EUREKA

Digital Audio Broadcasting

The Eureka 147 Project

DIGITAL AUDIO BROADCASTING SYSTEM

A Brief Description

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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 774 and 789. The System itself is recommended worldwide by the Inter-Union Technical Committee of the World Conference of Broadcasting Unions and now in ITU-R Draft New Recommendations 10/76 (rev.1) and 10/75, for terrestrial and satellite broadcasting respectively. The detailed specification of the Eureka DAB System (also known as ITU Digital System A) is given by the European Telecommunications Standards Institute (ETSI) in Final Draft pr ETS 300 401, September 1994.

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 perpetually irrelevant information from the audio source signal, then it 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 may be eliminated from the recovered signal in the receiver, even when working in conditions of severe multipath propagation, whether stationary or mobile.

Efficient spectrum utilisation 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 on the radiated frequency. The latter feature is known as the Single Frequency Network (SFN), and this may also employ the gap filling technique. In this case, a gap filler transmitter receives and re-transmits the signal on the same frequency, without intervening demodulation, to cover shadowed areas which may arise within the broadcasting network overall coverage area, provided by the main network transmitters.

Nevertheless, the relatively low co-channel protection ratio of the System also permits adjacent local coverage areas to be planned, on a continuously extending basis, with as few as four different frequency blocks.

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). The available services bit-rate ranges between about 1.7 Mbit/s and 0.6 Mbit/s.

The services may contain audio programme data or other data services, and a data service may or may 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) and 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 to 32 kbit/s. This enables the multiplex to be configured to provide typically 6 high-quality stereo audio programmes or up to, say, 20 restricted quality stereo mono programmes with moderately rugged error protection. An example table of multiplex options for audio services is given in Table 1.

Protection Level	3	4
mean code rate, R_{av}	~ 0.5	~ 0.6
coded audio rate, kbit/s	No. of audio channels	
64	18	20
192	6	7
224	5	6
256	4	5

Table 1 - Examples of audio service capacities in a DAB ensemble.

- Program Associated Data (PAD), embedded in the audio bit-stream, for data which are directly linked to the audio programme (e.g. dynamic range control, song lyrics, music/speech flag, etc.). The amount of PAD is adjustable (min. 667 bit/s), 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.
- 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; 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 each 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.

Outline of System Implementation

General

A conceptual block diagram of the DAB System is shown in Fig. 1; Fig. 1(a) 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 time-interleaving process. Finally, very rugged synchronisation 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.

Fig. 1(b) 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.

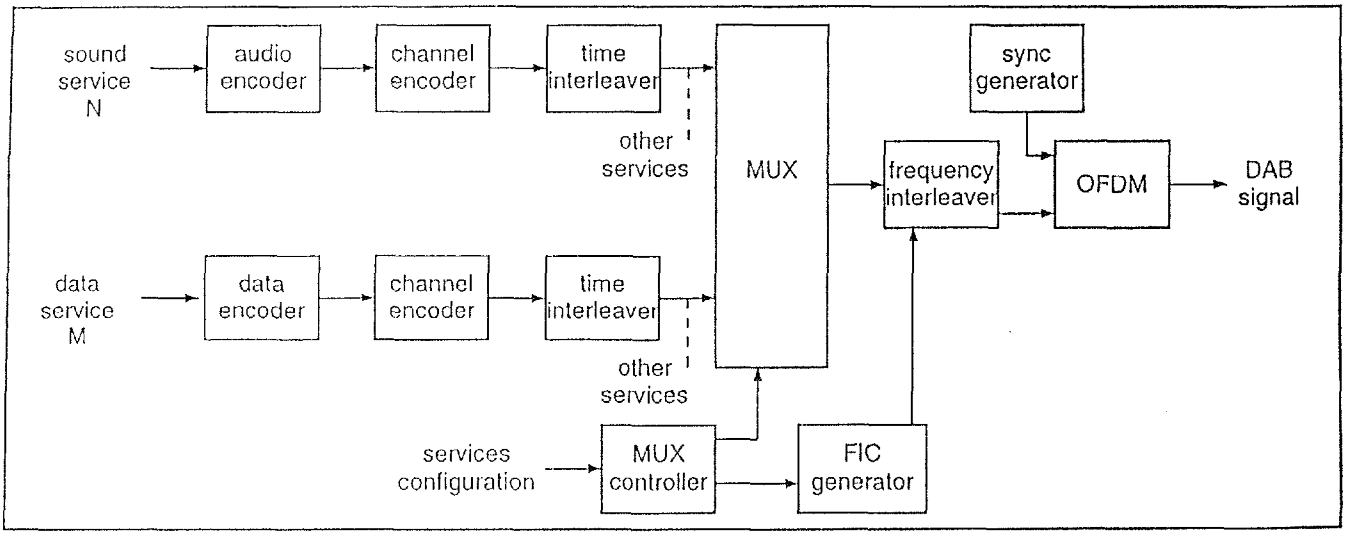


Fig. 1(a) - Conceptual DAB Transmitter Drive.

Thereafter, the receiver performs the transmitter operations of Fig. 1(a) in reverse order, having selected the wanted DAB ensemble and acquired synchronisation. Thus selection is done in the analogue tuner, which performs the tuning and filtering functions. The digitised output of the converter is first fed to the FFT (Fast Fourier Transform) stage and differentially demodulated. This is followed by time and frequency de-interleaving processes, and error correction to output the original coded services data. These data are 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.

Audio Services

The audio source coding method is a perceptually based, low bit-rate sub-band coding system for high quality audio signals, standardised by ISO/IEC under the heading

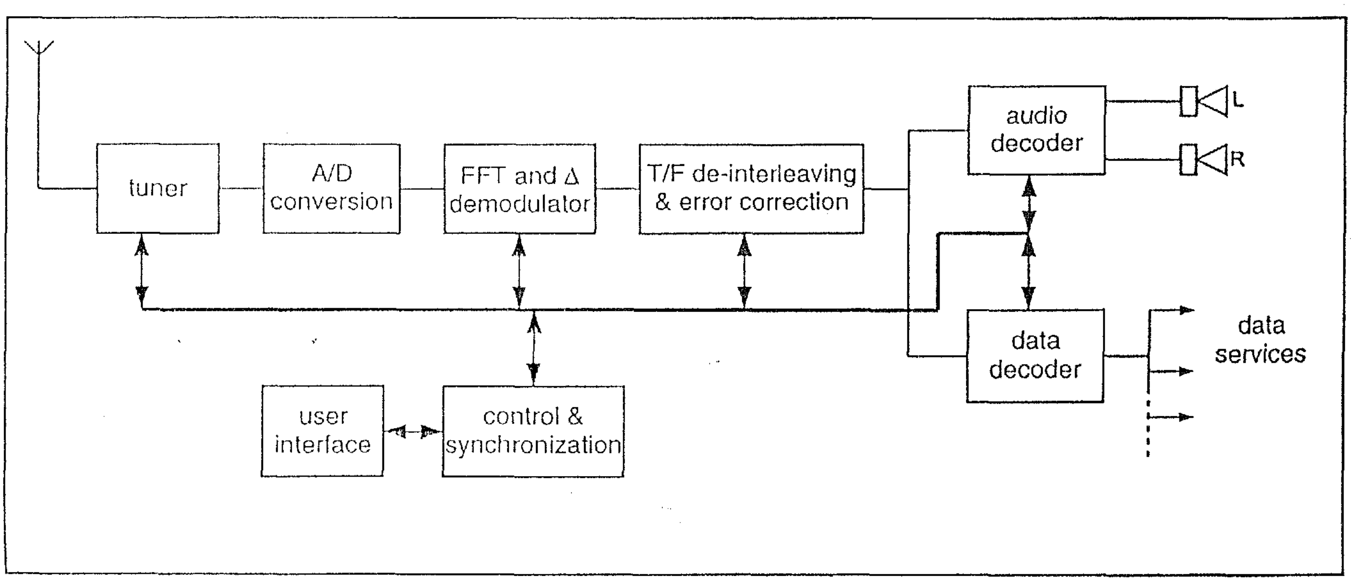


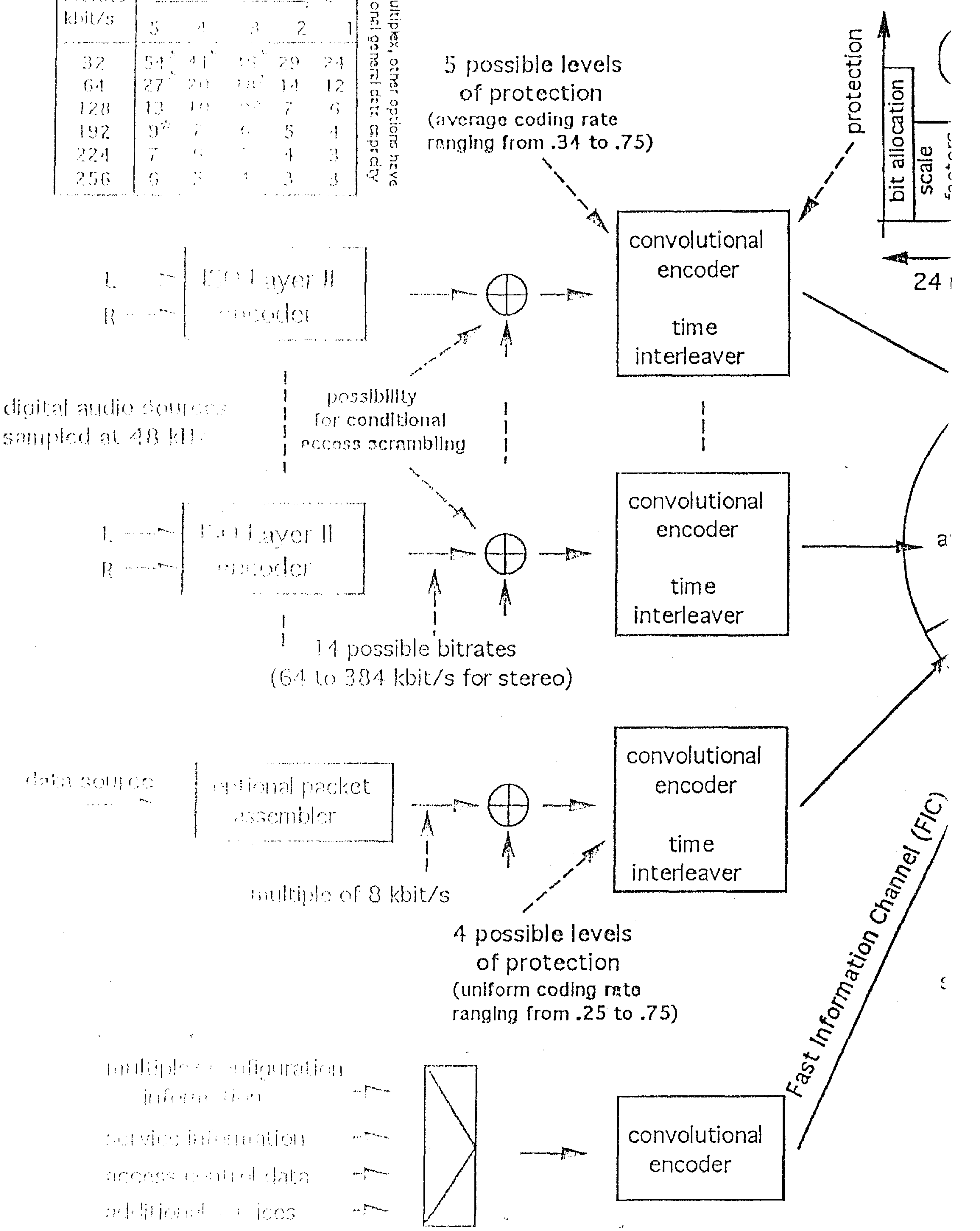
Fig. 1(b) - Conceptual DAB Receiver.

Eureka 147-DAB at

example DAB multiplexes:
maximum number of audio channels

audio bit-rate kbit/s	protection level (increasing protection)				
	5	4	3	2	1
32	54*	41*	[illegible]*	29	24
64	27*	20	18*	14	12
128	13	10	[illegible]*	7	6
192	9*	7	6	5	4
224	7	[illegible]	[illegible]	4	3
256	6	5	4	3	3

* full multiplex, other options have additional general data capacity



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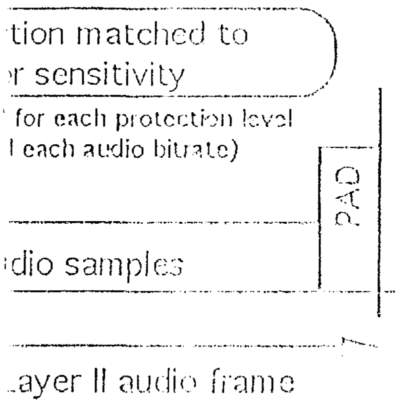
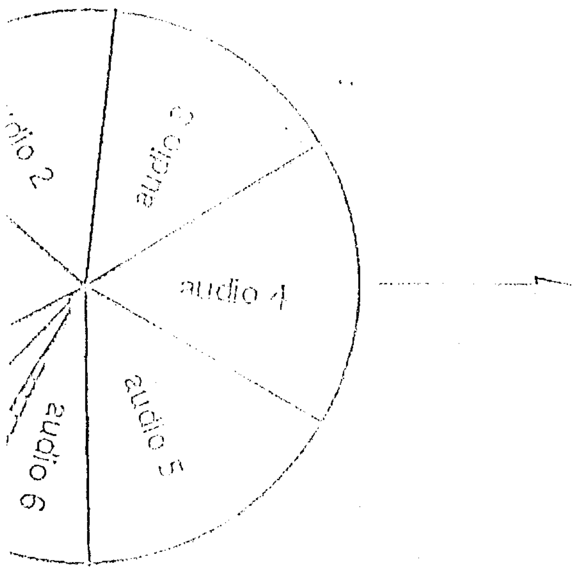


3 transmission modes

- mode I Δ ~246 µs, ts=1 ms, Frequency ≤ 375 MHz
- mode II Δ ~62 µs, ts=250 µs, Frequency ≤ 1.5 GHz
- mode III Δ ~31 µs, ts=125 µs, Frequency ≤ 3 GHz

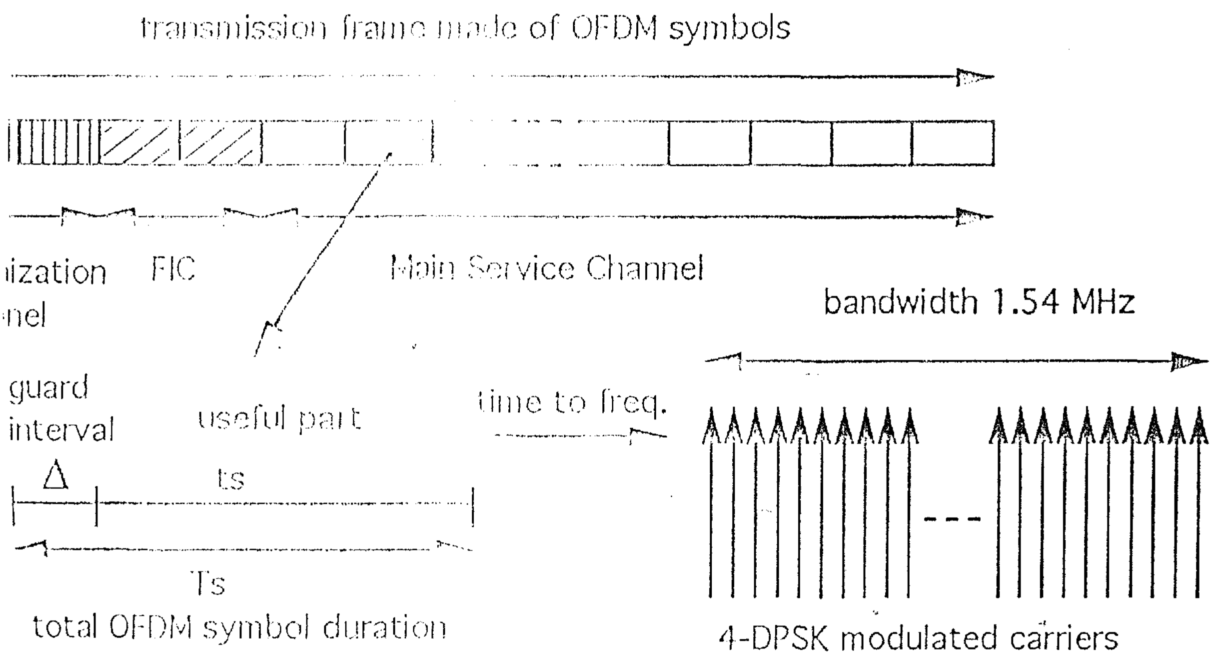
DAB ensemble
to transmitter(s)

it/s MULTIPLEX



OFDM signal generator

- synchronization channel generator
- multiplex to OFDM symbols mapper
- QPSK symbol mapper
- Frequency interleaver
- Differential modulator
- Transmitter identification generator



ISO/IEC 11172/3 (MPEG Audio Layer II). The DAB Specification permits use of the flexibility of Layer II except for the fact that only the standard studio sampling frequency of 48 kHz is used. Layer II is capable of processing mono, stereo and dual channel such as a bilingual programme, and different encoded bit-rate options are available (viz: 32, 48, 56, 64, 80, 96, 112, 128, 160 or 192 kbit/s per monophonic channel). In stereophonic or dual channel 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 may be conveyed in the stereo mode, or, in particular at lower bit-rates, in the joint stereo mode. This exploits the redundancy and interleaving of the two channels of a stereophonic programme to maximise the overall perceived audio quality.

Each audio service channel also contains a PAD (programme associated data) channel, having a variable capacity (minimum 0.667 kbit/s), which can be used to convey information which is intimately linked to the sound programme. This PAD channel is incorporated at the end of a DAB/ISO audio frame, and, therefore, cannot be subject to 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, may be conveyed in the PAD.

Data Services

In addition to the Programme Associated Data which may be carried with the audio general data may be conveyed as a separate service. This may be in the form of a continuous stream, segmented in 24 ms 'transport frames', or arranged as packet data services. The resource allocated to a data service is arranged in multiples of 8 kbit/s 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.

Service information

The following elements of Service Information (SI) can e 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 monitoring and control by the broadcasters, for example.

System Organisation & 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 are 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, in 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, may 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') may 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 the 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.

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. The average code rate, defined as the ratio between the number of source-encoded bits and the number of encoded bits after convolutional encoding, may take a value from 0.35 (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 may 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.

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 synchronised data from all of the programme services using the multiplex are brought together.

Transmission Frame and Modes

The System provides three 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 multipath echoes.

Table 2 *(overleaf)* gives the temporal guard interval duration and nominal maximum transmitter separation and frequency range for mobile reception. The noise degradation at the

System Parameter	Transmission Mode I	II	III
Guard interval duration	246 μs	62 μs	31 μs
Nominal maximum transmitter separation for SFN	96 km	24 km	12 km
Nominal frequency range (for mobile reception)	≤ 375 MHz	≤ 1.5 GHz	≤ 3 GHz

Table 2 - Limiting planning parameter values for each Transmission Mode.

highest frequency and in the most critical multipath condition, occurring infrequently in practice, is equal to approximately 1 dB at 100 km/h.

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), because it allows the greatest transmitter separations. Mode II is most suitable for local radio applications requiring one terrestrial transmitter, although it can also be used for a medium scale SFN; in fact, larger transmitter spacings can be accommodated by inserting artificial delays at the transmitters and by using directive transmitting antennas. Mode III is most appropriate for cable, satellite and complementary terrestrial transmission, since it is able to operate at all frequencies up to 3 GHz for mobile reception, and has the greatest phase-noise tolerance.

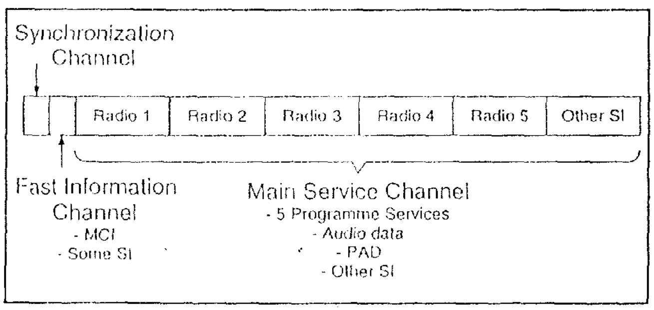


Fig. 2 - An example of a DAB Multiplex Frame.

In order to facilitate receiver synchronisation, the transmitted signal is built up with a frame structure having a fixed sequence of symbols. Each transmission frame, see Fig. 2, begins with a null symbol for coarse synchronisation (when no RF signal is transmitted), followed by a phase reference symbol for differential demodulation. 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 or 24 ms, depending on the transmission mode as given in Table 3. Each audio service within the MSC is allotted a fixed time slot in the frame.

System Parameter	Transmission Mode I	II	III
Frame duration T_F	96 ms	24 ms	24 ms
Null symbol duration T_{null}	1297 μs	324 μs	168 μs
Guard interval duration t_Δ	246 μs	62 μs	31 μs
Useful symbol duration t_s	1 ms	250 μs	125 μs
Total symbol duration T_s	1246 μs	312 μs	156 μs
No. of radiated carriers N	1536	384	192

Table 3 - DAB Transmission parameters for each Transmission mode.

Modulation with OFDM

The 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 multipath 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 multipath propagation will not cause inter-symbol interference. The large number, N, of orthogonal carriers (see Table 3), which can be conveniently generated by an FFT 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. Fig. 3 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 may be further reduced by additional signal conditioning when coupled with non-linear amplification in the transmitter. In the presence of multipath 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 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, multipath 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 multipath propagation can completely destroy a service.

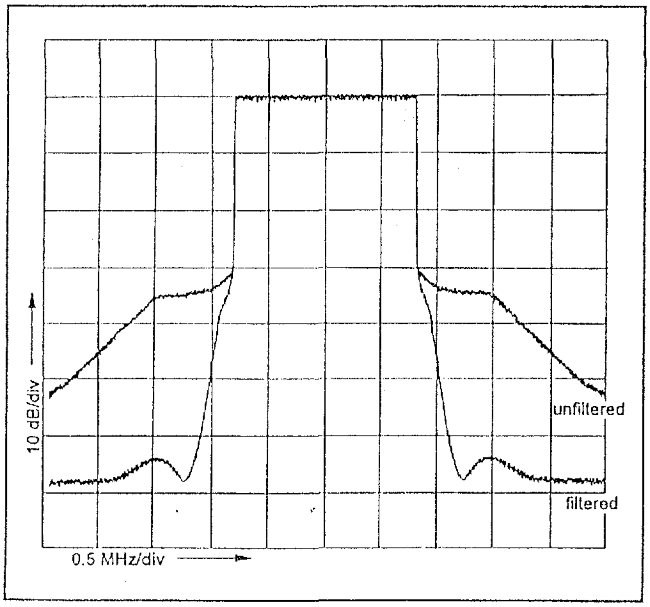


Fig. 3 - Transmitter Signal Spectrum and Output Filtering (VHF Band III).

Further Information

Further information about the Eureka DAB System can be found in the detailed specification available from ETSI as *Final Draft pr ETS 300 401, Radio Broadcasting System; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers*, European Telecommunications Standards Institute, Sophia Antipolis, September 1994; and from the current versions of the ITU Radiocommunications Sector (Geneva) Reports 1203 (terrestrial) and 955 (satellite), and Recommendations 774 (terrestrial) and 789 (satellite), where the Eureka DAB System is known as Digital System A.

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