Digital Audio Broadcasting

Towards the next generation of DAB receivers

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The first DAB receivers which accorded with the DAB European Telecommunication Standard became available in 1995. These were based on either a DAB channel-decoder chipset from the JESSI project, or on general-purpose DSPs. For a consumer product, the complexity of these early DAB receivers was much too high.

This article describes the features of the next generation of DAB receivers, which are due to be launched at the Berlin IFA in late August. These new receivers will be based on a second generation of channel-decoder chipsets, as developed within the JESSI AE-14 and AE-89 projects. Not only will they be much smaller (and cheaper) than the earlier models from 1995, the new generation of receivers will also support DAB transmission mode IV and the reception of low bit-rate MPEG-2 speech.

1. Introduction

The chances for a successful introduction of Digital Audio Broadcasting (DAB) [1] have increased significantly over the last three years. Several activities have contributed to this:

- professional receivers from Philips, Grundig and Bosch have become available since 1995, allowing large-scale European field trials to take place;
- these field trials have shown the feasibility of a terrestrial DAB network [2];
- official DAB transmissions have already begun in the UK, Sweden, Belgium and Canada;
- in Germany, several large-scale field trials are in operation;

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- the IFA 95 exhibition in Berlin demonstrated a clear move away from the technically-driven interest in DAB to a commercially-driven interest (confirmed by the lectures given at the last DAB symposium in Montreux [3]);

- The WorldDAB (formerly EuroDab) forum and the CEPT T-DAB planning meeting [4] in Wiesbaden, Germany, have prepared the way in Europe for a smooth introduction of terrestrial DAB in Band III;

- the recent addition of transmission mode IV in the DAB standard has facilitated the use of a hybrid transmission network in the (1.5 GHz) L-band [5].

In addition, many experiments have taken place with regard to the introduction of DAB data services. Mobile DAB services which carry still pictures, (MPEG) video and even HTML have been demonstrated on several occasions by certain broadcasters and companies. However, for a successful introduction of these non-audio services,

Abbreviations

A/D	Analogue-to-digital	**GSM**	Global system for mobile communications
AFC	Automatic frequency control	**HTML**	Hyper-text markup language
AGC	Automatic gain control	**IEC**	International Electrotechnical Commission
AM	Amplitude modulation	**ISI**	Inter-symbol interference
ASCII	American Standard Code for Information Interchange	**ISO**	International Organization for Standardization
ASIC	Application-specific integrated circuit	**JAVA**	(Sun Microsystems) programming language for the WWW
BER	Bit error rate	**JESSI**	Joint European Sub-micron Silicon Initiative
BMP	(Windows) bitmap	**JPEG**	(ISO) Joint Photographic Experts Group
CA	Conditional access	**LSF**	Low sampling frequency
CEPT	European Conference of Postal and Telecommunications Administrations	**MCI**	Multiplex configuration information
CIR	Channel impulse response	**MHEG**	(ISO) Multi- and Hyper-media coding Experts Group
CRC	Cyclic redundancy check	**MOT**	Media object transfer
CU	Capacity unit	**MPEG**	(ISO) Moving Picture Experts Group
CVF	Common virtual frame	**OFDM**	Orthogonal frequency division multiplex
DAB	Digital Audio Broadcasting	**QEF**	Quasi-error-free
DIN	*Deutsches Institut für Normung e.V.* (German industry standards organization)	**PAD**	Programme-associated data
DRC	Dynamic range control	**PCM**	Pulse code modulation
DSM-CC	(DVB) Digital storage media command control	**R-S**	Reed-Solomon
DSP	Digital signal processor	**RDI**	Receiver data interface
DVB	Digital Video Broadcasting	**RDS**	Radio Data System
EEP	Equal error protection	**SFN**	Single-frequency network
ETS	European Telecommunication Standard	**SI**	Service information
FFT	Fast Fourier transform	**TII**	Transmitter identification information
FIB	Fast information block (DAB)	**TS**	(MPEG) transport stream
FIC	Fast information channel (DAB)	**UEP**	Unequal error protection
FIG	Fast information group (DAB)	**ZIP**	An industry-standard file compression utility (PKWare Inc.)
GIF	Graphics interchange file		

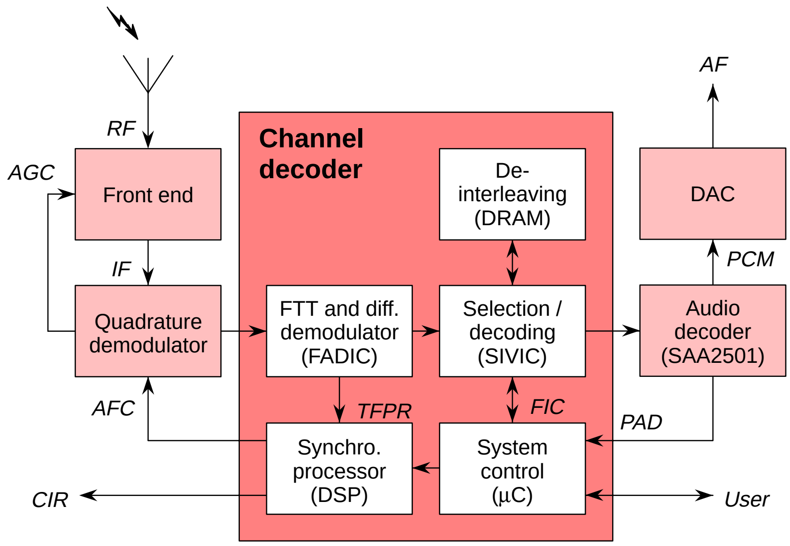


Figure 1
Block diagram of the Philips DAB452 receiver, based on the first-generation channel-decoder chipset.

some clear decisions will have to be made in the near future.

2. *A single-chip channel decoder*

Two application-specific IC (ASIC) concepts have been developed for DAB within the European JESSI AE-14 project.

The first concept is based on separate ICs from Philips for the demodulator (FADIC) and the decoder (SIVIC). A block diagram of a receiver that is based on these ICs [6] is shown in *Fig. 1*. Recently this concept has been extended by the German manufacturer, Temic, with the introduction of modules and ICs which support the transmission of DAB mode IV in L-band. This new mode, for satellite reception of DAB, fills the previous gap between mode I and mode II.

The second concept has also been extended with an ASIC which generates digital (I and Q) baseband signals from the analogue IF signal. A disadvantage of this concept is the need for an external DSP and the know-how to perform the synchronization functions of the receiver. With current implementations, the decoding rate is limited to 384 kbit/s, whereas the full DAB ensemble can carry up to 1843 kbit/s (with an error protection ratio of 4/5).

The second ASIC concept, as applied in the new Philips SAA3500, is illustrated in *Fig. 2*. The SAA3500 is a single chip which contains a digital IF mixer, the OFDM demodulator, deinterleaving, a full-speed Viterbi decoder as well as a time- and frequency-synchronization processor. A disadvantage of this level of integration might be the loss of flexibility in synchronization, although the on-chip synchronization has been further improved with respect to previous DAB receiver implementations.

The function of each block in this new ASIC is now described.

2.1. *Digital mixer*

The SAA3500 input signal is a 10-bit IF signal, (sub)sampled at 8192 kHz. The digital mixer provides down-conversion from the 2048 kHz IF frequency to baseband, with a resolution of better then 2 Hz, in order not to degrade the decoder performance due to inter-carrier interference. The IF frequency is adjusted digitally with a 22-bit AFC value derived from the frequency detector of the synchronization processor.

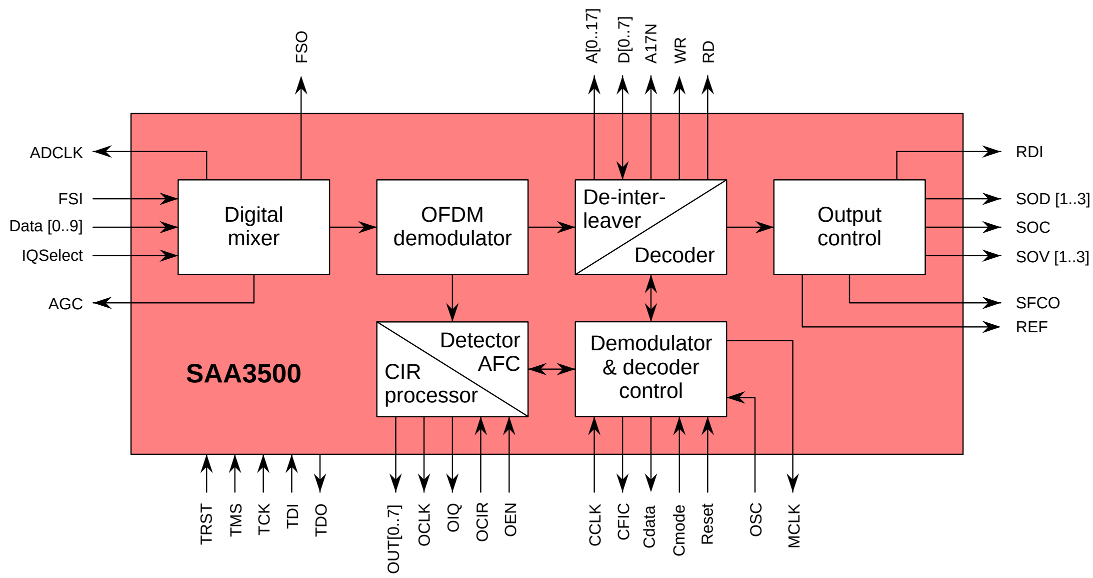


Figure 2
Block diagram of the SAA3500, a fully-integrated DAB channel decoder of the next generation.

An in-band AGC detector is provided to adjust the average digital IF input level at about 12 dB below the maximum level, in order to prevent harmonic distortion. Digital filters provide 48 dB selectivity and a pass-band ripple of less than 0.5 dB. In-band AGC is also applied internally at baseband level, with a step size of 1.5 dB and an amplification range of 48 dB, in order to normalize the input signal level to the demodulator.

2.2. Demodulator

The demodulator performs a Fast Fourier Transform (FFT) function in order to obtain the multiple OFDM carriers which hold the encoded data bits. The size *N* of the FFT is 2048, 1024, 512 or 256 points respectively for modes I, IV, II and III. The transform should be finished within an OFDM symbol period (including the guard interval period Δ), which is 1.24 ms for mode I. The operation of the FFT is defined by:

$$X_{k,l} = \frac{1}{\sqrt{N}} \sum_{n=-w}^{N-w-1} x_{n,l} . e^{-j2\pi\frac{kn}{N}}$$

$$= \frac{e^{-j2\pi\frac{kw}{N}}}{\sqrt{N}} \sum_{n=0}^{N-1} x_{n,l} . e^{-j2\pi\frac{kn}{N}}$$

where: X = the OFDM carrier spectrum
x = the baseband signal
k = the OFDM carrier index ($-K/2 \ldots K/2$)
l = the OFDM symbol index ($0 \ldots L$)
n = the baseband sample index for the current symbol ($0 \ldots N-1$)
w = the start of the FFT demodulation window ($0 \ldots \Delta-1$).

The values for *K*, *L*, *N* and Δ are given in *Table 1*. The equation shows that, due to the guard interval, the OFDM carrier spectrum *X* is independent of the demodulation window position *w*, apart from a linear phase shift. Given a Gaussian amplitude distribution of the baseband signal, a scale factor $1/\sqrt{N}$ is applied to obtain an FFT output signal whose amplitude is about equal to that of the input signal.

The number of (complex) multiplications required for a basic (radix-2) FFT is about $N/2.\log_2(N)$ [7]. For transmission mode I (where $N = 2048$), this corresponds to 11,264 multiplications in a 1 ms period. This leads to either a complex multiplier operating at 12.288 MHz (a multiple of the IF frequency) or, equivalently, a single multiplier operating at 49.152 MHz. To limit the memory-access bandwidth (and the number of multiplications), a radix-64 FFT is chosen. This means that, depending on the DAB mode, the *N*-point FFT is split into either 4, 8, 16 or 32 FFTs in units of 64 samples. This approach results in power savings with respect to the common radix-2 FFT.

After calculation of the FFT, differential demodulation is applied by means of a complex multiplication:

$$Y_{k,l} = X_{k,l} * X^{*}_{k,l-1}$$

The results are quantized to 4-bit metrics, with the encoded data bits being extended with a 3-bit reliability indication. Consequently, soft-decision error correction can be performed in order to improve the coding gain.

2.3. Synchronization core

Integrating the synchronization algorithms, which previously have been implemented in a general purpose DSP, was the final step towards a fully-integrated channel decoder. The synchronization core starts with a calculation of the Channel Impulse Response (CIR) from the reference symbol ($l = 1$), after inverse FFT processing:

$$h_n = \frac{1}{N} \sum_{k=-K/2}^{K/2} X_{k,1} Z_k^{*} e^{j2\pi\frac{kn}{N}}$$

$$= \frac{1}{N} \sum_{k=-K/2}^{K/2} Z_k e^{-j2\pi\frac{kw}{N}} Z_k^{*} e^{j2\pi\frac{kn}{N}}$$

$$= \frac{1}{N} \sum_{k=-K/2}^{K/2} e^{j2\pi\frac{k(n-w)}{N}} = \delta(w)$$

where: Z_k = the reference symbol as inserted at the transmitter side
$X_{k,1}$ = the received reference symbol, demodulated by an FFT with offset *w*.

Parameter	DAB transmission mode			
	I	IV	II	III
N	2048	1024	512	256
Δ	504	252	126	63
K	1536	768	384	192
L	76	76	76	153
Δf	1 kHz	2 kHz	4 kHz	8 kHz

Table 1
DAB mode parameters (Δf is the carrier spacing).

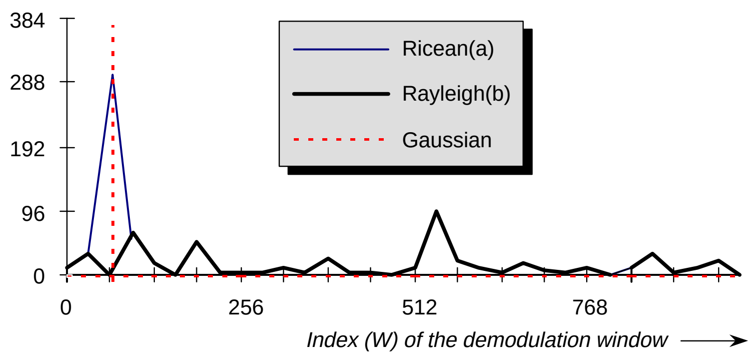
This operation corresponds to a time domain correlation of both these reference symbols, and the resulting h_n represents the CIR. In the case of an ideal transmission channel, this will be a single correlation peak δ, at the offset position w of the demodulation window. The detected peak position is used to adjust the demodulation window properly.

In order to obtain optimum receiver synchronization in a multipath environment, it is vital to implement a robust CIR peak-detection algorithm. The CIR peak position is used to adjust the index w of the demodulation window in such a way that the Inter-Symbol Interference (ISI) due to multipath reception is avoided whenever possible. It is hard to express in a simple formula the optimum peak position. In the case of a Gaussian or Ricean (e.g. line-of-sight) channel, it is sufficient to position the FFT window relative to the most significant CIR peak position detected (see *Fig. 3 plot a*). In a Rayleigh channel with no line-of-sight reception, a dominant peak may not be present, or significant reflections may precede it. In this case, an average peak position for the CIR will be determined (*Fig. 3 plot b*).

In order to determine if there is an integer offset in the carrier positions of the demodulated reference symbol, a correlation of carrier sequences is carried out. This is based on the fact that the reference symbol contains a repetition of modulated data sequences, each with a length of 32 and a constant amplitude. The demodulated sequences can be correlated with the original sequences to find the integer carrier offset i that provides maximum correlation. In addition to the integer offset, the fine or fractional frequency offset should also be determined. It can be obtained from a common phase-error detector on the output of the differential demodulator [8]. The addition of the integer and the fractional offsets results in an AFC value which is used to adjust the digital mixer.

It is relatively simple for the synchronization processor to deal with (optional) Transmitter Identification Information (TII) in the null symbol. The algorithm is similar to the integer carrier-offset detection algorithm, due to the fact that TII carrier-pairs are identical to the reference symbol carrier-pairs. If all the carriers in the null symbol are to be checked for the presence of TII, either real-time processing of the synchronization core, or buffering of the TII, is needed. In a simplified approach, as applied in the SAA3500, sequential processing is carried out where three TII identifiers are processed every other frame.

Figure 3
Examples of Channel Impulse Responses.



2.4. Decoder

Output metrics from the demodulator are grouped, independent of mode, into 864 Capacity Units (CUs). An inter-symbol frequency de-interleaver and a time de-interleaver with a depth of 16 frames take care of removing unwanted metric correlation. To constrain the chip area used, the de-interleaver makes use of an external RAM of either 1 MB (416 CUs) or 2 MB (864 CUs). After de-interleaving, the metrics are applied to a Viterbi decoder which has the possibility of de-puncturing the incoming metrics to handle code rates of between 8/32 and 8/9. Tables for control of the de-puncturing, in accordance with both the Equal Error Protection (EEP) and Unequal Error Protection (UEP) profiles as defined for DAB, are stored in a ROM. These include the recently-defined EEP code rates R of 4/5, 4/6, 4/7 and 4/9.

For the Viterbi decoder, a trade-off has been made between decoding capacity, on the one hand, and frame buffering on the other. If the Viterbi decoder is not able to perform real-time decoding of the incoming metrics, the incoming data has to be stored selectively on a logical frame basis. Given a decoding rate of up to 384 kbit/s and 4-bit metrics, this requires the storage of 24*384*4*4 bits at a protection ratio $R = {}^1/_4$. This corresponds to 144 kB (about 16 mm^2 in a 0.8µ technology). Due to the low decoding speed, the Viterbi decoder may use a sequential implementation method (about 10 mm^2) [9]. If the Viterbi decoder can perform real-time decoding (which requires a 2 Mbit/s decoding speed), input metric buffering is not needed. For decoding of an MPEG audio component, a double buffer is needed at the Viterbi output (see *Section 4.3*), which requires only about 384 kbit/s*48 ms = 18 kB (about 2 mm^2). In this case a more complicated parallel (trace-back) implementation is needed for the decoder (about 16 mm^2). The savings in memory area for real-time decoding compensate abundantly for the additional area needed for a high-speed Viterbi decoder implementation. In addition to this smaller area, full-capacity decoding can be exploited for fast switching between services within one ensemble, or for future high bit-rate services such as MPEG video.

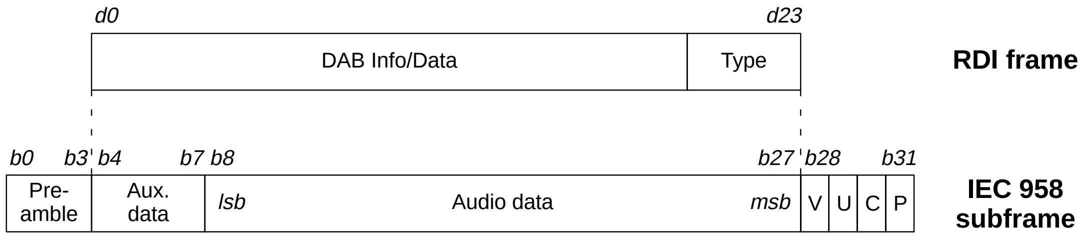


Figure 4
DAB ensemble data is embedded in the IEC-958 frames.

The Fast Information Channel (FIC) is decoded separately without time de-interleaving and at a fixed code rate of 1/3. The number of corrected bits (the error-flag count) for the FIC is registered as an indication for the received signal quality. During decoding of the FIC, a window signal is generated, which may be used by the external controller for synchronization to the decoded DAB frame. The FIC data is buffered on-chip to allow low-speed access on a DAB frame basis.

2.5. Demodulator and decoder control

Initially this block takes care of synchronizing the OFDM symbol processing to the null symbol detector, and verification of the selected transmission mode. Once synchronized, the demodulator and decoder control block handles the selection of symbols for demodulation, the CUs for de-interleaving, as well as the selection of the Error Protection profiles based on parameters which are transferred via the control bus. In the event of a DAB multiplex reconfiguration, these selections are updated with a well-defined timing relationship. The control block also handles the I2C and L3 bus-protocols associated with the external micro-controller communication.

2.6. Receiver Data Interface

With the increasing interest in the provision of additional data services on DAB, the need arose for a standardized Receiver Data Interface (RDI) which allows data transfer to an external terminal. Within the Eureka-147 project, an RDI has been defined [10], which can transfer all the data of the DAB ensemble to an external device in a standardized format that is based upon the IEC-958 audio interface [11]. Although intended for digital audio data, the IEC-958 interface definition comes very close to the needs of a DAB decoder interface. It is widely accepted as an interface for audio equipment, and offers the required bandwidth.

The IEC frames carry PCM data at up to 24 bits/sample, which corresponds to a bit-rate of 2304 kbit/s at a 48 kHz sampling frequency. This bit-rate equals the gross bit-rate of the DAB main service channel. At the output of the DAB receiver, the gross bit-rate has been reduced by the code rate, which means that all ensemble data can be transferred via the RDI. This includes some control data to indicate the beginning and the end of each sub-channel as well as FIC data. *Fig. 4* shows how the DAB data is embedded into an IEC-958 sub-frame. A different type of indication

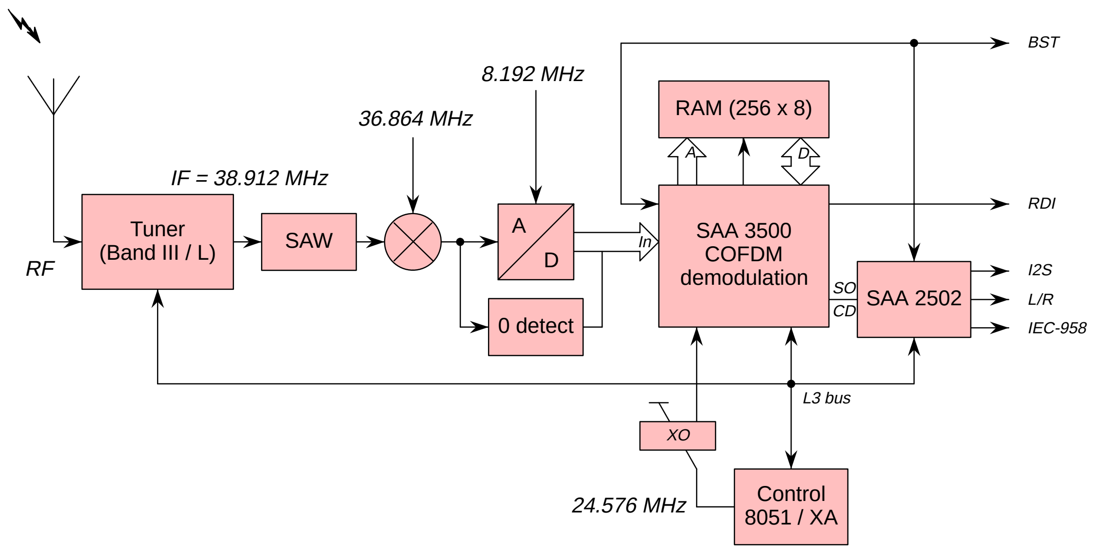
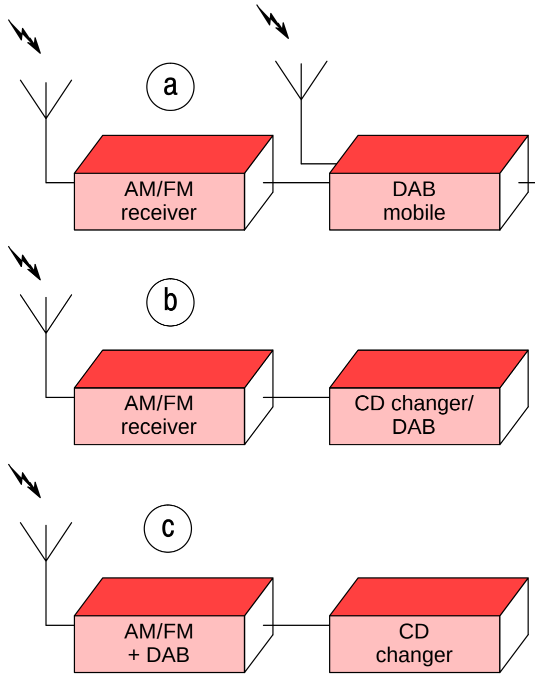


Figure 5
DAB receiver module, based on the next-generation channel-decoder IC.



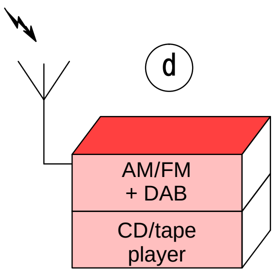
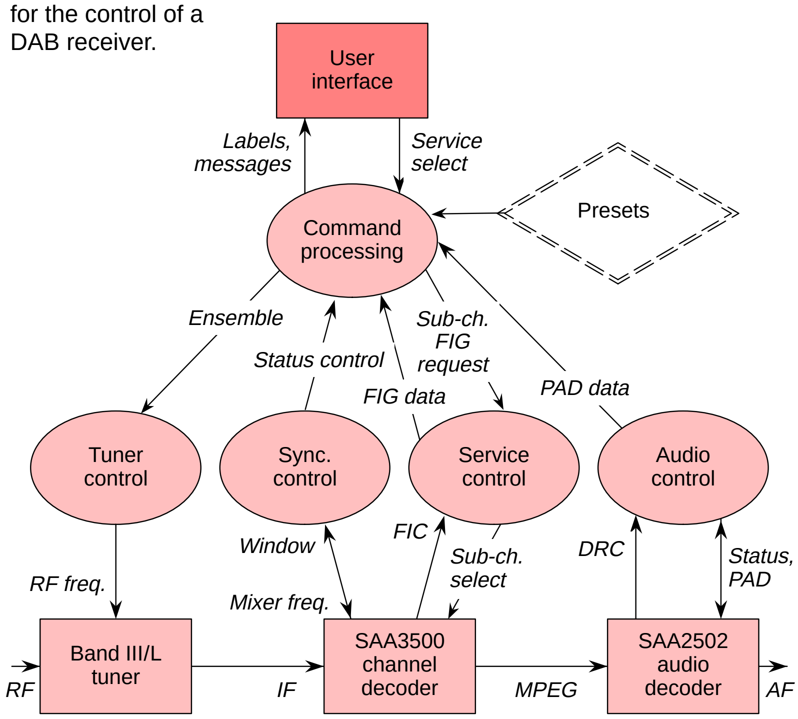


Figure 6
Car audio systems of tomorrow; each box represents a separate DIN-size module.

Figure 7
Software architecture for the control of a DAB receiver.



is used in the serial channel-status bit-stream (C) to distinguish DAB data from PCM audio data.

The User bit (U) will be used in the future as a Return Control Channel for data with a maximum bit-rate of 96 kbit/s. It can be used to exchange control data between the DAB receiver and one or more source decoders.

3. Next-generation DAB receivers

3.1. Hardware

Based upon the concept of a single-chip channel decoder, the receiver architecture becomes relatively simple as shown in *Fig. 5*. Recent component developments allow the complete tuner modules for both band III and L-band to be as small as 30 cm^2. For the DAB/MPEG-2 audio decoder, a single chip (the SAA2502) was already available; it includes the audio D/A converters and an IEC-958 interface.

The complete DAB receiver module will fit on a single car-radio-sized printed circuit board. A standard DIN-size car radio can accommodate the DAB receiver (instead of a cassette player) in combination with an AM/FM receiver. Several receiver configurations are possible for a car system as shown in *Fig. 6*. Price, ease of mounting (including the antenna) and its possible combination with other car system functions (such as a cassette player, a navigation system or a GSM telephone) will be prime considerations for making the right choice. It is worthwhile noting here that the number of CD changers on the market is increasing rapidly, and that more and more vehicles provide the dashboard space for mounting a double-height car-radio case (*Fig. 6d*).

If the receiver concept shown in *Fig. 5* is used and if it is operated with the appropriate control software, then the DAB receiver will be able to support all the features marked as essential for DAB car radios [12] by the former European Forum on Digital Audio Broadcasting (*EuroDab).*

3.2. Software

The control function for a DAB receiver module can be implemented on a separate micro-controller or it can be combined with an existing car-system controller. *Fig. 7* shows the basic control functions of the DAB receiver micro-controller (μC) and the associated data items.

Each function is described further in the following paragraphs.

3.2.1. Command processing

This function provides global control of the receiver. It takes care of translating a service selection into an ensemble and the sub-channel selections. It also handles the display of service-related data (such as ensemble, service and dynamic labels) and status messages (for example "Service not found"). Presets may be used to store service selections although it should be taken into account that some DAB services will be available only temporarily.

3.2.2. Tuner control and null detection

After tuning to a DAB ensemble, by setting the correct RF synthesizer frequency, the channel decoder should synchronize to the DAB ensemble and check the DAB mode setting by means of null symbol detections. The length of, or the distance between, detected null symbols should correspond to the selected mode. If this is the case, the channel decoder may exit its null detection state and start the symbol demodulation with the internal timebase synchronized to the null detector.

3.2.3. Synchronization control

During normal operation of a DAB receiver, the µC should check the synchronization status of the demodulator. The µC can read the frequency deviation (AFC) and the CIR peak position offset from the demodulator every frame, from calculations made by the on-chip synchronization processor. Offset values are used to update the digital mixer frequency and the FFT demodulation window position, in order to obtain and maintain proper synchronization. A good trade-off between synchronization speed and accuracy is obtained by adaptation of the feedback attenuation factors for the AFC and CIR during coarse synchronization. Initially a relatively low attenuation may be used and when the deviations are small the attenuation should be increased. If there is a significant frequency error, or if there are no significant CIR peaks below the noise threshold, the updating of the respective settings can be inhibited in order to retain the values set in the previous frame. If the inhibit conditions are found over a number of consecutive frames, a synchronization loss should be signalled and the receiver should restart with tuning or null detection.

3.2.4. FIC processing and selections of sub-channels

After proper synchronization, the receiver can start to process the FIC. The controller should capture the FIC from the channel decoder within the current transmission frame. The FIC is obtained as separate (32-byte) Fast Information Blocks (FIBs), from which the last 2 bytes reflect the result from a Cyclic Redundancy Check (CRC) verification. Data from a properly-received FIB (e.g. CRC = 0) can be processed further.

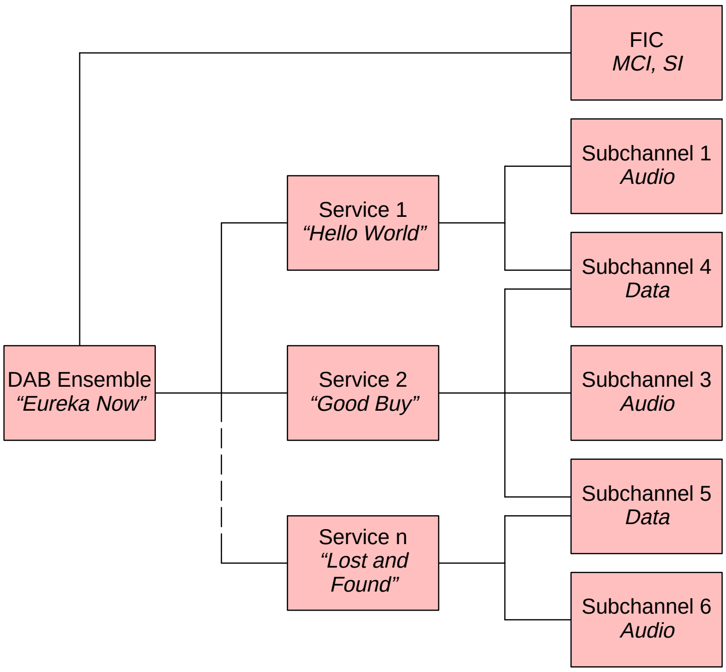
The Multiplex Configuration Information (MCI), which holds the key data needed for service handling, is obtained from the FIC [13]. The basic DAB service organization, as signalled in the MCI, is depicted in *Fig. 8*. The decoder itself only accepts sub-channel selections, so it is the task of the controller to extract the sub-channel organization and select the available (or, to save power, only the required) sub-channels. Sub-channels that are associated with a selected service should be directed to the corresponding source decoders.

Due to interleaving, there is a delay of about 400 ms between selecting the sub-channels and having the actual data available at the decoder outputs. In the event of a multiplex reconfiguration, the controller should apply the new sub-channel(s) organization to the decoder and signal the occurrence of new services or the disappearance of a selected service.

3.2.5. Audio control

Control of the MPEG audio decoder basically involves checking the proper functioning of the decoder (synchronized and no data CRC errors). In addition, this control block may capture the Program Associated Data (PAD), process the Dynamic Range Control (DRC) data and transfer the dynamic labels (if present).

Figure 8
Basic DAB service organization.



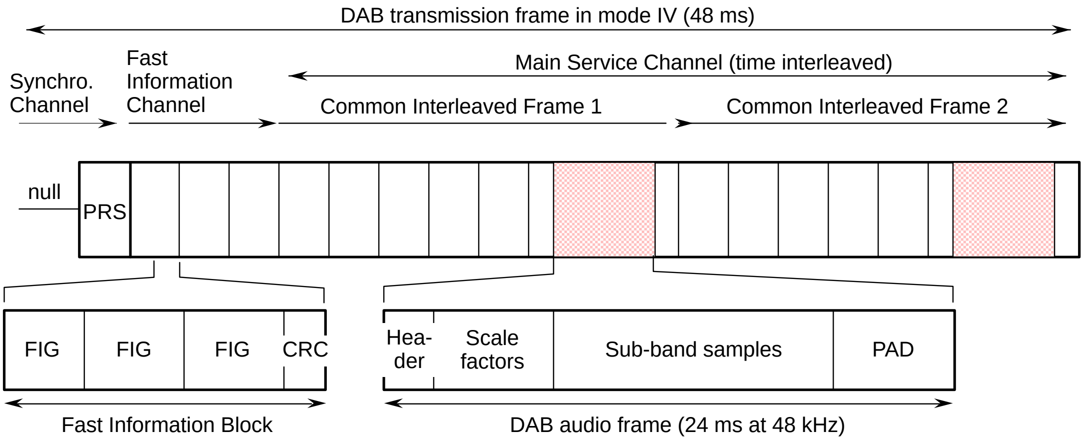


Figure 9
Mode IV frame structure (the phase reference symbol is indicated by PRS).

4. *Additional features*

New DAB receivers will be extended with several features that were recommended by the former EuroDab Forum [12]. Some of these are of particular interest to the broadcasters and have recently been added to the DAB standard [1]. The decoding-related parts of these features are supported by the new DAB decoder ASICs, but the availability of recommended service features will depend on the receiver software.

4.1. *Mode IV*

In the original ETS, there was a gap between mode I (with 1 kHz intercarrier spacing) and mode II (4 kHz spacing). Originally, mode II (or even mode III) was intended for hybrid satellite and terrestrial broadcasting in the (1.5 GHz) L-band, as required for large and thinly-populated countries such as Canada or Australia. Simulations and field tests have shown that the performance of mode II in these cases is above the initial expectations. This is due to the fact that frequency shifts caused by the Doppler effect are tracked by the receiver AFC as long as there is a line-of-sight signal (e.g. a Ricean channel). The new mode IV has an intercarrier spacing of only 2 kHz, which is more critical with respect to Doppler effects at 1.5 GHz. This is compensated by the fact that it allows a 40 km transmitter spacing, rather then just 20 km, and thus reduces significantly the costs for an SFN. The frame structure for mode IV is shown in *Fig. 9*.

	Audio bit-rate	
	Low sampling frequency (kbit/s)	**Normal sampling frequency (kbit/s)**
2	**16**	48
3	**24**	56
4	32	64
5	**40**	80
6	48	96
7	56	112
8	64	128
9	80	160
10	96	192
11	112	224
12	128	256
13	**144**	320
14	160	384

Table 2
MPEG audio bit-rates. New bit-rates for which no UEP has been defined are indicated in bold.

4.2. *Low bit-rate audio*

MPEG audio has been extended with a Low Sampling Frequency (LSF) of 24 kHz, to address the requirement of broadcasters who need to transmit speech services with a lower bit-rate, but with acceptable quality. The audio bit-rates which are allowed at the normal and low sampling frequencies are given in *Table 2*.

When the 24 kHz LSF is used, the duration of the audio frame doubles (to 48 ms) and thus one MPEG frame is distributed over two DAB common-interleaved frames. This means that the UEP profiles, as defined for a 48 kHz sampling frequency, no longer match the audio frame structure. However it is still quite valid to use them – except for the bit-rates where no UEP has been defined, in which case EEP should be used. In case of a reconfiguration involving a low-sampling-frequency audio sub-channel, the reconfiguration should take place at the boundary of a (48 ms) audio frame in order to prevent mismatches between the bit-rate signalled in the audio frame header and the actual audio frame length.

Since the type of data carried in a sub-channel is signalled separately at the service level, the channel decoder can handle the new audio bit-rates without special precautions. Since both the sampling frequency and the bit-rate are signalled in the audio frame header, the audio decoder will be able to decode the MPEG audio regardless of which protection profile has been used.

4.3. Multiplex reconfigurations

The multiplex configuration of the DAB signal can be changed dynamically at source to allow the insertion of new (or the removal of old) sub-channels by changing the start position and the size of the existing sub-channels. Such a change must not cause a (temporary) drop-out of the audio or data in any sub-channel, as this would irritate the listener. Due to time interleaving of the gross data, the reconfiguration process takes 15 frames. In practice this means that if a sub-channel size is reduced, the receiver should wait 15 frames before reducing the size of its time de-interleaver memory.

If the sub-channel size is enlarged, the increase in the receiver's time de-interleaver memory should occur simultaneously. However, to prevent this increased memory from still being used for a sub-channel that has been reduced in size, the capacity of the source should be reduced 15 frames in advance of the receiver reconfiguration moment (*Table 3*). The Common Virtual Frame (CVF) count on which the reconfiguration takes place is signalled in the MCI Fast Information Group (FIG 0/0) of the FIC. Apart from a sub-channel reconfiguration, a service reconfiguration may also be indicated in this FIG. In this case, it will be processed by the system controller.

If no precautions are taken in the channel decoder, then a change of the sub-channel start position in the baseband will also result in a time shift of that sub-channel in the output signal. In principle, the start position of a sub-channel may change from the beginning of the DAB frame to the end and vice versa. This may cause an overflow or an underrun of the input buffer of the (audio) source decoder. To prevent this, a double frame buffer is needed at the output of the channel decoder.

4.4. Transmitter Identification Information

Although Transmitter Identification Information (TII) has been adopted in the ETS from the beginning, so far it has not been supported at the receiver side. The TII signalling within the DAB null symbol (*Fig. 10*), is of interest in combination with other services such as (Traffic) Announcements and other kinds of regionally-related information. TII provides the possibility at the receiver of detecting from which actual transmitter(s) in an SFN it is receiving the signal(s). Once the TII is detected, the receiver knows in which region it is located and can select from the numerous service messages only those related to the region(s) of interest. This makes it easier for a motorist to interpret the messages without being too distracted.

4.5. Local windows in an SFN

Recent experiments have shown that local broadcasting of a service in an SFN is possible by using local windows in the DAB main service channel. Each transmitter should use the same position for this local window, which should start at the edge of two successive OFDM symbols. In order to signal the MCI properly for each local service, the complete FIC may be alternated cyclically by the transmitters involved [14]. As long as the on-air signal fulfils the DAB standard, proper reception of all services is possible. The price to be paid for having different local services is the occurrence of so-called "local mush areas" in the region between the transmitters involved.

At the receiver side, the controller that handles the service selection should check the MCI continuously, to detect whether or not the service content has changed without a notification. If, due to local signalling, such a change has taken place, a user message should be generated and the service selections adapted accordingly.

5. New applications

The number of possible applications of DAB is almost unlimited. The consequence is that, currently, numerous broadcasters and research institutes are setting up different examples of possible data transmissions. Most of them are using their own specific coding techniques and dedicated interfaces. For professional applications this might be of interest; for consumer applications, however, it

Reconfi-guration	CVF count		
	Source encoder	(De)Inter-leaver	Source decoder
Increase capacity	r	r	r + 15
Decrease capacity	r – 15	r	r
Change position	r	r	r

Table 3
Re-configuration of sub-channels in a DAB multiplex (r is the CVF count as signalled in the MCI).

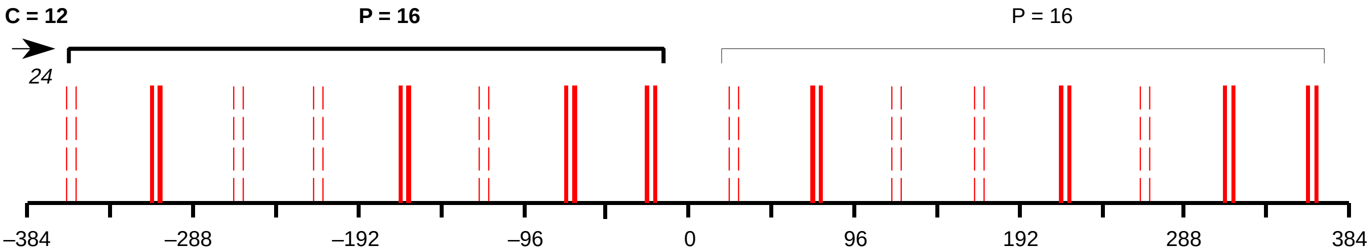


Figure 10
Example of TII (mode IV). The combination of four out of eight carrier-pairs determines the main identifier number (P). The position of the comb determines the transmitter's sub-identifier.

would be wise to search for some more commonality, in order not to confuse the already overwhelmed consumer. Some applications that might be of great interest to the consumer are described now.

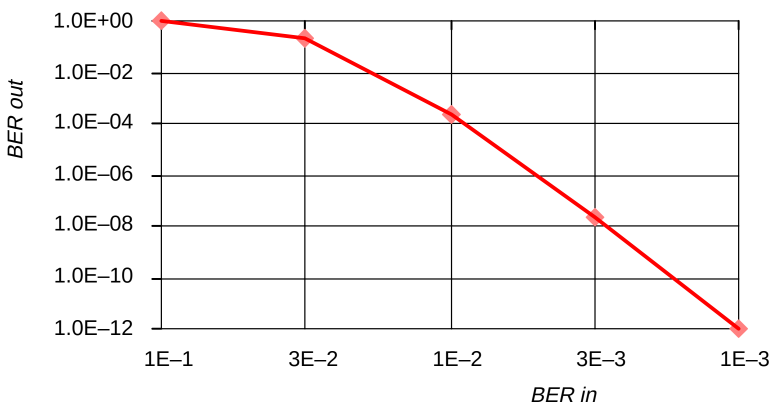
5.1. Picture radio

The attractiveness of audio services may be enhanced by adding still pictures to the service, such as CD covers or pictures of the artist(s) or the event(s). A good picture quality can be obtained with JPEG or GIF (CompuServe) image compression. Both formats are used widely on the World Wide Web (WWW). A bit-rate of about 16 kbit/s is already sufficient to update medium-size pictures within a period of 5 to 10 sec. This bit-rate can easily be carried by the audio PAD channel, in which case the synchronization at the receiver side is easily obtained. This feature requires at least a 256-colour LCD display, which probably is only of interest if it can be shared with other applications (for example a navigation system).

Table 4
Examples of possible bit-rates and protection levels for DAB video services.

R	DAB bit–rate	Blocks/ 24ms	MPEG bit-rate (184 bytes)
4/5	1843	27.1	1662 kbit/s
3/4	1728	25.4	1559 kbit/s
2/3	1536	22.6	1385 kbit/s
4/7	1316	19.4	1188 kbit/s

Figure 11
BER in versus BER out, using additional R-S protection.



5.2. Mobile video (MPEG)

Once digital audio has become available to the motorist, the next logical step is to have digital video. The Digital Video Broadcasting (DVB) system is intended to provide this service to the consumer, but it is primarily for stationary reception. The DAB system provides the necessary protection against multipath and Doppler-effects that are needed for robust mobile reception. The DAB ensemble bit-rate is high enough to carry one MPEG video service with a quality comparable to that of conventional TV.

Two problems arise when MPEG video data is transmitted via DAB. First of all the BER for Quasi Error Free (QEF) MPEG video reception should be below 10^{-10}, whereas the BER required for audio is about 10^{-5} only. Secondly, the frame duration for DAB audio is fixed (24 ms), whereas the frame duration for MPEG video is variable. Therefore it is not possible to apply the UEP profiles of MPEG audio to MPEG video. Additional error protection is required to obtain proper reception at acceptable input signal-to-noise ratios. For compatibility with DVB [15], it is recommended that an MPEG transport stream with a concatenated code scheme should be used. As a consequence, Reed-Solomon (R-S) coding should be added to the DAB convolutional coding. The common RS (204,188) decoder offers QEF video reception for an input BER as high as 2.10^{-3}, as shown in *Fig. 11*.

A block diagram of the system is given in *Fig. 12*. The use of an MPEG transport stream will allow future upgrading to MPEG-4 video coding, which will require a lower bit-rate for the same picture quality. When the full DAB ensemble is used, the configurations shown in *Table 4* are possible.

Note that the number of R-S blocks in a DAB frame is not an integer. In practice this is not a problem, since the MPEG transport stream has its own synchronization mechanism. A consequence of this is that the audio which accompanies the video is not compatible with DAB. Therefore it is not possible to extract the audio only with a DAB audio receiver, unless it is encoded separately as well. The alternative is to keep the audio separated from the video, but in that case special synchronization is needed. Also note that service signalling should be done both within the FIC channel of DAB and the TS of MPEG.

5.3. World Wide Web

Given the current interest in the World Wide Web, it is worthwhile considering the use of DAB, in conjunction with a notebook PC, to gain mobile access to the WWW. Clearly this is not a typical car system application, but much more a portable one: there is direct competition from mobile telephone applications, which already provide the bi-directional point-to-point connection that is required for the WWW. The GSM system, for example, provides a data-rate of about 13 kbit/s data, which is acceptable (except if large files need to be downloaded). In the case of DAB transmission, either a selection of Websites would need to be made on the broadcaster’s side or a return channel would need to be provided at the receiver side. Most probably there should be Conditional Access (CA) to charge for the service. By the time CA is widely provided, the mobile telephone network might already have won the race, at least if there is serious interest in this type of portable WWW access.

Within DAB, a Media Object Transfer (MOT) protocol has been proposed for the transmission of multimedia services. This could be used for access to the WWW. The MOT protocol allows the transfer of data services either in a packet-mode-formatted sub-channel, or in the PAD bytes of the audio frame. The PAD method is likely to be preferred by the service provider whereas the packet mode will be preferred by the DAB network provider. At the receiver side, PAD is easily accessible but, for packet data, a special packet mode decoder is required.

The MOT data consists of a header, which contains information such as a transport identifier, repetition rate, size and type, and a data field. Currently-defined data types are ASCII and HTML text, GIF, JPEG and BMP pictures, MPEG audio and video, H263 video and also ZIP, JAVA and MHEG files. In principle these allow the transfer of hypertext applications containing a mixture of text, graphics and audio. Additional data error protection will be needed for these applications, unless a very high protection level is used. This is not provided for within the MOT protocol and thus should be defined separately for each data type. Field and user tests should be carried out in order to confirm the reliability and usefulness of these applications.

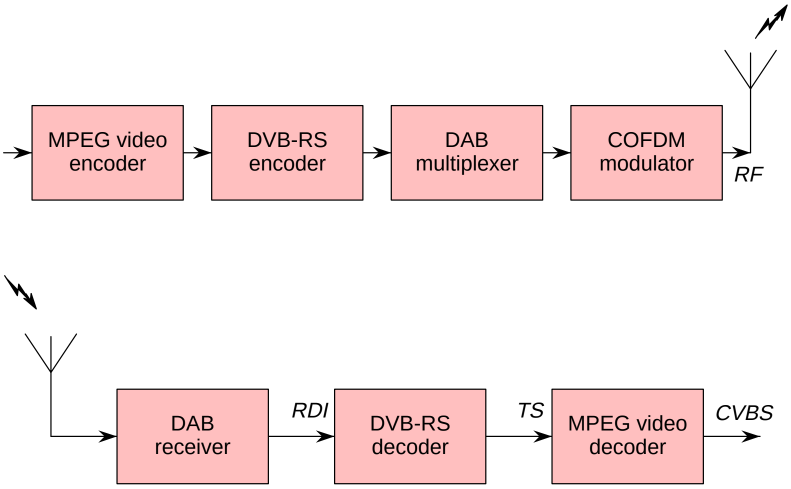


Figure 12
Basic components needed for DAB video services.

5.4. Multimedia applications (MHEG)

MHEG (Multi- and Hyper-media coding Experts group) is a standardized language for the description of interactive multimedia applications. It is currently applied to multimedia presentations and as a kind of multimedia successor to teletext in Digital Video Broadcasting. Since MHEG is already supported in these applications, it can be expected that a large number of such broadcast services will become available. As for the WWW services, a protocol is needed for embedding the MHEG applications into a DVB or DAB data-stream. For DVB, the Digital Storage Media Command Control (DSM–CC) protocol is used. For DAB this protocol can be considered as an alternative to the MOT protocol described in the previous paragraph. Additional error protection

Figure 13
Preview of the SAA3500 DAB channel-decoder ASIC.

may be obtained by employing Reed-Solomon coding as described in *Section 5.2*.

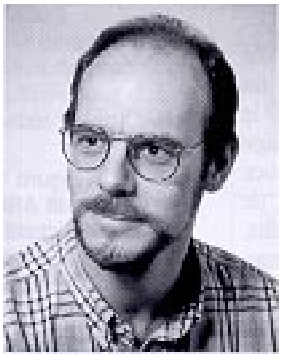
6. Conclusions

New DAB channel-decoder ICs will become available in 1997 (*Fig. 13*), enabling the high-volume manufacture of DAB receivers for the consumer market. The current level of integration allows one to consider the channel decoder as a black box which performs the full COFDM decoding functions. From now on, attention can be turned away from the technology towards the DAB services that could be implemented. However a further level of integration is possible which will allow the channel-decoder functions to be added to the IC (e.g. A/D conversion, interleaving memory, sub-channels selections and a packet decoder for data services). Together with current developments in DAB tuners and source decoding, it is now possible to combine DAB, an AM/FM radio and a cassette player – all within a standard-size (DIN) car radio case.

The main challenge for DAB still seems to be the identification of useful data services which would offer real added value over the existing FM (+RDS) radio services to the future radio listener. Closer co-operation between all the participants is needed, in order to obtain a more unified approach (e.g. to network planning). This is needed to implement low-cost receivers on the one hand and to avoid confusing the already overwhelmed consumer on the other hand. The introduction of features such as dynamic labels and regional service selections (based on TII) should seriously be considered, since they can already be realized with the current decoder hardware.

For data, video and multimedia services, it is desirable to have a common agreement on Reed-Solomon coding as an extension to the convolutional coding already specified for DAB audio.

Given the progress that has been made with the introduction of DAB transmissions and the development of consumer ICs for the receiver, it seems



Mr Frank van de Laar *received a computer engineering degree from Breda, the Netherlands, in 1984. He then joined the data transmission group of the Philips Research Laboratory in Eindhoven, where he worked on adaptive filters for acoustic echo cancellation.*

Since 1990, Mr van de Laar has been working at the Philips Consumer Electronics advanced development centre on Digital Audio Broadcasting. He has been involved in DAB standardization, on coding and modulation, and the prototype design of DAB channel decoders. Recently he has been involved in the specifications and validations of the SAA3500 channel-decoder ASIC.

Mr Norbert Philips *graduated from the Katholieke Universiteit in Leuven as an Electronics Engineer in 1976. He joined the Philips International Technology Centre in Leuven, Belgium, (ITCL) in 1983. Since 1989, he has been working on DAB as a systems engineer.*

Mr Philips has been involved in the realization of DAB test receivers, with a special focus on synchronization algorithms, and in the specification of channel-decoder ASICs. He is currently working on digital RF links for indoor applications.

Mr Jos Huisken *received his Masters degree in Electrical Engineering from the University of Twente, the Netherlands, in 1984. In that year he joined the Philips Research Laboratories where he was involved in the design of ICs for digital signal processing, including the architectural synthesis, logic synthesis and CAD aspects of the layout. Since 1991, the application of synthesis techniques has led to the design of DAB channel-decoder ASICs, resulting in the Philips SAA3500.*

Mr Huisken's other major areas of interest include low–power architectural synthesis and the integration of test and debug capabilities on silicon chips.

justified to expect that the Eureka-147 DAB system will truly be the radio system of the next era!

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World-wide convergence on HDTV production standards

The arguments about a common HDTV standard for television programme production have raged for over twelve years. Everyone agrees on the need for a common standard but the precise parameter values to choose have remained elusive – until recently. Now, the ITU-R has made major progress in breaking the deadlock. In April 1997, Working Party 11A agreed a draft new version of the relevant Recommendation (BT.709-2) which was subsequently ratified by its parent group, ITU-R Study Group 11.

The new Recommendation acknowledges the existence of the two existing HDTV systems – 1035 active lines/60Hz and 1152 active lines/50Hz. However, for new installations, it states a preference for the HD-CIF (High Definition – Common Image Format) system. This system has a new set of parameter values which use 1080 active lines, whether the field rate is 50Hz or 60Hz.

It would be unreasonable to say that equipment working to the old standards will become “non-standard” overnight. The intention is that there should be a clear and definite migration to the new common system over the next few years. This should help to lower HDTV equipment costs and accelerate the introduction of HDTV throughout the world.

The same sampling frequencies for interlace (74.25 MHz) and progressive (148.5 MHz) scanning are used in the 50 Hz and 60 Hz versions of the format. The hope is that single equipment sets will be made available which are switchable between the two field rates.

The draft Recommendation will be circulated to national Administrations for approval during the course of 1997. The HD-CIF format is supported by the Inter Union Technical Committee of the World’s Broadcasting Unions.

This could be the end of the beginning for the common HDTV production standard.

David Wood