**Possible configuration**	**Possible configuration**	**Possible configuration**	**Possible configuration**
4	4 x 256 S (+96 kbit/s)	4 x 256 S (+64 kbit/s)	4 x 256 S	4 x 224 S
	1 x 384 S 2 x 256 S 1 x 224 S (+8 kbit/s) (+p)	1 x 384 S 1 x 256 S 2 x 224 S (+8 kbit/s) (+p)	1 x 384 S 2 x 224 S 1 x 192 S (+16 kbit/s)	1 x 256 S 2 x 224 S 1 x 192 S (+p)
5	5 x 224 S	4 x 224 S 1 x 192 S (+p)	2 x 224 S 3 x 192 S (+16 kbit/s)	4 x 192 S 1 x 128 (+16 kbit/s) (+p)
	1 x 256 S 3 x 224 S 1 x 192 S (+p)	1 x 256 S 2 x 224 S 2 x 192 S (+p)	1 x 256 S 4 x 192 S (+16 kbit/s)	1 x 224 S 3 x 192 S 1 x 96[3] M (+16 kbit/s) (+p)
	2 x 256 S 1 x 224S 2 x 192 S (+8 kbit/s) (+p)	2 x 256 S 3 x 192 (+16 kbits)	4 x 224 S 1 x 128 S	2 x 224 S 2 x 192 S 1 x 56 M[2] (+16 kbit/s) (+p)
6	6 x 192 S	1 x 224 S 4 x 192 S 1 x 96 M (+24 kbit/s)	4 x 192 S 2 x 128 S (+16 kbit/s) (+p)	4 x 192 S 2 x 64[1] M (+16 kbit/s) (+p)
	1 x 224 S 4 x 192S 1 x 128 (+16 kbit/s) (+p)	5 x 192 S 1 x 128 S (+24 kbit/s) (+p)	4 x 224 S 2 x 64[1] M	1 x 224 S 2 x 192 S 3 x 96 M (+16 kbit/s) (+p)
	2 x 224 S 3 x 192 S 1 x 96 M (+16 kbit/s) (+p)	4 x 224 S 2 x 96 M[3] (+p)	3 x 224 S 1 x 192 S 1 x 96 M or S[2] 1 x 64 M (+8 kbit/s)	4 x 192 S 1 x 80 S[2] 1 x 64 M[1] (+8 kbit/s)
	1 x 256 S 1 x 224 S 2 x 192 S 2 x 128 S (+8 kbit/s) (+p)	4 x 256 S 2 x 32[2] M	4 x 224 S 2 x 64[1] M	4 x 192 S 2 x 64 M (+16 kbit/s) (+p)

Minimum Bit rate for Data service	**32 kbit/s (840 CUs available)**	**64 kbit/s (816 CUs available)**	**128 kbits (768 CUs available)**	**256 kbit/s (672 CUs available)**
Number of Audio Sub-channels	**Possible configuration**	**Possible configuration**	**Possible configuration**	**Possible configuration**
7	5 x 192 S 2 x 96 M	5 x 192 S 2 x 64 M[1] (+24 kbit/s) (+p)	4 x 192 S 1 x 128 S 2 x 64 (+16 kbit/s) (+p)	4 x 192 S 3 x 48[2] M (+8 kbit/s) (+p)
		1 x 224 S 4 x 192 S 1 x 96 M 1 x 24[2] M	4 x 192 S 2 x 96 M 1 x 64 M[1] (+16 kbit/s) (+p)	3 x 192 S 1 x 144 S[1] 3 x 56 M[2] (+24 kbit/s)
8	5 x 192 S 3 x 56[1] M (+16 kbit/s) (+p)	3 x 224 S 3 x 96 M or S[2] 2 x 64 M (+8 kbit/s)	3 x 192 S 3 x 128 S[2] 2 x 40 M[2]	
9	4x 224 S 5 x 40 M[1] (+16 kbit/s) (+p)			
16	2 x 192 S 13 x 56 M[2] 1 x 48 M[2] (+p)			
24	3 x 56 M[2] 21 x 48 M[2] (+p)			

NOTE 1: Figure in brackets denote additional data capacity.
NOTE 2: (+p) in a configuration shows that extra padding (of less than 8 kbit/s) is used to fill the multiplex.
NOTE 3: Protection level 3 (with an average code rate of 0,5) is used for all audio and data subchannels
1 = 24 kHz sampling frequency recommended
2 = 24 kHz sampling frequency only
3 = 48 kHz sampling frequency only

History

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