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Since 1998, the Technical Review has been published on-line, four times per year. This venture has been very successful because it has expanded the readership. It has been very gratifying to meet people who have "discovered" the on-line version. Some of these new readers are not directly connected with the EBU but they work in related areas, such as suppliers of hardware or software to the broadcasting industry.

Our electronic publishing service includes an archive of the Technical Review, dating back to 1992. You can access these archive articles by clicking on the "Archive / Thematic Index" link, in the navigator frame to the left of the screen. This archive contains a wealth of excellent articles on many different topics. Recently we added a "Hot Topics" section to the on-line version which brings together relevant articles from the archive on topics such as "HDTV in Europe" and "Broadcasting to Handhelds".

The on-line version also includes a list of abbreviations used in the Technical Review over the past 15 years or so. New terms are continually being added to the list which you can download as a PDF file by clicking on the "Abbreviations" link in the navigator frame.

At the beginning of the year 2000, the EBU abandoned the printed version of EBU Technical Review. Nevertheless, it was recognised that electronic publishing could not entirely replace "hard copies". This state of "Nirvana" will not arrive until we have computers that can match this paper publication in terms of portability, weight, readability, quality, price and (last, but not least) battery life!

As there is still considerable value in paper publication, the EBU decided to re-publish some of the articles in an annual printed edition of EBU Technical Review. Whereas the on-line version is available only in English, the annual edition is also available in French.

If you enjoy reading the articles in this publication, remember to consult the on-line version of EBU Technical Review at **www.ebu.ch/en/technical/trev/trev_home.html**

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Lieven Vermaele Director
EBU Technical Department

Implementation of the

Digital Dividend

– technical constraints to be taken into account

Jan Doeven
Doeven Radiocommunication Consultancy

At the RRC-06, a new Agreement and associated frequency plans for digital broadcasting and analogue TV broadcasting during the transition period were agreed (GE06). The next step is implementation of the new Agreement.

Broadcasting organizations, network operators, spectrum user forums and others have announced their opinions on the use of Bands III, IV and V. A term often used in relation to the implementation of the new Agreement is "digital dividend". This article describes the technical constraints to be taken into account when using released spectrum for several digital dividend applications.

1. Introduction

At the RRC-06, a new Agreement and associated frequency plans for digital broadcasting and analogue TV broadcasting during the transition period were agreed (GE06) [1]. The next step is the implementation of the new Agreement. Broadcasting organizations, network operators, spectrum user forums and others have announced their opinions on the use of Bands III, IV and V. A term often used in relation to the implementation of the new Agreement is "digital dividend". There may be many meanings of the term. For the countries in the European Union, the definition used by the Radio Spectrum Policy Group (RSPG) and the European Commission is most relevant.

"Digital Dividend" is, according to the RSPG, to be understood as the spectrum made available over and above that required to accommodate the existing analogue television services in a digital form in VHF (Band III: 174 - 230 MHz) and UHF (Bands IV and V: 470 - 862 MHz) [2]. It should be noted however that existing analogue television also makes use of Band I (47 - 68 MHz) and, after digital switchover, Band I spectrum could be considered as digital dividend too. Furthermore, Band III is also planned for T-DAB and many existing T-DAB services already make use of Band III. In addition, in a number of countries, non-broadcasting services make use of Bands III, IV and V.

Many possible applications of the digital dividend are under discussion. In its Communication on *"EU spectrum policy priorities for the digital switchover in the context of the upcoming ITU Regional Radiocommunication Conference 2006 (RRC-06)"* [3], the European Commission identified three categories:

1) Spectrum needed for the improvement of terrestrial broadcasting services: e.g. services with higher technical quality (notably HDTV), increased number of programmes and/or enhancement of TV experience (e.g. multi-camera angles for sports, individual news streams and other quasi-interactive options);

2) Radio resources needed for "converged" broadcasting services which are expected to be primarily "hybrids" of traditional broadcast and mobile communication services;

3) Frequencies to be allocated to new "uses" which do not belong to the broadcasting family of applications. Some of these potential new "uses"

of the spectrum dividend are future services and applications which are not yet marketed and others are existing ones which do not operate yet in these frequencies (e.g. extensions of 3G services, short-range radio applications).

This article describes the technical constraints to be taken into account when using released spectrum for several digital dividend applications. Considerations on the use of the digital dividend are also described in [4].

2. Size of the digital dividend

2.1. "Layers"

A term often used when considering national input requirements and results of RRC-06 is the number of "layers". A layer is not defined in the GE06 Agreement, nor was it defined at RRC-06, but for most European countries it may be described as a set of channels which can be used to provide full or partial nationwide coverage. The number of layers depends, among others, on the geographical situation, the level of accepted interference, transmission and reception characteristics and the way an Administration composes its layers out of the available Plan entries.

Administrations submitted their T-DAB and DVB-T requirements before RRC-06. Fulfilling these initial requirements would, in some areas, have required ten times the band capacity and, in most areas, two or three times. In defining input requirements, Administrations took into account their long-term broadcasting needs, their rights concerning use of other primary services operating in Bands III, IV or V (if

any) and maybe, in some countries also, possible future use of other applications. However, as the planning process at RRC-06 allowed Administrations to make input requirements only for T-DAB or DVB-T, other possible applications needed to be described as broadcast requirements. Another element in defining input requirements was the wish for all Administrations to have an equitable access to the frequency bands. Therefore the T-DAB and DVB-T input requirements do not always necessarily represent the current minimum market requirements. Furthermore, it should be noted that a national requirement may seem unrealistic from a frequency-planning point of view, or even from the point of view of a neighbouring Administration, but could be political reality in a country.

During RRC-06 there was a strong pressure on Administrations to reduce their requirements in accordance with the following guidance (see the table below).

Most European countries were successful in achieving the above-mentioned number of layers.

In most countries there are four analogue TV services and these can in general be accommodated into one DVB-T multiplex for which one DVB-T layer is needed. However countries with five or more analogue TV services and using DVB-T with a robust modulation, may need two DVB-T multiplexes and thus two layers for broadcasting their existing analogue TV services in digital format.

For a successful introduction of DVB-T, more multiplexes are needed than the number of channels containing the current analogue TV programmes (see Section 6.1) but, following the RSPG definition, in general out of the eight to nine achieved DVB-T layers, six to eight DVB-T layers

and the three T-DAB layers could be seen as digital dividend (Fig. 1).
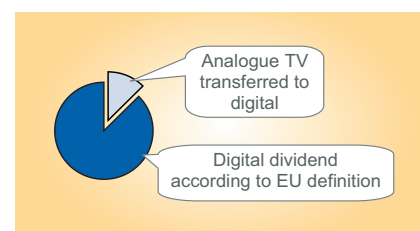
Figure 1
Band III, IV & V spectrum

2.2. Frequency bands

Band I (47 - 68 MHz) was not planned for digital broadcasting at RRC-06 and is regulated by the revised Stockholm Agreement [5]. The band is not included in the RSPG definition of digital dividend. However, after analogue TV has been switched off, it may also be considered for new applications, taking into account that there are already non-broadcasting services in a number of countries. Band I is less attractive than Bands III, IV or V for many services due to:

- its long wavelength, and therefore large antenna dimensions;

- its susceptibility to ionospheric interference from the Sporadic E-layer;

- the high levels of man-made noise at these frequencies [6].

In general, not much interest has been expressed for Band I. Currently some DRM (Digital Radio Mondiale) experiments take place in this band.

Band III (174 - 230 MHz) has been planned for T-DAB and DVB-T. A number of countries are considering implementing DVB-T only in Band IV/V, and to use Band III exclusively for T-DAB or multimedia applications making use of a T-DAB based system. There is currently no interest in applying new non-broadcasting services in this band.

For digital dividend applications, the band is considered as 1st and 2nd category (see Section 1).

Guidance for number of "layers"

Band III		Band IV/V
T-DAB	**DVB-T**	**DVB-T**
3	1	7 - 8

Band IV/V (470 - 862 MHz) is subject to most of the discussions on digital dividend, covering all three categories *(see Section 1)*. In addition to broadcasting, the UMTS lobby sees it as an attractive band for mobile communication systems.

3. GE06 Agreement

The Plan entries of GE06 will only become fully available after analogue switch-off. The European Union proposes to switch off analogue TV before 2012[7]. According to the GE06 Agreement, analogue TV will have no right of protection after 17 June 2015 (and, in some African and Middle East countries, after 17 June 2020 in the case of VHF transmissions).

The GE06 Agreement offers two options to achieve flexibility in the application of Plan entries (Article 5):

● **Different characteristics of a Plan entry** can be applied as long as the "conformity check" is fulfilled. The main criterion is that interference from the application is not more than that of the Plan entry. This mechanism can for instance be applied to convert a Plan entry into a Single Frequency Network or a different reception mode (see Section 4);

● **Alternative applications of a Plan entry** (that is other than DVB-T or T-DAB) are possible in the Broadcasting, Mobile and Fixed services if three conditions are fulfilled:
  ● band allocation in the Radio Regulations to the relevant service;
  ● not exceeding the spectral power density of the associated Plan entry;
  ● not claiming more protection than afforded to the associated Plan entry.

A more detailed description of the options for achieving flexibility is given in [8].

In addition, the GE06 Agreement contains a procedure for modification of the Plan (Article 4). Under this procedure, the agreement of all potentially-affected countries is needed to make a change to a Plan entry. The Article 4 procedure also needs to be followed in cases where services other than broadcasting, which have co-primary status, are introduced or modified.

Depending on the impact on the GE06 Agreement, two uses of the digital dividend can be distinguished:

● **Applications making use of Plan entries** which require no or limited modifications to the GE06 Plan;

● **Applications making use of a dedicated sub-band** with the consequence of considerable modifications to the GE06 Plan.

4. Applications making use of GE06 Plan entries

The GE06 Agreement has harmonised planning parameters for use of the 174 - 230 MHz band by T-DAB and DVB-T and the 470 - 862 MHz band by DVB-T. T-DAB has been planned for mobile and portable reception, DVB-T for rooftop and portable reception.

4.1. Reception mode

Each Plan entry has a specified reception mode. The most used for DVB-T are a set of characteristics for rooftop reception or portable outdoor reception. The latter term stands also for portable indoor or mobile reception at a lower coverage quality. *Fig. 2* shows the specified DVB-T reception mode for the European countries.

A transmission based on a Plan entry specified for rooftop reception can be used for portable reception if a reduced coverage area is acceptable. If it is not, a dense Single Frequency Network (SFN), that fulfils the conditions of the conformity check of Article 5 of the GE06 Agreement, is a possibility to improve coverage. It may also be necessary to seek international agreement for modifying the Plan entry with a higher power by applying the Article 4 procedure of the GE06 Agreement.

If a transmission based on a Plan entry that was specified for portable reception is used for rooftop reception, a larger coverage area will be obtained and there may be an overlap of (rooftop) coverage of two or more adjacent transmitters. In practice, portable coverage may be restricted to built-up areas. In the surrounding rural areas, the wanted field strength is likely to be sufficient for rooftop reception but the received interference levels associated with the portable Plan entry may be too high for full rooftop coverage of the area. There is likely to be more than one possible wanted transmitter, because of the overlap of coverage areas. In some cases, instead of directing the rooftop antenna towards the transmitter giving the highest signal strength, a better signal-to-interference ratio may be obtained by aligning the antenna on another transmitter. In some areas, a very directional (and hence, a much more expensive) rooftop antenna may be needed. It may also be necessary to optimize the transmitter characteristics, or the SFN, taking care that the conformity check of Article 5 of the GE06 Agreement is fulfilled.

The most-used basic characteristics of the DVB-T reception modes in Band IV/V are:

Reception mode	Rooftop	Portable
Capacity	≤24 Mbit/s	≤16 Mbit/s
Required field strength	56dBμV/m	78dBμV/m

[provided by the EBU]

Reception mode

- ▮ Portable
- ▮ Rooftop or portable
- ▮ Rooftop

Figure 2
Reception mode as specified in GE06

5.1. Replanning Band IV/V

A new non-broadcasting application needs to be agreed by all potentially-affected countries in accordance with the Article 4 procedure of GE06 and has to be incorporated in the "List" of Annex 5 of GE06 in order to be protected from GE06 Plan entries and further modifications of the GE06 Plans. In order to obtain agreement, the new application may be subject to restrictions because Plan entries of other countries need to be protected and interference from Plan entries of other countries accepted.

In general in the GE06 Plan, the frequencies at a given site or in a certain area are scattered over the whole band. A sub-band for Mobile services, such as UMTS, could therefore affect all DVB-T "layers" *(for the meaning of "layers" see Section 2.1)* as it would create "holes" in the layers (areas not covered because the frequencies are no longer available due to the sub-band).

The remaining part of the band will need to be re-planned in order to obtain the original envisaged DVB-T coverage, with a reduced number of layers, by applying the Article 4 procedure of the GE06 Agreement. This re-planning means in practice a re-doing of the GE06 Plan. However Plan entries of other countries need to be protected and interference

The field-strength requirements for handheld reception in Band IV/V range from 85 to 107 dBµV/m, depending on the modulation and reception conditions [9] and are higher than for portable reception. A transmission based on a Plan entry for portable reception can be used for handheld reception under comparable conditions to those indicated above for the case of a Plan entry for fixed reception being used for portable reception.

4.2. Different network topologies

If, in a given area, the network topology of one or more of the multiplexes is different to that of the other multiplexes (e.g. if dense networks are used for some multiplexes), adjacent channel interference may occur around non co-sited stations. Such interference may occur on the first, second and even third

adjacent channel on both sides of the wanted channel.

Adjacent channel interference is a local problem. A possible solution is co-locating fill-in transmitters at the site of the interfering transmitter. The question will arise however who will have to pay for these provisions.

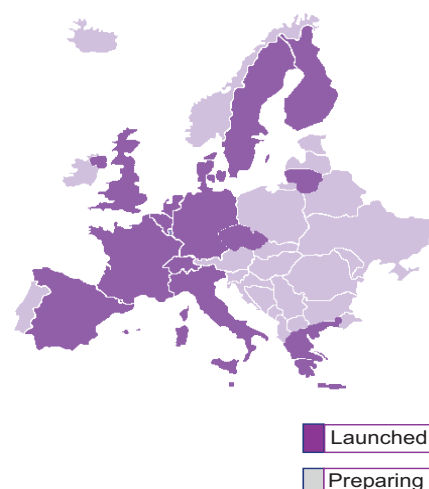## 5. Applications making use of a dedicated sub-band

For applications with up-links and different channelling schemes, dedicated sub-bands are considered [2][18] e.g. for UMTS. In the case of UMTS, an allocation to Mobile services in the Radio Regulations needs to be agreed at ITU WRC-07 or WRC-11 and a sub-band would be required from which the GE06 Plan entries are deleted.

- ▮ Launched
- ▯ Preparing

Figure 3
European DVB–T map *(from DigiTag)*

from those Plan entries of other countries accepted.

The re-planning process is likely to be complex and time-consuming and it is not guaranteed that the original coverage can be repaired. DVB-T has already been introduced in 14 European countries *(see Fig. 3)* and, by the time the process is completed, many more DVB-T transmitters will be in operation. A transition from the original GE06 Plan to a re-planned GE06 Plan will be necessary.

5.2. Guard bands

In order to avoid interference between uplink transmissions and adjacent (downlink) broadcast transmissions, guard bands are needed. The width of a guard band depends on many factors and, according to ongoing studies in ITU-R and elsewhere, may be more than 10 MHz. Also a guard band is needed between the uplink and the downlink sub-band. The total guard bands and thus the unused spectrum may add up to several DVB-T channels.

6. Spectrum use

Bands III, IV and V are the only available bands for obtaining wide-area DVB-T and DVB-H coverage. Following an "Opinion" of the Radio Spectrum Policy Group of the European Union on the introduction of multimedia services[10], CEPT has been mandated to identify appropriate technical and regulatory parameters for opening up the band 1452 to 1479.5MHz to allow flexible use by a wide range of mobile multimedia technologies. However the propagation characteristics and the width of this band (25.5 MHz) are in general not adequate to plan nationwide coverage in each of the European countries, even if a 5 MHz DVB-T or DVB-H bandwidth is chosen.

UMTS services can be operated in several bands and a series of possible extension bands have been identified including Band IV and V[11].

6.1. Broadcasting use of Band IV and V

In order to motivate consumers to buy a digital receiver for terrestrial services, an attractive broadcast package needs to contain 20 to 30 popular programmes. Such a number is also needed to provide better competition to satellite and cable delivery. A large number of programmes that are of high individual interest to only a few people can best be delivered by means of on-demand services: for example, via UMTS in the case of reception on small screens. This could be facilitated by means of a common Electronic Service Guide (ESG). These on-demand services could include public and commercial programmes received outside the national territory, for instance by travellers and tourists wishing to receive their homeland programmes.

The number of layers which can be provided by GE06 is very large and significantly increases the spectrum usage as compared to ST61; in fact, this usage exceeds the theoretical capacity of the frequency bands, at least for the technical conditions used at RRC-06. These "extra" layers have been achieved at the expense of accepting higher interference levels which may result in lower quality services and/or reduced coverage areas. To overcome these difficulties when implementing the Plan and in order to provide reliable services,

Abbreviations

CEPT	Conférence Européenne des Postes et Télécommunications (European Conference of Postal and Telecommunications Administrations)
CRT	Cathode Ray Tube
DAB	Digital Audio Broadcasting (Eureka-147) http://www.worlddab.org/
DRM	Digital Radio Mondiale http://www.drm.org/
DVB	Digital Video Broadcasting http://www.dvb.org/
DVB-H	DVB – Handheld
DVB-T	DVB – Terrestrial
EU	European Union
GE06	Geneva Frequency Plan of 2006
ITU	International Telecommunication Union http://www.itu.int
ITU-R	ITU – Radiocommunication Sector http://www.itu.int/ITU-R/publications/rec/index.asp
RRC	(ITU) Regional Radiocommunication Conference
RSPG	Radio Spectrum Policy Group
SAB	Services Ancillary to Broadcasting
SAP	Services Ancillary to Programme-making
SFN	Single-Frequency Network
ST61	Stockholm Frequency Plan of 1961
T-DAB	Terrestrial – DAB
UHF	Ultra High Frequency
UMTS	Universal Mobile Telecommunication System
VHF	Very High Frequency
WRC	(ITU) World Radiocommunication Conference

it may be necessary to deploy additional transmitters and additional frequencies.

In order to provide an acceptable video and audio quality on conventional displays, three to four programmes can be accommodated in a multiplex for portable reception (16 Mbit/s) and five to six programmes in a multiplex for rooftop reception (24 Mbit/s). The average data capacity allocated to each programme could be from 3 to 4 Mbit/s depending on the DVB-T variant used and depending on the statistical multiplexing, if used [12].

It should be noted that quality requirements need to be increased with the advent of flat-panel screens. These kinds of screens are very popular and are, or will be, used soon in many households. EBU investigations have shown that flat screens are more sensitive to artefacts and, for a good picture, require about twice the bitrate needed for Cathode Ray Tubes (CRTs) [13][14].

The video compression system MPEG-4 will enable a lower bitrate compared to MPEG-2, while maintaining the same quality. Use of MPEG-4 could therefore compensate for the higher bitrate demand of flat-panel displays. A number of countries, where DVB-T is yet to start, are considering or have already decided to use MPEG-4 with DVB-T. However

countries that have already introduced DVB-T will need to use additional layers for introducing MPEG-4, with the exception of France where some multiplexes have already been introduced with MPEG-4 (for pay-TV).

Currently DVB-H pilot transmissions are taking place in several countries and are already operational in Italy [15]. In planning DVB-H services, a balance needs to found between, on the one hand, the radiated power and number of transmitters required to obtain the wanted coverage and, on the other hand, the available bitrate. As the reception conditions are very demanding [8], most operators tend to choose a robust system variant with the consequence of a limited net bitrate. Therefore 10 to 15 programmes may be accommodated in a DVB-H multiplex.

It is expected there will be more than 50 million HD-ready TV sets in Europe by 2010 [16] and, consequently, there will also be a high demand for HDTV programmes. Currently HDTV programmes are delivered by satellite but many European broadcasters are planning to transmit HDTV on terrestrial networks. EBU studies [17] indicate that two HDTV programmes can be accommodated in a DVB-T multiplex for rooftop reception (24 Mbit/s). HDTV is not compatible with standard definition TV reception and therefore HDTV

needs to be transmitted in parallel to DVB-T multiplexes.

6.2. UMTS considerations

One option for use of the digital dividend, which is being considered by CEPT, is UMTS. The UMTS Forum considers that 2 x 30 MHz of paired spectrum, based on 5 MHz channelling, would provide a viable minimum coverage extension band for UMTS [18]. This requirement includes a guard band between the uplink and downlink sub-bands and would also require guard bands of 10 to 16 MHz between it and the adjacent sub-bands used for DVB-T.

6.3. Other uses

Assignments to other services having primary status in the Radio Regulations have been taken into account at RRC-06 if so requested by the Administrations concerned. These services include radio navigation and fixed or mobile services for military applications and are shown in the "List" of Annex 5 of GE06. In any re-planning process, if so required, these services need to be taken into account.

In addition there are services with secondary status in the Radio Regulations in Band IV/V. These services are not taken into account when primary services are planned. However, on a national basis, these services could be of great importance, for instance the Radio Astronomy Service in channel 38 and Services Ancillary to Broadcasting and Programme making (SAB/SAP).

SAB/SAP services are of increasing importance because an increase in the number of broadcast programmes means also an increase in the need for facilities to produce broadcast programmes. This is true in spite of the fact that the use of SAB/SAP in Band IV/V is becoming more restricted since the band is densely planned for DVB-T, leaving less room for SAB/SAP transmissions.

7. Digital Dividend choices

From a technical point of view there are two alternative options for digital dividend applications:

Either,

- Applications making use of Plan entries that require no or limited modifications to the GE06 Agreement, such as DVB-T, HDTV, DVB-H
  - Some restrictions may be expected because of power limitations and interference levels of the corresponding GE-06 Plan entries.
  - In most cases the services can be implemented under Article 5 of the GE-06 Agreement and no international agreement is needed.
  - In some cases plan modifications may be needed by applying Article 4 of the GE-06 Agreement, requiring the agreement of potentially affected countries.
  - Adjacent channel problems may occur if different network topologies in Band IV/V are used in the same area. These problems need to be solved nationally.

Or,

- Applications making use of a dedicated sub-band with the consequence of considerable modifications to the GE06 Plan.
  - In case of uplink transmissions, an allocation in the Radio Regulations for Mobile services would be needed. In addition, guard bands are needed.
  - For new applications, Article 4 of the GE06 Agreement needs to be applied. Restrictions are to be expected in order to protect the GE06 Plan entries of other countries; interference from Plan entries of other countries needs to be accepted.
  - Some technical constraints may arise if different network topologies and systems co-exist in the same bands. Feasibility studies are needed.
  - Re-planning of the remaining part of the band is needed for DVB-T, requiring application of Article 4 of the GE06 Agreement.
  - A transition from the original GE06 plan to a modified plan is needed.
  - Re-planning and transition to a modified plan will be a complex and time-consuming process requiring several years of intense international coordination.

In several European countries, five or six multiplexes for DVB-T or DVB-H have been licensed or will be licensed soon. This means that in those countries a considerable part of the digital dividend will be used for categories 1 and 2 (see Section 1).

After having licensed five or six multiplexes for DVB-T or DVB-H, in general one or two layers remain. These could in principle be considered for all three digital dividend categories.

For application of the third category only (a new use such as 3G), a dedicated sub-band must be considered and consequently a re-planning process. However it raises the following questions:

- Would WRC-07 or WRC-11 indicate Band IV/V as a 3G extension band when there are so many alternative bands, while Bands III, IV and V are the only possibilities for wide-area coverage of DVB-T and DVB-H?

- Would Administrations be interested in involving themselves in another intensive period of re-planning for digital broadcasting with unpredictable results after having experienced the two sessions of the RRC and more than six years of preparing for these?

- Would broadcasters and network operators be willing to bear the nuisance and the costs of another transition period without the benefit of transmitting additional services?

Only the future will tell!

Jan Doeven received a bachelor degree in Electrical Engineering in 1971. All through his career, he held leading positions in frequency management and the application of new technologies for broadcasting. He worked for Nozema and KPN Broadcast Services in the Netherlands as Strategic Technology Advisor until his retirement in August 2007 and he is now an independent consultant.

He has participated in EBU activities in the field of radio and television broadcasting for 30 years and was chairman of the Broadcast-technology Management Committee (BMC) from 1997 to 2007.

Since the early nineties, Jan Doeven has been deeply involved, nationally and internationally, in the planning and implementation of digital broadcasting networks. He chaired the European preparatory groups for RRC-04 and RRC-06 (CEPT Project Team FM24 and CEPT Working Group RRC-06 respectively) and during RRC-06 he was the overall CEPT coordinator and vice chairman of the Conference.

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Report No. 38 from the UMTS Forum

The evolution of
DAB

Frank Herrmann
Chairman, WorldDMB Technical Committee

Larissa Anna Erismann
Chairperson, WorldDMB Marketing Committee

Markus Prosch
Chairman, WorldDMB TC Task Force on Audio Systems

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.

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.

Figure 2
The world of Eureka-147

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)*.

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
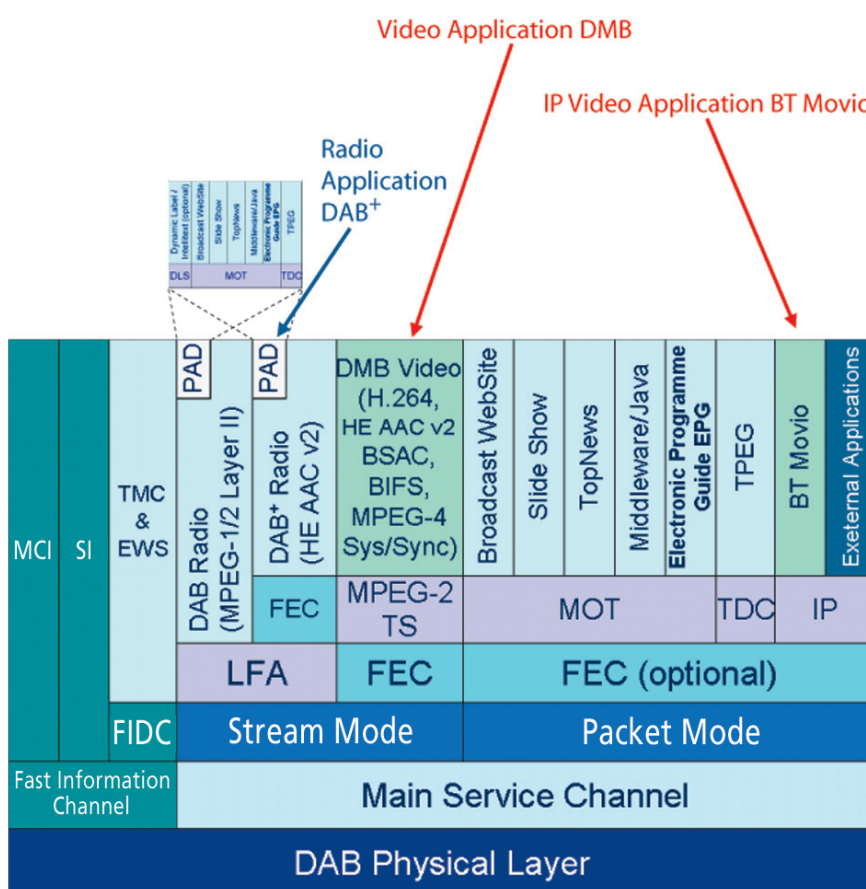

Figure 1
DAB System Protocol Stack

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.

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

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.

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
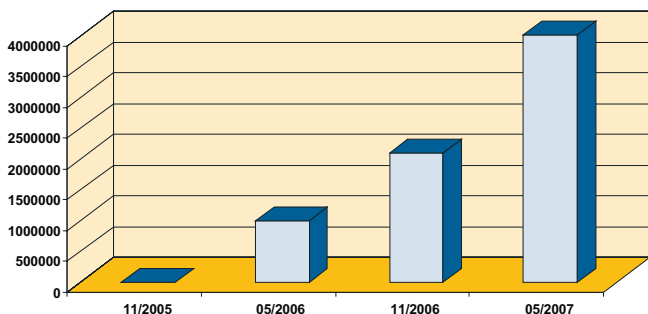

Figure 3
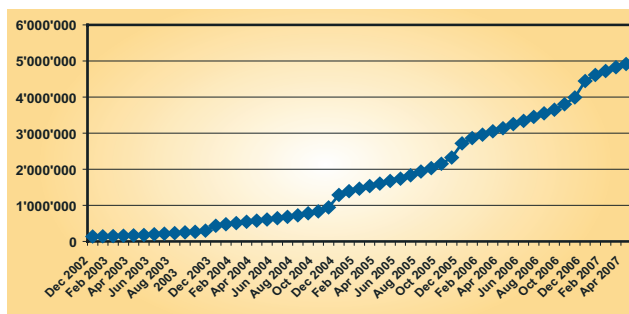**DAB receiver sales in the UK (accumulative)**

Figure 4
Sales of DMB receivers in Korea
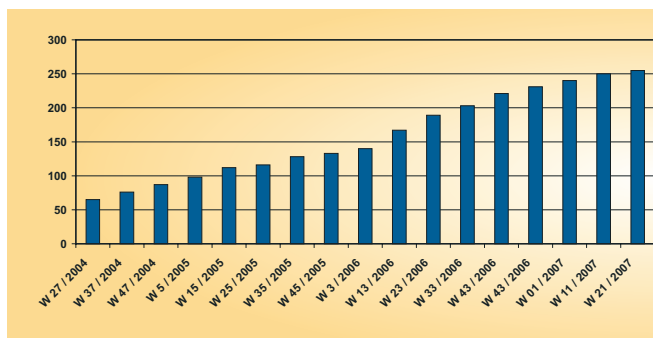

Figure 5a
Manufacturers of DAB/DMB receivers
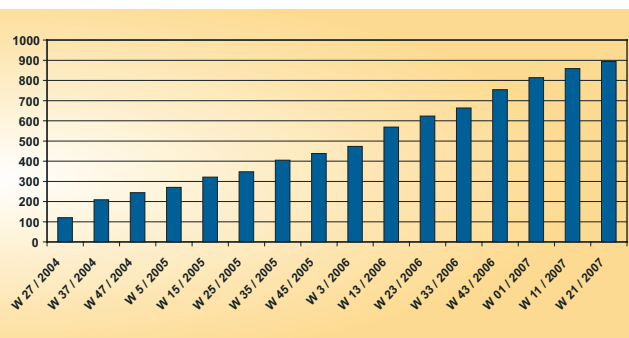*(Source: www.dab-digitalradio.ch, May 2007)*

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

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 built up as a hierarchical system *(see Fig. 6 and the text box)*, 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 hence the implementation of HE-AAC v2 is mandatory.
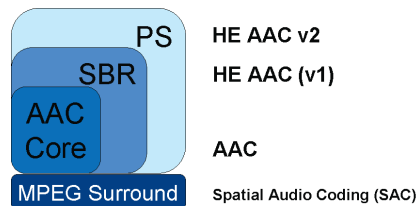
In light of the fact that audio coded with MPEG Layer II will remain on-air for many years to come, a new DAB receiver needs to cover both coding algorithms – MPEG-1/2 Layer II and HE-AAC v2.

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

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

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.

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 overhead of 8.3% – lead to an ability of correcting up to five erroneous bytes in those 120 bytes *(Fig. 9)*.
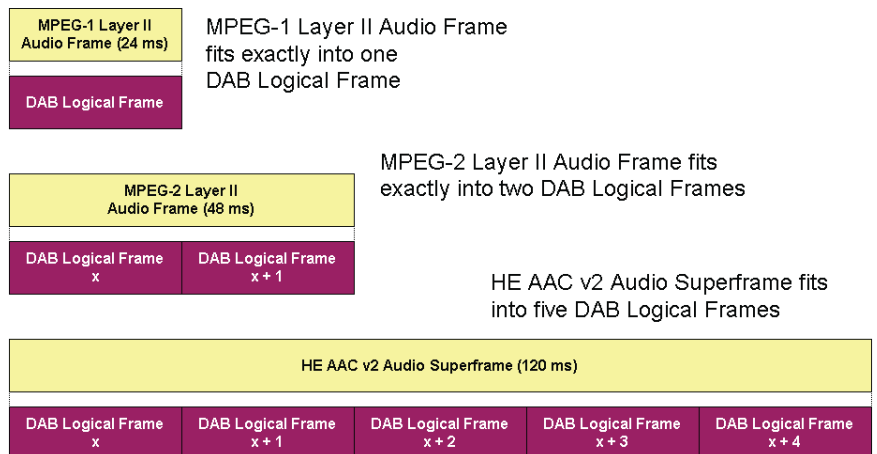
For test purposes, the new algorithms have already been implemented in both transmitting and receiving equipment. Therefore the step towards mass production is a small one for those who have already invested effort and resources in the standardization exercise.
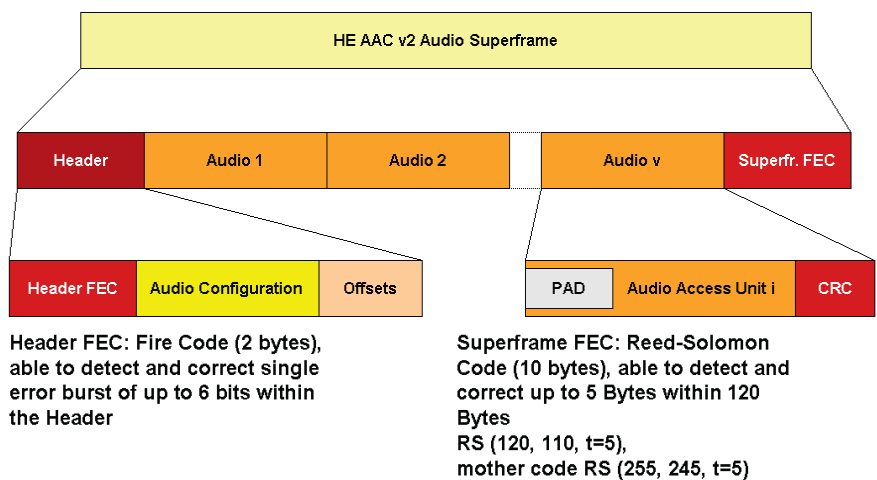
Features of DAB+

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

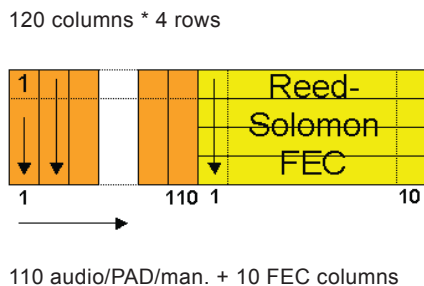- service following (e.g. to FM or other DAB ensembles);

- 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..

MPEG-1 Layer II Audio Frame fits exactly into one DAB Logical Frame

MPEG-2 Layer II Audio Frame fits exactly into two DAB Logical Frames

HE AAC v2 Audio Superframe fits into five DAB Logical Frames

Figure 7

DAB Logical Frame Alignement for Layer II and DAB+

Header FEC: Fire Code (2 bytes), able to detect and correct single error burst of up to 6 bits within the Header

Superframe FEC: Reed-Solomon Code (10 bytes), able to detect and correct up to 5 Bytes within 120 Bytes
RS (120, 110, t=5),
mother code RS (255, 245, t=5)

Figure 8

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

120 columns * 4 rows

110 audio/PAD/man. + 10 FEC columns

1. Fill in audio, PAD and management data columnwise from top to bottom and from left to right

2. Calculate RS parity bytes over 110 audio/PAD/management bytes (row-wise)

3. Transmit data from left to right, top to bottom for all 120 columns

Figure 9

Error-control code calculation and virtual interleaving in a 32kBit/s sub-channel

Abbreviations

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

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.

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.

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

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

Figure 10
C/N for threshold of audibility
(Source: Communications Research Centre Canada)

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 *(Fig. 10)*.

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*.

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 was known in the UK as BT Movio and was provided as a wholesale service by the incumbent telecom operator. This application put a source coding algorithm – that was not specified with an open standard – on top of IP. In fact, the whole application was 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.

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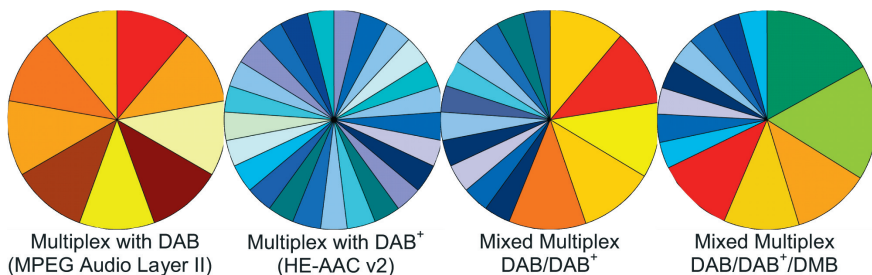
Altogether, this chain represents a classical combination of MPEG-4 elements transported by an MPEG-2 Transport Stream. BIFS, as one of those MPEG elements/norms, represents quite a powerful tool for data provision and interactivity.

3.2 BT Movio

Unlike DMB, BT Movio was not fully standardized by WorldDMB / ETSI, but made 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 applied protocols and source coding algorithms designed by Microsoft. It should be noted that, in the meantime, 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 unfragmented 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.

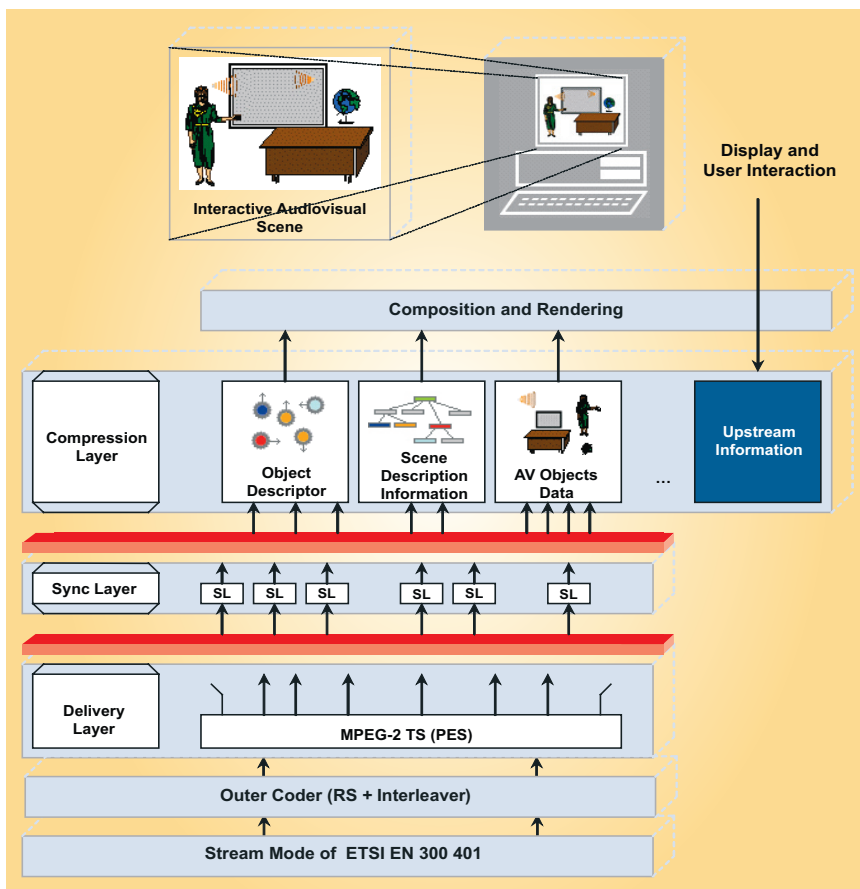**Figure 11**
Various implemantations of the DAB system

Multiplex with DAB (MPEG Audio Layer II)
Multiplex with DAB+ (HE-AAC v2)
Mixed Multiplex DAB/DAB+
Mixed Multiplex DAB/DAB+/DMB

Figure 12
Terminal processing chain for the DAB application, DMB
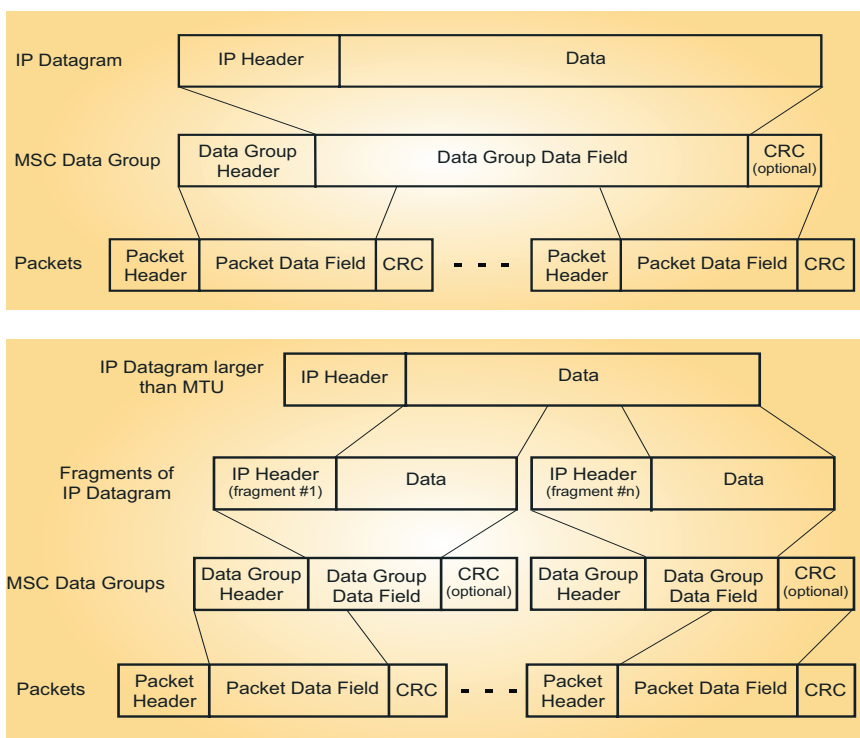

Figure 13
DAB IP Tunnelling

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 was absolutely vital to employ the second layer of error-control coding.

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

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

Since this article was first published the BT Movio service has closed down.

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.

Figure 14
HTC/Qtek Lobster, as used with the BT Movio DAB-IP service
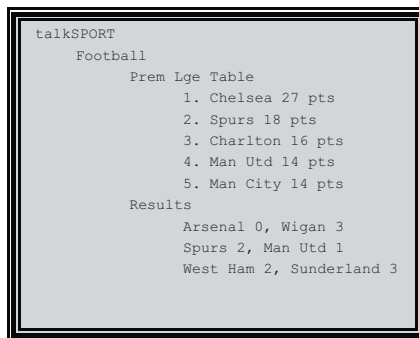
Figure 15
Exemple of display of Intellitext message data

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

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.

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.

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
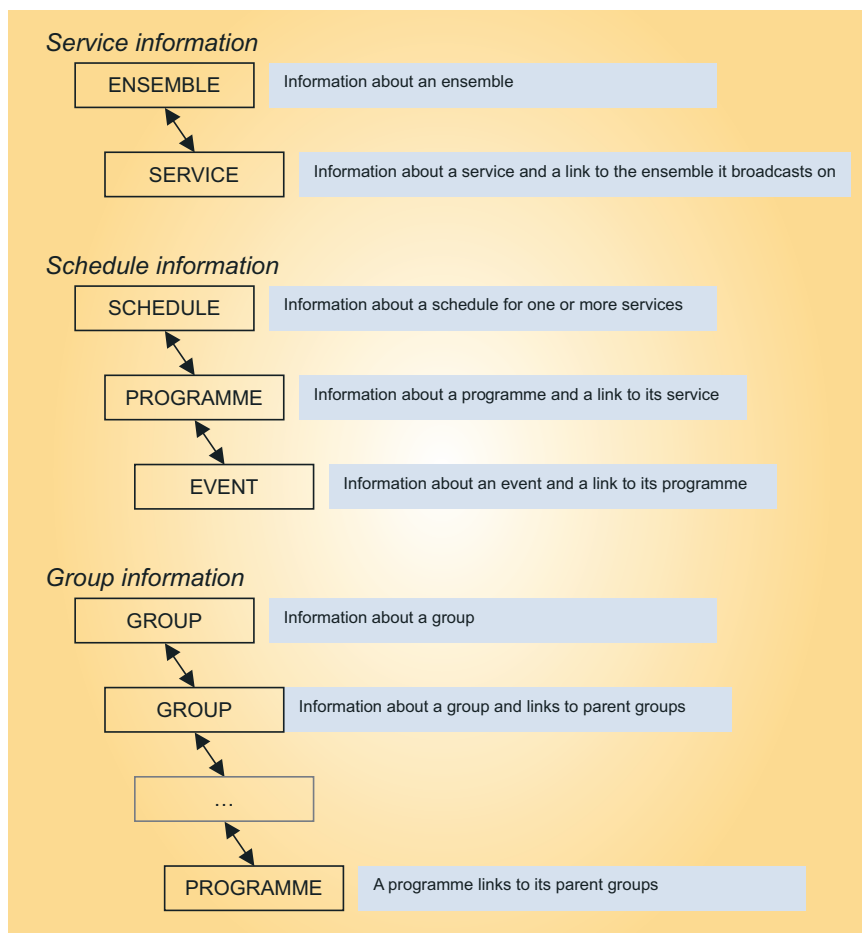
Figure 17
EPG entities and hierarchies

Figure 18
EPG-enabled DAB Receivers

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.

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.
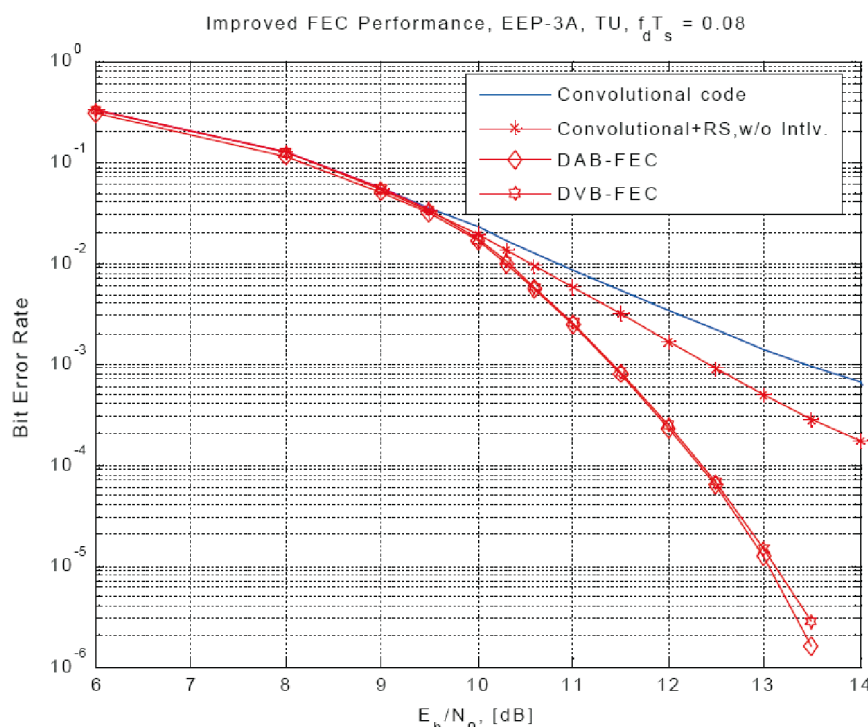
Figure 19
Error behaviour of DAB with cascaded error-control coding
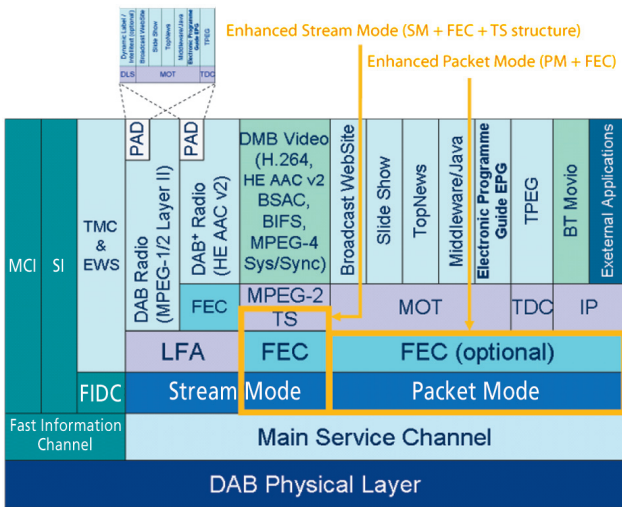
Figure 20
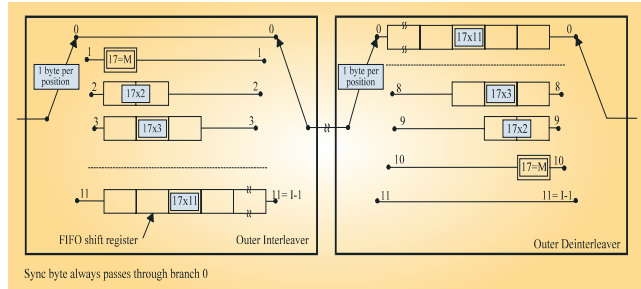**Position of Enhanced Stream and Packet**
Modes in the DAB protocol stack
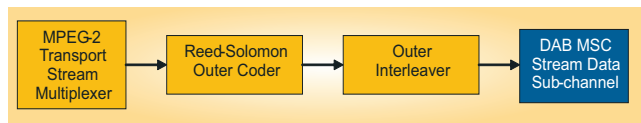
Figure 21
Adapting Transport Streams to existing DAB infrastructure

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

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.

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.
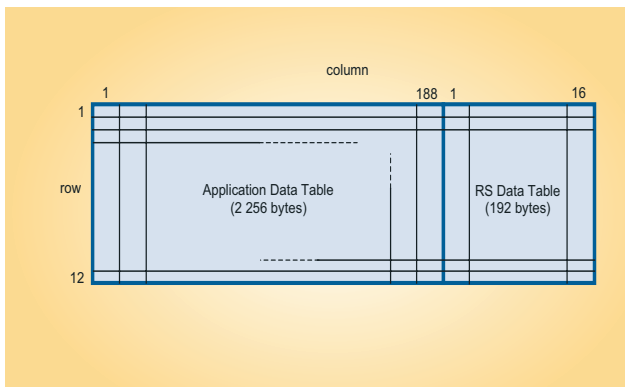
Figure 23
Building the FEC Frame
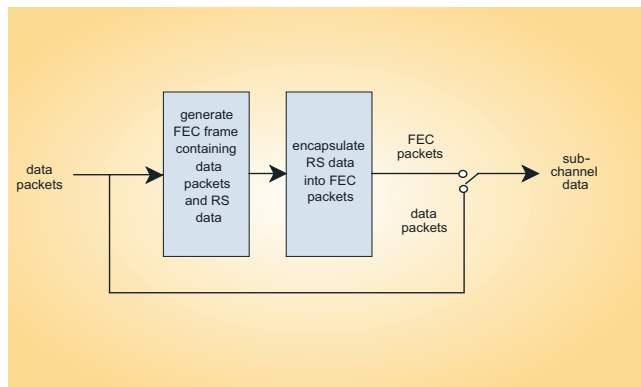
Figure 24
Adapting EPM to the existing DAB infrastructure

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

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%.

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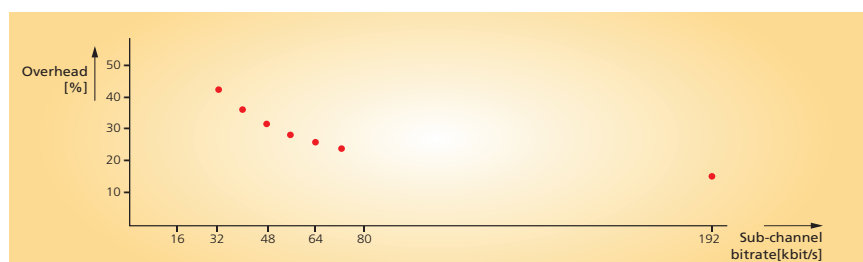
Figure 25
MPEG-2/MPEG-4 system layer and FEC overhead

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*.

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.

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: Syste m 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.

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HDMI & HDCP

– the manufacturers' perspective

Dietrich Westerkamp
Thomson, EICTA HDTV Issue Manager

HDTV signals offer great opportunities to broadcasters, but there is also the negative side – a high risk of piracy. In order to protect prime content against illegitimate use, content-protection mechanisms can be used.

For the digital HDMI interface between an HDTV set-top box and an "HD ready" display device, HDCP technology is chosen. This is a tool that can be used at the discretion of the broadcaster who can activate it by means of a switching signal. In the case of a piracy attack, the technology offers a revocation mechanism whereby a list of revoked devices is transmitted in a safe way to the receiver, where it is stored.

The availability of a content protection mechanism – being a mandatory requirement of the EICTA "HD ready" logo – does not mean that the display device always needs to be fed in a protected manner. Free-to-air signals that are transmitted in the clear are always displayed.

The high quality of digitally transmitted HDTV offers the broadcaster big opportunities – but also brings along some risks not to be neglected: pirates use the high-quality signals to illegally copy them and start their own business, thereby neglecting the copyright of the originator.

One of the links that are open to attacks is the digital baseband interface between a receiving set-top box (STB) and an HDTV display device. Here, either the *Digital Visual Interface* (DVI) [1] or the *High-Definition Multimedia Interface* (HDMI) [2] is in use. In order to protect high-quality digital signals on these interfaces, a technology called *High-bandwidth Digital Content Protection* (HDCP) [3] is used. The European CE industry association, EICTA [4], made HDCP part of their minimum requirements for an HD-capable display device that is labelled with the *HD ready* logo. This article explains the function of HDCP and the way it is implemented. It also highlights the different positions of European broadcasters concerning the control of the copy protection mechanism.

As of today, the application of any content-protection mechanism is mainly controlled by the content owner. The broadcaster or pay-TV operator is obliged by its licence contracts to ensure adequate content protection by switching on an appropriate mechanism, and the receiving/recording/displaying devices must have implemented it.

High-bandwidth Digital Content Protection (HDCP)

Fig. 1 sketches a digital transmission system for HDTV signals. The HDTV
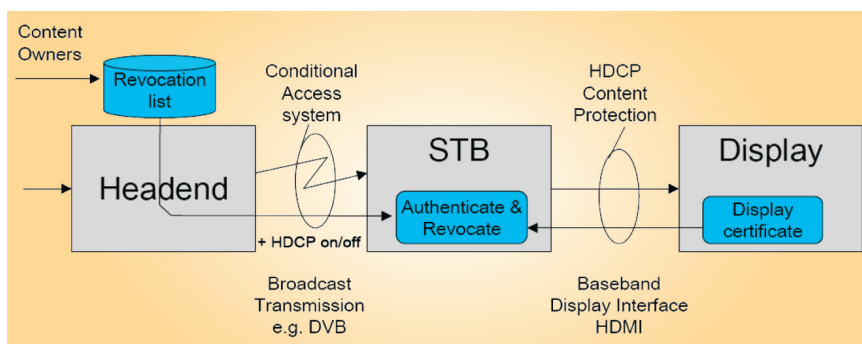
Figure 1
Concept diagram of a digital transmission system with Conditional Access and HDCP copy protection of the diplay interface

signal from the head-end is sent to a set-top box (STB). In many cases a Conditional Access (CA) system is used to enable the protection of the content as well as the subscription management. Once the STB has received and decoded the signal, it needs to be forwarded to a suitable display. In the case of HDTV signals, the digital connection between the STB and the display will be either HDMI or DVI *(Figs 2 and 3)*, with the former being the most up to date.

If the content owner requests the broadcaster to protect the content against piracy, there must be a mechanism in place that prevents someone from tapping the interface between the STB and the display and making an illegal copy.

For this purpose, the HDCP scheme has been developed. Using this mechanism, the content on the interface between the STB and the display device is scrambled in order to make it useless for pirates.

Once a display device is hooked up to a source device (here, the STB), an initial authentication/negotiation procedure between the source and the display is started. In the course of this authentication procedure, keys are exchanged and validated, and the scrambling mechanism is activated.

Authentication is also needed in order to have the possibility of taking action in case any of the devices involved have been compromised in a way that could be used for piracy. In those cases, the content owners can signal via so-called *revocation lists* that the compromised devices are black-listed and shall no longer be permitted to transport signals using the HDCP scrambling mechanism. By this method, content owners can render such devices useless and hence "plug the piracy holes".

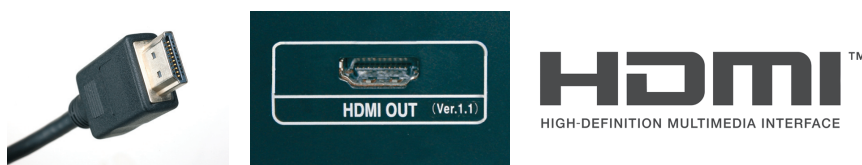The responsibility for putting together these revocation lists is with the content

owners. The broadcasters as well as the equipment manufacturers are obliged to transmit the lists and react accordingly, based on the licence contracts they have signed for using HDCP. In order to protect these lists from being tampered with on their way to the receiver, they are transmitted with a digital signature.
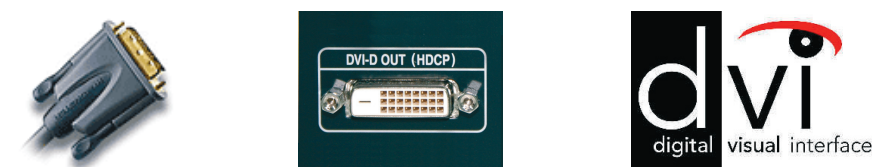
HDCP switchable, programme-by-programme

Content protection on the display interface may not be needed for all the programmes broadcast by a particular TV channel; there may even be TV channels that do not request any content protection. In those cases, the HDCP mechanism can be switched off and the content can be transmitted in the clear as a high-bitrate baseband video and audio signal.

At present, such a switching mechanism is realised within the different CA systems. In the same channel that transmits the programme in protected form, the information is transmitted to the STB whether any copy protection is needed on the display interface (DVI/HDMI).

There are currently various implementations in use that differ in their ways of controlling the HDCP on/off switch. It goes without saying that control over this switch is sensitive and will not be made available to all potential users of the STB … including a potential pirate!

The way it is used is defined by the operator who specified the set-top box. In fact, the implementation in most cases is part of the Conditional Access system implementation. Based on conditions set by the content owners, copy-control mechanisms are even wider than the simple on/off switching of HDCP on the digital interface. Almost all set-top boxes have analogue as well as digital outputs, including one or more SCART plugs for hooking up standard-definition devices.

In the case where HDCP on the digital interface is enabled (for protecting a

Figure 2
High-Definition Multimedia Interface (HDMI): plug, socket and logo *(courtesy of HDMI.org)*

Figure 3
Digital Visual Interface (DVI): plug, socket and logo *(courtesy of DDWG.org)*

proof device. One of the questions that needed to be answered was the necessity of implementing copy protection.

The European CE, IT and communications industry association, EICTA, defined the "HD ready" and "HD TV" logos *(Fig. 4)*. While HD ready defines the minimum requirements for display devices, the HD TV logo does the same for HDTV receiving equipment. Details can be found on the EICTA website [4].

high-quality HDTV signal), the analogue interfaces may behave in several different ways:

- They could be copy-protected by an analogue system but, at present, such systems only exist for standard definition;

- The HD component interface could be switched off, with only the SD interface (SCART) delivering a copy-protected SDTV signal;

- All analogue interfaces could deliver a signal but only in standard definition – sometimes this can even be recordable;

- All analogue interfaces could be switched off.

It is very important to note that the behaviour of the analogue interfaces is defined by the body that specifies the set-top box and has nothing to do with the HDCP mechanism described above. HDCP does not deal with any analogue signals.

Free-to-air content and copy protection

Almost all HDTV set-top boxes on the European market are put there by pay-TV operators. At the end of 2006, there were approximately 500'000 STBs in consumer households. This number is quickly heading towards one million boxes, as further HDTV services get launched in various European countries. An intense debate has occurred around the way these boxes should handle free-to-air content.

All DVB set-top boxes defined for pay-TV are also capable of receiving free-to-air content. In the case of HDTV, the decoded signal is fed to the display device preferably by the HDMI interface in order to best preserve the high quality of the pictures. But the free-to-air broadcasters currently have no influence to control the way HDCP is used (or not) on that interface. These rights are defined by the party that specified the set-top box – the pay-TV operator. That being said, there is also no obligation on free-to-air broadcasters to deal with the transmission of revocation lists.

In fact, the existing boxes in Germany, the UK and France handle the HDCP switching differently: some boxes leave HDCP on at all times whereas others switch HDCP on only for specific programmes such as first-run movies. In both cases, the free-to-air signals will be displayed on the connected HD-ready device and the viewer would not even know whether copy protection is active or not.

Obviously there is one exception ... once the display device has been misused for piracy activities and is consequently put on the revocation list, it will not receive any further images when HDCP is switched on.

HD ready and HD TV logos and copy protection

When HD-capable display devices became available on the market place, discussions started on which features needed to be implemented in order to have a future-

Figure 4
EICTA logos:
(left) **"HD ready"** for display devices and *(right)* **"HD TV"** for receiving devices

The HD ready minimum requirements include analogue as well as digital interfaces. The latter – which could be DVI or HDMI – necessarily needs to have HDCP implemented. This was made mandatory in order to ensure that the consumer will always see an HDTV picture, even if the broadcaster or content provider decides to use copy protection on the output of the receiving device.

After a lengthy debate, EICTA decided not to make HDCP mandatory for all receiving equipment. This pays tribute to the fact that, in future, there might be free-to-air receivers without any CA system that simply do not offer the technical means needed for HDCP implementation (i.e. a secure transmission channel for switching information and revocation lists).

When the first HD ready devices came on the market, there was a campaign in the technical press that the HD ready logo would simply be an industry action to have copy protection made mandatory in all cases. This definitely is not the case, because all interfaces always accept signals that are offered without copy protection. However, the logo assures the consumer that he will always see a picture ... unless
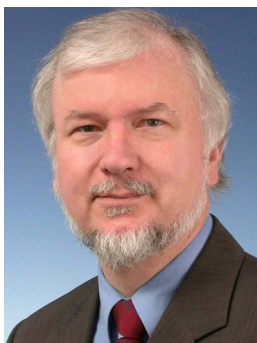
Dietrich Westerkamp graduated from the University of Hannover. He then worked for several years as a research assistant on data compression for images. Since 1985 he has been with Thomson, currently as Director Standards Coordination. Over the past 20 years, Mr Westerkamp has been involved in many projects to improve TV systems, ranging from HD-MAC and PALplus via DVC (the standard for digital camcorders) to MPEG and DVB. In the DVB Consortium, he has served for several years as an industry member of the Steering Board. Within EICTA, he chairs the HDTV issue group that developed the HD ready and HD TV product logos.

his display device has been misused for piracy action and has been revoked.

Handling of revocation lists

In the current implementations, the revocation list is stored in the STB. The receiving device gets the information via the broadcast channel as defined by the licensing authority, DCP LLC. Whenever a new version of the revocation list is issued, the information stored in the receivers will be updated.

Conclusions

The HDMI interface is the best choice for delivering HDTV content from a receiving device to a modern display device. It maintains the quality of the image at the highest possible level, by avoiding unnecessary cascaded A/D and D/A conversions. The high quality of the signal on the interface makes it a target for signal pirates to make illegal copies. HDCP is the means to prevent this.

EICTA has made HDCP part of the minimum requirements for HD ready display devices in order to assure the consumer that he will always get a high-quality HDTV picture on his display. It needs to be underlined here that the HDMI interface of the display also accepts non-copy-protected signals. Once the connected set-top box uses HDCP all the time, free-to-air content will also always be displayed, even if it is transmitted via the copy-protected link because the pay-TV operator who sponsored the set-top box has decided so.

There is an ongoing debate at the level of European standardization on whether there is a possibility of defining a secure switching mechanism that would allow every broadcaster to decide whether or not to activate HDCP. Looking at the current HD TV set-top boxes in the market place, it can be seen clearly that they all implement HDCP and are using different concepts on how to control the use of HDCP. Independent of that, all of these boxes can handle free-to-air signals and deliver them to the connected display. In all cases the consumer can enjoy the HDTV pictures … unless he has misused his display device for piracy actions and the device has been put on the revocation list. In that case, the screen will remain dark.

References

[1] DVI DDWG, "DVI Visual Interface", rev. 1.0, April 2, 1999 as further qualified in EIA/CEA-861 rev. B, "A DTV Profile for Uncompressed High Speed Digital Interfaces" May 2002 http://www.ddwg.org

[2] HDMI HDMI Licensing, LLC, "High-Definition Multimedia Interface", rev. 1.3, November 10, 2006 http://www.hdmi.org

[3] HDCP Intel, "High-Bandwidth Digital Content Protection System", rev. 1.2, June 13, 2006 http://www.digital-cp.com

[4] EICTA http://www.eicta.org

Note from the Editor

This article outlines the views of EICTA – the European CE equipment manufacturers association – on HD content protection using HDCP. The views of several European broadcasters are presented in a separate article published in this edition.

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HDCP

– the FTA broadcasters'perspective

Jean-Pierre Evain
European Broadcasting Union

The first HD services have now been deployed on pay-TV platforms using content-protection measures such as HDCP, in accordance with contractual obligations mandated by the production studios. Before long, free-to-air TV platforms will also become involved in HDCP.

This article provides technical information on the HDCP system, which is used to protect the HDMI link from a set-top box to a display device (HDMI is the HDTV equivalent of the familiar "SCART" connector used with standard-definition television). The article also explains "what HDCP is" and "what it is not", and outlines the views of several different European broadcasters on methods for controlling content protection.

HDCP over HDMI: a de facto standard

HDMI – which has now supersed DVI in consumer electronic products – is a high-bandwidth interface between an HDTV transmitter (e.g. a set-top-box) and an HDTV repeater/receiver (e.g. a display device). Such interfaces are often referred to as "display links", with DVI more commonly being found on personal computers. The HDMI interface can transmit HD digital video at bitrates up to 2.23 Gbit/s[1] at 720p or 1080i resolution, and up to eight channels of digital audio,

sampled at 192 kHz with 24 bits per sample.

Although technically challenging, HDMI is clearly of interest to pirates for accessing high-quality content sources in order to produce unauthorised copies. This is where HDCP comes in: it protects the content by encrypting the signal that is being carried over the HDMI (or, indeed, DVI) link to the display device.

HDCP is a proprietary technology from Intel Corporation, described in a specification that can be implemented under licence from the "Digital Content Protection LLC" (a subsidiary of Intel). The specification and licensing conditions can be found at **www.digital-cp.com.**

As shown in *Fig. 1*, up to 128 devices can be used simultaneously, provided that each

piece of equipment is (1) HDCP-compliant and (2) recognized (authenticated) as a valid secure implementation. In the context of broadcasting, the *Upstream Content Control Function* is the signalling information delivered from the broadcast stream (e.g. DVB's free-to-air signalling for content protection and copy management – CPCM).

HDCP is based on linear Authentication and Key Exchange (AKE), a process familiar to cryptologists. The AKE process involves the exchange of secret keys that are unique to each and every device. The authentication process assesses the validity of these keys including a revocation control. If the AKE process succeeds, content is encrypted by the transmitter over the link and delivered to the receiver which decrypts it according to rules securely set up during the authentication

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1. *The current version of HDMI has a maximum bitrate limit of 4.95 Gbit/s but that figure will be extended to 10.2 Gbit/s in a later version of the interface.*
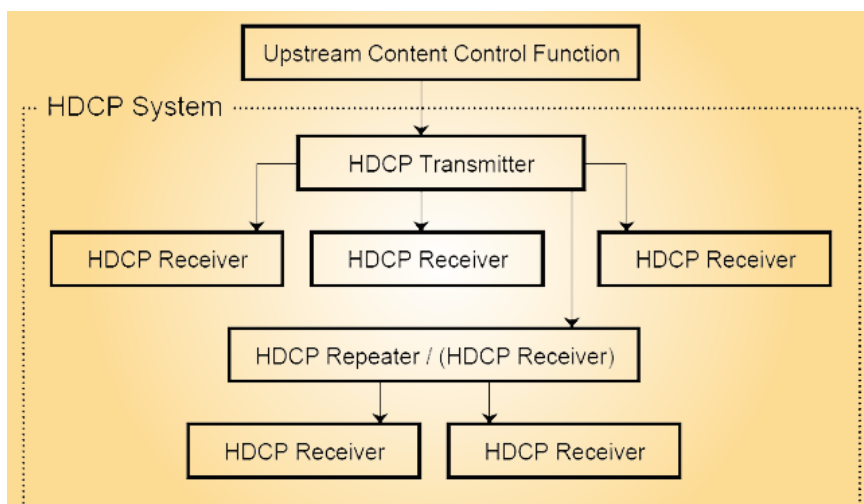
Figure 23
Building the FEC Frame

process, and displayed. If the AKE process fails, the display will probably remain black. Other options are possible such as downscaling the content resolution, which doesn't seem to be widely implemented today.

HDCP is a de facto standard as most manufacturers have licensed the technology from the Digital Content Protection LLC group and abide by contract to a certain number of implementation rules and obligations. DVB has adopted HDMI with HDCP as the associated protection mechanism. Furthermore, HDCP is mandated by EICTA in order to obtain the right to use the "HD Ready" logo.

HDCP content protection

Why?

The main reason for using HDCP is to prevent content being exposed and accessed in the clear, over high-bandwidth high-quality digital interfaces from which material could be extracted e.g. to produce unauthorised copies.

What?

HDCP is a security tool for "content protection". It is not a "copy management mechanism", used to carry and enforce usage restrictions. A copy management

mechanism may in turn require the use of security tools such as HDCP to "protect" content. The fact that HDCP is activated has no other meaning than "this content can only be accessed by compliant and authenticated devices" and shall not be subject to interpretation of derived usage restrictions (e.g. "copy never" or "do not redistribute over the Internet"). It is essential to understand, without any ambiguity, the precise nature and specific role of HDCP.

Example: Let's imagine an interface (e.g. other than HDCP) connecting a set-top box to a PVR. In the case where "copy never" applies to some content, a compliant PVR will not allow copying of this content, by means of deactivating the recording function. Conversely, content may be encrypted over the link between the two devices to prevent tampering with it for unauthorised copying purposes. However, although content might be protected over this link, e.g. if no copy restriction applies, it shall still be possible to make a copy of this content. Hence "content protection" is not the same as "copy management".

The actual usage restriction associated with the activation of HDCP is "unauthenticated access to content through this interface is not allowed". However, a content protection axiom would state that HDCP should be activated whenever content is subject to a usage restriction.

By whom?

The decision to apply or not any content protection and copy management is the decision of the content owner, which subsequently becomes a contractual obligation when content is licensed to service providers e.g. free-to-air or pay-TV broadcasters. Broadcasters are themselves often owners of the content that they produce and to which they may decide not to systematically, if at all, apply content protection and copy management. One should know the potential implications of activation or deactivation of HDCP on user access to "protected" content. The conditions under which HDCP might be used and how it might be used is subject to different circumstances and needs.

As a first example, this article focuses on free-to-air broadcasting but it is interesting to note that certain pay-TV operators wish to have the flexibility to activate HDCP on a content-by-content basis, while it is *deactivated* by default! Other pay-TV operators have specified their proprietary set-top boxes with HDCP being *activated* by default.

As far as free-to-air is concerned, different positions have been expressed that correspond to different market and regulatory situations:

Scenario 1

"Free-to-air" (FTA) or "clear-to-air" (CTA). In both cases, access is granted but limited to a particular geographical location when FTA content is delivered in scrambled form. FTA content that has been "protected" for delivery can remain protected after acquisition through the activation of HDCP, which could occur through signalling in the conditional access system (as for pay-TV), or by default in the receiver. There is also a need to be able to deactivate HDCP (and subsequently any similar content protection mechanism) for some content. Content could remain in the clear after geographical delivery unless otherwise instructed through proper "DVB free-to-air signalling information".

<div style="background:#F5A623">

Abbreviations

AKE — Authentication and Key Exchange

CPCM — (DVB) Content Protection and Copy Management

CTA — Clear-To-Air

DCP-LLC — Digital Content Protection LLC licensing group

DTCP — Digital Transmission Copy Protection

DVB — Digital Video Broadcasting
http://www.dvb.org/

DVI — Digital Visual Interface

EICTA — European Information, Communications and Consumer Electronics Technology Industry Association

FTA — Free-To-Air

HDCP — High-bandwidth Digital Content Protection

PVR — Personal Video Recorder

SRM — System Renewability Message

</div>

Scenario 2

For CTA content delivered in the clear, some EBU members want HDCP being deactivated by default on CTA-capable devices. If a set-top box gives access to CTA content and pay-TV content, independently of each other, it should be possible to activate or deactivate HDCP according to the default state originally set unless otherwise instructed through proper "DVB free-to-air signalling information". HDCP deactivation should preferably be the default condition for such CTA set-top boxes in a horizontal market.

Scenario 3

Some CTA broadcasters would prefer HDCP being activated by default with the flexibility to deactivate it for certain content through proper "DVB free-to-air signalling information".

Scenario 4

If FTA/CTA content is delivered as part of a pay-TV service to pay-TV set-top boxes, the default HDCP state will be defined by the pay-TV operator as well as the possibility and mechanisms to activate or deactivate HDCP.

The above valid, but diverse, scenarios illustrate the need for HDCP (and similar content protection mechanisms) to be switchable on a content-by-content basis from one initial state (either "on" or "off" by default) to another.

When?

It seems logical to activate HDCP content protection when usage restrictions – such as limited access, copying, redistribution and consumption – apply, because unauthenticated access to content in the clear would allow circumventing these restrictions.

Conditional Access (CA) systems can play the role of Upstream Content Control Function that activates or deactivates HDCP content protection. In some cases, the simple fact that content is delivered in a scrambled form is sufficient to require the activation of HDCP. In other CA configurations, the same channel also carries usage restriction messages, which allows more flexibility such as the activation of HDCP on a content-by-content basis in set-top boxes with HDCP "off" by default, or for deactivating HDCP for FTA content after acquisition.

DVB considers that CTA content shall be considered as "protected" as long as DVB free-to-air signalling is delivered alongside this content within the broadcast stream. DVB has specified free-to-air signalling to allow or prevent:

1) the redistribution of content over the Internet (control_remote_access_over_the_internet);

2) the scrambling of content (do_not_scramble);

3) the use of revocation lists (do_not_apply_revocation).

If the "do_not_scramble" flag is set to "true", HDCP should be deactivated. It is acknowledged that, although originally designed to control DVB Content Protection and Copy Management (DVB CPCM) scrambling, this signalling should equally apply to HDCP and similar protection mechanisms independently of the implementation of DVB CPCM.

But when does it really become essential to control content protection over a high-bandwidth "display link"? The answer to that question lies principally in two key implementation features of HDCP, i.e. legacy compliance and revocation.

HDCP compliance

In a perfect world where all devices are HDCP compliant, the "normal" honest user experience would be unaffected by content flowing over the HDCP interface in a scrambled form or not. But there will be a legacy of early adopters with displays without HDCP or, not to be underestimated, displays with "early and not fully-compliant" HDCP implementations.

One of the reasons pay-TV operators switch HDCP "off" by default may have been to ensure access to owners of early displays and to overcome potential early interoperability problems.

FTA broadcasters should share the same concern.

The evolution of the HDCP specification might generate a new legacy … and, in particular, a greater interoperability challenge – managing the "revocation" lists.

The revocation dilemma

In a fully HDCP-compliant world, having protection "on" by default wouldn't be such an issue if there weren't the additional burden of revocation which, in turn, would be less problematic if

managed on a content-by-content basis as recommended by DVB. But HDCP (and other similar protection mechanisms such as DTCP) currently makes this more complicated.

Revocation consists of identifying devices that have been compromised and could be misused as a sink to access content and generate unauthorised copies. A device is "compromised' when (1) a device private key has been cloned and replicated in pirate devices or (2) the private key of that device has been made public (e.g. after being lost or stolen).

"Compromised" devices are identified by their individual keys, compiled into revocation lists which are typically distributed with the content (in the signal or with removable medias) in signed / authenticated "System Renewability Messages" (SRMs) but can also be embedded into new devices. This list is consulted during the HDCP authentication procedure and although the AKE process is successful, a device would not be granted access to content if blacklisted.

The Content Participant Agreement defines the conditions under which content owners who have signed the agreement may request revocation of devices. The responsibility for putting together these revocation lists is with the content owners. Broadcasters are obliged to transmit the lists and react accordingly by the licence contracts they have signed for HDCP.

Although version 1.1 of the HDCP specification was not specific about revocation list management, version 1.2 defines a "device-based" revocation mechanism. This means that revocation lists must be permanently stored on devices. Revocation lists are updated each time a device receives a more recent list either with the content or when interconnected with another device (e.g. a new device with a preloaded revocation list) either directly or through a home network. According to this specification, revocation is "per device" and not "per content".

SRMs are signed using a public key delivered by the Digital Content Protection LLC group. They do not require particular protection to be transmitted. FTA/CTA broadcasters should be asked to collaborate in the delivery of such lists if they require the activation of HDCP.

A buffer of 5 KBytes restricts the number of keys that can be stored in a device to one Vector Revocation List (the individual 40-bit keys of 128 devices), which has a limiting effect on the bandwidth needed to carry the SRMs and its cost for broadcasters. One key of one device can actually deactivate thousands of devices sharing a compromised key.

Crypto-analysis has demonstrated that HDCP could be considered "broken" if 40 keys are compromised. A new version is in preparation, which would justify the handling of more than 128 devices, as envisaged in the HDCP specification. But the use of this new version may raise compatibility and legacy issues.

Why is device revocation dangerous for FTA broadcasters?

If a receiving device that gives access to both free-to-air and pay-TV services has been instructed to blacklist some equipment (e.g. a display) for pay-TV content, then "per device" revocation would result in turning the screen black for pay-TV but also free-to-air services. In this context, the black-screen threat is not in favour of HDCP being set "on" by default. However, a solution has been agreed within DVB by defining the free-to-air signalling flag "do_not_apply_revocation", which allows deactivating revocation on a "per content" basis for the associated FTA/CTA content. Obviously, this solution requires being implemented by HDCP to be effective.

Summary

Like pay-TV operators, FTA/CTA broadcasters across Europe see different possible uses of HDCP but would like the flexibility to activate or deactivate it on a "per content" basis. This is a requirement already endorsed by DVB for more generic "content protection and copy management".

HDCP is only "content protection" and not a "copy management" scheme. Usage restrictions cannot be derived or interpreted from the activation of HDCP but, in principle, HDCP would be activated when usage restrictions apply to content.

Jean-Pierre Evain joined the EBU's Technical Department in 1992 to work on "New Systems and Services", having spent six years in the R&D laboratories of France-Télécom (CCETT) and Deutsche Telekom. Mr Evain manages all EBU metadata activities. He represents the EBU in several DVB groups regarding metadata as well as Copy Protection and Digital Rights Management. He also represents the EBU in the IPTC consortium (news metadata).

HDCP is a de facto standard that has been implemented differently in various proprietary implementations for pay-TV. Meeting the needs of "FTA" broadcasters in the long term, in a horizontal market, may require some adaptation to those currently developed for pay-TV.

In a fully HDCP-compliant world, having content protection "on" by default would not be a problem, notwithstanding the additional burden of revocation. This in turn would be less problematic if managed on a "per content" basis. But HDCP (and other similar protection mechanisms such as DTCP) has opted for "device-based" revocation. In such conditions, pay-TV set-top-boxes that are revoked to protect pay-TV premium content will no longer deliver FTA content to users unless using the DVB FTA switching flag. This must not prevent FTA broadcasters being involved in the revocation decision-making process – to counter-balance the market impact of such actions. FTA broadcasters would be asked to collaborate in the delivery of revocation messages if they require the activation of HDCP.

DVB has agreed a "free-to-air signalling scheme", which offers a solution to several of the key issues mentioned in this article and, more particularly, concerning HDCP activation and "per content" revocation. It is strongly advised that future HDCP implementations respond to such signalling, if not already.

One issue of serious concern to potential FTA broadcaster-users of HDCP is the lack of stability of the specification. The specification has already changed from version 1.1 to version 1.2 and 1.3. There are critical legacy and interoperability issues. The value of HDCP will be weakened if the specification and compliance rules are being changed without open consultation.

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Note from the Editor

This article outlines the views of several European broadcasters on HD content protection using HDCP. The views of EICTA – the European CE equipment manufacturers association – are presented in a separate article published in this edition.

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Multiple Description
Coding

– a new technology for video streaming over the Internet

Andrea Vitali
STMicroelectronics

The Internet is growing quickly as a network of heterogeneous communication networks. The number of users is rapidly expanding and bandwidth-hungry services, such as video streaming, are becoming more and more popular by the day. However, heterogeneity and congestion cause three main problems: unpredictable throughput, losses and delays. The challenge is therefore to provide: (i) quality, even at low bitrates, (ii) reliability, independent of loss patterns and (iii) interactivity (low perceived latency) ... to many users simultaneously.

In this article, we will discuss various well-known technologies for streaming video over the Internet. We will look at how these technologies partially solve the aforementioned problems. Then, we will present and explain *Multiple Description Coding* – which offers a very good solution – and how it has been implemented and tested at STMicroelectronics.

Packet networks [1] [2]

Heterogeneity adds up with errors and congestion: backbone and wired links have an increasing capacity while, at the same time, more and more low-bandwidth error-prone wireless devices are being connected.

Throughput may become unpredictable. If the transmission rate does not match the capacity of the bottleneck link, some packets must be dropped. The delivery system may provide prioritisation: the most important packets are given a preferential treatment, while the least important packets are dropped first. However, usually networks will drop packets at random. Packet loss probability is not constant; on the contrary, it can be wildly varying, going from very good (no loss) to very bad (transmission outages).

This makes the design of the delivery system very difficult. Usually there are two options:

- the system can be designed for the worst case;

- or it can be made adaptive.

If it is designed for the worst case, it will be inefficient every time the channel is better than the worst case, i.e. most of the time. Conversely, if it is designed to be adaptive, it will most probably adapt too late.

Data-independent content delivery technologies

ARQ: Automatic Repeat reQuest

One of the most effective techniques for improving reliability is the retransmission of lost packets: Automatic Repeat reQuest, or ARQ. TCP-based content delivery is based on this.

If losses are sporadic, this technique is very efficient: packets are successfully sent only once. On the other hand, if losses are frequent, retransmissions can even increase congestion and also the loss rate, in a vicious cycle (this is avoided in TCP-based content delivery).

Retransmission is very useful in point-to-point communications where a feedback channel is available. However, when broadcasting to many receivers, the broadcaster cannot handle all the independent retransmission requests.

The added delay of the retransmission is at least one round-trip transport time. But each retransmission can also be lost, and the delay can be arbitrarily large. This is critical for streaming video: the delay of a retransmitted packet may be much longer than inter-arrival times and, as a consequence, streaming may suffer from stalls. This delay adds up in the receiver buffer which must be large enough to compensate for variations in the inter-arrival times (jitter).

FEC: Forward Error Correction / Erasure Recovery

Another very effective technique is channel coding, i.e. the transmission of redundant packets that allow recovery of erroneous / lost packets at the receiver side: Forward Error Correction / Erasure Recovery, or FEC.

If the loss rate is known, the added redundancy can be made just enough to compensate. Unfortunately, in the real world not only the amount of losses is not known, but also it is wildly time-varying. This, coupled with the fact that this technique has an all-or-nothing performance, makes its use very difficult: either errors are too much or they are less than expected.

If losses are too much, the recovery capability will be exceeded. Added redundancy will not be enough and the losses will not be recovered. Decoded quality will be very bad (cliff effect). Because of this, to be safe, broadcasters typically consider the worst case and choose to increase the amount of redundancy at the expense of the video. The video is compressed more heavily, lowering the final decoded quality.

If errors are less than expected, which is probable when the system is designed for the worst case, the losses will be recovered. The decoded quality will be guaranteed, unaffected by loss patterns. However capacity is wasted: less redundancy could be used leaving room for a higher-quality lightly-compressed video. Adaptation could be used in principle to dynamically balance the added redundancy and video compression, but it is rarely done because of the difficulty. Decoded quality is lower than it is possible to get.

The complexity can be very high: encoding and decoding of redundant packets requires memory and computational power. Efficient schemes for error correction / erasure recovery require processing of a large number of video packets. Therefore the added delay is not arbitrarily large, but it can be significant.

Data-dependent content delivery technologies

Robust source coding

The more efficient the video encoder, the more important a video packet is. When compression efficiency is very high, the loss of a packet has potentially a devastating effect. Then, a heavy recovery mechanism, such as complex FEC codes, must be used to reduce the probability of this happening. Conversely, when the compression efficiency is low, the loss of a packet has little effect. In this case, concealment techniques do exist that can reduce or even completely hide the effect of the loss. Consequently, a light recovery mechanism can be used.

Therefore, compression efficiency should be tuned carefully, taking into account the effect of losses, the effectiveness of concealment techniques and the effectiveness of the recovery mechanism. The available bandwidth can then be optimally split between the video data and redundant data.

Said in other words, it is always useful to optimize the parameters of the source encoder and of the channel encoder jointly (a technique known as "joint source-channel coding"). In the case of multimedia communications, this means exploiting the error resilience that may be embedded in compressed multimedia bitstreams, rather than using complex FEC codes or complex communication protocols.

Video encoders use a bunch of techniques to efficiently squeeze the video: prediction (also known as motion estimation and compensation), transform, quantization and entropy coding. Prediction is one of the most important techniques from the point of view of compression efficiency: the current video is predicted from the previously transmitted video. Because of this, video packets are dependent on previous packets. If these packets have not been successfully received, then the current packet is not useful. This is known as loss propagation. Compression efficiency can be a trade-off for robustness by reducing the amount of prediction (i.e. more intra coding): dependencies will be reduced, stopping the loss propagation effectively.

Transmission delay can also be a trade-off for robustness. Video packets can be reorganized (in so-called "interleaving buffers") so that consecutive video packets do not represent neighbouring video data. This is done to delocalise the effect of losses and ease the concealment. A long burst of lost packets will affect portions of the video which are far apart from each other. Lost portions can then be concealed effectively by exploiting neighbouring video data.

Concealment is usually done blindly at the receiver side. However, the transmitter

can encode hints (concealment data) that increase its effectiveness. Obviously this consumes part of the available bandwidth.

All these techniques are very effective, but it is very difficult to choose an optimal set of parameters. It is especially difficult when there are many receivers which experience different channel conditions.

Multiple Description Coding [3] [4]

Multiple Description Coding (MDC) can be seen as another way of enhancing error resilience without using complex channel coding schemes. The goal of MDC is to create several independent descriptions that can contribute to one or more characteristics of video: spatial or temporal resolution, signal-to-noise ratio, frequency content. Descriptions can have the same importance (balanced MDC schemes) or they can have different importance (unbalanced MDC schemes).

The more descriptions received, the higher the quality of decoded video. There is no threshold under which the quality drops (cliff effect). This is known as "graceful degradation".

The robustness comes from the fact that it is unlikely that the same portion of the same picture is corrupted in all descriptions. The coding efficiency is reduced depending on the amount of redundancy left among descriptions; however channel coding can indeed be reduced because of enhanced error resilience. Experiments have shown that Multiple Description is very robust: the

delivered quality is acceptable even at high loss rates.

Descriptions can be dropped wherever it is needed: at the transmitter side if the bandwidth is less than expected; at the receiver side if there is no need, or if it is not possible to use all descriptions successfully received. This is known as "scalability". It should be noted that this is a side benefit of Multiple Description Coding which is not designed to obtain scalability; instead it is designed for robustness.

Descriptions of the same portion of video should be offset in time as much as possible when streams are multiplexed. In this way a burst of losses at a given time does not cause the loss of the same portion of data in all descriptions at the same time. If interleaving is used, the same criterion is to be used: descriptions of the same portion of video should be spaced as much as possible. In this way a burst of losses does not cause the loss of the same portion of data in all descriptions at the same time. The added delay due to the amount of offset in time, or the interleaving depth, must be taken into account.

Layered Coding

Layered Coding (LC) is analogous to Multiple Description Coding (MDC). The main difference lies in the dependency. The goal of LC is to create dependent layers: there is one base layer and several enhancement layers that can be used, one after another, to refine the decoded quality of the base layer.

Layers can be dropped wherever required but they cannot be dropped at random:

the last enhancement layer should be dropped first, while the base layer must never be dropped. If the base layer is not received, nothing can be enhanced by the successive layers. Layered Coding is designed to obtain this kind of scalability.

Repair mechanisms are needed to guarantee the delivery of at least the base layer. Moreover: because of the unequal importance of layers, repair mechanisms should unequally protect the layers to better exploit Layered Coding. However not all networks offer this kind of services (prioritization).

Recovery mechanisms and Layered / Multiple Description Coding

Channel coding is needed by Layered Coding. However channel coding can also be used with Multiple Description Coding. Generally speaking, it is better to adapt the protection level of a given description / layer to its importance, a technique commonly known as "unequal error protection".

Unequal error protection is better even in the case of equally-important descriptions (balanced MDC). In fact, armouring only one description may be more effective than trying to protect all descriptions. If this is done, there is one description which is heavily protected. If the channel becomes really bad, this description is likely to survive losses. Then the decoder will be able to guarantee a basic quality, thanks to this description.

Summary of reviewed technologies and their characteristics

To summarize, here is an overview of the technologies that can be used for video streaming over packet networks, to compensate for losses due to errors and congestion:

Abbreviations

ARQ	Automatic Repeat reQuest
FEC	Forward Error Correction
IF-PDMD	Independent Flux – Polyphase Downsampling Multiple Description
LC	Layered Coding
MD	Multiple Description
MDC	Multiple Description Coding
TCP	Transmission Control Protocol

Data-independent content delivery technologies

- **Automatic Repeat Request** (ARQ): suitable only for point-to-point, needs feedback, added delay arbitrarily large.

- **Forward Error Correction** (FEC): no feedback required, all-or-nothing performance (cliff effect), waste of capacity when tuned for worst case, complexity, significant added delay.

Data-dependent content delivery technologies

- **Robust Source Coding**: difficult to choose optimal parameters

- **Multiple Description Coding** (MDC): no cliff effect (graceful degradation), no prioritisation needed, allows scalability, very robust even at high loss rates

- **Layered Coding** (LC): requires prioritisation and recovery mechanisms, allows efficient scalability

It should be noted that packet networks are designed to deliver any kind of data: a data-independent technique is therefore always needed. The best option is Forward Error Correction / erasure recovery (FEC).

For multimedia data, such as video (and audio as well), several smart techniques exists. In this case the best option is Multiple Description Coding (MDC).

Standard-compatible Multiple Description Coding [6] [8]

Losses due to errors and congestion do cause visible artefacts in decoded video: loss concealment techniques may help, but they are rarely effective, as can be seen in *Fig. 1*. This explains the need for an effective technique to recover losses and/or ease the concealment.

Figure 1

On the left, errors are not concealed. On the right, state-of-the-art concealment has been applied

Automatic Repeat reQuest (ARQ) is suitable only for point-to-point communications and cannot be easily scaled to broadcast scenarios; furthermore, it requires a feedback channel which may not be available. FEC is effective only if complex (which means: more power, delay, etc) and it has a threshold which yields an all-or-nothing performance (the cliff effect).

Robust source coding is difficult to use, as parameters are difficult to be tuned. Layered Coding is not designed for robustness and relies on the aforementioned recovery mechanisms. Conversely, Multiple Description Coding does not require a feedback channel and does not have an all-or-nothing behaviour: instead it has graceful degradation (more descriptions, more quality), plus it offers free scalability (to transmit as many descriptions as possible, receive as many as needed).

The question is: if Multiple Description Coding does serve the purpose well (robustness, effectiveness), then what is the price to be paid when implementing this solution (efficiency, bandwidth, quality, complexity, compatibility with legacy systems).

Standard compatibility

It is not easy to design and implement a Multiple Description video coding scheme. There are many established video coding standards deployed in the real world: e.g. MPEG-2, MPEG-4, H.263 and H.264. It is difficult to impose yet another standard which is more complex.

There are many other techniques available for creating multiple descriptions: multiple description scalar or vector quantization, correlating transforms and filters, frames or redundant bases, forward error correction coupled with layered coding, spatial or temporal polyphase downsampling (PDMD).

The best choice can be found by following this criteria:

- **Compatibility**: the possibility to use standard encoders for each description and the possibility of being compatible with legacy systems;

- **Simplicity**: minimum added memory and computational power;

- **Efficiency**: for a given bandwidth and when there are no losses, the minimum loss of decoded quality with respect to the best quality delivered by standard coding.

Among the aforementioned techniques, polyphase downsampling is particularly interesting as it is very simple and it can be easily implemented using standard state-of-the-art video encoders.

The sequence to be coded is subdivided into multiple subsequences which can then be coded independently. This is done in a pre-processing stage

(Fig. 2). At the decoder side, there is a post-processor stage *(Fig. 3)* in which decoded subsequences are merged to recreate the original one. This simple scheme is also known as "Independent Flux Polyphase Downsampling Multiple Description" coding (IF-PDMD).

This scheme is completely independent of the underlying video encoder.

Subdivision to create descriptions can be done along the temporal axis (e.g. by separating odd and even frames) or in the spatial domain (e.g. by separating odd and even lines). As encoding of each description is independent from others, there can be slight differences in the decoded quality. When temporal subdivision is used a potentially annoying artefact may arise: the difference among odd and even frames may be perceived as "flashing".
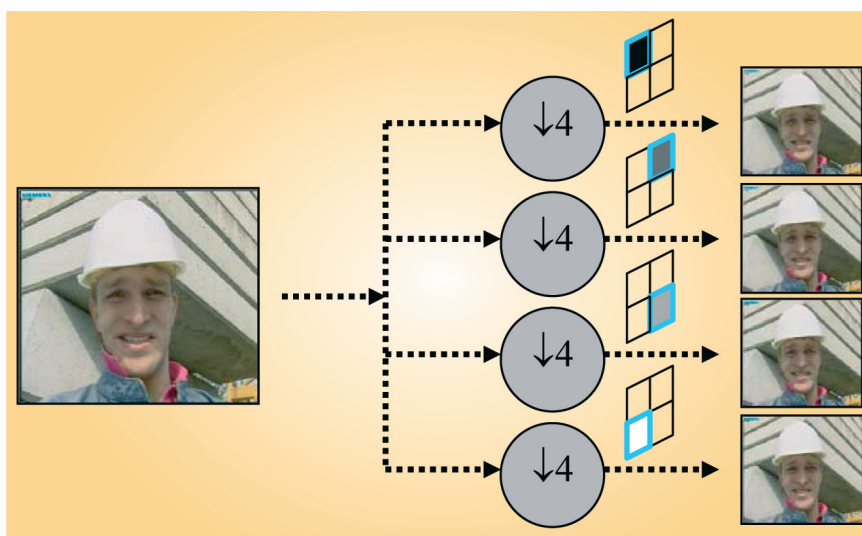
Figure 2

Pre-processing stage: downsampling in spatial domain. Odd and even lines are separated, the same is done for columns. Four descriptions are created.
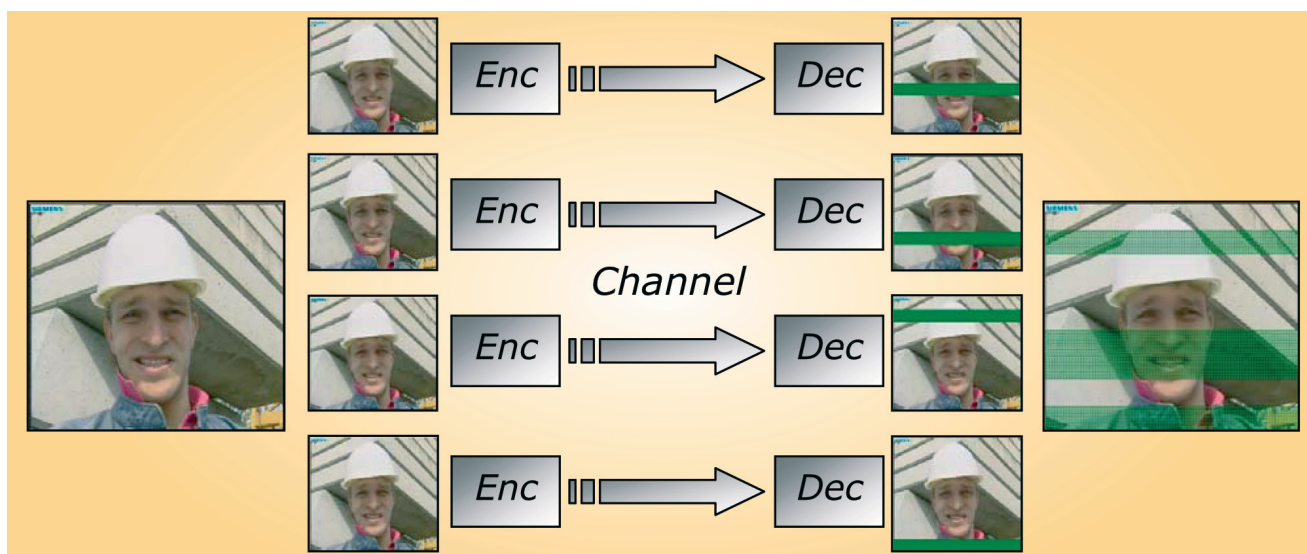
Figure 3

The whole chain: pre-processing, encoding, transmission, decoding, post-processing

On the contrary, when spatial subdivision is used *(see Fig. 4)*, a potentially pleasant artefact may arise: the difference between descriptions may be perceived as dithering, a known technique applied in graphics to hide encoding noise.

Spatial subdivision has two more advantages:

- Two descriptions can be created by separating odd and even lines: interlaced video is then reduced to

two smaller progressive video streams which may be easier to encode.

- Four descriptions can be created by separating odd and even lines, and then separating odd and even columns: high definition video (HDTV) is then reduced to four standard definition video streams which can be encoded using existing encoders.

It should be noted that keeping Multiple Description Coding decoupled from the

underlying codec prevents it from giving its best. To get maximum quality and to encode the descriptions with least effort, joint or coordinated encoding could be used. Also, to exploit the redundancy and to maximize the error resilience, joint Multiple Description decoding is recommended.

As an example, video encoders can share expensive encoding decisions (motion vectors) instead of computing them; also they can coordinate encoding decisions
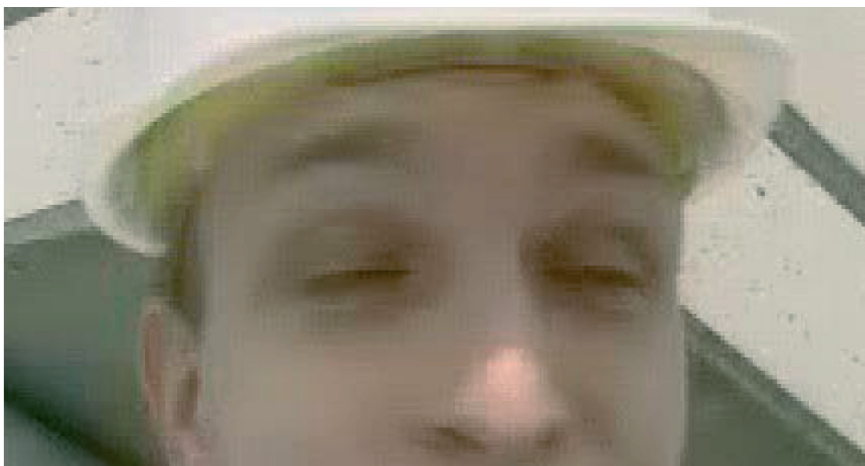
Figure 4
Dithering effect as a result of spatial downsampling: 4 descriptions are created by separating odd/even lines and taking every other pixel. As encoding of each description is independent from others, the decoded quality may differ slightly.
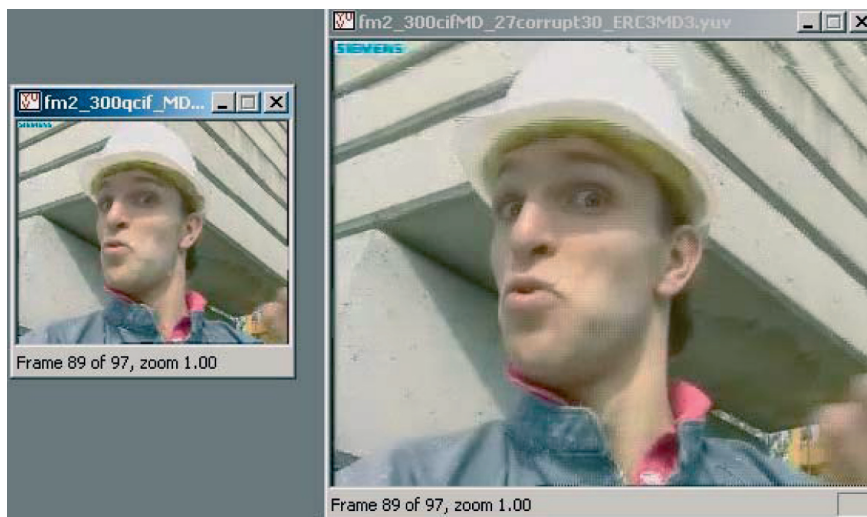
Figure 5
30% packet loss; *left*: **the output of a standard decoder, not aware of Multiple Description, has been instructed to see descriptions as replicas of the same packet (fake standard encoding);** *right*: **the output of a Multiple Description decoder.**

(quantization policies) to enhance the quality or resilience (interleaved multi-frame prediction policies, intra-refresh policies). Decoders can share decoded data to ease error concealment; also they can share critical internal variables (anchor frame buffer) to stop error propagation due to prediction.

It is worth mentioning that, if balanced descriptions are properly compressed and packed, any losses can be recovered before the decoding stage. In this case, decoders are preceded by a special processor in which lost packets are recovered

by copying similar packets from other descriptions. Similar packets are those that carry the same portion of video data.

The scheme is also compatible with systems not aware of Multiple Descriptions *(see Fig. 5)*.

In fact, each description can be decoded by a standard decoder, which need not be MD-aware in order to do this. Of course, if spatial MD has been used, the decoded frame has a smaller size … while if temporal MD has been used, the decoded sequence has a lower frame rate.

Moreover, MD encoding can even be beneficial. In fact, multiplexed descriptions can be marked so that old decoders believe that they are multiple copies of the same sequence.

As an example, when four descriptions are transmitted, the old decoder will believe that the same video packet is transmitted four times. Actually, they are four slightly different packets, but this does not matter. The decoder can be instructed to decode only the first copy and, if this copy is not received correctly, it can be instructed to decode another copy.

Why use Multiple Description Coding?

Firstly: increased error resilience. Secondly: we get scalability for free.

Robustness

Multiple Description Coding is very robust, even at high loss rates *(see Fig. 6)*. It is unlikely that the same portion of a given picture is corrupted in all the descriptions. It's as simple as that!

A more sophisticated point of view is to note that descriptions are interleaved. In fact, when the original picture is reconstructed, descriptions are merged by interleaving pixels. A missing portion in one description, will results in scattered missing pixels. These pixels can easily be estimated by using neighbouring available pixels.

It is assumed that errors are independent among descriptions. This is true only if descriptions are transmitted using multiple and independent channels. If one single channel is used instead, descriptions have to be suitably multiplexed. If this is done, error bursts will be broken by the demultiplexer and will look random, especially if the burst length is shorter than the multiplexer period.

Scalability

There are many scenarios where scalability can be appreciated. With mobile terminals
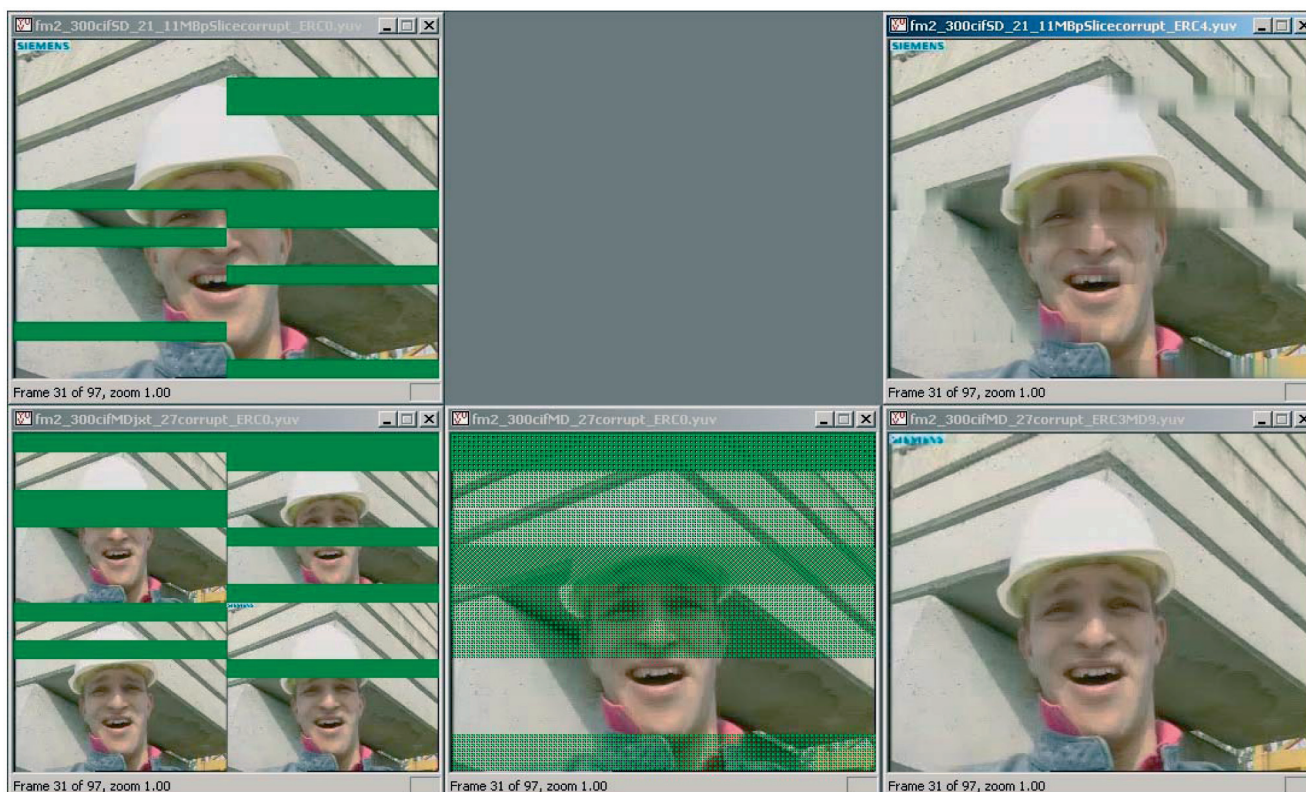
Figure 6
Same aggregate bandwidth, number of packets and average packet size, and with 30% packet loss rate.
Top row: **standard coding.** *Bottom row:* **four multiple descriptions generated by separating odd/even lines and taking every other pixel; before and after concealment.**

in mind, when standard coding is used, the whole bitstream should be decoded and downsized to adapt it to the small display. Power and memory are wasted. Conversely, when Multiple Description is used, a terminal can decode only the number of descriptions that suits its power, memory or display capabilities.

Also, when the channel has varying bandwidth, it would be easy to adapt the transmission to the available bandwidth. Descriptions may simply be dropped. Instead, a non-scalable bitstream would require an expensive transcoding (re-encoding the video to fit the reduced available bitrate).

This kind of scalability should be compared to the scalability provided by Layered Coding: think about losing the base layer while receiving the enhancement. It happens that the received enhancement is useless and bandwidth has been wasted. Usually, in order to avoid this, the base layer is given

a higher priority or is more protected than the enhancement layer.

When MD coding is used, there is no "base" layer. Each description can be decoded and used to get a basic quality sequence. More decoded descriptions lead to higher quality. There is no need to prioritise or protect a bitstream.

Finally, it must be noticed that at very low bitrates the quality provided by Multiple Description Coding is greater than that provided by standard coding. This happens because the low bitrate target can easily be reached by simply dropping all descriptions except one. On the contrary, with standard coding a rough quantization step must be used. Artefacts introduced

Figure 7
Site foreman, CIF resolution (352x288 pixels) at 155 kbit/sec using MPEG-4/10 encoder. Standard coding on the *left***, one out of four multiple descriptions on the** *right***.**

by heavy quantization are more annoying than artefacts introduced by dropping descriptions *(see Fig. 7)*.

Why not use Multiple Description Coding?

At a given bitrate budget, there is a quality loss with respect to standard (single description) coding. The loss depends on the resolution (the lower the resolution, the higher the loss) and on the number of descriptions (the more the descriptions, the higher the loss).

Descriptions are more difficult to encode. Prediction is less efficient. If spatial downsampling is used, pixels are less correlated. If temporal downsampling is used, motion compensation is not accurate because of the increased temporal distance between frames.

Also, syntax is replicated among bitstreams. Think about four descriptions. There are four bitstreams. Each holds data for a picture which has 1/4th the original size. When taken all together, the four bitstreams hold data for the same quantity of video data as the single description bitstream. The bit-budget is the same. However, the syntax is replicated, therefore there is less room for video data.

However, it must be noted that it is not fair to compare the decoded quality of Multiple Description Coding with standard (single description) coding – when there are no losses.

Standard coding has been designed for efficiency while Multiple Description Coding has been designed for robustness. If there are no losses, this increased error resilience is useless. A fair comparison would be to compare error-resilient standard coding with Multiple Description Coding. As an example, the standard bitstream can be made more error resilient by reducing the amount of prediction (increased intra refresh).

The intra refresh should be increased until the quality of the decoded video is
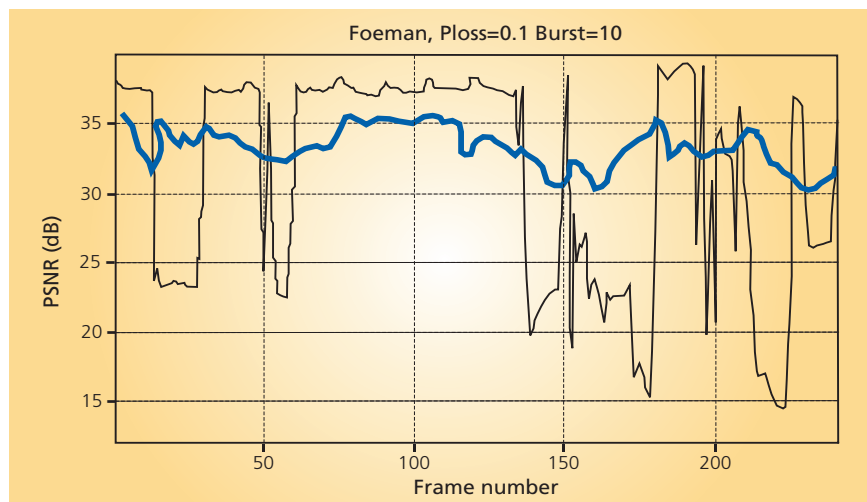
Figure 8

Quality frame-by-frame: the black line corresponds to standard coding protected by Reed-Solomon forward error correction (all-or-nothing behaviour),
the blue line corresponds to two Multiple Descriptions (slightly lower average quality, but much lower variance).

equal to the quality of decoded Multiple Description. Then it would be possible to evaluate the advantage of using Multiple Description by letting the packet loss rate increase and see which coding is better.

Experiments have shown [5] that Multiple Description is still superior when compared to error-resilient standard coding, even if the packet loss rate is very low (~1%). Simulations have been done at the same aggregate bitrate and same decoded quality using one of the most efficient FEC schemes: Reed-Solomon (R-S) codes *(see Fig. 8)*.

From a higher point of view, we might decide to reduce channel coding and use part of its bit-budget for Multiple Descriptions bitstreams, therefore increasing the quality of the decoded Multiple Descriptions.

Foreseen applications of Multiple Description Coding

- Divide-and-rule approach to HDTV distribution: HDTV sequences can be split into SDTV descriptions; no custom high-bandwidth is required.

- Easy picture-in-picture: with the classical solution, a second full decoding is needed plus downsizing; with MDC/LC, it is sufficient to decode one description or the base layer and paste it on the display.

- Adaptation to low resolution/ memory/power: mobiles decode as many descriptions/layers as they can – based on their display size, available memory, processor speed and battery level.

- pay-per-quality services: the user can decide at which quality level to enjoy a service, from low-cost low-resolution (base layer or one description only) to higher cost high-resolution (by paying for enhancement layers / more descriptions).

- Easy cell hand-over in wireless networks: different descriptions can be streamed from different base stations exploiting multi-paths on a cell boundary.

- Adaptation to varying bandwidth: the base station can simply drop descriptions/layers; more users can easily be served, and no trans-coding process is needed.

- Multi-standard support (simulcast without simulcast): descriptions can be encoded with different encoders (MPEG-2, H.263, H.264); there's no waste of capacity as descriptions carry different information.

- Enhanced carousel: instead of repeating the same data over and over again, different descriptions are transmitted one after another; the decoder can store and combine them to get a higher quality.

Application to P2P (peer-to-peer) networks

In P2P networks users help each other to download files. Each file is cut into pieces. The more popular a file is, the greater the number of users that can support a given user by transmitting the missing pieces.

Streaming however is a different story. The media cannot easily be cut into pieces, and in any case the pieces should be received in the correct order from a given user to be useful for the playout.

Also, a typical user has greater downlink capacity than uplink capacity. Therefore (s)he is not able to forward all the data (s)he receives and cannot help other users that are willing to receive the same stream.

One of the most effective solutions for live streaming has been implemented by Octoshape [7]. This is their scheme:

- A video that would require 400 kbit/s is split into four streams of 100 kbit/s each.

- Therefore N redundant 100 kbit/s streams are computed, based on the original four streams; the user is able to reconstruct the video given any four streams out of the available streams (the four original and the N redundant streams) – this can be done using an (N,4) Reed-Solomon FEC.

Following this scheme, the typical user is able to fully use the uplink capacity even if it is smaller than the downlink capacity. Each user computes and forwards as many redundant streams as possible, based on the capacity of its uplink.

A very similar scheme can be implemented using Multiple Description Coding:

- Four descriptions can be created by separating odd and even lines and taking every other pixel; each subsequence is encoded in 1/4 of the bitrate that would have been dedicated to full resolution video.

- Redundant descriptions can be created by further processing the video data; e.g., by averaging the four aforementioned descriptions, and so on. This is known as frame expansion.

Frame expansion can easily be explained by this simple example: 2 descriptions can be generated by separating odd and even lines as usual; a 3rd description can be generated by averaging odd and even lines. It is clear that perfect reconstruction (except for quantization noise) is achieved if any 2 descriptions out of 3 are correctly received. Frame expansion can be seen as equivalent to a Forward Error Correction code with rate 2/3: one single erasure can be fully recovered (except for the quantization noise). However, unlike FEC, there is no threshold: if there is more than one erasure, received descriptions are still useful. Moreover, the redundancy can be controlled easily by quantizing the third description more heavily.

Conclusions

Two data independent content delivery techniques have been presented: Automatic Repeat reQuest (ARQ) and Forward Error Correction (FEC). The latter is preferable as it does not require a feedback from receivers and is then suited to broadcast. However this technique has an all-or-nothing performance: when the correction capability is exceeded the quality of decoded video drops.

Three data dependent content delivery techniques have been presented: robust source coding, Multiple Description Coding (MDC) and Layered Coding (LC). The latter is also known as Scalable Video Coding (SVC) as it allows efficient scalability: layers can be decoded one after another, starting from the base layers; layers have different importance and require prioritisation

Andrea L. Vitali graduated in Electronics at the Milan Polytechnic in 1999. He then joined STMicroelectronics as a designer of digital multi-standard decoders for analogue TV. In 2000 he moved to the company's System Technology labs to work on real-time hardware prototyping for video algorithms. He has also worked on non-standard still picture / bayer-pattern compression and on Multiple Description video coding, and has published several papers on these topics. He holds patents granted in Europe and the USA in digital video processing, still picture compression, digital modulators and silicon sensors (automotive market).

Mr Vitali is now working in the field of robust source coding, joint source channel coding, adaptive multimedia play-out, metadata for multimedia signals, and graphical interfaces. He gave lectures on Digital Electronics at Pavia Polytechnic in 2002 and, since 2004, has also been an external professor at Bergamo University, Information Science department, where he is teaching Microelectronics.

which may not be available in the network. Robust source coding exploits the resilience that can be embedded in the bitstream by tuning coding parameters; however it is very difficult to optimize. Multiple Description Coding allows scalability (transmit or decode as many descriptions as possible), does not require prioritisation, it is very robust (it is unlikely to lose all descriptions) and has no all-or-nothing behaviour (decoded descriptions all contribute to decoded video quality).

A standard-compatible Multiple Description Coding scheme has been presented: descriptions are created by spatial downsampling in a pre-processing stage prior to encoding, they are merged after decoding in a post-processing stage. MDC performance has been compared to standard coding protected by state-of-the-art FEC: peak quality of the decoded video is lower but it is much more stable (absence of cliff effect). Several foreseen applications have been listed, including applications in peer-to-peer networks.

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Network structures – the internet, IPTV and

QoE

Jeff Goldberg and Thomas Kernen
Cisco Systems

How would a broadcaster transmit TV transported over IP packets rather than using traditional broadcast methods?

This article introduces a view of a generic Service Provider IP distribution system including DVB's IP standard; a comparison of Internet and managed Service Provider IP video distribution; how a broadcaster can inject TV programming into the Internet and, finally, how to control the Quality of Experience of video in an IP network.

Transport of broadcast TV services over Service Provider managed IP networks

The architecture of IP networks for the delivery of linear broadcast TV services looks similar to some traditional delivery networks, being a type of secondary distribution network. The major components are:

- Super Head-End (SHE) – where feeds are acquired and ingested;

- Core transport network – where IP packets route from one place to another;

- Video Hub Office (VHO) – where the video servers reside;

- Video Serving Office (VSO) – where access network elements such as the DSLAMs are aggregated;
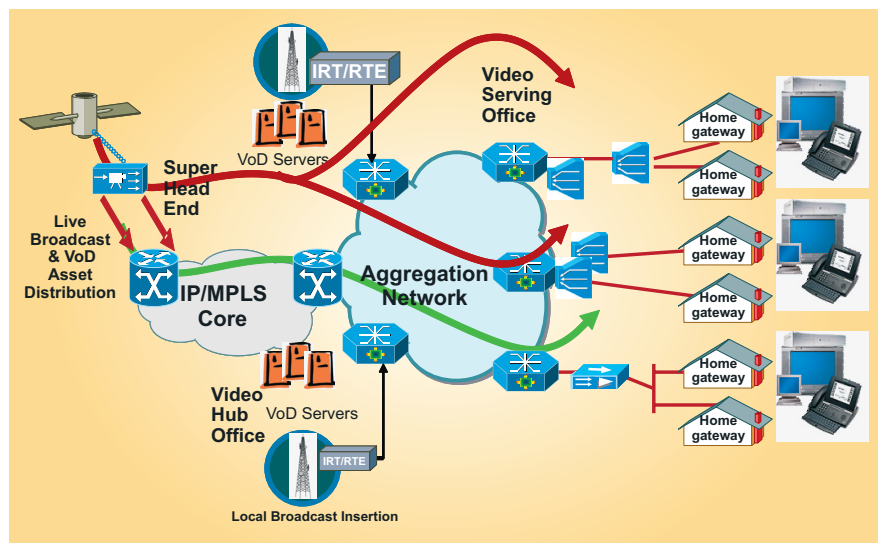
- Access network – which takes the data to the home – together with the home gateway and the user's set-top box (STB).

The whole network, however, is controlled, managed and maintained by a single Service Provider (SP) which allows him to control all the requirements

Figure 1
Broadcast TV over an SP-managed IP network

Thomas Kernen is a Consulting Engineer working for Cisco Systems' Central Consulting Group in Europe. He works on Video-over-IP with broadcasters, telecoms operators and IPTV Service Providers, defining the architectures and video transmission solutions. Before working for Cisco, he spent ten years with different telecoms operators, including three years with an FTTH triple play operator, for which he developed their IPTV architecture.

Mr Kernen is a member of the IEEE, SMPTE and active in the AVC group within the DVB Forum.

Jeff Goldberg is a Technical Leader working for a Chief Technology Officer within Cisco. He has been working on IPTV, IP STB design and home networking since 1999, and has been working for Cisco since 1994. He was part of the founding group of DVB-IPI and has been working on it ever since, particularly on the home networking, reliability, Quality of Service and remote management parts. Before working for Cisco he designed handheld devices and PC software.

needed to deliver a reliable service to the end point. These requirements are, for example, IP Quality of Service (QoS), bandwidth provisioning, failover paths and routing management. It is this management and control of service that separates a managed Service Provider IP delivery of video streams transported over the public Internet.

The Service Provider acquires the video source in multiple ways, some of which are the same as in other markets, such as DVB-S. This results in significant overhead as the DVB-S/S2/T/C IRDs and SDI handoffs from the broadcasters form a large part of the acquisition setup. It is therefore preferable to acquire content directly from another managed network using IP to the head-end, something that is more efficient and becoming more common.

Once the content has been acquired, descrambled and re-encoded, it is then carried as MPEG-2 Transport Streams (TS) encapsulated into IP packets instead of the traditional ASI. The individual multicast groups act as sources for the services which are then routed over the infrastructure, though in some highly secure cases, these may go through IP-aware bulk scramblers to provide content protection. If security is important, then routers at the edge of the SHE will provide

IP address and multicast group translation to help isolate the head-end from the IP/MPLS core transport network.

The core network lies at the centre of transporting the stream to its destination but it is the recent developments of high speed interfaces that have made it possible. The low cost and widely available Gigabit Ethernet, the more expensive 10 Gigabit Ethernet and the swift 40 Gigabit interface now provide the ability for the core to transport both contribution and distribution video streams. The modern optics used in these interfaces deliver Bit Error Rates (BERs) and latency that is lower than those of traditional transports such as satellite. These advantages, combined with an application layer Forward Error Correction (FEC) scheme – such as the Pro-MPEG Forum Pro-MPEG Code of Practice 4 (CoP4) and IP/MPLS Traffic Engineering (TE) – allow for redundant paths across the transport infrastructure. These paths can be designed in such a way that the data flows without ever crossing the same node or link between two end points, and delivers seamless failover between sources if the video equipment permits it. In addition, Fast Re-Route (FRR) and Fast Convergence (FC) reduce the network re-convergence time if a node or link fails to allow for swift recovery, should a path fail.

The transport stream can also use the characteristics of any IP network to optimize the path and bandwidth usage. One of these characteristics is the ability of an IP network to optimally send the same content to multiple nodes using IP Multicast, in a similar manner to a broadcast network. This characteristic has many applications and has proven itself over a long time in the financial industry, where real-time data feeds that are highly sensitive to propagation delays are built upon IP multicast. It also allows monitoring and supervision equipment to join any of the multicast groups and provide in-line analysis of the streams, both at the IP and Transport Stream level. These devices can be distributed across the network in order to provide multiple measurement points for enriched analysis of service performance.

The Video Hub Office (VHO) can act as a backup or a regional content insertion point but also may be used to source streams into the transport network. This sourcing can be done because of a novel multicast mechanism called IP Anycast, which enables multiple sources to be viewed by the STB as one single and unique source, using the network to determine source prioritization and allowing for source failover without the need of reconfiguration.

Primary and secondary distribution over IP

The bandwidth of individual or collective services in primary distribution between a studio or a playout centre and the secondary distribution hubs is traditionally limited by the availability and cost of bandwidth from circuits such as DS-3 (45 Mbit/s) or STM-1 (155 Mbit/s). This has restricted the delivery of higher bitrate services to such hubs that may benefit from a less compressed source.

The flexibility of IP and Ethernet removes these limitations and enables services to be delivered using lower compression and/or with added services. This means that delivery over an IP infrastructure is now possible:

● to earth stations for satellite (DVB-S/ S2) based services;

● IPTV (DVB-IPI) or cable (DVB-C) head-ends;

● terrestrial (DVB-T) or handheld (DVB-H) transmitting stations.

We shall now look at two examples of this: firstly, Cable distribution and, secondly, IP distribution via DVB's IPI standard.

Example 1: Cable distribution

Cable distribution typically follows a similar pattern to primary and secondary distribution, with the major exception being the use of coaxial cable over the last mile. IP as a transport for secondary distribution in systems such as DVB-C has already been deployed on a large scale by different networks around the globe. Multiple Transport Streams (MPTS) are run as multicast groups to the edge of the aggregation network where edge "QAMs" receive the IP services and modulate them onto RF carriers for delivery to cable STBs.

The modulation onto RF carriers can be done in one of two ways: by translating a digital broadcast channel to the STB or by using a cable modem built into the STB to deliver it directly over IP. In the latter case, as it is a true IP system, the distribution could use DVB IPI described previously without any modification.

Today, almost all of the STBs have no cable modem internally so the IP stream terminates in the hub-site closest to the STB and even if they did, the data infrastructure is often separate from the video infrastructure. This separation is beginning to change as cable data modems become much cheaper and the data infrastructure costs become lower. An in-between stage is emerging where most of the broadcast channels are as before, but some of the little-used channels are sent via IP, known as "Switched Digital Video" (SDV). The consumer notices little difference between a Switched Digital Video

channel and a standard digital cable channel since the servers and QAMs in the hub and/or regional head-ends do all the work. The SDV servers respond to channel-change requests from subscriber STBs, command QAM devices to join the required IP multicast groups to access the content, and provide the STBs with tuning information to satisfy the requests. The control path for SDV requests from the STB is over DOCSIS (DSG), or alternatively over the DAVIC/ QPSK path. In some designs, encryption for SDV can also take place at the hub in a bulk-encryptor, so minimizing edge-QAM encryption-key processing and thus speeding up the channel-change process.

Example 2: IP distribution to the STB via DVB IPI

DVB has had a technical ad-hoc committee (TM-IPI) devoted to IP distribution to the STB since 2000 with a remit to provide a standard for the IP interface connected to the STB. In contrast to other standards bodies and traditional broadcast methodology, it is starting at the STB and working outwards.

In the time since TM-IPI started, many groups around the world have discovered IP and decided to standardize it *(see Fig. 2)*. The standards bodies shown are:

● DLNA (Digital Living Network Alliance) for the home network – see also the section "The Home Network and IP Video";

● HGI (The Home Gateway Initiative) for the standards surrounding the residential gateway between the broadband connection and the in-home network;

● ISMA (The Internet Streaming Media Alliance) for the transmission of AVC video over IP;

● DSL Forum for the standards surrounding DSL and remote management of in-home devices including STBs and residential gateways;
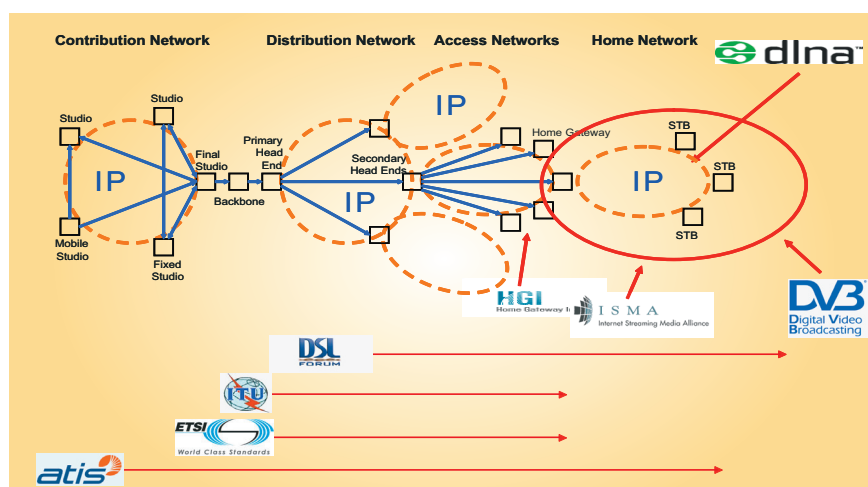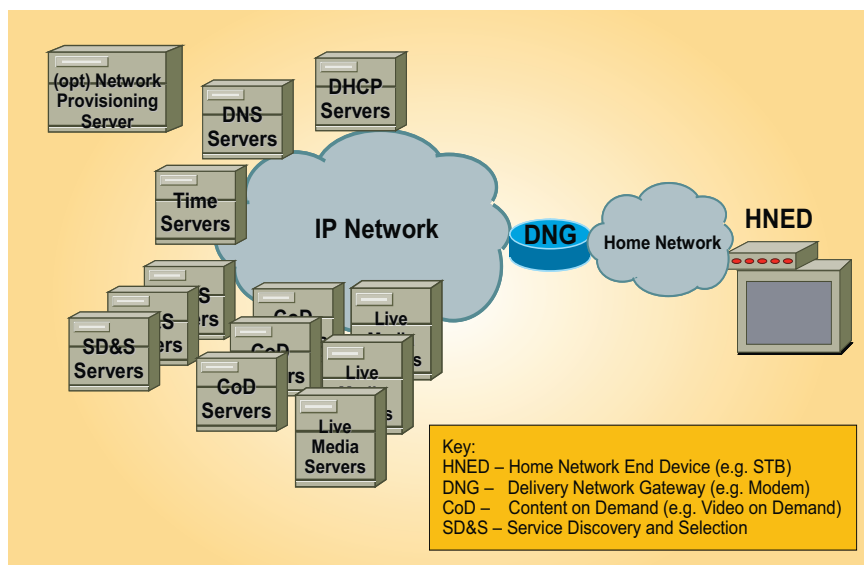

Figure 2
IPTV-related activities of selected standardization bodies

Figure 3
DVB-IP version 1.3 Architecture

- ITU which, via the IPTV Focus Group, is standardizing the distribution and access network architecture;

- ETSI which, via the NGN initiative, is standardizing the IP network carrying the IPTV;

- ATIS which, via the ATIS IPTV Interoperability Forum (ATIS-IIF), is standardizing the end-to-end IPTV architecture including contribution and distribution.

Nevertheless, the DVB-IPI standard does mandate some requirements on the end-to-end system *(see Fig. 3)*, including:

- The transmission of an MPEG-2 Transport Stream over either RTP/UDP or over direct UDP. The method of direct UDP was introduced in the 1.3.1 version of the handbook. Previous versions only used RTP, and the use of AL-FEC requires the use of RTP.

- Service Discovery and Selection either using existing DVB System Information, or an all-IP method such as the Broadband Content Guide.

- Control of content on demand using the RTSP protocol.

- The use of DHCP to communicate some parameters such as network time, DNS servers etc. to the STB.

It is normal in IPI to use single-programme transport streams (SPTS) as the content are normally individually encoded and not multiplexed into MPTS as they would be for other distribution networks. This provides the added flexibility of only sending the specifically-requested channel to the end user, which is important when the access network is a 4 Mbit/s DSL network as it reduces bandwidth usage.

IPTV and Internet TV convergence

The two worlds of managed STB and unmanaged Internet TV are coming together with sites like YouTube or MySpace showing user-generated content and excerpts from existing TV programming. Internet TV demonstrates what can be done with an unmanaged network across a diversity of different networks, including one in the home. In this section we'll cover what the home network will look like, compare IPTV to Internet TV, and show how a broadcaster can place content on the Internet via an Internet Exchange.

The Home Network and IP Video

Improving technologies of wireless networks, increases in hard-disk-drive sizes and the increasing number of flat-screen TVs in European households, makes the home network inevitable in the near future. Unfortunately the home network still remains more of promise than reality for high-quality broadcast TV transmission, mainly because the standards and interoperability are some way behind.

DVB has just released a Home Network reference model which is the first part of a comprehensive specification which will be completed in 2008. The home network consists of several devices *(See Fig. 4)*:

- **Broadband Gateway Device** (BGD) – The residential gateway or modem connected to the IP Service Provider, usually via either cable or DSL.

- **Uni-directional Gateway Device** (UGD) – A one way device that converts broadcast TV to a stream on the home network. For example a DVB-T tuner that converts the stream to IP and sends it wirelessly over the home network.

- **Home Network End Device** (HNED) – The display, controlling and/or storage device for the streams received either via the BGD or UGD.

- **Home Network Node** (HNN) – The device, for example a switch or Wireless Access Point, that connects the home network together.

The Home Network Reference Model, available as a separate DVB Blue Book, is based on work done by the DLNA (Digital Living Network Alliance). DLNA already has existing devices that do stream video over the home network but from sources within the network. The DVB Home Network is the first that integrates both programming from broadcast TV and in-home generated video.
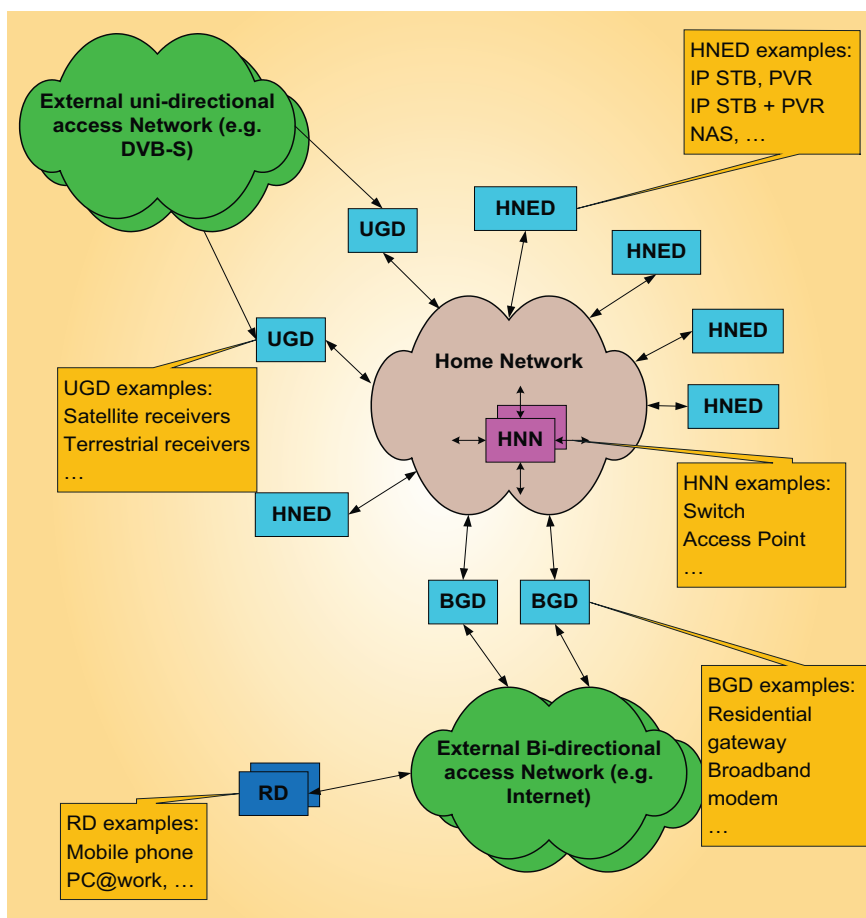
Figure 4
DVB-IPI Home Network Reference Architecture

to request data from the closest and least congested servers or nodes, to reduce latency and packet loss. In the peer-to-peer model, lack of available bandwidth from the different nodes, due to limited upstream bandwidth to the Internet, enforces the need for larger and more distant "supernodes" to compensate which, overall, makes the possibility of packet loss higher so increasing the chance of a video artefact.

The decoding devices in the uncontrolled environment of Internet TV also limit encoding efficiency. The extremely diverse hardware and software in use to receive Internet video services tend to limit the commonalities between them. H.264, which is a highly efficient codec but does require appropriate hardware and/ or software resources for decoding, is not ubiquitous in today's deployed environment. MPEG-2 video and Adobe Flash tend to be the main video players that are in use, neither being able to provide the same picture quality at the equivalent bitrates to H.264.

Challenges of integration with Internet Video services

Internet Video services are growing very fast. The diversity of the content on offer, the ease of adding new content and the speed with which new services can be added is quite a challenge for managed IPTV services. This leads to the managed IPTV service providers wanting to combine the two types of IP services on the same STB.

The most natural combination is the "Hybrid" model which has both types of services, probably by integrating the peer-to-peer client within the SP's STB. This would allow for collaboration between the two services and would benefit the users by allowing them to view the Internet video content on a TV rather than a computer. The Service Provider would then make sure that the Internet video streams obtain the required bandwidth within the network, perhaps even hosting nodes or caching content within the Service Provider network to improve

Comparison of Internet video and IPTV

Although IPTV and Internet-based video services share the same underlying protocol (IP), don't let that deceive you: distribution and management of those services are very different.

In an IPTV environment, the SP has a full control over the components that are used to deliver the services to the consumer. This includes the ability to engineer the network's quality and reliability; the bitrate and codec used by the encoder to work best with the limited number of individually managed STBs; the ability to simplify and test the home network components for reliability and quality; and prevention of unnecessary wastage of bandwidth, for example by enabling end-to-end IP Multicast.

Control over the delivery model doesn't exist with Internet video services. For example, IP Multicast deployments on the Internet are still very limited, mostly to research and academic networks. This means that Internet-streamed content services use either simple unicast-based streams between a given source and destination or a Peer-to-Peer (P2P) model which will send and receive data from multiple sources at the same time.

One of the other main differences is the control of the required bandwidth for the delivery of the service. A Service Provider controls the bitrate and manages the QoS required to deliver the service, which allows it to control the buffering needed in an STB to ensure the audio and video decoders don't overrun or underrun, resulting in artefacts being shown to the end user. Internet video cannot control the bitrate so it must compensate by implementing deeper buffers in the receiver or attempting

delivery. They may even transcode the Internet TV content to provide a higher quality service that differentiates itself from the Internet version.

This "Hybrid" model offers collaboration but may still incur some limitations. The Internet TV services might be able to be delivered to the STB but the amount of memory, processing and increased software complexity might make it too difficult within the existing STB designs. This would increase the cost of the unit and therefore impact the business models, whilst competition between such services may lock out specific players from this market due to exclusive deals.

How can a broadcaster get content into an Internet Video service?

First some Internet history: Today, the Internet is known worldwide as a "magical" way to send e-mails, videos and other critical data to anywhere in the world. This "magic" is not really magic at all, but some brilliant engineering based on a network of individual networks, so allowing the Internet to scale over a period of time to cover the entire world, and continue to grow. This network of networks is actually a mesh of administratively independent networks that are interconnected directly or indirectly across a packet switching network based on a protocol (IP) that was invented for this purpose.

The Internet model of a network of networks with everyone connected to everyone individually was fine until the cost and size of bandwidth became too high, and the management of individual links became too difficult. This started the movement towards Internet Exchange Points (IXP) which minimized connections and traffic going across multiple points by allowing the Service Providers to connect to a central point rather than individually connecting to each other. One of the first was at MAE-East in Tyson's Corner in Virginia, USA, but today they exist across Europe with LINX in London,

AMS-IX in Amsterdam and DE-CIX in Frankfurt being among the largest and most established ones.

The Internet Exchange Point, by interconnecting directly with other networks, means that data between those networks has no need to transit via their upstream SPs. Depending on the volume and destinations, this results in reduced latency and jitter between two end points, reducing the cost of the transit traffic, and ensuring that traffic stays as local as possible. It also establishes a direct administrative and mutual support relationship between the parties, which can have better control over the traffic being exchanged.

Being at the centre of the exchange traffic means that IXPs can allow delivery of other services directly over the IXP or across private back-to-back connections between the networks. Today, this is how many Voice-over-IP and private IP-based data feeds are exchanged.

This also makes the IXP an ideal place for Broadcasters to use such facilities to establish relationships with SPs to deliver linear or non-linear broadcast services to their end users. The independence of the IXP from the Service Provider also allows content aggregation, wholesale or white-labelled services, to be developed and delivered via the IXP. For example,

the BBC in collaboration with ITV is delivering a broadcast TV channel line-up to the main broadband SPs in the UK. They also provide such a service for radio in collaboration with Virgin Radio, EMAP and GCA. This service has been running for a couple of years and has been shortlisted for an IBC 2007 Award within the "Innovative application of technology in content delivery" category.

Quality of Experience

The Quality of Experience (QoE), as defined by ETSI TISPAN TR 102 479, is the user-perceived experience of what is being presented by a communication service or application user interface. This is highly subjective and takes into account many different factors beyond the quality of the service, such as service pricing, viewing environment, stress level and so on. In an IP network, given the diversity and multiplicity of the network, this is more difficult and therefore more critical to success than in other transports (see Fig. 5).

Subjective and Objective requirements

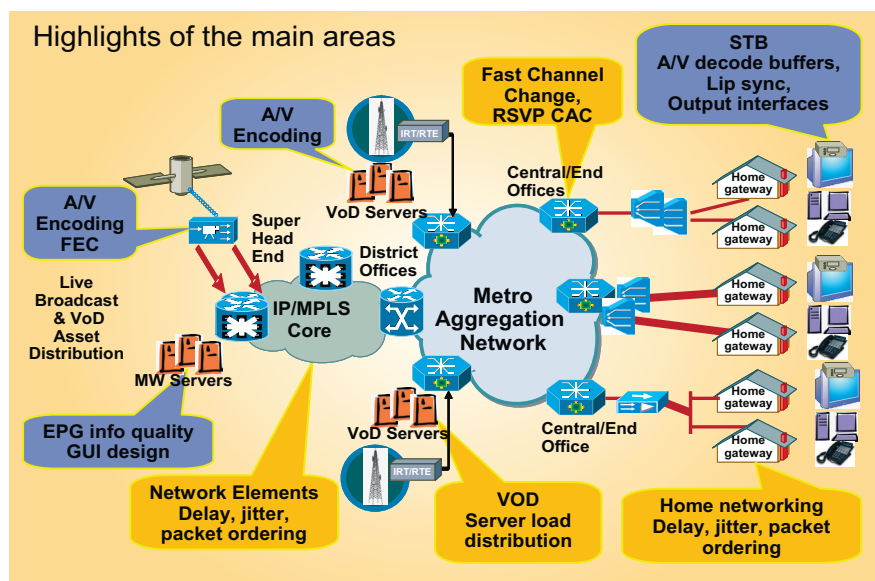Subjective measurement systems, such as ITU-R BT.500-11, provide a detailed

Figure 5
IPTV QoE in the end-to-end model

Abbreviations

AL	Appliation Layer		**IPI**	Internet Protocol Infrastructure
ASI	Asynchronous Serial Interface		**IPTV**	Internet Protocol Televison
ATIS	Alliance for Telecommunications Industry Solutions (USA) http://www.atis.org/		**IRD**	Integrated Receiver/ Decoder
AVC	(MPEG-4) Advanced Video Coding		**ISMA**	Internet Streaming Media Alliance http://www.isma.tv/
BER	Bit Error Rate		**ITU**	International Telecommunication Union http://www.itu.int
BGD	Broadband Gateway Device		**IXP**	Internet eXchange Point
CBR	Constant Bit-Rate		**MDI**	Media Delivery Index
CoP4	(Pro-MPEG) Code of Practice 4		**MLR**	Media Loss Rate
DAVIC	Digital Audio-Visual Council		**MPLS**	Multi Protocol Label Switching
DHCP	Dynamic Host Configuration Protocol		**MPTS**	Multi Programme Transport Stream
DLNA	Digital Living Network Alliance http://www.dlna.org/home		**NGN**	Next Generation Network
DNS	Domain Name System		**NMS**	Network Management System
DSG	(CableLabs) DOCSIS Set-top Gateway		**QAM**	Quadrature Amplitude Modulation
DSL	Digital Subscriber Line http://www.dslforum.org/		**QoE**	Quality of Experience
DVB	Digital Video Broadcasting http://www.dvb.org/		**QoS**	Quality of Service
DVB-C	DVB – Cable		**QPSK**	Quadrature (Quaternary) Phase-Shift Keying
DVB-H	DVB – Handheld		**RF**	Radio-Frequency
DVB-S	DVB – Satellite		**RSVP**	ReSource reserVation Protocol
DVB-S2	DVB – Satellite, version 2		**RTP**	Real-time Transport Protocol
DVB-T	DVB – Terrestrial		**RTSP**	Real-Time Streaming Protocol
ETSI	European Telecommunication Standards Institute http://pda.etsi.org/pda/queryform.asp		**SDI**	Serial Digital Interface
			SDV	Switched Digital Video
			SHE	Super Head End
			SP	Service Provider
FC	Fast Convergence		**SPTS**	Single Programme Transport Stream
FEC	Forward Error Correction		**STB**	Set-Top Box
FRR	Fast Re-Route		**TE**	Traffic Engineering
GUI	Graphical User Interface		**TS**	(MPEG) Transport Stream
HGI	Home Gateway Initiative http://www.homegatewayinitiative.org/		**UDP**	User Datagram Protocol
			UGD	Uni-directional Gateway Device
HNED	Home Network End Device		**VBR**	Variable Bit-Rate
HNN	Home Network Node		**VHO**	Video Hub Office
IP	Internet Protocol		**VoD**	Video-on-Demand

model for picture-quality assessment by getting a panel of non-expert viewers to compare video sequences and rate them on a given scale. This requires considerable resources to set up and perform the testing, so it tends to be used for comparing video codecs, bitrates, resolutions and encoder performances.

An IP network operator cannot have a team of humans sitting looking at pictures to assess picture quality, particularly with the number of channels these days. They therefore test quality with automated measurement systems which provide real-time monitoring and reporting within the network and services infrastructure. The measurement systems usually use some subjective human input to correlate a baseline that objective measurement methods can be mapped to. An operator usually deploys probes at critical points in the network which report back to the Network Management System (NMS) a set of metrics that will trigger alarms based on predefined thresholds.

When compared to a traditional broadcast environment, video services transported over an IP infrastructure introduce extra monitoring requirements. The two main categories of requirements are:

- **IP transport network**
  Whilst transporting the services, IP packets will cross multiple nodes in the network(s) – possibly subjected to packet delay, jitter, reordering and loss.

- **Video transport stream**
  (MPEG-2 TS)
  Traditional TS-monitoring solutions must also be used to ensure the TS packets are free of errors.

The two categories are also usually in different departments: the IP transport monitoring is within the Network Operations Centre, and the video transport stream monitoring within the TV distribution centre. One of the keys to a good Quality of Experience in IP is sometimes just good communication and troubleshooting across the different departments.

Finally, although this is beyond the scope of network-based management, additional measurements should be taken into account in a full system, such as the following:

- **Transactional** – GUI and channel change response time, service reliability.

- **Payload (A/V compression)** – Compression standards compliance, coding artefacts.

- **Display (A/V decoding)** – Colour space conversion, de-blocking, de-interlacing, scaling.

Measurement methods

The main measurement methodology for the IP transport network is the Media Delivery Index (MDI) as defined in IETF RFC 4445. MDI is broken down into two sub-components: Delay Factor (DF) and Media Loss Rate (MLR) which are both measured over a sample period of one second. The notation for the index is DF:MLR.

DF determines the jitter introduced by the inter-arrival time between packets. This shouldn't be viewed as an absolute value but is relative to a measurement at a given point in the network. Jitter can be introduced at different points by encoders, multiplexers, bulk scramblers, network nodes or other devices. It is important to know what the expected DF value should be, which can be determined by a baseline measurement in ideal operating conditions. The value can change dependent on the stream type: Constant Bitrate (CBR) streams should have a fixed inter-arrival time whilst Variable Bitrate (VBR) streams will have a varying value. Once a baseline value has been determined, you normally set a trigger significantly above this value before alerting via an alarm.

MLR provides the number of TS packets lost within a sample period. This is achieved by monitoring the Continuity Counters within the TS.

If the stream contains an RTP header, the sequence number can be used for identifying out-of-sequence or missing packets without the need to examine the IP packet payload. This will reduce the computational requirements and speed up the monitoring process. It is normal therefore to distribute MDI probes across the IP forwarding path to allow supervision on a hop-per-hop basis. This helps troubleshoot potential issues introduced by a specific network element.

To complement the IP packet metrics, DVB-M ETSI TR 101 290 (ETR 290) is used to provide insight within the transport stream itself. This operates in the same way as in a traditional ASI-based infrastructure.

The combination of MDI and ETR 290 delivers a scalable and cost-effective method for identifying transport-related issues. By triggering alarms at the IP and TS level, these can be aggregated and correlated within the NMS to produce a precise reporting tool between different events and their insertion point within the network infrastructure.

Improving QoE with FEC and retransmission

DVB has considerable experience in error-correction and concealment schemes for various environments, so it was natural – given the difficulty of delivering video over DSL – that the IPI ad-hoc group should work in this area. They spent a significant time considering all aspects of error protection, including detailed simulations of various forward error correction (FEC) schemes and quality of experience (QoE) requirements.
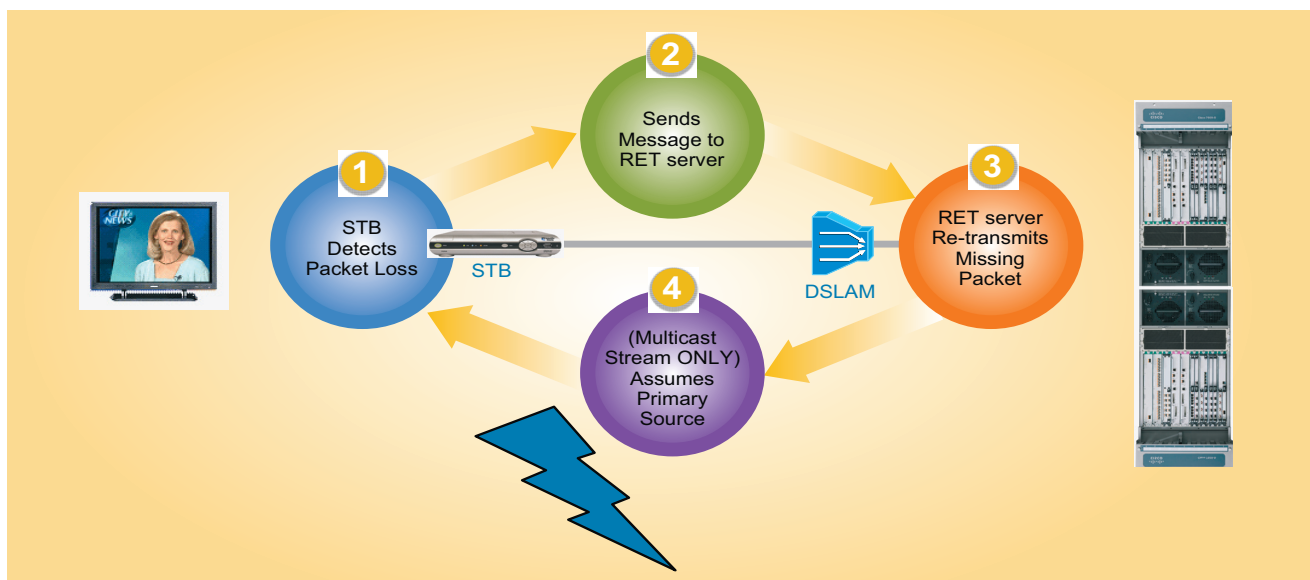

Figure 6
IPTV QoE in the end-to-end model

The result is an optional layered protocol, based on a combination of two FEC codes – a base layer and one or more optional enhancement layers. The base layer is a simple packet-based interleaved XOR parity code based on Pro-MPEG COP3 (otherwise known as SMPTE standard 2022-1 via the Video Services Forum, see http://www.videoservicesforum.org/activities.shtml) and the enhancement layer is based on Digital Fountain's Raptor FEC code (http://www.digitalfountain.com). It allows for simultaneous support of the two FEC codes which are combined at the receiver to achieve error correction performance better than a single code alone.

FEC has been used successfully in many instances; however, another technique in IP can also be used to repair errors: *RTP retransmission*. This works via the sequence counter that is in every RTP header that is added to each IP packet of the video stream. The STB counts the sequence counter and if it finds one or more missing then it sends a message to the retransmission server which replies with the missing packets. If it is a multicast stream that needs to be retransmitted then the retransmission server must cache a few seconds of the stream in order to send the retransmitted packets *(see Fig. 6)*.

Bandwidth reservation per session

One of the advantages of IP is the ability to offer content on demand, for example Video on Demand (VoD). This is resulting in a change in consumer behaviour: from watching linear broadcasts to viewing unscheduled content, thus forcing a change in network traffic. This makes corresponding demands on the IP infrastructure as the number of concurrent streams across the managed IPTV infrastructure can vary from thousands to hundreds of thousands of concurrent streams. These streams will have different bandwidth requirements and lifetime, dependent on the nature of the content which is being transported between the source streamers playing out the session, across the network infrastructure to the STB.

The largest requirement is to prevent packet loss due to congestion, which can be prevented if the network is made aware of these sessions and makes sure enough bandwidth is available whenever setting up a new stream. If there isn't enough bandwidth, then the network must prevent the creation of new streams – otherwise all the connected users along that path will have a degraded viewing experience *(Fig. 7)*.

RSVP CAC (based on RFC2205, updated by RFC2750, RFC3936 and RFC4495) allows for per-session bandwidth reservation to be established across the data path that will carry a given session. Step 1 & 2 in *Fig. 7* show the VoD session starting between the STB and the middleware. The authorization credentials will be checked to make sure that the customer can play the content, based on a set of criteria such as credit, content rating, geography and release dates. Once these operations are authorized by the middleware and billing system, the middleware or VoD system manager identifies the VoD streaming server for this session. In step 3, the server initiates a request for an RSVP reservation path between the two end points across the RSVP-aware network infrastructure. Finally, in step 4, if the bandwidth is available then the session can be initiated; otherwise a negative response will be sent to the middleware to provide a customized response to the customer.

Conclusions

Delivery by IP of broadcast-quality video is here today and is being implemented by many broadcasters around the world. The nature of IP as a connectionless and non-deterministic transport mechanism makes planning, architecting and managing the network appropriately, which can be done with careful application of well-known IP engineering. When the IP network is the wider Internet, the lack of overall control makes guaranteed broadcast-level quality difficult to obtain, whereas on a managed IP network, Quality of Service techniques, monitoring and redundancy can be used to ensure broadcast-level quality and reliability.

The techniques to monitor video are similar to the ones used for any MPEG-2 transport stream. However, these need to be related to the IP layer, for example using MDI, as debugging the problem will often require both network and video diagnostics.
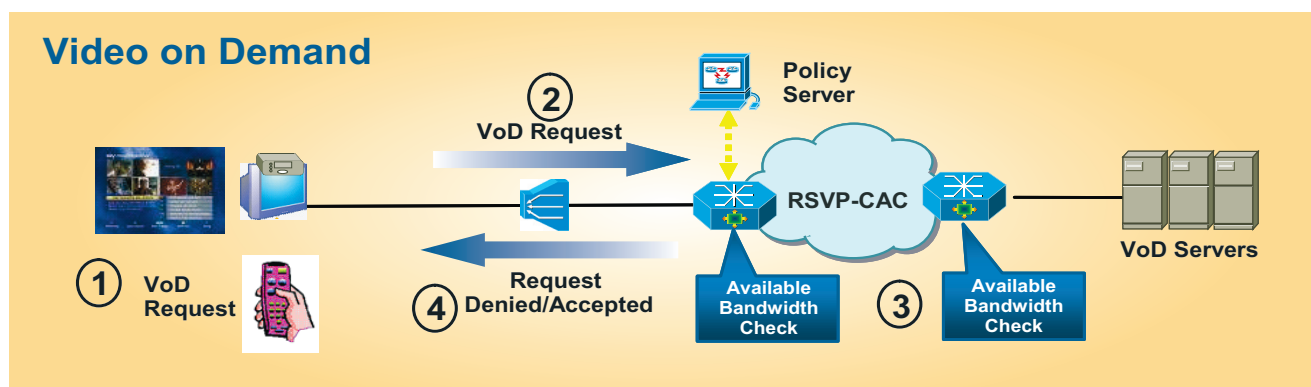
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Figure 7
Connection Admission Control

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Editorial

Technological neutrality and service neutrality
Philip Laven

Content Protection

HDMI & HDCP — the manufacturers' perspective
Dietrich Westerkamp 📄 *(202 kB)*

HDTV signals offer great opportunities to broadcasters, but there is also the negative side – a high risk of piracy. In order to protect prime content against illegitimate use, content-protection mechanisms can be used.

For the digital HDMI interface between an HDTV set-top box and an "HD ready" display device, HDCP technology is chosen. This is a tool that can be used at the discretion of the broadcaster who can activate it by means of a switching signal. In the case of a piracy attack, the technology offers a revocation mechanism whereby a list of revoked devices is transmitted in a safe way to the receiver, where it is stored.

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