The evolution of DAB

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DAB – already covering 500 million people in 40 countries around the world – represents the fully mobile and narrowband (1.7 MHz) terrestrial branch of COFDM broadcasting technologies.

Although the family of DAB standards has been growing continuously from its beginnings in the early 90s, several major milestones have been reached by the WorldDAB / WorldDMB Forum, especially within the last three years. The most prominent examples are certainly DMB and DAB$^+$. Those two and further applications, as well as the necessary framework created, are illustrated in this article.

The technical perspective is accompanied by an economic one, visualising the growth underway and the promising prospects that lie ahead, based on the substantially extended DAB toolkit.

1. Introduction

1.1. Flexibility and Reliability – keys to the success of Eureka-147

In the digital age, broadcasting technical standards need to balance the benefits of *stability* and *innovation*. Stability gives confidence to broadcasters, manufacturers and consumers. And yet, enhancing a standard to take advantage of technological innovation can offer new benefits and protect a standard's competitiveness in a rapidly changing market place.

The international organization responsible for the Eureka-147 standards, WorldDMB (formerly known as the WorldDAB Forum) has carefully monitored developments in audio and multimedia broadcasting over the last decade and has kept up to date with state-of-the-art coding and transport systems. Although WorldDMB remains a strong advocate of stability, innovation in the interest of efficiency and diversity is an important issue in today's highly competitive market, and the Eureka-147 family of standards has easily managed to keep on top of the ever-increasing speed of developments in the digital world.

Several challenges have been met over the years and overcome by the addition of new features to the Eureka-147 family, each time increasing its flexibility even further whilst ensuring a continuing robustness and reliability.

When the original **DAB** (Digital Audio Broadcasting) system was first developed in the late 1980s, it was based on MPEG Audio Layer II coding, which was then state-of-the-art and is still a commonly used coding technology in digital radio broadcasting. Since then, MPEG Audio Layer III, better known as mp3 has conquered the market of digital music players and radio streams. Even though still the most successful technology on the market, mp3 has already been overtaken in efficiency and performance by MPEG-4 AAC (Advanced Audio Coding). This development has called for an additional audio coding system in DAB which would allow for more efficiency at lower bitrates – hence the birth of **DAB$^+$**.

Another important innovation has been the addition of video/multimedia capabilities to DAB, allowing it to become a digital mobile television platform – called **DMB** (Digital Multimedia Broadcasting) – as well as a digital radio platform.

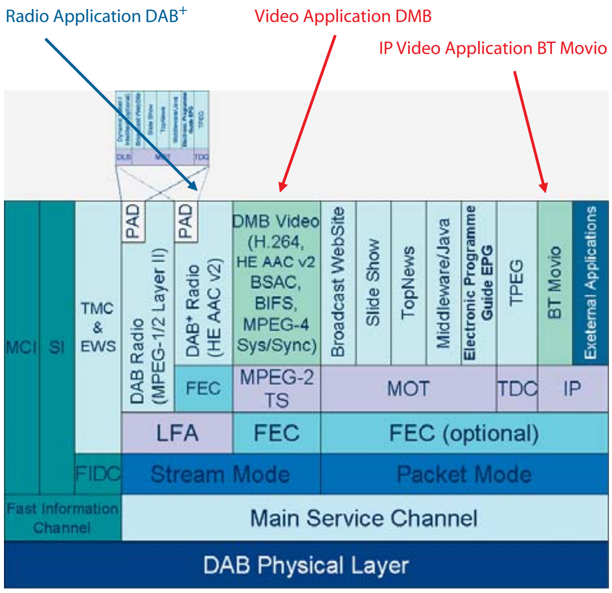

Figure 1
DAB System Protocol Stack

Both for DMB and DAB$^+$, the technical basis remains with DAB. In other words, the physical layer is still the same ... just new applications, new transport protocols and a second error-control coding layer have been added *(Fig. 1)*.

New challenges will continue to be addressed by the WorldDMB Technical Committee, which will ensure DAB remains a very attractive, flexible and market-ready standard for digital audio, mobile and multimedia broadcasting.

One of the strengths of the Eureka-147 DAB standards is that not only different applications can co-exist within the same multiplex, but also different transport protocols and individual convolutional code rates for each sub-channel respectively.

1.2. Tested, trialled and rolled out all over the world

Thanks to this flexibility and robustness, Eureka-147 standards have managed to conquer more than 40 countries all over the world *(Fig. 2)*. DAB digital radio, for example, has been tested, trialled and rolled out in most European countries, among them significant markets such as the UK, Germany, Spain, Italy, France, Denmark, Norway and Switzerland. Digital Audio Broadcasting has travelled further overseas to numerous countries in Asia, Africa and America, and has also arrived in Australia and New Zealand.

Mobile television using DMB, on the other hand, was successfully introduced in Korea in November 2005 and has since become the biggest mobile television market in the world. Only 18 months after its launch, the DMB receiver market in Korea passed the 4-million mark at the end of March 2007 *(Fig. 3)*.

Figure 2
The world of Eureka-147

In Germany (DMB) and the UK (DAB-IP), mobile TV is currently in its initial stages of roll-out, and there have been numerous tests and trials on DMB in other countries, including France, China, Norway, Denmark, India, Germany and the UK.

Also, the additional audio codec used by DAB$^+$, even though it has only just been issued as a standard, has already raised much interest, especially in markets where digital radio is about to be first rolled out. The two most prominent DAB⁺ bidders are Australia, where the introduction is planned to start in 2008/9 in the 11 most populated cities, and Malta where licences for digital radio using DAB⁺ have recently been acquired by DigiB, a network operator. Many other countries are planning trials and tests for DAB⁺, among them Switzerland, Italy, Luxembourg, Belgium, France, India and South Africa.

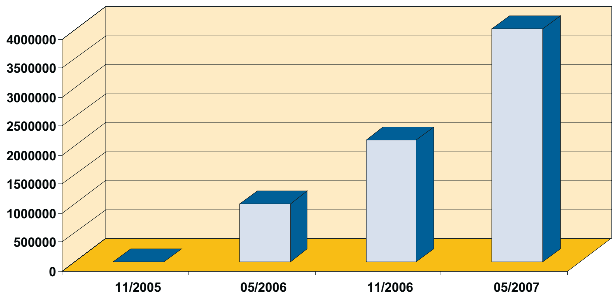

Figure 3
Sales of DMB receivers in Korea

1.3. DAB/DMB receiver market – on a steady rise

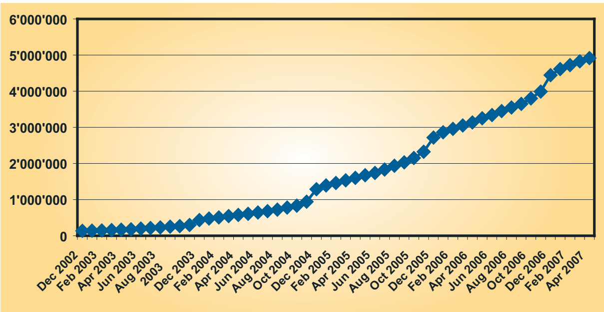

Figure 4
DAB receiver sales in the UK (accumulative)

It is not surprising, therefore, that the DAB/DMB receiver market has developed rapidly over the last five years. Apart from the four million DMB receivers that have been sold in Korea since the commercial launch of T-DMB in November 2005 *(Fig. 3)*, there are also more than five million DAB radios now in European households, most of them in the UK *(Fig. 4)*.

DAB/DMB receivers are available in all price ranges, and the latest figures *(Fig. 5)* show that the choice of different receiver models is simply staggering. Over 250 manufacturers offer a total of almost 900 different receiver models, and the end of this growth is not yet in sight – the offer is now almost twice as large as it was just over a year ago.

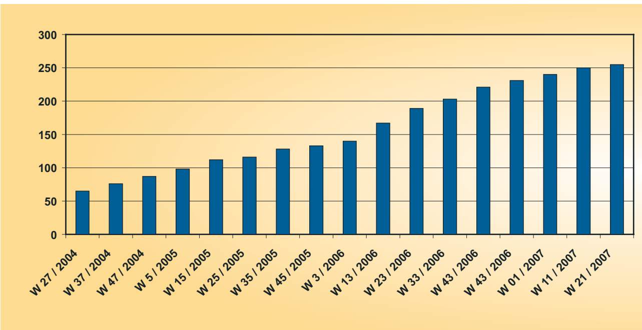

Figure 5a
Manufacturers of DAB/DMB receivers
(Source: www.dab-digitalradio.ch, May 2007)

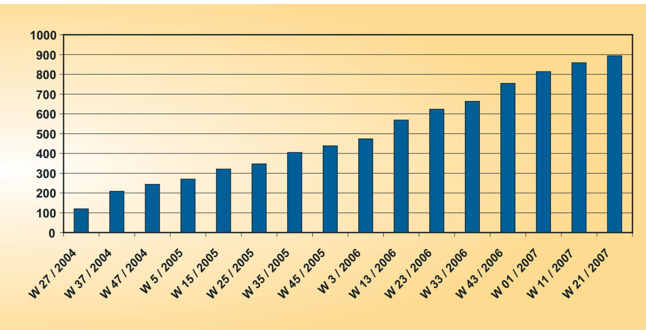

Figure 5b
Number of DAB/DMB receiver models
(Source: www.dab-digitalradio.ch, May 2007)

2. Audio: the original DAB and the additional DAB⁺

2.1. DAB

During the development of DAB in the early nineties, MPEG-1 (sampling rate = 48 kHz) and MPEG-2 (sampling rate = 24 kHz) Layer II audio coding were selected as the most appropriate algorithms at that time. MP3 (i.e. Layer III) was refused, because a better performance could not be verified at that time, but higher processing power was required. Layer II was very robust against the errors imposed by the broadcast channel and was protected well enough by the convolutional channel coding and the time interleaving introduced as part of the physical layer of the OFDM-based broadcasting system DAB.

2.2. DAB+

After many years of repeated discussions and consideration, the point of time was reached in 2005 for making use of the remarkable margin that audio compression developments had led to within a decade. Pressure in this direction was generated especially by markets about to start with the roll-out of DAB. Naturally they were not ready to ignore the developments sketched above. In addition, existing DAB markets were also looking for expansion through efficiency enhancements. Another element of the arising pressure was the fact that some providers were considering or had already started to make use of the application DMB for carrying just audio services.

For classical radio, DMB with its state-of-the art audio codecs – HE-AAC v2 and MPEG-4 ER BSAC – looks attractive on first view, but audio services can be realised in a much more efficient and smarter way.

So the WorldDAB Forum decided in June 2005 to start the development of an alternative audio system for DAB – the Technical Committee set up the Task Force, *New Audio System*. The result of 1.5 years of enthusiastic work – the norm *"Transport of Advanced Audio Coding (AAC) audio"* – was published by ETSI in February 2007 and was announced publicly as DAB+ at the same time.

The significantly increased efficiency, which is discussed in more detail later, offers benefits for Governments and Regulators (even better spectrum efficiency), broadcasters (lower costs per radio station) and consumers (a bigger choice of stations). It is designed to provide the same functionality as the current MPEG Audio Layer II radio services.

In some countries where DAB digital radio has already been launched, broadcasters are committed to continuing to use MPEG Audio Layer II. However, in countries planning to launch digital radio, the arguments in favour of launching DAB+ are compelling.

It is worth noting that this is not the first time HE-AAC v2 has been included in the Eureka-147 family of standards. Already, the DMB standard allows for HE-AAC v2 audio as part of the video services. However, DMB – designed for mobile television – naturally lacks some of the functionality required for pure radio services.

Other broadcast technologies such as DVB-H (digital video broadcasting to handheld devices), DRM (Digital Radio Mondiale; i.e. digital long-, medium- and short-wave) or Qualcomm's MediaFLO technology also use HE-AAC v2 audio coding and are able to carry multiple audio services in the digital capacity needed for a single radio station using MPEG Audio Layer II.

2.2.1. Technical overview of DAB+

The corresponding Call for Technologies resulted in just one family of audio codecs desired by the group of applicants – AAC. Since AAC is build up as a hierarchical system *(see Fig. 6 and the text box on Page 5)*, it was self-explaining to decide in favour of the most recent development – HE-AAC v2. It still enables the application of, for example, just the core codec for high fidelity radio at the higher bitrates. Providers have the choice of using just the core, the core plus SBR ... or the core plus SBR plus PS. Of course, the receivers must be prepared for all cases and

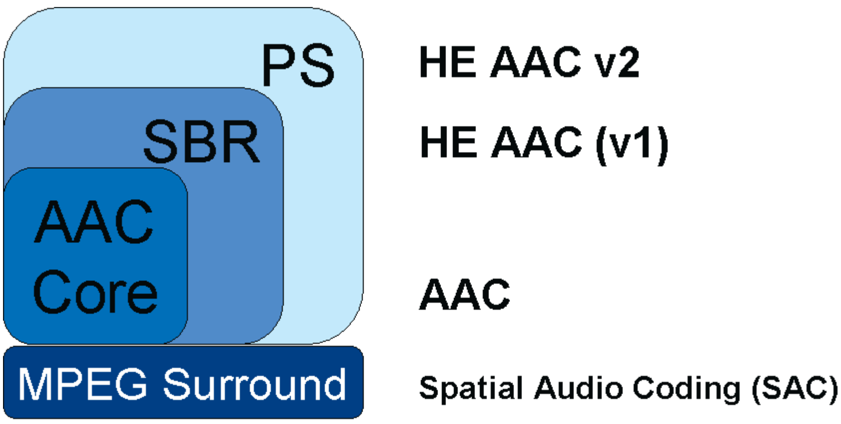

Figure 6
Modular set-up of the AAC family

hence the implementation of HE-AAC v2 is mandatory.

In light of the fact that audio coded with MPEG Layer II will remain on-air for many years to come, a new DAB receiver needs to cover both coding algorithms – MPEG-1/2 Layer II and HE-AAC v2.

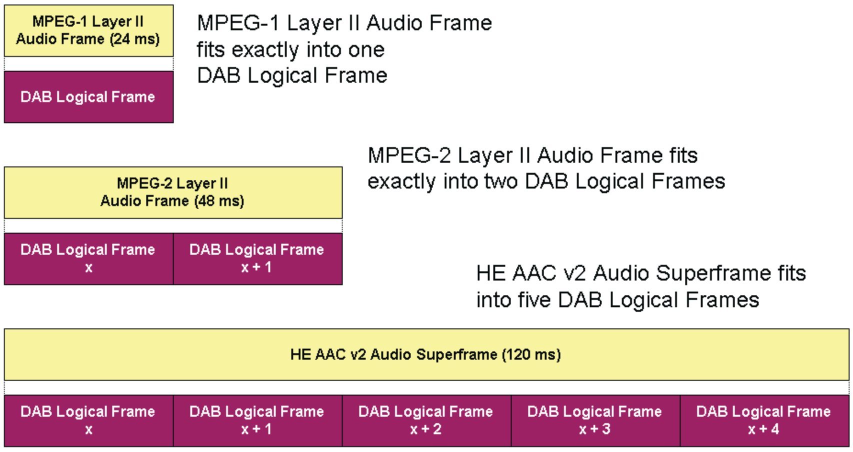

Figure 7
DAB Logical Frame Alignment for Layer II and DAB+

DAB was originally designed around MPEG-1 layer II structures – best reflected by the fact that the DAB logical frames were of identical length in time (24 ms) as the MPEG-1 layer II audio frames. The step to MPEG-2 Layer II, with half the sampling rate, was simple – one audio frame per two logical frames. And DAB+ uses the common denominator of all permitted lengths of AAC Access Units (of length 20, 30, 40 or 60 ms) with a 120 ms long superframe equivalent to five logical frames *(Fig. 7)*. It should be noted here that this quite short length, and hence the quick zapping from one service to another, was realised through the adoption of the AAC variant with 960 samples per Access Unit (as used for Digital Radio Mondiale).

Due to the high efficiency of the new coding algorithms, the impact of lost bits is more significant. In other words, better protection is needed. Already introduced for DMB – more precisely for DAB Enhanced Stream and Packet Mode – the concatenation of the inner convolutional coding (Viterbi), being an element of the original DAB set-up, and an outer block code in the form of Reed-Solomon (R-S) coding was chosen as the most appropriate solution.

HE-AAC v2

DAB+ uses MPEG-4 High Efficiency AAC v2 profile (HE-AAC v2). This audio codec is the most efficient audio compression scheme available worldwide. It combines three technologies:

- The core audio codec **AAC** (Advanced Audio Coding).
- A bandwidth extension tool **SBR** (Spectral Band Replication), which enhances efficiency by using most of the available bitrate for the lower frequencies (low band) of the audio signal. The decoder generates the higher frequencies (high band) by analysing the low band and side information provided by the encoder. This side information needs considerably less bitrate than would be required to encode the high band with the core audio codec.
- **Parametric Stereo** (PS): a mono down-mix and side information is encoded as opposed to a conventional stereo signal. The decoder reconstructs the stereo signal from the mono signal using the side information.

HE-AAC v2 is a superset of the AAC core codec. This superset structure permits the use of (i) plain AAC for high bitrates, (ii) AAC and SBR (HE-AAC) for medium bitrates or (iii) AAC, SBR and PS (HE-AAC v2) for low bitrates. Therefore HE-AAC v2 provides the highest level of flexibility for the broadcaster. A detailed description of HE-AAC v2 is available on the EBU website [1]. An introduction to MPEG-4 is available on the MPEG Industry Forum website[2].

HE-AAC v2 provides the same perceived audio quality at about one third of the subchannel bitrate needed by MPEG Audio Layer II. The same audio coding is also used in DRM and DMB e.g. for television audio. Devices, which also include DMB or DRM can benefit from the fact that the audio coding for this range of technologies is essentially the same.

1. EBU Technical Review: **MPEG-4 HE-AAC v2 – audio coding for today's digital media world** (2006)
   http://www.ebu.ch/en/technical/trev/trev_305-moser.pdf
2. An MPEGIF White Paper: **Understanding MPEG-4: Technologies, Advantages, and Markets**
   http://www.m4if.org/public/documents/vault/MPEG4WhitePaperV2a.zip

The structure applied *(Fig. 8)* consists of super-frames covering a fixed number of AAC access units. Each Access Unit (AU) carries its PAD (Programme Associated Data) part in a similar way as for MPEG Layer II audio frames. The required additional error protection is realised with virtual interleaving and an R-S scheme (120, 110, t=5) derived from the same mother code as the R-S schemes for Enhanced Stream and Packet Mode. The ten parity bytes per 110 data bytes – equivalent to an over-head of 8.3% – lead to an ability of correcting up to five erroneous bytes in those 120 bytes *(Fig. 9)*.

For test purposes, the new algorithms have already been implemented in both transmitting and receiving equipment. Therefore the step towards mass production is a small one for those who have already invested effort and resources in the standardization exercise.

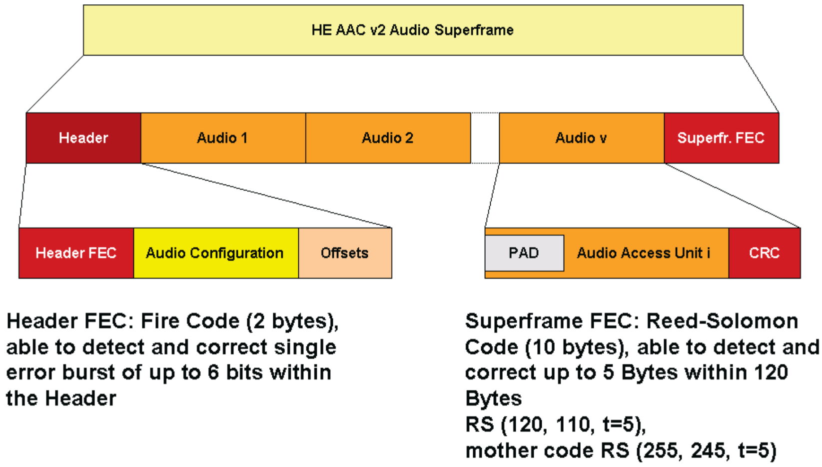

Figure 8
Superframe structure used for transport of HE AAC v2 audio in DAB

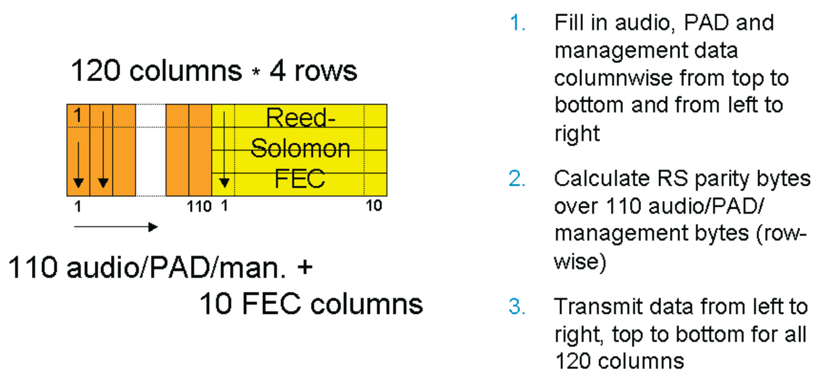

Figure 9
Error-control code calculation and virtual interleaving in a 32 kbit/s sub-channel

Features of DAB+

All the functionality available for MPEG Audio Layer II services is also available for DAB+:

- service following (e.g. to FM or other DAB ensembles);
- traffic announcements;
- PAD multimedia (Dynamic Labels such as title, artist information or news headlines, still images such as weather charts, and other multimedia content);
- service language and programme type information (e.g. Classical Music, Rock Music, Sport);
- etc..

The multimedia information carried in the PAD of an HE-AAC v2 radio service is as well protected against data losses as the audio itself, both enjoying the cascaded error-control coding. In order to ensure that the PAD data of a radio service using MPEG Audio Layer II also takes advantage of the new developments, a backwards-compatible and optional FEC layer will be added here as well.

An important design criterion for DAB+ was a short “zapping” delay. Both the time it takes to switch from one radio station to another station on the same DAB ensemble, as well as the time it takes to tune to a radio station on another DAB ensemble, was minimized.

Currently all DAB radio stations are mono or stereo. However, DAB+ also provides the means to broadcast surround sound in a backwards compatible way. Using MPEG Surround it is possible to broadcast a stereo signal together with surround side information (e.g. 5 kbit/s side information). Standard stereo radios will ignore this side information and decode just the stereo signal. MPEG Surround receivers will evaluate the side information and reproduce surround sound. So at a comparatively low additional bitrate, the broadcaster can increase the audio experience on surround sound receivers, and still provide high quality sound to all other radios.

2.2.2. Performance of DAB+

During the standardization process, simulations were undertaken by the Communications Research Centre, Canada. With typical reference channel models for DAB environments, a gain of 1.7 to 6.7 dB for the new system, compared to the existing Layer II system, was determined.

Typical Urban (Cost 207) | Rural Area (Cost 207)

System / Speed	4.2 km/h	84 km/h	251 km/h
Layer II, 192 kbit/s	15.7	12.2	16.7
HE AAC v2, 40 kbit/s	12.5	9.0	10.0
HE AAC v2, 96 kbit/s	14.0	9.5	10.2
Layer II, 192 kbit/s	19.2	13.2	15.7
HE AAC v2, 40 kbit/s	17.0	10.0	10.9
HE AAC v2, 96 kbit/s	17.0	10.0	11.0

Source: Communications Research Centre Canada

Figure 10
C/N for threshold of audibility

Field tests conducted in the UK and Australia confirmed the results of the simulations. They showed that the geographical coverage area of radio services using HE-AAC v2 is slightly larger than that for radio services using MPEG Audio Layer II.

Audio services using HE-AAC v2 performed about 2 - 3 dB better at the threshold of audibility. This means that in some areas close to the coverage area limit, where MPEG Audio Layer II services already showed audible artefacts, HE-AAC v2 radio services showed no audible artefacts.

The error behaviour of MPEG Audio Layer II is different to that of HE-AAC v2. With MPEG Audio Layer II, the weaker the DAB signal gets, the more audible artefacts can be heard.

HE-AAC v2 produces no audible artefacts, but when the signal gets too weak, an increased number of audio frames will be lost and this causes short periods of silence (fade-out and fade-in). Test listeners preferred this error behaviour. Compared to radio services using MPEG Audio Layer II, radio services using HE-AAC v2 will fail later (they can cope with a slightly lower DAB signal quality), but the margin from error-free reception to loss of reception is smaller.

2.2.3. Implementation scenarios

Thanks to the flexible structure of the DAB system, radio services encoded with MPEG Layer II can co-exist with radio services encoded with HE-AAC v2. Examples of multiplex implementations are given in *Fig. 11*.

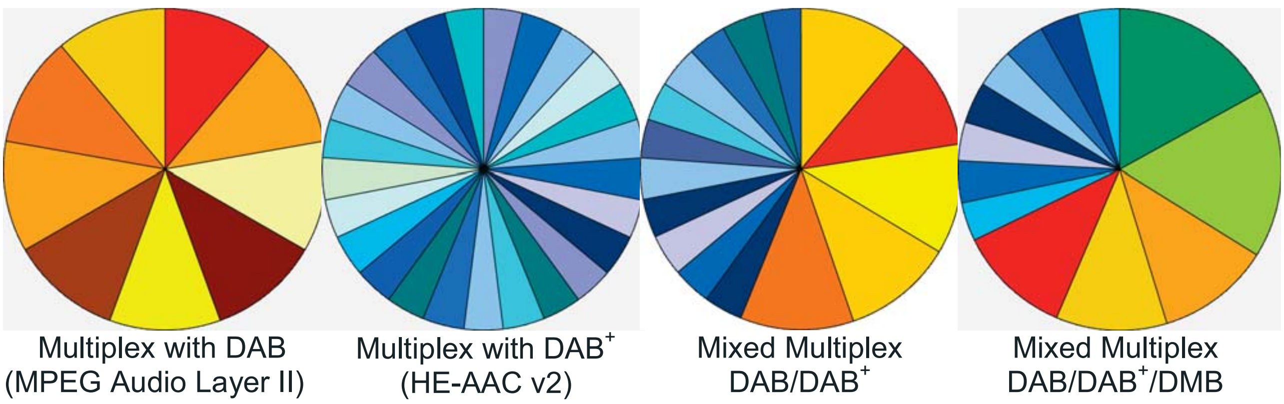

Figure 11
Various implementations of the DAB system

1) *First on the left:* this is the classical set-up with, say, nine MPEG Layer II encoded radio services.
2) *Second from the left:* in contrast to the classical set-up, a progressive constellation is shown here. It does not allow for legacy receivers not understanding the new coding algorithm and no fewer than 28 DAB+ radio services can find space in such a multiplex arrangement.
3) *Second from the right:* this is a migration scenario that moves slowly from Layer II to AAC. It is shown still providing five Layer II services, but already bringing 11 AAC-coded services on air.
4) *First on the right:* this is another way of benefitting from the saved Ensemble capacity. With three Layer II services and eight AAC services, there is still enough capacity left for two mobile TV services using DMB.

3. Mobile TV: DMB and BT Movio

In early 2007, there were two different variants for providing video applications via DAB – DMB and DAB IP Tunnelling. An example for the latter path is known in the UK as BT Movio [1] and is provided as a wholesale service by the incumbent telecom operator. This application puts a source coding algorithm – that is not specified with an open standard – on top of IP. In fact, the whole application is a proprietary one.

3.1. Digital Multimedia Broadcasting (DMB)

DMB uses H.264/MPEG-4 AVC (Advanced Video Coding), HE-AAC v2 or BSAC (Bit-Sliced Arithmetic Coding) and BIFS (Binary Format for Scenes), respectively, as the encoders for video, audio and content-related data services. All of these encoded Elementary Streams are multiplexed into MPEG-2 Transport Stream (TS) packets.

To increase the necessary robustness – especially for mobile reception – an additional block coding scheme (Reed-Solomon Coding) and convolutional interleaving is applied to the MPEG-2 Transport Stream – in line with DVB structures. The byte-interleaved and error-protected TS packets are transmitted through the Eureka-147 stream mode.

T-DMB obtained official approval as a European ETSI standard in July 2005.

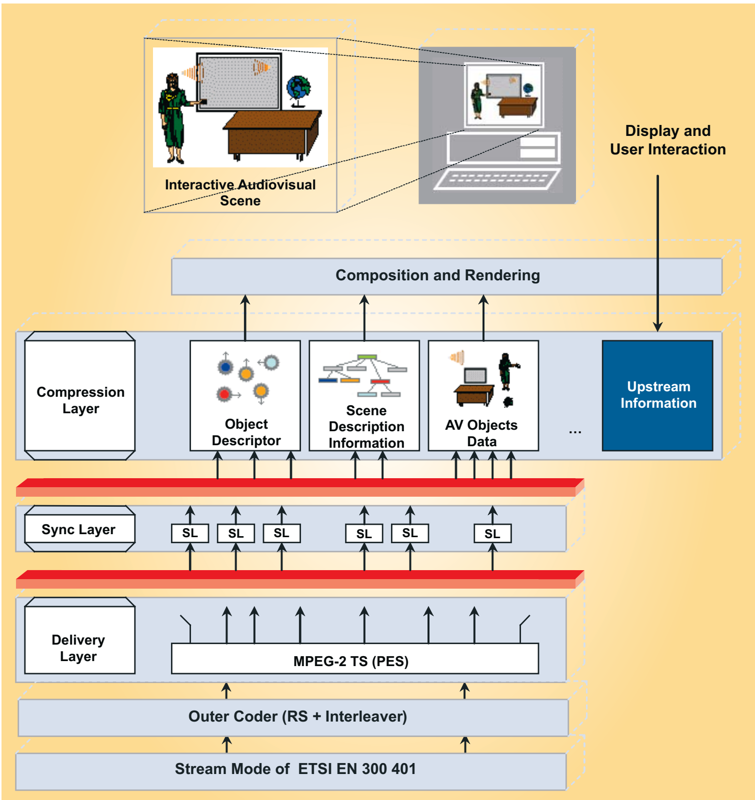

Figure 12
Terminal processing chain for the DAB application, DMB

1. E. Lloyd, R. Maclean and A. Sterling: **Mobile TV — results from the BT Movio DAB-IP pilot in London** EBU Technical Review No. 306, April 2006
http://www.ebu.ch/en/technical/trev/trev_306-movio.pdf

Extraction, error-control decoding, stripping of Elementary Streams and synchronization – both temporally and spatially – as well as source decoding and reproduction are shown in *Fig. 12* for the terminal side.

Altogether, this chain represents a classical combination of MPEG-4 elements transported by an MPEG-2 Transport Stream. BIFS, as one of those MPEG elements/norms, represents quite a powerful tool for data provision and interactivity.

3.2. BT Movio

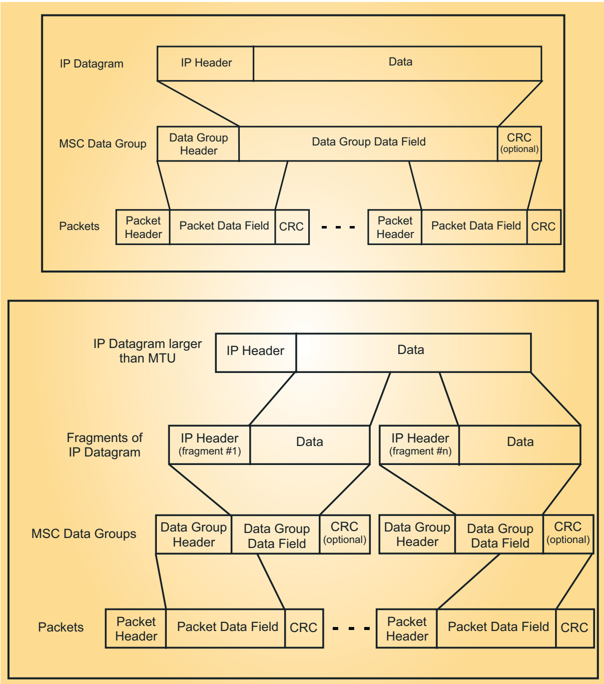

Figure 13
DAB IP Tunnelling

Unlike DMB, BT Movio is not fully standardized by WorldDMB / ETSI, but makes use of a hook that was designed exactly for that purpose – DAB IP Tunnelling. Based on this transport system for IP datagrams via DAB, the provider has applied protocols and source coding algorithms designed by Microsoft. It should be noted that, in the mean-time, all of these specifications (ASF and VC-1) are in the public domain apart from one – WMA.

As with most DAB data formats, IP Tunnelling is based on (Enhanced) Packet Mode – *see Figs 1 and 20*. The encapsulation of the IP datagrams in DAB MSC Data Groups (DGs) – either unfrag-mented or fragmented – is shown in *Fig. 13*.

For the unfragmented case, the size of a single MSC Data Group data field, carrying always exactly one IP datagram, lies in the range 576 to 8191 bytes. This is given by the minimum size of an IP Datagram according to RFC 791 and the largest MSC Data Group size according to ETSI standard EN 300 401.

Figure 14
HTC/Qtek *Lobster*, as used with the BT Movio DAB-IP service

For the fragmented case, the MSC DGs might be even smaller. Mapping of the Data Groups onto Packets is done as usual, using large packet sizes as far as possible and limiting the padding – both measures for reducing the overhead.

Especially for streaming services such as BT Movio, it is absolutely vital to employ the second layer of error-control coding.

On top of IP, BT Movio uses UDP and ASF. Source material is encoded with Windows Media Audio and Video codecs (the latter one is equivalent to VC-1). BT Movio services are all digital-rights managed according to a Microsoft DRM specification. They are enhanced with the DAB Electronic Programme Guide.

The BT Movio device marketed in the UK is the HTC/Qtek *Lobster (Fig. 14)*. Of course, this device is also equipped for the reception and reproduction of DAB radio services.

4. Further applications

4.1. *Intellitext*

Intellitext is the youngest "offspring" of the Eureka-147 family of standards. It extends the well-known Dynamic Label in a backwards-compatible and structured way and allows for the provision of text elements, enabling a hierarchy of detail. Intellitext is transported in PAD exclusively.

The data is compiled into a simple Tele/Videotext-like database of information which the user of any DAB radio, equipped with this application, can browse on demand. Intellitext messages are a special form of Dynamic Label messages, formatted in such a way that receivers not supporting Intellitext will continue to function normally.

Since not all transmitted DL messages will be Intellitext messages, Intellitext-capable receivers need to determine whether a received DLS message is an Intellitext message or not, in order to process the received message appropriately. Intellitext messages are parsed and stored. The stored messages are updated and deleted to ensure that the data is appropriately maintained.

The Intellitext system provides a means for broadcasters to control the lifetime and basic formatting of broadcast information, while the display of information is user-driven.

```
talkSPORT
    Football
          Prem Lge Table
                1. Chelsea 27 pts
                2. Spurs 18 pts
                3. Charlton 16 pts
                4. Man Utd 14 pts
                5. Man City 14 pts
          Results
                Arsenal 0, Wigan 3
                Spurs 2, Man Utd 1
                West Ham 2, Sunderland 3
```

Figure 15
Example of display of Intellitext message data

The Intellitext system allows the broadcaster to dictate the structure and design of menus, including menu naming. The information provided by each service provider is stored in such a way that it cannot be altered by any other service provider.

Navigation is usually via a simple up/down/select interface, with the actual display being tailored to the resources available to a given receiver.

Intellitext messages consists of a category, a sub-category and some data. Within a given category, the sub-categories may be ordered by using a numerical index; similarly the data items are ordered within the sub-category. An example of the type of user display is shown in *Fig. 15*.

Figure 16
Receiver with Intellitext Capability:
PURE digital – "One"

Once again, the UK is the first market introducing the new technology: it will be possible for a number of existing receivers *(Fig. 16)* to be updated for reproducing this more attractive, but still simple to use, text application.

4.2. *EPG*

Who could imagine television today in the absence of an EPG? And for radio it's even more important. The DAB Electronic Programme Guide – also suitable for Digital Radio Mondiale – is available in two variants, binary and XML. It provides an overview of the Programme Items currently on-air and the ones that will be on-air within a given time period, e.g. over the next 24 hours. It is also applicable to Mobile TV services. The EPG might cover the tuned Service, several or all Services on the tuned Ensemble or it can even include Services being broadcast on other Ensembles.

With this technical tool at hand, the next step – pre-programmed Service selection and/or recording (nowadays on SD Cards) – is the logical way towards a state-of-the-art receiver.

EPGs are transported with the Multimedia Object Transfer (MOT) protocol and might be compressed for broadcast efficiency purposes – see *Fig. 1* above.

Fig. 17 illustrates the hierarchies of the different sorts of information that can be provided – Service Information, Schedule Information and Group Information.

On the provider side, EPG is already in use to a wide extent and its coverage is getting larger continuously.

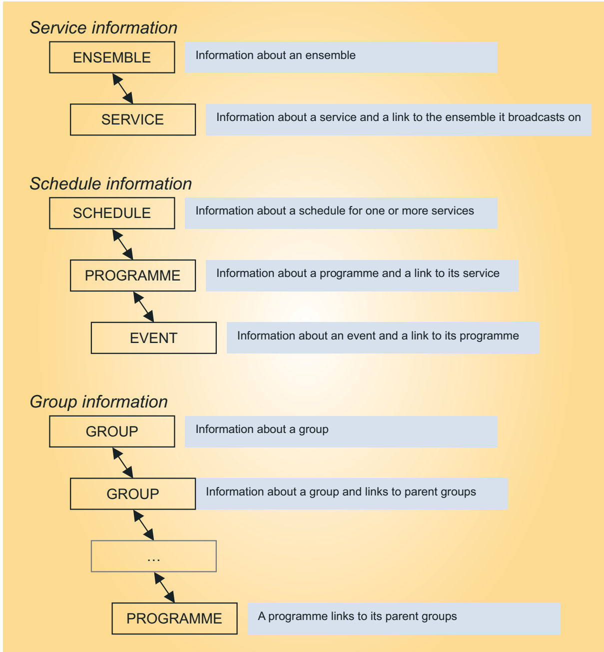

Figure 17
EPG entities and hierarchies

Figure 18
EPG-enabled DAB Receivers

What are the advantages for the providers?

- EPG enhances the listening experience, and is a marketing tool for Digital Radio stations:
- EPG enables additional media spend, revenue stream, opportunities;
- EPG content can be generated without requiring dedicated production teams.

Fig. 18 shows some consumer receivers that feature the application EPG, e.g. in the form of displaying programme schedules for the next seven days. Timed recording is enabled as well – see the user interface at the top of the figure.

5. Second error-control coding layer

The more efficient state-of-the-art source-coding algorithms are naturally more sensitive to transmission errors. Here the original algorithms were significantly more robust and error-tolerant. H.264 video coding requires an average BER as low as 10^{-8} at the input to the decoder *(Fig. 19)*. In contrast, DAB was originally designed for a BER of 10^{-4} at the input to the MPEG Layer II coder.

After a thorough simulation and field-test project, this issue was solved in the end through the application of a second error-control coding layer – resulting in a cascaded coding arrangement with convolutional Viterbi coding as the inner coding and Reed-Solomon block codes as the outer coding – as in DVB. Here, with an overhead of 7.8%, a really dramatic improvement could be reached.

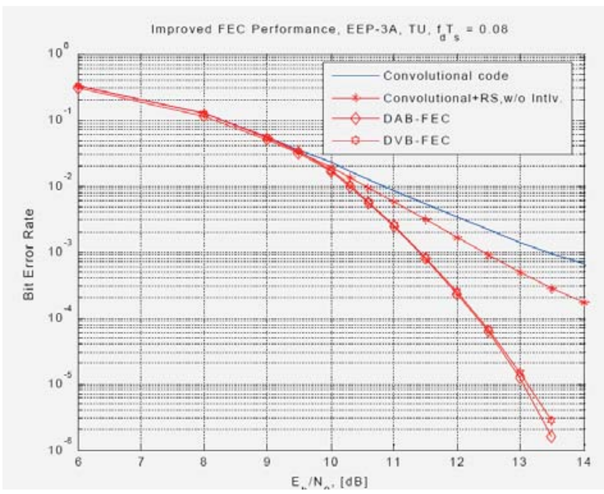

Figure 19
Error behaviour of DAB with cascaded error-control coding

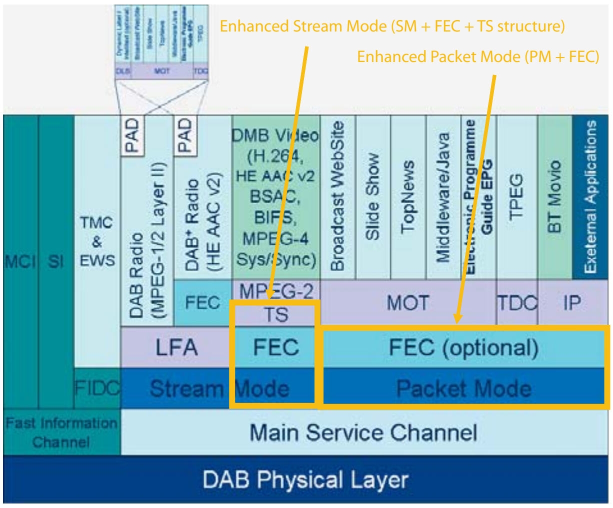

Figure 20
Position of Enhanced Stream and Packet Modes in the DAB protocol stack

5.1. *Enhanced Stream Mode (ESM)*

This Transport Mode – an evolution of what is called "MSC Stream Data" in the central DAB standard, ETSI EN 300 401 – is in fact an additional Packet Mode, consisting of a structure of 188-byte long packets with 16 Reed-Solomon parity bytes attached.

Furthermore, a Forney interleaver is applied to those FEC'ed 204-byte long packets. This structure is in use for DMB with the MPEG-2 Transport Stream – see ETSI TS 102 427.

Originally this structure was introduced with all variants of the DVB system.

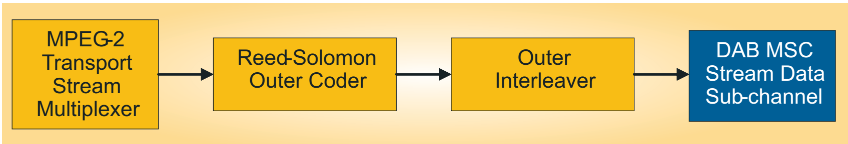

Figure 21
Adapting Transport Streams to existing DAB infrastructure

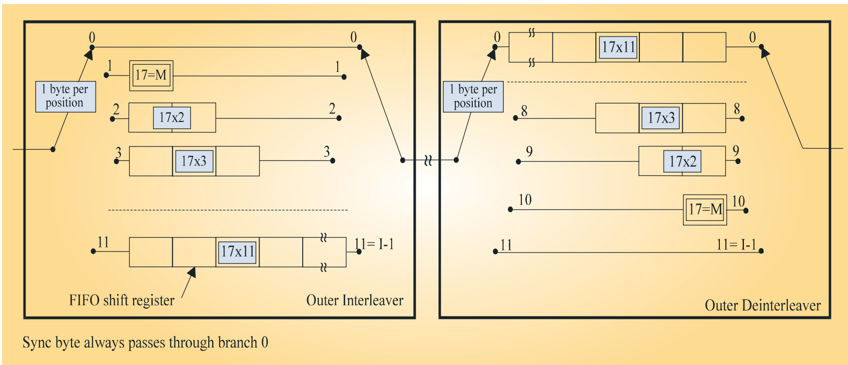

Figure 22
Forney Interleaver for the transmission of Transport Streams via DAB

5.2. *Enhanced Packet Mode (EPM)*

In a similar way as for the Enhanced Stream Mode described above, the existing Packet Mode ("MSC Packet Data" in ETSI EN 300 401) was extended and improved with another layer of error-control coding. In the case of EPM, virtual time interleaving is applied.

Two figures might illustrate the improved structure. *Fig. 23* presents the FEC frame that is filled vertically, packet by packet. Fully filled with an integer number of packets, the Reed-Solomon parity bytes are calculated horizontally over the 188 bytes in the same row. The same R-S scheme as for Enhanced Stream Mode led to a similar performance and was hence reused.

All application data columns are read out vertically as they are filled and are transmitted followed by the R-S parity bytes also read out vertically.

Fig. 24 illustrates the simple set-up of the equipment on the transmitter side.

Legacy receivers ignore the FEC packets, because they are different from ordinary data packets in two ways:

a) FEC packets carry a Packet Address that doesn't correspond to a Service Component, and;
b) they are of a different structure, including the position of the CRC.

The Enhanced Packet Mode can be used for all application data defined for Packet Mode, because EPM is a fully back-wards-compatible extension and shall therefore be applied to regular transmissions without exceptions.

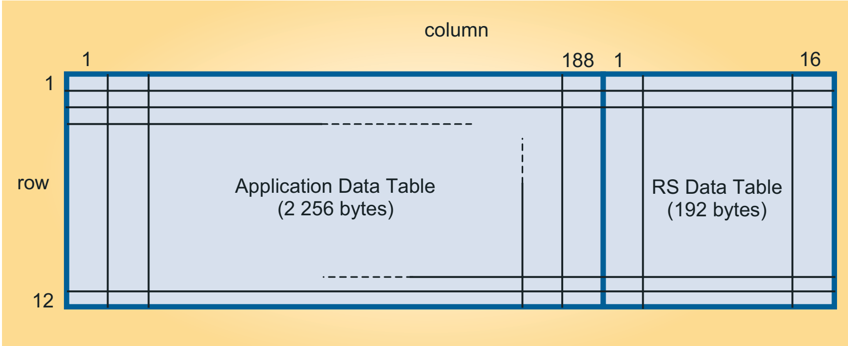

Figure 23
Building the FEC Frame

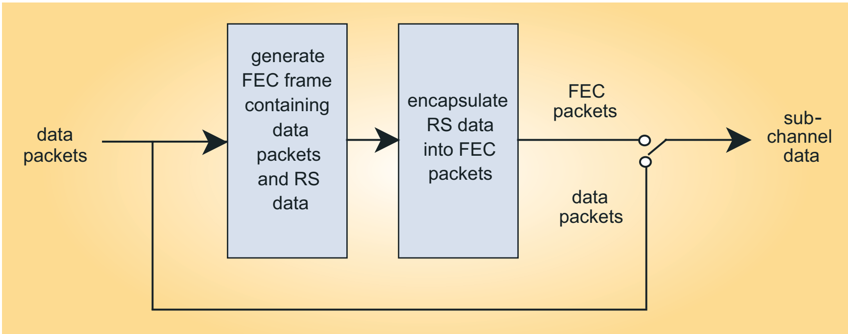

Figure 24
Adapting EPM to the existing DAB infrastructure

6. MPEG-2/MPEG-4 system and FEC overhead demystified

Every broadcasting system requires the addition of a particular overhead on top of the main content to be transported. Typical examples for such an overhead are synchronization signalling, error-control coding as well as service-parameter signalling and metadata. In particular, regarding the specific case of transporting narrowband applications with DMB, the figures given range from "a few percent" to half of a stream. In order to give such discussion a reliable scientific basis, an example case is discussed here in detail – a streaming application.

Let's grab a time slice from a narrowband DAB sub-channel used for DMB, consisting of a single stream. Because it's a common denominator of the entities we want to discuss, let it be seven seconds long.

The sub-channel bitrate for this example (other examples can be derived from this exercise, ideally eased by a few lines of source code) is 40 kbit/s. Here "k" for "kilo" is equivalent to 1000. An MPEG-2 Transport Stream fills the sub-channel completely, which means that within the seven seconds 171.57 MPEG-2 TS packets – all with 16 Reed-Solomon parity bytes attached – can be transported.

Starting with the PSI/SI signalling required for this case, we need to transport the two tables PAT (Programme Association Table) and PMT (Programme Mapping Table) every 500 ms. Hence within seven seconds, 28 of those tables occur and 143.57 TS packets are left for other purposes – assuming that each of the tables can be accommodated by one single TS packet. The MPEG-4 Initial Object Descriptor (IOD) is part of the PMT.

For a complete description of the object transported – i.e. the application data stream – the MPEG-4 system layer entity OD (Object Description) gets its PID and corresponding TS packets that are repeated every 500 ms. So 14 packets are assigned to OD and the remaining amount sums up to 129.57 TS packets. These packets offer 184 bytes each for the payload. So altogether 23,840.63 bytes are available for the payload to be transmitted.

For controlling the 27 MHz receiver clock accurately, the "Programme Clock Reference" PCR parameter is required every 100 ms. PCR is provided within the so-called adaptation field being part of a TS packet and located right after the 4-byte header. The PCR carries the same PID as the accompanied stream and can herewith be transported in the TS packets carrying the payload. PCR travels in the Adaptation field and occupies eight bytes per occurrence. In total, 560 bytes will be consumed by PCR in seven seconds. With this, 23,280.63 bytes are left.

MPEG-4 Access Units (AU) carry the MPEG-4-encoded content. Each AU is embedded in a Synchronization Layer (SL) packet and the SL packet in a PES packet. Insertion of PES packets into TS packets can be done in a fragmented way.

Assuming the extreme case of a length in time of 60 ms for each MPEG-4 AU, 116.67 of them need to be transported within seven seconds. This is equivalent to the number of SL and PES packets employed for the transport. The PES packet overhead is five bytes per packet and the SL packet overhead is one byte. Hence the complete overhead for seven seconds is 700 bytes.

Due to the fact that, every 700 ms, the Object Clock Reference (OCR) for synchronization of MPEG-4 objects and the Composition Time Stamp (CTS) – each of them 33 bits long – are repeated, every eleventh PES/SL packet additionally carries 66 bits (nine bytes) of this overhead, which sums up to 795.46 bytes within seven seconds.

Subtracting these 795.46 bytes from the 23,280.63 above, there are 22,485.17 bytes available for the transport of naked Access Units. This value can be converted to a bitrate of 25.70 kbit/s remaining for the Access Units. It is equivalent to 64.24% of the sub-channel bitrate of 40 kbit/s. So the overhead for the example discussed is 35.76%.

Abbreviations

AAC Advanced Audio Coding
ASF (Microsoft) Advanced Streaming Format
AU Access Units
BER Bit-Error Ratio
BIFS BInary Format for Scene description
BSAC Bit Sliced Arithmetic Coding
C/N Carrier-to-Noise ratio
CRC Cyclic Redundancy Check
CTS (MPEG) Composition Time Stamp
DAB Digital Audio Broadcasting (Eureka-147)
http://www.worlddab.org/
DAB+ DAB using the AAC codec
DAB-IP DAB - Internet Protocol
DLS Dynamic Label Segment
DMB Digital Multimedia Broadcasting
http://www.t-dmb.org/
DRM Digital Radio Mondiale
http://www.drm.org/
DRM Digital Rights Management
DVB Digital Video Broadcasting
http://www.dvb.org/
EPG Electronic Programme Guide
EPM Enhanced Packet Mode
ETSI European Telecommunication Standards Institute
http://pda.etsi.org/pda/queryform.asp
FEC Forward Error Correction
FM Frequency Modulation
GE06 Geneva Frequency Plan of 2006
HE-AAC High Efficiency AAC
IOD Initial Object Descriptor
IP Internet Protocol
MOT Multimedia Object Transfer
MPEG Moving Picture Experts Group
http://www.chiariglione.org/mpeg/
MSC Main Service Channel
OCR Object Clock Reference
OD Object Description
OFDM Orthogonal Frequency Division Multiplex
PAD Programme-Associated Data
PCR Programme Clock Reference
PES Packetized Elementary Stream
PID Packet IDentification number
PMT Programme Map Table
PS Parametric Stereo
PSI Programme Service Information
R-S Reed-Solomon
RDS Radio Data System
http://www.rds.org.uk/
SBR Spectral Band Replication
SD Secure Digital
SI Service Information
SL Synchronization Layer
TS Transport Stream
UDP User Datagram Protocol
WMA (Microsoft) Windows Media Audio
XML eXtensible Markup Language

With this calculation and the related assumptions applied to several more sub-channel bitrates we get :

Sub-channel bitrate [kbit/s]	Overhead [%]
32	42.24
40	35.76
48	31.43
56	28.34
64	26.02
72	24.22
.	.
.	.
.	.
192	15.21

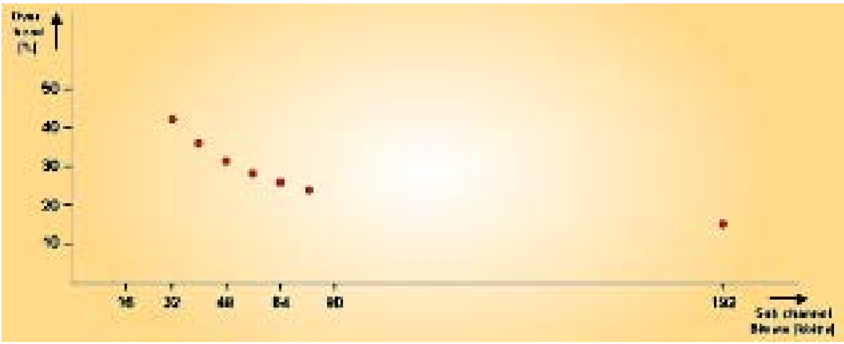

Figure 25
MPEG-2/MPEG-4 system layer and FEC overhead

In summary, it is recognized that for applications requiring low sub-channel bitrates, the combination of MPEG-2 and MPEG-4 system layers leads to quite a significant overhead.

For higher data rates the overhead is less significant.

7. Outlook

DAB has been refurbished in a way that assures its further success in the near future. The application of new coding algorithms has been enabled through a second layer of error-control coding in a way that is widely implemented for DVB already. Existing structures can easily co-exist with the new ones illustrated above.

Further to the elements described here, the new DAB middleware approach as well as voice-related applications are awaiting their completion shortly. Furthermore, the PAD content of the MPEG-1/2 Layer II radio services shall be protected more thoroughly. Once again, the well-established R-S scheme will be applied (same mother code).

As far as text applications are concerned, the DAB equivalent to RDS RadioText +, i.e. Dynamic Label +, will be adopted shortly.

From the industry's point of view, the different digital broadcasting systems shall be aligned more closely in the future. Corresponding convergence activities are about to be started and will build a core subject for the next few years in digital broadcasting.

Making IP the universal and common layer for more or less all communication systems, all sides can make use of existing applications and there is less and less need for developing bearer-specific upper-layer elements.

In 2007, it is also time to consider backwards-compatible amendments to the physical layer of DAB in order to extend DAB's spectral efficiency. Competition between broadcasting systems over the coming decade will put emphasis on this aspect. Clearly, such a step does not come for free, but will require higher C/N ratios. This means, first of all, that conformity with GE06 planning parameters must be assured for each such step. In addition, European regulations will not allow for high field strengths. So it is necessary to detect the borderline and move closer to it – for sure, the necessary investments will pay off after quite a short time period.

DAB will build further on its strengths – flexibility and reliability.

Frank Herrmann – born 1963 – received an engineering diploma from the Technical University of Braunschweig in Germany. Having led Clarion into the Eureka-147 DAB project, digital broadcasting became the core element of his professional career. At Panasonic, he initially managed the DAB pilot project *Hessen*. Within the EU-sponsored TPEG project, he chaired the Industry Group and built the basis for commercial services now realised within the mobile.info project.

Today Mr Herrmann is Project Leader for DVB-T2 and DAB/DMB at Panasonic's Frankfurt labs and head of the Digital Broadcasting Team. Within the DVB-T2 group, he recently accepted responsibility for forming a team to deal with the design of the system layer.

Frank Herrmann belongs to the "oldest members" of WorldDMB and became TC Chairman in 2004. He has put emphasis on a thorough refurbishment of this broadcasting system and co-initialised WorldDMB's contacts to external Fora – with priority today on forging links with the DVB Project.

Larissa Anna Erismann works for Swiss Satellite Radio, a branch of the Swiss public broadcaster (SRG SSR idée suisse) which produces three ad-free, music-only radio stations: Radio Swiss Classic, Radio Swiss Jazz and Radio Swiss Pop. These three stations form an important part of the DAB-only offer in Switzerland.

Ms Erismann took charge of the B2B and B2C aspects of DAB marketing for all three stations in 2004 and subsequently launched www.dab-digitalradio.ch, the only web platform to offer a comprehensive range of business and consumer information – in four languages – on the DAB situation in Switzerland.

Larissa Anna Erismann is an acknowledged expert on Switzerland's DAB market and is continuously expanding her familiarity with DAB in other countries. In July 2006, she acceded to the chair of the WorldDAB Marketing Committee after participating in a number of the Committee's projects and heading a task force dedicated to the promotion of DAB in Germany.

Markus Prosch studied computer science at the University of Erlangen and gained his Dipl.-Ing. degree in 1994. In 1995 he joined the Fraunhofer Institute for Integrated Circuits IIS. Since 1996, he has been deeply involved in the specification of DAB standards. He is responsible for the development of the Fraunhofer MMDS (Multimedia DataServer), a system designed to insert all kinds of data services onto DAB.

Starting in 2001, Mr Prosch has also been involved in the specification of Digital Radio Mondiale (DRM) standards. He is responsible for the development of the Fraunhofer DRM-CS (DRM ContentServer), Fraunhofer's solution for the DRM broadcast chain. He now represents Fraunhofer IIS within WorldDMB. In 2006, he was the Chairman of the WorldDMB task force on *audio coding*.

Related standards

- ETSI EN 300 401 V1.4.1 (2006-06): **Radio Broadcasting Systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers**.
- ETSI EN 301 234 V2.1.1 (2006-06): **Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) Protocol**.
- ETSI ES 201 735 V1.1.1 (2000-09): **Digital Audio Broadcasting (DAB); Internet Protocol (IP) datagram tunnelling**.
- ETSI TS 101 498-1 V2.1.1 (2006-01): **Digital Audio Broadcasting (DAB); Broadcast website; Part 1: User application specification**.
- ETSI TS 101 498-2 V1.1.1 (2000-09): **Digital Audio Broadcasting (DAB); Broadcast website; Part 2: Basic profile specification**.
- ETSI TS 101 498-3 V2.1.1 (2005-10): **Digital Audio Broadcasting (DAB); Broadcast website; Part 3: TopNews basic profile specification**.
- ETSI TS 101 499 V2.1.1 (2006-01): **Digital Audio Broadcasting (DAB); MOT Slide Show; User Application Specification**.
- ETSI TS 101 756: V1.3.1 (2006-02): **Digital Audio Broadcasting (DAB); Registered Tables**.
- ETSI TS 101 759 V1.2.1 (2005-01): **Digital Audio Broadcasting (DAB); Data Broadcasting - Transparent Data Channel (TDC)**.
- ETSI TS 101 993 V1.1.1 (2002-03): **Digital Audio Broadcasting (DAB); A Virtual Machine for DAB: DAB Java Specification**.
- ETSI TS 102 368 V1.1.1 (2005-01): **Digital Audio Broadcasting (DAB); DAB-TMC (Traffic Message Channel)**.
- ETSI TS 102 371 V1.2.1 (2006-02): **Digital Audio Broadcasting (DAB); Digital Radio Mondiale (DRM); Transportation and Binary Encoding Specification for Electronic Programme Guide (EPG)**
- ETSI TS 102 427 V1.1.1 (2005-07): **Digital Audio Broadcasting (DAB); Data Broadcasting - MPEG-2 TS Streaming**.
- ETSI TS 102 428 V1.1.1 (2005-06): **Digital Audio Broadcasting (DAB); DMB video service; User Application Specification**.
- ETSI TS 102 563 V1.1.1 (2007-02): **Digital Audio Broadcasting (DAB); Transport of AAC audio**.
- ETSI TS 102 818 V1.3.1 (2006-02): **Digital Audio Broadcasting (DAB); Digital Radio Mondiale (DRM); XML Specification for DAB Electronic Programme Guide (EPG)**.
- ETSI TS 102 iii V1.1.1 (2007-ii): **Digital Audio Broadcasting (DAB); Intellitext**.
- ETSI TR 101 496-1 V1.1.1 (2000-11): **Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 1: System outline**.
- ETSI TR 101 496-2 V1.1.2 (2001-05): **Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 2: System features**.
- ETSI TR 101 496-3 V1.1.2 (2001-05): **Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 3: Broadcast network**.

Appendix A: WorldDMB policy statement on DAB/DAB^{+}/DMB

The WorldDMB policy statement relating to how the standards for DAB/DAB+ and DMB are to be interpreted and supported makes a clear differentiation between implementation of *video* **and** *audio* **services:**

The WorldDMB Forum recommends that:

- *DAB (ETSI EN 300 401) or DAB+ (ETSI TS 102 563) should be used for radio-centric services;*
- *DMB (ETSI TS 102 428) should be used for services which include a video component.*

Frank Herrmann's note: For video services based on DAB, DAB-IP might be employed as an alternative to DMB.

Last Update: *17 July 2007*