COMMISSION OF THE EUROPEAN COMMUNITIES
DIRECTORATE GENERAL

Telecommunications, Information Market and Exploitation of Research

# Digital Video Broadcasting

*A Volume of Technical Papers Accompanying the Commission's Communication*

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## INTRODUCTION TO THIS VOLUME

This accompanying volume provides additional technical information beyond what it was possible to include in the Commission's Communication to the Council on digital video broadcasting. The papers it contains have for the most part been written by leading experts in the different specialities that combine to make digital video broadcasting possible. This volume also contains several contributions previously published by other organisations, reproduced here with the permission and encouragement of the authors.

Collectively, the contributions present a rich diversity of ideas and opinions. This is entirely appropriate to an emerging activity which offers so many fertile possibilities within broadcasting, as well as linkages with other applications, based on the synergies achievable with the computing and telecommunications sectors through the use of common, "generic" technologies. However, readers should understand that the opinions expressed are those of the authors and do not necessarily reflect the official view of the European Commission.

Consequently, no attempt has been made to edit the papers so that they collectively provide a coherent, integrated view. The intention is rather to illustrate the extent of the debate in Europe. This accompanying volume therefore provides a series of snapshot analyses of digital television taken from different perspectives. There are inevitably some duplications and contradictions. It is hoped that interested readers will be stimulated and not deterred by them.

Part 1 contains three chapters which describe the principal technical functions of a digital broadcasting system. Its introduction is of interest to non-specialists because it provides a simplified description of the whole broadcasting chain.

Part 2 is a single paper which investigates implementation issues for digital television in Europe (chapter 4).

Part 3 is a survey of digital broadcasting research in Europe (chapter 5).

Part 4 examines the institutions and issues relative to standardisation in Europe and globally, focusing especially on the prospects for commonality with other applications (chapter 6).

Part 5 reproduces two important documents of the European Group for Digital Video Broadcasting (chapters 7, 8).

# *PART 1 - TECHNICAL DESCRIPTION OF DIGITAL VIDEO BROADCASTING*

# INTRODUCTION TO PART I
# WHAT IS DIGITAL VIDEO BROADCASTING?

## 1. ELEMENTS OF THE SYSTEM

The digitalisation of television signals today is a well-known and widely implemented process. It consists basically of the representation of a picture - and the accompanying sound - by a binary bit-stream, a series of '0's and '1's. However, compression and transmission of these signals through a communications channel - satellite, terrestrial or cable - becomes practical only after the raw digital data has been subject to a series of processes. And therefore, if it is wished to be able to interconnect digital TV equipment from different suppliers, or to receive such transmissions satisfactorily, it is necessary to define each of these processes clearly and completely.

Figure 1 sets out the whole chain of processes involved in digital television transmission. Each of them is then discussed in greater detail below.

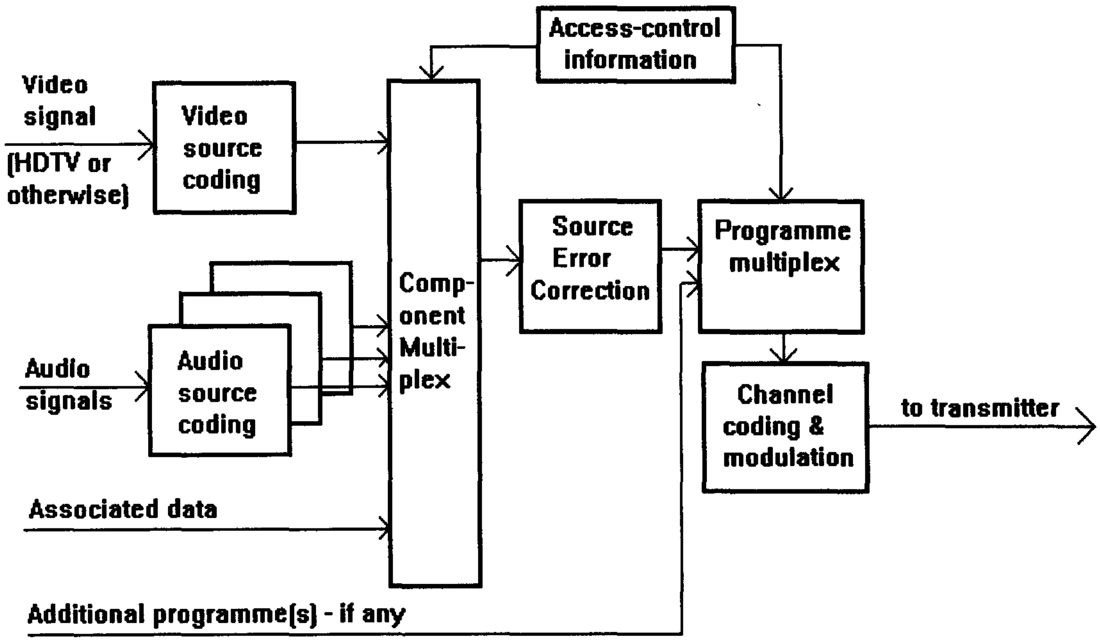


*Figure 1: Basic structure of digital television system*

### 1.1 Source coding

Source coding is the process whereby the bit-stream containing the picture information is compressed in order to eliminate most of the redundancy in the video information while still ensuring that the original picture can be regenerated afterwards by the corresponding expansion process. This process is based on the assumption that each frame of the image and the next frame are similar most of the time, and therefore less information must be transmitted in practice if in general it represents only the differences between the predicted image and the real one. Where large areas of the picture are in coherent motion, this assumption fails, and in this case the system detects and describes the characteristics of the moving areas so that the receiver can apply suitable compensation. As the amount of redundancy in the image signal depends on the picture contents, and the transmission channel generally has a maximum bit-rate, the reproduced quality obtained is highest with still pictures and decreases as the level of motion increases beyond a threshold, somewhat like human sight.

In both analogue and digital television broadcasting, the optical image within the camera is initially scanned in a series of horizontal lines which together comprise one frame; in current practice, the odd-numbered lines are scanned during the first half of each frame, followed by the even-numbered ones, giving an *interlaced* scan. In the Americas, with a few exceptions, and some Pacific Rim countries, notably Japan, there are 525 lines per frame and $\pm 60$ frames/second; elsewhere, notably in Europe, 625 lines and 50 frames/second is the standard.

Like the existing CCIR Recommendation 601 for digital video signals in television studios, MPEG-2 source coding is designed to handle both of these line and frame rates as well as the two aspect ratios, namely 4:3 and 16:9. For this reason, it is likely that the MPEG-2 source coding will not only be adopted formally as a world-wide standard, but also used widely in practice. However, it must be appreciated that the use of MPEG-2 source coding does not in itself solve the problems arising from the co-existence of these two incompatible scanning standards; when the digital signal is decoded back to a form suitable for display, this will have the same line and frame rates as at the input to the source coder.

## 1.2 Multiplexing

As the use of digital compression allows several digital television signals to be carried within a single broadcast TV channel, these signals must be combined in such a way that they can be separated again conveniently afterwards. In practice, a *time-division multiplex* (TDM) in which packets consisting of a fixed number of bits from each signal, together with identification bits, are sent sequentially. The MPEG-2 standard defines the ways in which these multiplexes can be created, reconfigured and split up; if so desired; the implementation may include provision for compatibility among several members of the MPEG-2 family. It should be noted that in practice each TV programme consists of a multiplex of sound, vision and data signals, and that some or all of certain progrmmes may be scrambled before the final multiplex is created, in order to enable access to them to be controlled.

These first two processes, source coding and multiplexing, are presently being defined by the Motion Picture Experts Group (MPEG), which is a joint working group of the International Electrotechnical Commission (IEC) and of the International Standardisation Organisation (ISO). Some 120 of the world's leading companies in TV, video, film-making, computing and consumer electronics, including those in most EC member-States, are participating in MPEG. This set of standards is intended to be *generic*, usable across many applications and sectors, without being limited by the special requirements of any particular application. The MPEG philosophy is discussed fully in Chapter 1 below.

## 1.3. Modulation and forward error correction

In order to enable the multiplex to be broadcast, it must be used to modulate one or more carrier signals, at frequencies that have the required propagation characteristics. Modulation is therefore the process of combining the multiplex of television programmes with the carrier signal which will convey it to the home. Propagation characteristics are the same whether the modulation system is analogue or digital; but the choice of modulation technique determines key economic criteria, like the amount of transmission power required to emit the signal for satisfactory reception in the home. The choice of the appropriate modulation technique - Amplitude Modulation (AM), Frequency Modulation (FM), Phase Modulation (QPSK etc.) and Coded Orthogonal Frequency-division Modulation (COFDM) - depends on the application, with AM and COFDM being preferred for cable and terrestrial broadcasting, for example. COFDM is profiled in Chapter 3 and the operational impact of different modulation schemes is covered in Chapter 4.

With digital transmission, it is customary to provide protection against some loss or corruption of the data by deliberately introducing additional data derived from the multiplex signal at the sending end. This data protection is part of a process known as *channel coding*; it reduces the transmit power requirement, which is high with analogue systems. However, because each type of broadcasting system is liable to be affected by different types of impairments, it is necessary to adopt different forward error correction techniques to provide optimum protection in each case. There is obviously a trade-off between the amount of additional data required for this protection and the total transmission capacity; instead of improving the overall received bit-error rate beyond the threshold of visibility, it may be preferable to provide extra protection to only part of the data-stream, so that, for example, the main sound channel(s) remain usable after the picture has failed, human hearing being more sensitive to interference than human vision.

## 1.4 Conditional access

Conditional access is not a single technical process but a combination of systems and management functions which determines who has the right to receive a service or programme. There are two main reasons for broadcasters to use conditional access, first where payments must be made for a service; and second where the service provider has purchased the right to broadcast programmes only in a particular territory. In practice, conditional access involves the operation of a Subscriber Management Centre (SMC). This is an office which deals with the commercial aspects of the system, such as promotion, tariffs and the processing of requests for authorisation to watch a particular service or programme. Authorisation is called *entitlement* in the business; customers pay for entitlements to receive the service or programme. Some systems require *smart cards* to be issued to customers. The SMC is normally responsible for this. When inserted into a reader/decoder, smart cards permit the customer to watch the programme or service, once the SMC has instructed the transmission centre to add the relevant entitlement to the control messages included in the broadcast multiplex. The combination of smart card and control message therefore provides a key for *descrambling* the programme.

The two technical functions used in conditional access systems are *scrambling* and *encryption*. Scrambling is the process whereby the data is made unintelligible, be it video or audio. It typically causes some degradation when applied to analogue television signals; but in the digital domain, secure scrambling can easily be performed in a way that causes no impairment to ultimate audio or video quality. An encryption algorithm prevents unauthorised viewers from finding out how the scrambling techniques are being used and reversing them. A cipher key is needed to start the descrambling process and to verify it periodically. Chapter 2 offers an extensive overview of all these topics.

Chapter 1

## Source coding and multiplexing

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## 1. INTRODUCTION

During the past decades digital techniques have made numerous inroads into different fields, thanks to the progress in data processing, stimulated by the large investments made by the big, mainly North American and Japanese, players in the field. This was possible thanks to the rapid obsolescence of data processing machines that forced customers into a rapid turn-over of equipment. Not equally so in the field of telecommunications, here meant to include broadcasting, where replacement or even upgrade of the infrastructure takes understandably a much longer time.

The attractiveness of digital techniques has not been foreign to the field of telecommunications, but has largely been restricted to the backbone network. Adoption of these techniques, conversely, has been slow in the secondary distribution network, mostly because of the constraint of the old analogue-based infrastructure and the cost of the terminal equipment for telecommunication services employing digital techniques.

The last few years have seen rapid progress, such that audio-visual information in digital form can now be transported to the end user with a better spectrum utilisation and with overall higher quality than was previously possible. The huge markets opened up by these concrete possibilities have made the expected cost of the terminal equipment fall by orders of magnitude compared with the equipment developed so far mostly for professional applications, and is now of the same order as that of conventional analogue terminal equipment.

Many companies and organisations are thinking these days of adopting this new technology for upgrading existing services or introducing new ones. The introduction of digital audio-visual services, however, is a great opportunity not to be lost. This is because, in contrast with the incompatible systems put in place in the analogue world, this epochal conversion from one technology to another gives the possibility to define a syntax representing audio-visual information that is universal in space, time and applications.

The purpose of this report is first to discuss the role of compression systems in audio-visual services, the principal techniques considered during the research phase and to make a comparison between the most promising candidates. The report then details the reasons why the emerging MPEG-2 standard is able to accommodate the need of international, inter-application inter-working and gives a comprehensive account of the technical content of the standard in its three components: video coding, audio coding and multiplexing. Lastly the report highlights the likely avenues of development in the area of audio-visual information coding. Some considerations on the role of recording systems are given at the end.

## 2. ROLE OF COMPRESSION SYSTEMS

Compression is an essential element in the practical use of audio and video in digital form. The reason lies in the extremely high data rate of these signals: a net bit-rate of 166 Mbit/s for 625/525-line video digitised according to CCIR Recommendations 601/656 and between 664 to 995 Mbit/s for HDTV, depending upon which of the conceivable formats (see Table 1) is being considered.

| | Format | Rate (Mpel/s) | Picture size | Comment |
|---|---|---|---|---|
| USA | | | | |
| | 1280h/720v/60 | 55.3 | 0.92 | square pels |
| | 1728h/960v/30 | 49.8 | 1.66 | square pels |
| | 1440h/960v/30 | 41.5 | 1.38 | rectangular pels |
| | 1920h/1080v/30 | 62.2 | 2.07 | square pels |
| Europe | | | | |
| | 2048h/1152v/25 | 59.0 | 2.34 | square pels |
| | 1920h/1152v/25 | 55.3 | 2.21 | rectangular pels |
| | 1440h/1152v/25 | 41.5 | 1.66 | rectangular pels |
| Japan | | | | |
| | 1920h/1035v/30 | 59.6 | 1.99 | rectangular pels |
| | 1840h/1035v/30 | 57.2 | 1.90 | square pels |
| | 1440h/1035v/30 | 44.7 | 1.49 | rectangular pels |

*Table 1: Some HDTV formats*

Audio for studio applications is usually sampled at 48 kHz (44.1 for storage on compact disc) with 16 bits per sample yielding a net bit-rate of 768 kbit/s. Two-channel stereo sound requires 1.536 Mbit/s and 5-channel sound 3.84 Mbit/s.

Such high bit-rates, in addition to being unmanageable for applications where delivery across time or space is required, are not really justified. Very simple processing of average pictures shows that one can represent them with half that bit-rate without any loss. It has, therefore, been the constant assumption since the very early days of audio and video digitisation that by exploiting the fact that, since digital video and audio are not sequences of purely random samples but carry structured information, it should be possible to represent those signals with a much lower bit-rate, possibly with some loss of information, but such that the loss is perceptually irrelevant or, at least, not annoying.

"How much lower ?" has been the question that research has tried to answer in the past three decades and a brief summary will be given in the next section.

## 3. HISTORY OF COMPRESSION SYSTEMS

For obvious reasons compression has first been applied to audio signals. The one-dimensional nature of audio, the possibility to create models of different complexity of the signal to be encoded and the requirement to have a low coding delay made the choice of DPCM a natural one.

### 3.1. Differential pulse code modulation (DPCM)

DPCM is a straightforward extension of one simple method to provide lossless bit-rate reduction, viz. the transmission of differences between samples. The major departures from that simple scheme are that the difference is quantised with a possibly non-linear quantiser (taking into account the fact that human sensors are more sensitive to errors when the signal is smaller and less so when its value is higher) and the use of subtraction from a "predicted" value instead of the actual value.

Around this basic principle a number of adaptations have been made over the years, particularly in the area of speech coding: adaptivity of predictor and quantiser, sub-band split with application of different DPCMs etc. While a number of ITU standards which are based on DPCM have been produced, the equivalent did not happen for video. The main reason lies in the limited amount of compression achievable with this technique, typically 4:1.

### 3.2. Discrete cosine transform (DCT)

More powerful algorithms were needed to make digital video progress possible. One of them, introduced as early as about 25 years ago, was transform coding. The basic principle is the use of a linear transformation basis to transform a set of correlated samples, each with the same energy, into a set of uncorrelated samples with a large dynamic range of values. Those values with highest variance are represented with more bits and a finer quantiser, while those with the smaller variance are represented with fewer bits and a coarser quantiser. The process of finding an optimum transformation basis is mathematically known as the search for eigenvectors of a given correlation matrix and many transformation bases (Hadamard, Fourier, Slant, DCT, Karhunen-Loève) were tried before settling on the DCT (Discrete Cosine Transform) as the optimum compromise between the complexity of the transform and its decorrelation properties. With DCT it is possible to achieve compression ratios in excess of 8:1, with the added advantage that one can easily get very fine steps of variation of the compression ratio, something that is of primary importance when these algorithms, originally conceived for one- or two-dimensional signals, are extended to three-dimensional signals, i.e. to sequences of pictures.

### 3.3. Other techniques

Another algorithm introduced in the research phase is Vector Quantisation (VQ). The basic idea is that if a set of N samples each represented with M bits is taken, the probability distribution in the N-dimensional space is usually not uniform. The N-space can then be quantised in a non-uniform way with a compression factor that depends on the number of possible quantisation cells allowed in the N-space.

Sub-band coding, i.e. the splitting of the spectrum of a signal in a number of bands, is a technique that has been employed with success in the audio field. Its application to video coding has been slow to take off and has attracted some interest of late because it provides a technically pleasing solution to the problem of hierarchical coding, thanks to the possibility to derive different sub sampled versions of the original signal by using filters. Critics of this method complain about losing the decorrelating properties of the DCT scheme, and indeed sub-band-coded pictures, as of today, look no better than DCT coded pictures - even in hierarchical mode, where they are supposed to yield their best performance.

Wavelet coding is a further generalisation of DCT and sub-band coding. If used as a replacement for the DCT block in a video coding scheme, it may add a better local match to the local property of the picture because of the possibility to provide a choice between better frequency or spatial analysis. This flexibility, however, has not been fully exploited yet.

An old scheme that is undergoing a revival these days is Singular Value Decomposition. This gives theoretically the best local adaptation to picture content in the sense that the transformation basis is what is to be found and coded on a block-by-block basis.

### 3.4. Extensions

Several extensions of some of the techniques mentioned above have been tried for the purpose of removing the inter-frame redundancy:

- The DPCM technique uses the time dimension to increase the effectiveness of the prediction;
- The linear transformation is also applied in the time dimensions;
- Motion detection (only areas in a picture that are "changed" compared with the previous picture are coded - this technique was standardised by the CCITT for video conference applications in the Recommendations of the H.100 series);
- Motion estimation (in general, from one picture to the next, only part of the information is new; the rest can be predicted using motion information and only the prediction error is transmitted);
- Interpolation (if two pictures are known, the pictures in between can be interpolated using motion information);

## 4. COMPARATIVE EVALUATION OF THE DIFFERENT APPROACHES

There is no doubt that coding of audio and video information is an exciting area of research. The capability of digging deep inside the audio signal and detecting the information buried in it by the sound producing process (human being or musical instrument) or of understanding the content of a scene by processing sequences of samples constitutes a great attraction. This, coupled with the obvious interest in exploiting these techniques for audio-visual products and services, explains why the audio-visual coding area has seen active participation from telecommunications, broadcasting and audio-visual equipment manufacturers, computer manufacturers and, of course, academia. No doubt, more and more ideas will be produced for many years to come.

Having said that, if we have to decide whether we should wait for another round of evolution in audio-visual coding or we can take what is being offered today and make good use of it, one has to assess the situation at this particular historical moment.

Even though the answer to this question goes beyond the purely technical domain it does not belong entirely to policy making either. This because only certain audio-visual applications fall under regulatory constraint and not to the same extent in all countries.

### 4.1. Digital video

Digital video is not a world awaiting to be created anew, as it has existed for quite some time in different forms. CCITT Recommendation H.120 (dated 1984) for video conference at 1.5/2

Mbit/s was probably the first official recognition of its existence and this was later followed by CCITT Recommendation H.261 (video conference/video telephony at px64 kbit/s, in 1990), by CCIR Recommendation 723 (contribution codec at 34/45 Mbit/s, again in 1990) and ISO/IEC IS 11172 (the so-called MPEG-1 standard in 1993), not to mention the almost countless proprietary implementations of video codecs for different applications stretching from video telephones to fully fledged HDTV codecs.

Because the technology enabling people to make digital television products is here and because many are eager to put such products to good uses there is hardly time today for all-encompassing philosophies of digital television.

A common feature of most of the existing implementations of digital video is the gradual increase in image resolution and, correspondingly, the bit-rate involved. This fact should not be a surprise since it is merely a translation of the evolution of the technological capabilities of signal processing. Lastly, a feature almost universally supported by implementations is the use of motion-compensated predictive/interpolated DCT as the basic ingredient for the reduction of spatial and temporal redundancy of the video signal.

### 4.2. DCT versus other techniques

Motion-compensated/interpolated DCT is *the* solution, not because it is "optimum" (what is optimum?) but because this technique has been the object of countless refinements in the research world, countless hardware implementations, some for trial and some for actual products in the market now, and the VLSI technology implementing it has gone through a number of refinements. The degree of compression offered by this technique is such to enable the offering of digital television services on a variety of distribution media: cable, satellite, terrestrial emission, telephone twisted pairs, optical digital storage media, digital magnetic tapes. Lastly, also some functionalities claimed to be needed by some delivery media, such as scalability, are supported by this technique.

What is then offered by competing techniques? Certainly a better theoretical explanation of their working. Sub-band coding gives added freedom in the design of the frequency analysis kernel so that it becomes possible to design better filters, but then one loses the recognised decorrelating capabilities of the DCT. Wavelet coding gives a further dimension in the design of filters by letting the designer decide on a block-by-block basis the optimum trade-off between frequency and space analysis, but convincing demonstrations that the system works are still awaited, not to mention descriptions of how it can be actually implemented. Singular-value decomposition gives the largest freedom of all by permitting every single block to be represented by the best possible transformation bases but the level of calculation needed is staggering.

There is a further element to be added to the picture. The description of the main compression algorithms given above has been mostly confined to the "intraframe" coding part of the compression work. The high reduction factors than can be achieved today (say 40:1, without noticeable loss of quality) can be obtained because of clever ways to reduce the "interframe" redundancy. The different algorithms introduced after DCT only provide different ways to reduce the intraframe redundancy, but most of them do not provide any alternative means to reduce interframe redundancy. Hence one well known statement,"Decorrelation method (DCT, sub-band, wavelet etc.) itself is irrelevant; what matters is the clever way you reduce interframe redundancy, control your buffer etc".

In the field of audio, predictive techniques are still predominant in the speech-coding field, particularly because of the tenet held so much in respect by ITU-TSS WP 15/1, but for high-quality audio more advanced techniques have been used. Sub-band coding schemes and DCT schemes have been proposed, with better performance shown by DCT schemes. Implementation complexity considerations have prompted the choice of sub-band as the preferred solution. A DCT-sub-band solution has been selected for more performance-demanding applications.

## 5. APPLICATIONS AND SYNERGIES WHICH MPEG-2 OFFERS

One of the reasons of the success of the MPEG standards lies in the decorrelation between the application domains and the "coding" function. The joint consideration of these two aspects has been the traditional approach adopted by other standards committees.

### 5.1. Separation of coding from application

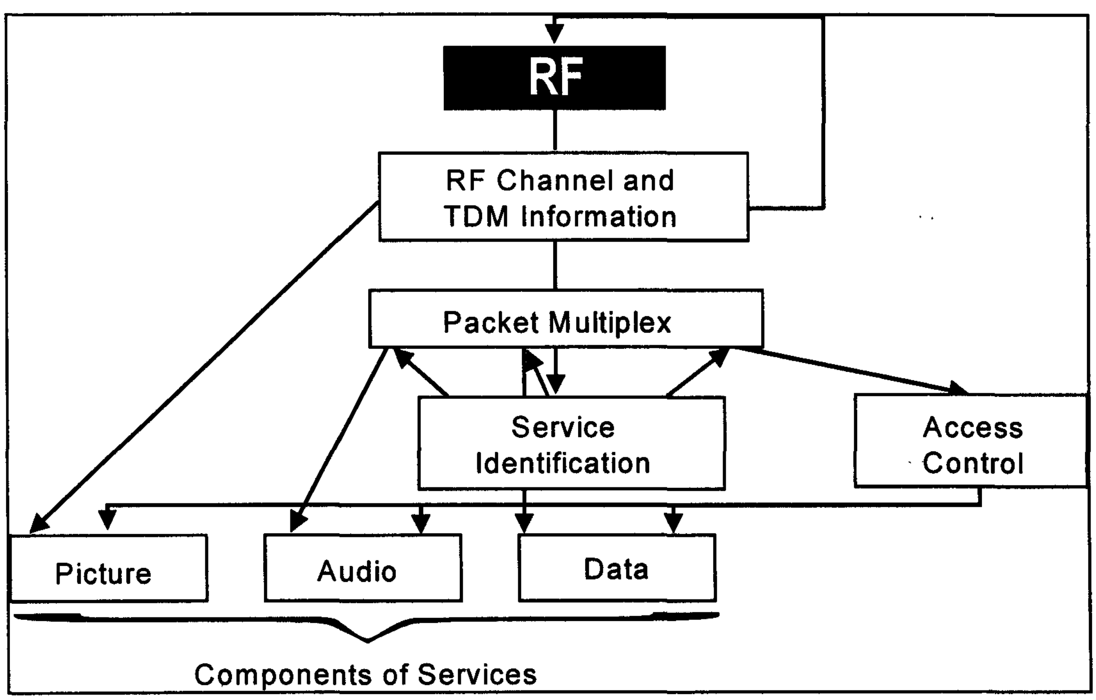



*Figure 1: Example of an "old" audiovisual system: D2-MAC2*

The figure above gives a schematic representation of the traditional approach and is taken from the D2-MAC specifications. Every component that is needed by the system is specified, viz.:

- The DPCM technique uses the time dimension to increase the effectiveness of the prediction;
- The linear transformation is also applied in the time dimensions;
- Motion detection (only areas in a picture that are "changed" compared with the previous picture are coded - this technique was standardised by the CCITT for video conference applications in the Recommendations of the H.100 series);

- Motion estimation (in general, from one picture to the next, only part of the information is new; the rest can be predicted using motion information and only the prediction error is transmitted);
- Interpolation (if two pictures are known, the pictures in between can be interpolated using motion information);

The specifications go as far as to specify the character set to be used by D2-MAC.

Incidentally, even if in the digital world this approach is no longer adopted, one has to remember that a broadcasting system has to have all its components specified, in particular the character set to be used. And if the digital solution promises to be universal - and audio-video coding does - one should remember that there are other parts that need to be universally agreed.

## 5.2. A layered reference model

The reference scheme of the digital audio-visual world should be based on a layered reference model as depicted in the figure below. If audio-visual (and other) information is represented (coded) in a single way, opportunities for interworking between applications via different delivery media are maximised.

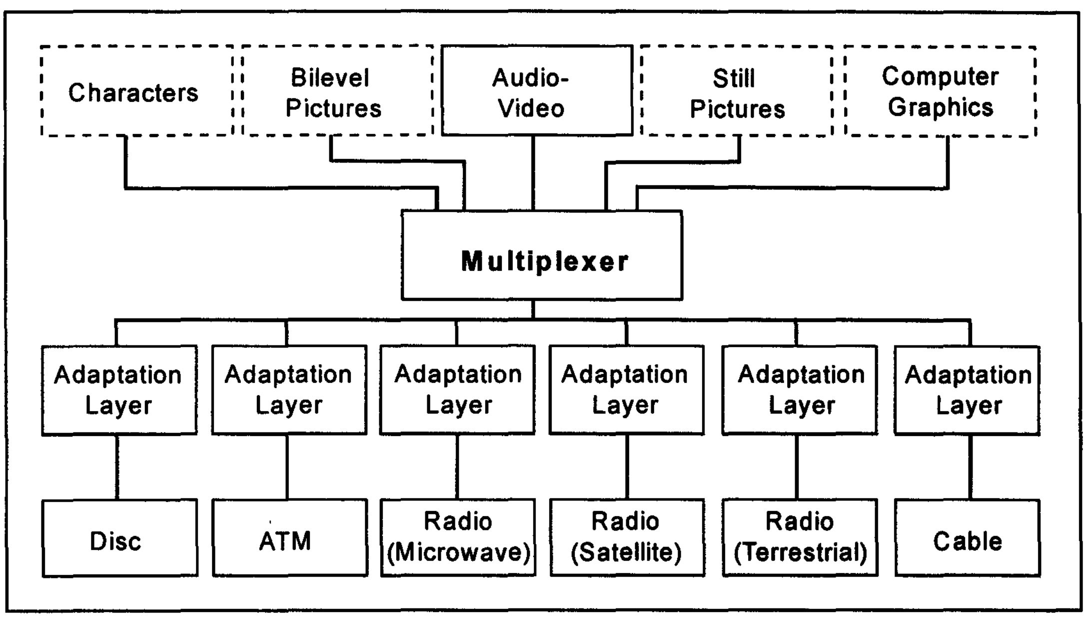



*Figure 2: Layering for audiovisual communication*

The preceding figure can be taken as a reference for the way MPEG has approached the problem of audio-visual coding. In MPEG, video coding is defined as a function operating on a 3-dimensional array of pixels, where the first two are associated with a frame and the third corresponds to the time. This is one reason why the MPEG standards are called generic. Only very general limits to the parameters associated with the array are given. In the case of MPEG-1 the maximum size of a picture is 4,096x4,096 pixels and the interval between pictures ranges from 1/24 to 1/60 seconds.

## 5.3. The MPEG-1 approach

This approach serves the purpose of satisfactorily decorrelating the definition of the compression coding from the application dependent parameters. It does not, however, give a simple way to specify how equipment should be constructed. Taking the example of MPEG-1, it is clear that if all equipment complying with the standard has to be equipped with as much RAM as necessary to deal with 4Kx4K pictures and with a frame rate of 60 Hz, the standard would never become, as it is intended to do, a mass market product.

This is the reason why in MPEG-1 the standard specifies a set of parameters, called the Constrained Parameter Set, set out in the table below.

| Parameter | Value | Comments |
|---|---|---|
| Horizontal size | <=768 | |
| Vertical Size | <=576 | |
| No. of macro block/picture | <=396 | =288/16x352/16 |
| No. of macro block/second | <=9900 | =396x25 |
| Picture rate | <=30 Hz | |
| Interpolated pictures | <=2 | |
| Bit-rate | <=1856 kbit/s | |

*Table 2: MPEG-1/video constrained parameter set*

This table highlights the fact that an implementation of MPEG-1 is constrained by:

1. the size of the RAM (a multiple of 288x352)

2. the number of memory accesses (a multiple of 288x352x30)

3. the input bit-rate and

4. the number of pictures that need to be interpolated.

But, apart from that, it is absolutely irrelevant if the pictures are from, e.g., an NTSC or PAL source.

Of course the actual selection of parameters is indeed important, but this is only at the application level. This is the reason why an MPEG-1/Video bit stream signals the value of several parameters called application - dependent parameters. They are given in the following table.

| Parameter | Values allowed |
|---|---|
| Pixel aspect ratio | square |
| | 16:9 (625) |
| | 16:9 (525) |
| | 4:3 (625) |
| | 4:3 (525) |
| | 8 other values |
| | 1 reserved value |
| picture rate | 23.976 |
| | 24 |
| | 25 |
| | 29.97 |
| | 30 |
| | 50 |
| | 59.94 |
| | 60 |
| | 7 reserved values |
| samples/line | up to 4096 |
| lines/picture | up to 4096 |

*Table 3: MPEG-1/video application - dependent parameters*

## 5.4. The MPEG-2 approach

This philosophy has been further extended in the case of the MPEG-2 standard. Here the situation is further complicated by the fact that the coding algorithm does not materialise in a single instance, but in different "profiles": scalable (Next), non-scalable (Main) and reduced (Simple). To each of these "levels" have been allocated, this word being used in a very similar way as the constrained parameter set above. At this moment three levels have been identified:

- Low Level, roughly corresponding to the CIF or SIF;
- Main Level, corresponding to conventional television and

- High Level, corresponding to HDTV.

While the first two levels do not constitute a particular problem, thanks to the fact that there is a considerable convergence of those values that are important for an implementation of the standard (it helps remember that 576x720x25 = 480x720x30), the situation is much more complex for HDTV, due to the large variation of parameter values (see Tab. 1 on different HDTV formats). It has to be decided whether there will be a single High Level or if it will have to be split into two levels.

The task of developing a generic standard enables one to get rid of certain nasty problems but does indeed create new ones. This because the work has to be done in such a way that applications requirements are taken into account, without having the resulting work specifically matched to them.

The approach adopted has been to identify the likely applications needing an MPEG standard. The table below lists the applications identified so far.

| | |
|---|---|
| BSS | Broadcasting Satellite Service (to the home) |
| CATV | Cable TV Distribution on optical networks, copper cables, etc. |
| CDAD | Cable Digital Audio Distribution |
| DAB | Digital Audio Broadcasting (terrestrial and satellite broadcasting) |
| DTTB | Digital Terrestrial Television Broadcast |
| EC | Electronic Cinema |
| ENG | Electronic News Gathering (including SNG, Satellite News Gathering) |
| FSS | Fixed Satellite Service (e.g. to headends) |
| HTT | Home Television Theatre |
| IPC | Interpersonal Communications (videoconferencing, videophone, etc.) |
| ISM | Interactive Storage Media (optical disks, etc.) |
| MMM | MultiMedia Mailing |
| NCA | News and Current Affairs |
| NDB | Networked Database Services (via ATM, etc.) |
| RVS | Remote Video Surveillance |
| SSM | Serial Storage Media (digital VTR, etc.) |
| TTV | Terrestrial TV Broadcasting |

*Table 4: Applications target of the MPEG-2 standard*

On the basis of requirements issued by representatives of the different application areas, the following list of general requirements has been produced:

**System**

1. Scalability of audio and video
2. Robustness to bit errors and cell loss
3. MPEG-1 and H.261 backward compatibility
4. Random access / channel hopping
5. Special (or trick) modes
6. Multi-channel and multi-lingual audio
7. Multiple programs
8. Elementary stream identification signals
9. Encryption and scrambling
10. Editing
11. ATM support
12. Variable bit rate operation

**Video**

1. Picture format
2. Picture quality
3. Flexibility in bit rates
4. Coding decoding delay
5. Random access/channel hopping
6. Bit stream scalability
7. Complexity flexibility
8. Compatibility
9. Editing encoded bit streams
10. Trick mode
11. Repetition of coding and decoding
12. Adaptation to storage and transport methods
13. Video windowing

**Audio**

1. Multi-channel audio
2. Multi-lingual audio
3. Extension of 11172-3 to a lower sampling frequency

Accommodating all the requirements listed above into a single coding syntax would yield a very complex standard, one that would probably fall into the non-zero set of wishful standards. But because of the disparate application fields addressed, not all implementations need all the tools, given that in general inclusion of a tool gives rise to additional hardware costs (more memory, faster processing etc.) which another application is unlikely to be willing to support if it does not make use of it. Still, such a lack of uniformity has a great impact on the possibility to interchange information.

## 5.5. The concept of profiles

A compromise has been adopted consisting of the definition of a family of harmonised subsets of the full syntax, called profiles. The concept of profiles is well-established within ISO/IEC JTC1 and is defined as *sets of one or more base standards, and, where applicable, the identification of chosen classes, subsets, options and parameters of those base standards, necessary for accomplishing a particular function.* Because, among others, picture resolution is an important cost element, each profile is usually associated with several levels, a decoder of level N being able to decode bit streams up and including that level.

At the moment three profiles are likely to be defined:

- ***The main profile.*** This is characterised by the capability to provide very high quality pictures with very compact implementations. In fact this profile is targeted to satisfy a large number of major initial applications. Within the main profile, the main level is particularly important. This corresponds to the resolution of conventional television and is therefore likely to satisfy a large number of major initial applications.
- ***The simple profile.*** This is a simplified version of the Main Profile requiring less memory.
- ***The scalable profile.*** This gives the possibility of providing "nested" pictures in an encoded form such that, e.g. a standard TV decoder can extract useful information from an HDTV bit-stream. Scalability takes one of three main forms:
    1) Spatial scalability that provides, among other features reverse compatibility with MPEG-1 and H.261 decoders. The attribute "spatially scalable" refers to the fact that this nesting of bit-streams can be repeated at different spatial resolutions.
    2) S/N scalability that gives the capability to provide pictures with different levels of quality but at the same resolution
    3) Frequency scalability where pictures of different resolutions can be obtained by operation in the frequency domain.

The profile approach enables MPEG to keep its promise to support all application requirements in the sense that a specific requirement is at least supported by one profile. What is still open to discussion is which set of requirements is supported by which profile(s), bearing in mind that to allow every possible set of requirements to become a profile would mean allowing the number of profiles to grow beyond what is practically acceptable and desirable. This would certainly harm the degree of interoperability that would otherwise be possible.

The figure below gives a graphical representation of the concept of profiles and levels.

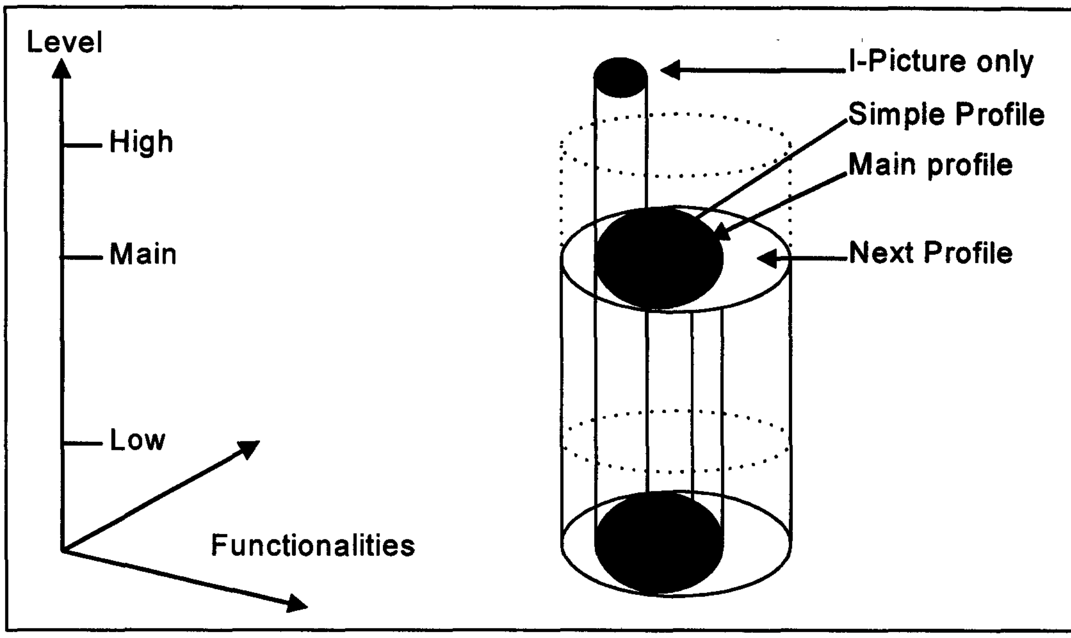



*Figure 3: MPEG profiles and levels*

### 5.6. MPEG targets functions and interworking

An additional feature, possibly less known, of the MPEG standards (all three parts of them) is the fact that the standards specify only the functions that a decoder (be it a demultiplexer, a video or audio decoder) has to perform on the incoming bit-stream. This approach has shaken many a deep-rooted belief in some audio-visual fields, where people were accustomed to be told exactly what to do and how well a certain piece of equipment would work.

The approach followed by MPEG is that a standard should specify things to the extent that they are needed to enable interworking. This means that the bit-stream syntax is specified, but not the way the bit-stream is produced. This has the benefit of allowing the application of the standards to be open to future improvements, insofar as encoders may undergo successive phases of optimisation, possibly of a proprietary nature.

This approach can be clearly observed in the MPEG-1/Video standard. The compression algorithm was developed using a rough model (called Simulation Model), selected for the purpose of comparing the different improvements proposed. In the 18 months since the algorithm has been frozen, the quality of pictures has constantly improved, thanks to different encoder optimisations and is now largely superior to VHS.

## 6. DETAILED DESCRIPTION OF THE MPEG-2 SOURCE CODING

### 6.1. Video

Since MPEG-2/Video is basically an extension of MPEG-1/Video, the latter will be briefly described first.

MPEG-1/Video is the short name for ISO/IEC IS 11172-2, i.e. part 2 of International Standard 11172. ISO/IEC reflects the fact that the environment in which the standard has been developed - JTC1 - is a Joint ISO/IEC Technical Committee.

MPEG-1/Video specifies a coded representation that can be used for compressing video sequences to bit-rates around 1.5 Mbit/s. Its use means that motion video can be manipulated as a form of computer data and can be transmitted and received over existing and future networks. The coded representation can be used with both 625-line and 525-line television and provides flexibility for use with workstation and personal computer displays.

MPEG-1/Video was developed to operate principally from storage media offering a continuous transfer rate of about 1.5 Mbit/s. Nevertheless it can be used more widely than this because the approach taken is generic.

The coded representation achieves a high compression ratio while preserving good picture quality. The algorithm is not lossless, as the exact pel values are not preserved during coding. The choice of the techniques is based on the need to balance a high picture quality and compression ratio with the requirement to allow random access to the coded bit stream. Obtaining good picture quality at such low bit-rates demands a very high compression ratio, which is not achievable with intraframe coding alone. The need for random access, however, is best satisfied with pure intraframe coding. This requires a careful balance between intra- and interframe coding and between recursive and non-recursive temporal redundancy reduction.

### 6.1.1. MPEG-1 compression techniques

A number of techniques are used to achieve a high compression ratio. The first, which is almost independent from the standard, is to select an appropriate spatial resolution for the signal. The algorithm then uses block-based motion compensation to reduce the temporal redundancy. Motion compensation is used for causal prediction of the current picture from a previous picture, for non-causal prediction of the current picture from a future picture, or for interpolative prediction from past and future pictures. Motion vectors are defined for each 16-pel by 16-line region of the picture. The difference signal, the prediction error, is further compressed using the discrete cosine transform (DCT) to remove spatial correlation before it is quantised in an irreversible process that discards the less important information. Finally, the motion vectors are combined with the DCT information, and coded using variable length codes.

Because of the conflicting requirements of random access and highly efficient compression, three main picture types are defined. Intra-coded pictures (I-Pictures) are coded without reference to other pictures. They provide access points to the coded sequence at which decoding can begin, but are coded with only a moderate compression ratio. Predictive coded pictures (P-Pictures) are coded more efficiently using motion compensated prediction from a past intra or predictive-coded picture and are generally used as a reference for further prediction. Bidirectionally-predictive coded pictures (B-Pictures) provide the highest degree of compression but require both past and future reference pictures for motion compensation. Bidirectionally-predictive coded pictures are never used as references for prediction. The organisation of the three picture types in a sequence is very flexible. The choice is left to the encoder and will depend on the requirements of the application. The figure below illustrates the relationship between the three different picture types.

The fourth picture type defined, the D-picture, is provided to allow a simple, but limited quality, fast-forward playback mode.

The choice of 16 by 16 macroblocks for the motion-compensation unit is a result of the trade-off between increasing the coding efficiency provided by using motion information and the overhead needed to store it. Each macroblock can be one of a number of different types. For example, intra-coded, forward-predictive-coded, backward-predictive coded, and bidirectionally-predictive-coded macroblocks are permitted in bidirectionally-predictive coded pictures. Depending on the type of the macroblock, motion vector information and other size information are stored with the compressed prediction error signal in each macro block. The motion vectors are encoded differentially with respect to the last coded motion vector, using variable-length codes. The maximum length of the vectors that may be represented can be programmed, on a picture-by-picture basis, so that the most demanding applications can be met without compromising the performance of the system in more normal situations.

It is the responsibility of the encoder to calculate appropriate motion vectors and the standard does not specify how this should be done.

Both original pictures and prediction error signals have high spatial redundancy. MPEG-1/Video uses a block-based DCT method with visually weighted quantisation and run-length coding. Each 8 by 8 block of the original picture for intra-coded macroblocks or of the prediction error for predictive-coded macroblocks is transformed into the DCT domain where it is scaled before being quantised. After quantisation many of the coefficients are zero in value and so two-dimensional run-length and variable-length coding is used to encode the remaining coefficients efficiently.

The standard does not specify an encoding process. It specifies the syntax and semantics of the bit-stream and the signal processing in the decoder. As a result, many options are left open to encoders to trade-off cost and speed against picture quality and coding efficiency.

The input video signal must be digitised and represented as a luminance and two colour difference signals (Y, $C_b$, $C_r$). This may be followed by pre-processing and format conversion to select an appropriate window, resolution and input format. The standard requires that the colour difference signals ($C_b$ and $C_r$) are sub-sampled with respect to the luminance by 2:1 in both vertical and horizontal directions and are reformatted, if necessary, as a non-interlaced signal.

The encoder must choose which picture type to use for each picture. Having defined the picture types, the encoder estimates motion vectors for each 16 by 16 macro block in the picture. In P-Pictures one vector is needed for each non-intra macro block and in B-Pictures one or two vectors are needed.

If B-Pictures are used, some reordering of the picture sequence is necessary before encoding. Because B-Pictures are coded using bidirectional motion-compensated prediction, they can only be decoded after the subsequent reference picture (an I or P-Picture) has been decoded. Therefore the pictures are reordered by the encoder so that the pictures arrive at the decoder in the correct order for decoding. The correct display order is recovered by the decoder.

The basic unit of coding within a picture is the macro block. Within each picture, macroblocks are encoded in sequence, left to right, top to bottom. Each macro block consists of six 8 by 8 blocks: four blocks of luminance, one block of Cb chrominance, and one block of Cr chrominance. See the figure below. Note that the picture area covered by the four blocks of luminance is the same as the area covered by each of the chrominance blocks. This is due to sub-sampling of the chrominance information to match the sensitivity of the human visual system.

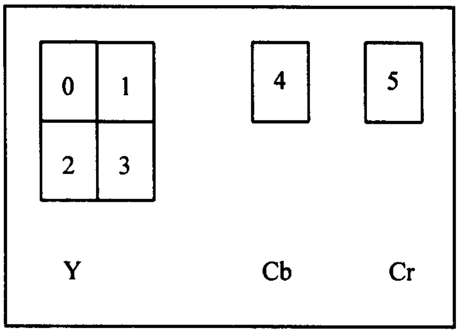



*Figure 4: Macro block structure*

Firstly, for a given macroblock, the coding mode is chosen. It depends on the picture type, the effectiveness of motion-compensated prediction in that local region, and the nature of the signal within the block. Secondly, depending on the coding mode, a motion-compensated prediction of the contents of the block based on past and/or future reference pictures is formed. This prediction is subtracted from the actual data in the current macroblock to form an error signal. Thirdly, this error signal is separated into 8 by 8 blocks (4 luminance and 2 chrominance blocks in each macroblock) and a discrete cosine transform is performed on each block. Each resulting 8 by 8 block of DCT coefficients is quantised and the two-dimensional block is scanned in a zigzag order to convert it into a one-dimensional string of quantised DCT coefficients. Fourthly, the side-information for the macroblock (mode, motion vectors etc) and the quantised coefficient data are encoded.

For maximum efficiency, a number of variable-length code tables are defined for the different data elements. Run-length coding is used for the quantised coefficient data.

A consequence of using different picture types and variable-length coding is that the overall data rate is variable. In applications that involve a fixed-rate channel, a FIFO buffer may be used to match the encoder output to the channel. The status of this buffer may be monitored to control the number of bits generated by the encoder. Controlling the quantisation process is the most direct way of controlling the bit-rate. The standard specifies an abstract model of the buffering system (the Video Buffering Verifier) in order to constrain the maximum variability in the number of bits that are used for a given picture. This ensures that a bit-stream can be decoded with a buffer of known size.

At this stage, the coded representation of the picture has been generated. The final step in the encoder is to regenerate I-Pictures and P-Pictures by decoding the data so that they can be used as reference pictures for subsequent encoding. The quantised coefficients are dequantised and an inverse 8 by 8 DCT is performed on each block. The prediction error signal produced is then added back to the prediction signal and limited to the required range to give a decoded reference picture.

### 6.1.2. Decoding

Decoding is the inverse of the encoding operation. It is considerably simpler than encoding as there is no need to perform motion estimation and there are many fewer options. The decoding process is defined by the standard.

For fixed-rate applications, the channel fills a FIFO buffer at a constant rate with the coded bit stream. The decoder reads this buffer and decodes the data elements in the bit stream according to the defined syntax.

As the decoder reads the bit stream, it identifies the start of a coded picture and then the type of the picture. It decodes each macroblock in the picture in turn. The macroblock type and the motion vectors, if present, are used to construct a prediction of the current macroblock based on past and future reference pictures that have been stored in the decoder. The coefficient data are decoded and dequantised. Each 8 by 8 block of coefficient data is transformed by an inverse DCT, and the result is added to the prediction signal and limited to the defined range.

After all the macroblocks in the picture have been processed, the picture has been reconstructed. If it is an I-picture or a P-picture it is a reference picture for subsequent pictures and is stored, replacing the oldest stored reference picture. Before the pictures are displayed they may need to be reordered from the coded order to their natural display order. After reordering, the pictures are available, in digital form, for post-processing and display in any manner that the application chooses.

### 6.1.3. MPEG-1 syntax

A syntax for a coded video bit stream is specified by the standard. This syntax contains six layers, each of which supports either a signal processing or a system function:

***Sequence layer****:* corresponds to the start of a new video sequence; it contains a header specifying global parameters matching the statistics of the particular sequence being coded.

***GOP layer****:* the coder/decoder process the input pictures in groups called Group Of Pictures (GOP), usually but not mandatorily composed of 10 to 15 pictures. This feature introduces advantages at resynchronisation level (when transmission is made in a non-error-free environment) and for the random access possibility. Within a GOP, pictures are classified according to the irrespective coding type: Intra (I), Predicted (P) and Bidirectional (B) pictures.

***Picture layer****:* this layer contains the information at picture level having three main optional coding modes - intra, forward prediction and interpolated prediction. At this level, the bit allocation is performed based on a rather complex pre-analysis step in order to maintain a near-constant quality coded sequence.

***Slice layer****:* this layer is the main resynchronisation point in the bit-stream corresponding to a small area of the picture. One slice correspond to 16 lines of the picture in either interlaced or progressive formats. The coding rate control is made at this level (slice granularity) by defining the quantisation parameter (Qp) as function of the output buffer fullness.

***Macroblock layer:*** it corresponds to the first "signal processing" function - the motion prediction and compensation (MC). MC methods used in MPEG assume that locally the current picture may be well predicted by translation of the picture at some previous (or future) time. This means that the "amount" and direction of the movement need not be the same everywhere in the picture. The basic coding unit used in the MC process is the macro block, 16 pixels x 16 lines of picture area, in interlaced or progressive format. This area size results from the trade-off between the coding gain provided by motion compensation and the overhead introduced to code the motion vector data. It is also possible, at this level, to modulate the Qp (producing the mquant), adapting it to the macro block "activity and criticality" in order to optimize the quantisation of the 16x16 area being coded.

***Block layer***: in this layer the two-dimensional DCT transform is performed on 8x8 data blocks, followed by quantisation and entropy coding processes.

### 6.1.4. MPEG-2 developments

The base algorithm in the MPEG-2/Video standard is a hybrid DCT/Motion Compensation/Interpolation having a similar structure to MPEG-1. The two main areas addressed by MPEG-2/Video are the efficient handling of interlaced video sources and the support of scalability. Further the algorithm has been conceived to be generic, i.e. independent of the application/transport medium.

Scalability became a key concept in the recent developments on video coding research motivated by the compatibility of MPEG-2 scheme with existing algorithms, consolidated in terms of international standardisation bodies (H.261, MPEG-1), but also as a powerful technique to deal with more general questions like those raised by multi-resolution video source formats, variable bit-rate coding and ATM network interfacing, error concealment techniques for noisy channel transmission, coexistence/expandability with/to different, new generation coding schemes, etc.

By scalability is meant the ability to have more than one picture resolution and/or quality level readily available in the bit-stream feeding a target decoder. Scalability may be performed according to different techniques summarised in the following:

***Spatial scalability:*** these techniques perform spatial resolution conversions in the pixel domain. The (forward and backward) compatibility is a specific implementation of spatial scalability. A particular case of scalability (called noise or SNR scalability) is obtained by using two layers with the same spatial resolution where the main objective is no longer multi-resolution handling but the efficiency of a layered coding algorithm for quality optimisation.

***Frequency scalability:*** to produce a lower resolution layer in the frequency domain (DCT), a subset of the DCT coefficients is coded as a base layer. This base information may be used as a prediction for the upper layers within a hierarchical pyramidal structure. Alternative schemes based on frequency scanning simply encode the complementary high frequency subsets until all coded DCT coefficients are transmitted (band selection).

***Rate scaling:*** in this particular technique the bit stream contains a multiplicity of temporal resolutions, (e.g. coding separately even and odd parity fields of a 601 frame according to the MPEG-1 algorithm). Hybrid techniques, using the above mentioned scalability modes, may be implemented, which increases the coder/decoder complexity.

The transmission of scalable bit streams is intimately related to the *layered coding* concept. It may be divided in two transmission modes:

***Simulcast mode***, which divides the available bit-rate, keeping all layers independent. This scheme does not use the base layer for prediction and gives less efficient bandwidth/quality trade-off. It is generally used as a benchmark for performance evaluation and it represents the least complex mode to achieve compatibility.

***Embedded mode***, which uses syntax extensions ("gateways") to enable the unambiguous decoding of the different resolution layers. This code may be further divided in two sub-modes:

***Tightly coupled:*** by this concept it is intended that both layers (in the simple case of a two-layer codec) must belong to the same *family* or more generally, it imposes constraints on the coding schemes used in the different coding layers. This case is typical of a frequency scalable bit-stream which produces information based on the DCT coefficients constraining both layers to use the same *base technique* to represent the information (in this case DCT).

***Loosely coupled,*** on the other hand, allows a great flexibility concerning the coding algorithm on both layers and is typical of spatially scalable schemes where the decimation/interpolation made in the pixel domain is transparent to the coding techniques used. This particular scalable/multi-layer coding mode is quite flexible concerning compatibility with existing and future-generation algorithms and has been proposed as one promising way to fulfil the many different requirements (application dependent) which cannot be satisfied by a single coding algorithm.

## 6.2. Audio

Since MPEG-2/Audio is a multi-channel coding extension of MPEG-1/Audio, the latter will be described briefly.

### 6.2.1. MPEG-1 audio

MPEG-1/Audio is the short name for ISO/IEC IS 11172-3, i.e. part 3 of International Standard 11172. ISO/IEC reflects the fact that the environment in which the standard has been developed - JTC1 - is a Joint ISO/IEC Technical Committee.

The encoder processes the digital audio signal and produces the compressed bit-stream for storage. The encoding algorithm is not standardised, and may use various means for encoding, such as estimation of the auditory masking threshold, quantisation, and scaling. However, the encoder output must be such that a decoder conforming to the specifications will produce audio suitable for the intended application.

Input audio samples are fed into the encoder. Samples are mapped, i.e. they are filtered and sub-sampled. The mapped samples may be called either sub-band samples (as in Layer I or II, see below) or transformed sub-band samples (as in Layer III). A psycho-acoustic model creates a set of data to control the quantiser and coding. These data are different depending on the actual coder implementation. One possibility is to use an estimation of the masking threshold to do this quantiser control. The quantiser and coding block creates a set of coding symbols from the mapped input samples. Again, this block can depend on the encoding system. The block "frame packing" assembles the actual bit stream from the output data of the other blocks, and adds other information (e.g. error correction) if necessary.

There are four different modes possible: single channel, dual channel (two independent audio signals coded within one bit stream), stereo (left and right signals of a stereo pair coded within one bit stream), and Joint Stereo (left and right signals of a stereo pair coded within one bit stream with the stereo redundancy exploited).

Depending on the application, different layers of the coding system with increasing encoder complexity and performance can be used. An MPEG-1/Audio Layer N decoder is able to decode bit stream data which has been encoded in Layer N and all layers below N.

Layer I

- This layer contains the basic mapping of the digital audio input into 32 sub-bands, with fixed segmentation to format the data into blocks, a psycho-acoustic model to determine the adaptive bit allocation, and quantisation using block companding and formatting. The theoretical minimum encoding/decoding delay for Layer I is about 19 ms.

Layer II

- This layer provides additional coding of bit allocation, scale factors and samples. Different framing is used. The theoretical minimum encoding/decoding delay for Layer II is about 35 ms.

Layer III

- This layer introduces increased frequency resolution based on a hybrid filterbank. It adds a different (non uniform) quantiser, adaptive segmentation and entropy coding of the quantised values. The theoretical minimum encoding/decoding delay for Layer III is about 59 ms.

Joint Stereo coding can be added as an additional feature to any of the layers.

The decoder accepts the compressed audio bit-stream in the syntax, decodes the data elements, and uses the information to produce digital audio output.

Bit-stream data is fed into the decoder. The bit-stream unpacking and decoding block does error detection if error-check is applied in the encoder. The bit-stream data are unpacked to recover the various pieces of information. The reconstruction block reconstructs the quantised version of the set of mapped samples. Inverse mapping transforms these mapped samples back into uniform PCM.

Performance of MPEG-1/Audio was tested in July 1991 with excellent results. Ten very demanding audio sequences were selected and coded at 256 kbit/s. For no sequence was the subjectively assessed quality lower than 4.6 on the CCIR scale.

### 6.2.2. MPEG-2 audio

The MPEG-2/Audio goal is to provide a multichannel compatible extension of MPEG-1/Audio in the sense that an MPEG-1/Audio decoder can understand the two-channel component of an MPEG-2/Audio bit-stream (backward compatibility) and MPEG-2/Audio decoder can successfully decode an MPEG-1/Audio bit stream, of course by producing two-channel sound (forward compatibility). MPEG-2/Audio is also designed for use in different applications.

As regards stereophonic presentation, specialist groups of BR, SMPTE and EBU recommend the use of an additional centre loudspeaker channel C and two surround loudspeaker channels LS and RS, augmenting the front left and right loudspeaker channels L and R. This reference sound format is referred to as "3/2-stereo" (3 front / 2 surround loudspeaker channels), and requires the transmission of five appropriately-formatted sound signals.

For sound applications accompanying picture (e.g. HDTV), the three front loudspeaker channels ensure sufficient directional stability and clarity of the picture-related frontal images, according to the common practice in the cinema. The dominant benefit is the "stable centre", which is guaranteed at any location of the listener and important for most of the dialogue.

Additionally for audio-only applications the 3/2-stereo format has been found to be the appropriate improvement of two-channel stereophony. The addition of one pair of surround loudspeaker channels allows improved realism of auditory ambience.

A hierarchy of sound formats providing a lower number of loudspeaker channels and reduced presentation performance (down to 2/0 stereo or even mono) and a corresponding set of downwards matrixing equations are recommended in BR Recommendation 775: "Multichannel stereophonic sound system with and without accompanying picture", November 1992. Useful altemative lower level sound formats are 3/1, 3/0, 2n, 2/1, 2/0, 1/0, which may be used in circumstances where economic or channel capacity constraints apply. Corresponding loudspeaker arrangements are 3/2, 3/0, 2/2, 2/2, 2/0, 1/0.

For several applications it is the intention to improve the existing 2/0-stereo sound system step-by-step, by transmitting additional sound channels (centre, surround), without making use of simulcast operation. This provision of backwards compatibility with existing receivers implies the use of compatibility matrices: the decoder of the previous generation must reproduce the two conventional basic stereo signals Lo/Ro, and the multichannel decoder produces the complete 3/2-stereo presentation L'/C'/LS'/RS' from the basic stereo signal, and the extension signals 1.

Particularly for HDTV applications, multi-channel stereo performance on the one hand and bilingual programs or multi-lingual commentaries on the other are required. MPEG-2/Audio provides alternative sound channel configurations in the five-channel sound system, for example a bilingual 2/0-stereo program or one 2/0, 3/0-stereo sound plus accompanying services (e.g. "clean dialogue" for the hard-of-hearing, commentary for visually impaired people, multi-lingual commentary etc). An important configuration is the reproduction of one selected commentary dialogue or dialogue channel (via a centre loudspeaker together with the common music/effect stereo down-mix (examples are documentary films, sports reportage).

In basic terms, the transmission of the five audio signals of a 3/2 sound system requires five transmission channels (although, in the context of bit-rate reduced signals, these are not necessarily totally independent). In order that two of the transmitted signals can provide a stereo service on their own, the source sound signals are generally combined in a linear matrix prior to encoding. These combined signals (and their transmission channels) are identified by the notation T1, T2, T3, T4 and T5.

### 6.2.3. Compatibility issues

Backwards and forwards compatibility with an MPEG-1/Audio decoder is provided. For a multi-channel audio bit stream, backwards compatibility means that an ISO 11172-3 IS audio decoder properly decodes the basic stereo information, consisting of basic left and right channels. The signals in these channels constitute an appropriate down-mix of the audio information in all channels. The appropriate down-mix equations are given below.

Lo= L+x*C+y*LS

Ro=R+x*C+y*RS

Forward compatibility means that an MPEG-2 multi-channel audio decoder is able to decode properly an MPEG-1/Audio bit stream.

The following combinations are possible:

| Basic Lo, Ro Stereo | Multi-channel Extension |
|---|---|
| Layer I | Layer II MC |
| Layer II | Layer II MC |
| Layer III | Layer II MC |

MPEG-2/Audio describes the combination of Layer I, II and *m* for the basic Lo, Ro stereo signal and of Layer MC and Layer III MC for the multi-channel extension.

The MPEG-2/Audio Multi-channel system provides full compatibility with MPEG-1/Audio. This compatibility is realised by using the compatibility matrix equations above in the composite coding mode and by exploiting the ancillary MPEG-1 data field in the audio frame.

The complete MPEG-1/Audio frame incorporates basically four different types of information:

- Header information within the first 32 bits of the MPEG-1 audio frame.
- Cyclic Redundancy Code check (CRC), consisting of 16 bits, just after the header information.

- Audio data, consisting of bit-allocation (BAL), scale-factor select information (SCFSI), scale factors (SCF), and the biggest part of the audio information, the sub-band samples.
- Ancillary data. Due to the large number of different applications which will use the standard, the length and usage of this field are not specified.

The lack of specified field length and usage of ancillary data opens the possibility of packing the complete extension information of the channels T3/T4/T5 into the first part of the ancillary data field. An MPEG-1/Audio decoder just ignores these data in the ancillary data field. Nevertheless it still can have access to ancillary data transmitted at the end of the audio frame, if the MC encoder does not use up all of the ancillary data field for the multi-channel extension information. This remaining part of the ancillary data field can be decoded by an MPEG-1/Audio decoder irrespective of the variable length of the ancillary data field. The bit-rate required for the multi-channel extension information may vary on a frame-to-frame basis, depending on the sound signals. However, the overall bit-rate is constant and in accordance with the bit-rates specified in MPEG-1/Audio

### 6.2.4. Encoder and decoder options

The audio input/output format has the following characteristics

Sampling frequencies: 48, 44.1 or 32 kHz.

Quantization: up to 24 bits/sample PCM resolution.

Possible encoder input configurations are:

a) Five channels, using the 3/2 configuration L, C, R plus two-channel surround LS, RS
b) Four channels, using the 3/1 configuration L, C, R plus single-channel surround S
c) Four channels, using the 2/2 configuration L, R plus two-channel surround LS, RS
d) Four channels, using the bilingual 2/0+2/0 configuration L, R first program plus L2, R2 of second program.
e) Three channels using the 3/0 configuration L, C, R without surround
f) Three channels using the 2/1 configuration L, R without surround
g) Two channels, using the 2/0 configuration L, R stereo (as in MPEG-1/Audio)
h) One channel, using the 1/0 configuration Mono (as in MPEG-1/Audio)

Possible decoder output configurations for the main service presentation modes are:

a) Five channels, using the 3/2 configuration

   Front: Left (L) and right (R) channels plus centre channel (C)

   Surround: Left surround (LS) and right surround (RS)

b) Four channels, using the 2/2 configuration

   Front: Left (L) and right (R) channel

Surround: Left surround (LS) and right surround (RS)

c) Three channels, using the 3/0 configuration

Front: Left (L) and right (R) channel plus centre channel (C)

Surround: No surround

d) Two channels, using the 2/0 configuration

Front: Left (L) and right channel (R)

Surround: No surround

e) One channel output, using the 1/0 configuration

Front: Mono channel (Mo)

Surround: No surround

A low-frequency enhancement channel (LFE channel) can optionally be added to any of these configurations. The purpose of this channel is to enable listeners, if they wish, to extend the low frequency content in terms of both frequency and level. In this way it is the same as the LFE channel proposed by the film industry for their digital sound systems.

## 7. DESCRIPTION OF MPEG-2 MULTIPLEXING

As of today the MPEG-2/System specifications have extended the MPEG-1/System specifications to provide more functionalities. For this reason the latter is first described in some detail.

The systems specification addresses the problem of combining one or more data streams from MPEG-1 video and audio with timing information to form a single stream. Once combined into a single stream, the data are in a form well-suited to digital storage or transmission. The syntactical and semantic rules imposed by this systems specification enable synchronised playback without overflow or underflow of decoder buffers under a wide range of stream retrieval or receipt conditions. The scope of syntactical and semantic rules set forth in the systems specification differ: the syntactical rules apply to systems layer coding only, and do not extend to the compression layer coding of the video and audio specifications; by contrast, the semantic rules apply to the combined stream in its entirety.

The systems specification does not specify the architecture or implementation of encoder or decoders. However, bit-stream properties do impose functional and performance requirements on encoders and decoders. For instance, encoders must meet minimum clock tolerance requirements. Notwithstanding this and other requirements, a considerable degree of freedom exists in the design and implementation of encoders and decoders.

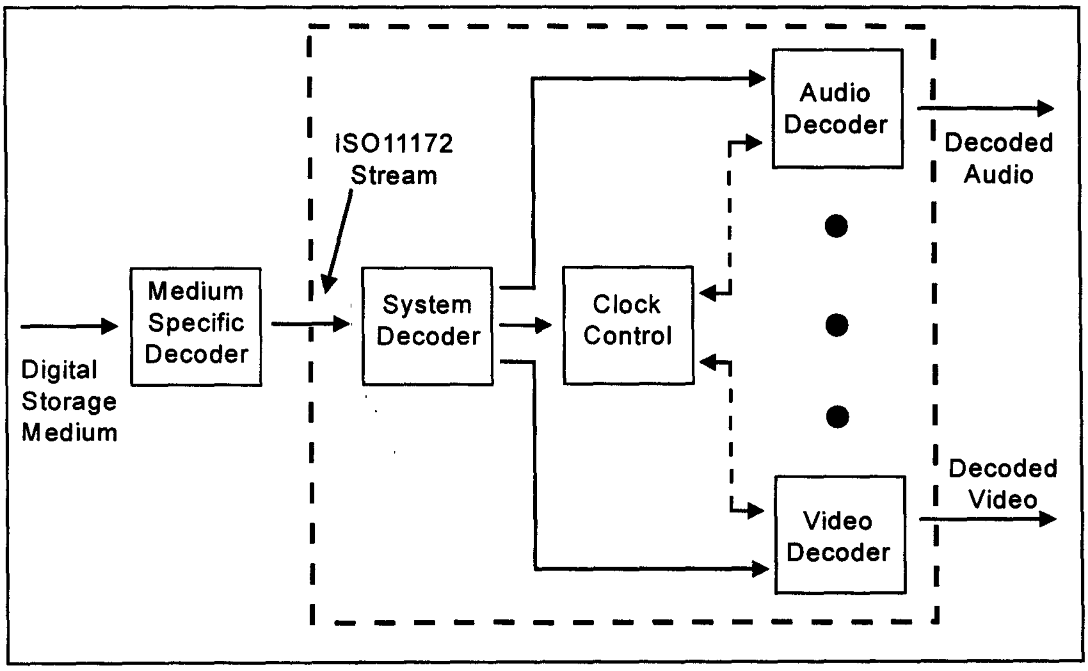
### 7.1. A prototype MPEG-1 decoder

A prototypical audio/video decoder system is depicted in the figure below to illustrate the function of an MPEG-1 decoder. The architecture is not unique, however, since System Decoder functions including decoder timing control might equally well be distributed among elementary stream decoders and the Medium Specific Decoder. The prototypical decoder design does not imply any normative requirement for the design of an MPEG-1 decoder. Indeed non-audio/video data is also allowed, but not shown.

The prototypical MPEG-1 decoder is composed of System, Video, and Audio decoders conforming to Parts 1, 2, and 3, respectively, of ISO/IEC 11172. In this decoder the multiplexed coded representation of one or more audio and/or video streams is assumed to be stored on a digital storage medium (DSM), or network, in some medium-specific format. The medium specific format is outside of the MPEG-1 specifications, nor is the medium-specific decoding part of the prototypical MPEG-1 decoder.

The prototypical decoder accepts as input an MPEG-1 multiplexed stream and relies on a System Decoder to extract timing information from the stream. The System Decoder demultiplexes the stream, and the elementary streams so produced serve as inputs to Video and Audio decoders, whose outputs are decoded video and audio signals. Included in the design, but not shown in the figure, is the flow of timing information among the System Decoder, the Video and Audio Decoders, and the Medium Specific Decoder. The Video and Audio Decoders are synchronised with each other and with the DSM using this timing information.

MPEG-1 multiplexed streams are constructed in two layers: a system layer and a compression layer. The input stream to the System Decoder has a system layer wrapped around a compression layer. Input streams to the Video and Audio decoders have only the compression layer.




*Fifure 5: A reference scheme for DIS 11172*

Operations performed by the System Decoder either apply to the entire MPEG-1 multiplexed stream ("multiplex-wide operations"), or to individual elementary streams ("stream-specific operations"). The MPEG-1 system layer is divided into two sub-layers, one for multiplex-wide operations (the pack layer), and one for stream-specific operations (the packet layer).

Multiplex-wide operations include the co-ordination of data retrieval from the transport medium, the adjustment of clocks, and the management of buffers. The tasks are intimately related. If the rate of data delivery from the medium is controllable, then medium delivery may be adjusted so that decoder buffers neither overflow nor underflow; but if the medium rate is not controllable, then elementary stream decoders must slave their timing to the medium to avoid overflow or underflow.

MPEG-1 multiplexed streams are composed of packs whose headers facilitate the above tasks. Pack headers specify intended times at which each byte is to enter the system decoder from the transport medium, and this target arrival schedule serves as a reference for clock correction and buffer management. The schedule need not be followed exactly by decoders, but they must compensate for deviations about it.

An additional multiplex-wide operation is a decoder's ability to establish what resources are required to decode an MPEG-1 multiplexed stream. The first packet of each MPEG-1 multiplexed stream conveys parameters to assist decoders in this task. Included, for example, are the stream's maximum data rate and the highest number of simultaneous video channels.

The principal stream-specific operations are

1) demultiplexing, and

2) synchronising playback of multiple elementary streams.

These topics are discussed next.

## 7.2. Demultiplexing and synchronising playback

On encoding, MPEG-1 multiplexed streams are formed by multiplexing elementary streams. Elementary streams may include private, reserved, and padding streams in addition to MPEG-1 audio and video streams. The streams are temporally subdivided into packets, and the packets are serialised. A packet contains coded bytes from one and only one elementary stream. Both fixed and variable packet lengths are allowed, subject to some constraints.

On decoding, de-multiplexing is required to reconstitute elementary streams from the MPEG-1 multiplexed stream. This is made possible by *stream-id codes* in packet headers.

Synchronisation among multiple streams is effected with presentation time stamps in the MPEG-1 multiplexed stream. The time stamps are in units of 90 kHz. Playback of N streams is synchronised by adjusting the playback of all streams to a master time base rather than by adjusting the playback of one stream to match that of another. The master time base may be one of the N decoders' clocks, the transport medium or channel clock, or it may be some external clock.

Because presentation time-stamps apply to the decoding of individual elementary streams, they reside in the packet layer. End-to-end synchronisation occurs when encoders record time-stamps at capture time, when the time stamps propagate with associated coded data to decoders, and when decoders use those time-stamps to schedule presentations.

Synchronisation is also possible with the transport medium's time-stamps in the multiplexed data stream.

The packet layer is independent of the compression layer in some senses, but not in all. It is independent in the sense that packets need not start at compression layer start codes. For example, a video packet may start at any byte in the video stream. However, time stamps encoded in packet headers apply to presentation times of compression layer constructs.

MPEG-1/System employs a "system target decoder," (STD) to provide a formalism for timing and buffering relationships. Because the STD is parameterised in terms of fields defined in the standard (for example, buffer sizes), each MPEG-1 multiplexed stream leads to its own parameterisation of the STD. It is up to encoders to ensure that the bit streams they produce will play in normal speed, forward play on corresponding STDs. Physical decoders may assume that a stream plays properly on its STD; the physical decoder must compensate for ways in which its design differs from that of the STD.

The systems specification addresses the problem of combining one or more video and audio streams as well as other data into a single or multiple streams which are suitable for storage or transmission. System coding following the syntactical and semantic rules imposed by the standard provides the necessary and sufficient information to enable synchronised playback without overflow or under flow of decoder buffers under a wide range of stream retrieval or receipt conditions and with various forms of system coding.

## 7.3. The program and transport streams

MPEG-2/System coding is specified in two forms: the Program Stream and the Transport Stream. Each is optimised for a different set of applications. Both the program and transport streams defined in the specification provide system level coding which is necessary and sufficient to synchronise the decoding and presentation of the video and audio information, while ensuring that coded data buffers in the decoders do not overflow or underflow. Such information is coded in the syntax using time stamps for the decoding and presentation of coded audio and visual data and time stamps for the delivery of the data stream itself. Both stream definitions are packet-oriented multiplexes.

The program stream is analagous and similar to that of MPEG-1 and combines one or more elementary streams which have a common time base into a single stream. The program stream definition can also be used to encode multiple audio and video elementary streams into multiple program streams which all have a common time base and which, like the single program stream, can be decoded with synchronisation between the various elementary streams. The program stream is designed for use in relatively error-free environments and for applications which involve processing, possibly in software, of the streams. Program stream packets may be of variable and relatively great length.

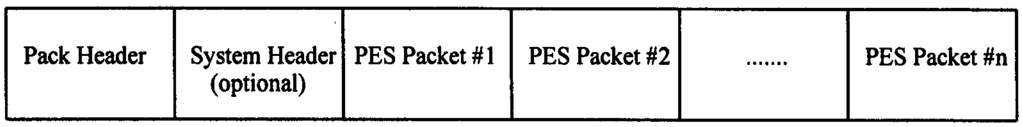
The transport stream combines one or more elementary streams with one or more distinct time bases into a single stream. The transport stream is designed for use in environments where errors are likely, such as storage or transmission in lossy or noisy media. Transport stream packets are of fixed and relatively short length.

As the program and transport streams are designed for different applications, their definitions do not follow strictly a layered model. It is possible and reasonable to convert from one to the other; however, one is not a subset or superset of the other. In particular, extracting the contents of a program from a transport stream and creating a program stream is straightforward, but not all of the fields of the program stream are contained literally within the transport stream; some must be derived. The transport stream may be used to span a range of layers in a layered model, and is designed for efficiency and ease of implementation in wide bandwidth applications.

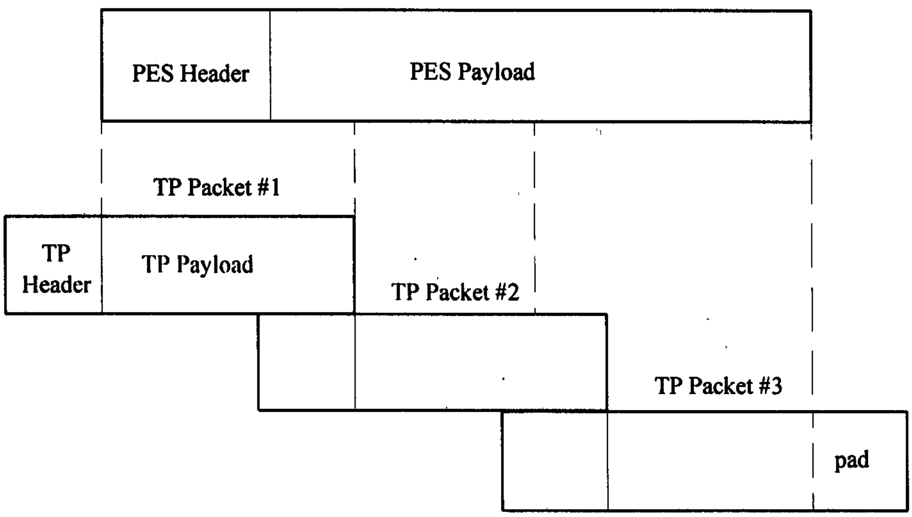
The scope of syntactical and semantic rules set forth in the systems specification differ: the syntactical rules apply to systems layer coding only, and do not extend to the compression layer coding of the video and audio specifications; by contrast, the semantic rules apply to the combined stream in its entirety.

The systems specification does not specify the architecture or implementation of encoders or decoders, nor those of multiplexers or demultiplexers. However, bit-stream properties do impose functional and performance requirements on encoders, decoders, multiplexers and demultiplexers. For instance, encoders must meet minimum clock tolerance requirements. Notwithstanding this and other requirements, a considerable degree of freedom exists in the design and implementation of encoders, decoders, multiplexers and demultiplexers.

The figures below depict the general structure of the Program and Transport Streams respectively; both are based on the definition of the PES (Packetised Elementary Stream) Packets, described in figure 6.




Program Stream structure




Transport Sream structure

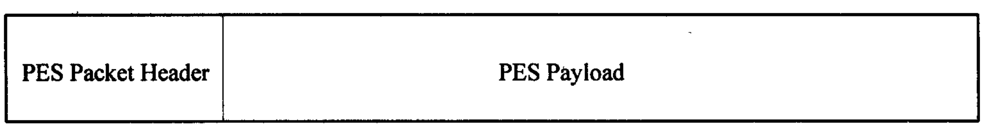



*Figure 6: PES packet structure*

The Program Stream structure derives directly from the MPEG-1 System standard: only a few fields have been added in the System Header and in the PES Packet Header (simply: Packet Header in the MPEG-1 terminology), to provide facilities to scramble and to handle separately the different program components. The System Header carries information concerning the whole program, while the PES Packet Headers describe only their PES.

The Transport Stream uses fixed-length Transport Packets to segment the PES Packets. A Packet ID is assigned to each Transport Packet: every Transport Packet tagged with a certain Packet ID may carry information related to only one PES. Global information concerning a whole program (something similar to the System Header in the Program Stream) may be conveyed in Transport Packets as well. Mechanisms are also provided to multiplex the Program Directory in the MPEG-2 Transport Stream and to describe the programs themselves, with hooks for possible expansions (e.g: MHEG, for the presentation of complex programs).

## 8. FUTURE POSSIBILITIES BEYOND MPEG-2

There is tendency in some environments to believe that MPEG, by providing a generic audio-visual coding standard, has solved all problems related to coding and that nothing is needed beyond the MPEG-2 standard.

Indeed the possibility for industry to finalise the developments that will lead in some months or few years to products and services capable of reaching users without artificial barriers is a landmark event. But we should not forget what MPEG-2 enables us to do: to transform one or more television signals and one or more audio signals into a sequence of bits.

Below are some areas that can be considered as next steps:

### 8.1. Binocular vision.

A sense of depth can be easily created if a scene taken by two cameras is displayed using an appropriate display system. This does not need to be based on 3D display techniques, since it can be easily realised by special monitors. The shortcomings is that the sense of depth is perceived only by those sitting in front of the monitor within a rather narrow angle, something that can restrict the use of the system for personal television.

The interesting part of this evolution lies in the possibility to provide a compatible upgrade of the current MPEG-2 standard (of course in addition to the straightforward one of sending the two television signals). Techniques of disparity and motion compensation can be used to predict the second view from the first one. This is being explored by the RACE project DISTIMA.

### 8.2. 2-D display of 3-D scenes

This responds to the natural evolution of viewers wishing to see programmes tailored to their own needs. A number of conventional cameras (say three) shoot a scene and extract the full description of the 3-D space. Information is coded in such a way that a user, upon receiving the bit-stream, can select his/her own viewpoint, which is not restricted to any one of the cameras. The scene is then displayed on a conventional monitor.

This is an entirely new field of research and is likely to lead to a totally new standard.

### 8.3. 3-D display of 3-D scenes

This is an area of research for the long term. As long as viable 3-D displays are not produced, 3-D display of 3-D scenes will remain a dream.

### 8.4. Analysis-synthesis coding

The time is ripe to attempt the merging of coding techniques with synthesis techniques. This applies to both audio and video. The next MPEG Work Item, also known as MPEG-4, will attempt to produce a standard in 1998 with some of these features.

## 9. CONSUMER RECORDING - TECHNOLOGICAL ISSUES

The field of television entertainment is strongly connected to recording. The role of this important piece of equipment must certainly considered during the move from analogue to digital systems.

### 9.1. Digital video tape recorder

Digital tape recorders (DVTR) are one of the unattained targets of consumer electronics. After nearly a decade of R&D with technical solutions found (intraframe DCT at 12-15 Mbit/s) and with several generations of equipment developed, including the necessary microelectronics, there are no immediate prospects of a safe introduction of this new product to the market.

This gloomy conclusion has been reached from the assumption that it involves introducing a new device that would provide beautiful pictures, although it is not clear what the consumer really needs. Indeed the audio field and the video fields are psychologically very different and comparing DVTR/VHS substitution with the CD/vinyl record case history is hard to defend.

The introduction of digital television distribution in the mass market has the potential to change many parameters; so the above decision may have to be reconsidered. In this case the DVTR should be considered also (or more?) as a "data logger" where consumers can store the bit-streams they receive via the various delivery media. The problem arises when consumers wish to play back the stored material, since the received bit streams, being highly compressed, do not allow a straightforward implementation of Fast Forward and fast reverse (FF/FR) modes, considered essential by experts.

This "data logger" approach has materialised in MPEG as a new attitude on the part of consumer electronics manufacturers towards the problem of how to deal with a DVTR standard. The current approach is to define some support within the picture coding that is not burdensome to the decoder used for playing video under normal operation, or the decoder that is not intended to provide FF/FR functions.

### 9.2. Disk recording

DVTR is not the only means for storing of digital television. Many consider it a more effective tool to transport information because of better bandwidth utilisation, but digital television enables linking the content (software) with the possibility of processing it (hardware) and this is best exploited when information is stored on disks.

The most straightforward possibility is offered by digital read-only optical technologies. The current limitation is set by the bit-rate which is limited to 1.4 Mbit/s. Higher bit-rates (say four times) with the same capacity (duration) are already possible in the laboratories but the critical factor is the development of a cheap blue laser, which is not just around the corner.

Chapter 2

# Subscriber management systems, conditional access and encryption

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## 1. SUMMARY

This paper describes the past development of Pay Television services and looks forward to the issues that will affect the development of future Pay-TV systems for new digital television broadcast services.

Section 2 provides a general introduction to the subject of Pay-TV systems. We examine the use of Pay-TV systems to fund the newer types of television service, such as cable and direct-to-home (DTH) satellite networks. We note the current development of digital compression techniques for television signals and the impact they could have on the number and type of new services available to the television viewer. We surmise that, as with analogue cable and satellite networks, Pay-TV technology will have a role to play in the funding and development of these new services. We also note that Pay-TV systems can provide free-access channels, and that there is nothing to stop a standardised Pay-TV system being used to provide a complete range of both free and pay channels.

We identify the various elements that make up the Pay-TV system and provide a description of the functionality of each of them. We note that it is the "conditional access" element which provides the real functionality and control within the Pay-TV system. This includes those features related directly to the revenue enforcement function of the system (e.g. subscription and pay-per-view operations), and those providing ancillary features, such as parental control and on-screen help messages.

We also describe the interfaces between the viewer and the Pay-TV system. The first is in the viewer's decoder, and consists of the "on-screen" messages and other functions that allow easy operation of the system. The second interface is provided by the customer management centre, where applications for access to services are received from the viewer, processed and the appropriate authorisation messages issued to the conditional access system. We highlight the importance of the information stored in the customer management database for marketing operations and for monitoring business performance. We note how the technology has developed to provide greater levels of support for the viewer, as well as allowing new forms of Pay-TV operation.

Section 3 covers the evolution of conditional access systems. We believe that it is important to understand the way in which the industry has developed when considering the requirements for new digital systems. This is particularly true for the security of the system and its resistance to piracy. We describe the scrambling techniques employed in the present generation of conditional access systems: those utilising "analogue" and those utilising "digital" scrambling technologies. In general we do not believe that the analogue scrambling of analogue signals can provide adequate security or a high level of concealment.

We review scrambling and encryption technologies for new digital television formats, and the use of renewable security systems. We conclude that systems should be upgradeable and based on the use of secure processors, possibly in the form of smart cards.

In section 4 we examine the use of Pay-TV systems with various transmission media, including satellite, cable and terrestrial distribution systems. We look at the contrasting requirements imposed by the differing characteristics of the transmission media, and some of the distortion and interference issues that affect the choice of scrambling technique. In particular, we consider the implications for the transmission of digital television services over terrestrial channels.

Sections 5 and 6 both cover the question of "open" versus "closed" conditional access systems. In the open system any broadcaster would be able to provide Pay-TV services to the installed base of decoders. Similarly, any manufacturer would be able to produce decoders for the market. In the closed system, crucial parts of the technology are proprietary to a particular manufacturer, so that broadcasters have to negotiate for the supply of new decoders, or to gain access to existing ones.

The technological issues associated with open and closed networks are covered in section 5. An open network implies that broadcasters and manufacturers alike have access to the cryptographic keys and algorithms necessary to generate and receive the scrambled television service. However, public knowledge of this information runs counter to the requirements for total system security. We explore possible solutions for this fundamental dilemma, none of which are perfect, but which offer the possibility of various degrees of openness in the system.

The existence of a workable technical solution is not the only requirement for an open system. In section 6 we examine the commercial realities of the market place. We note how European Pay-TV service providers are closely associated with the conditional access system suppliers. Although there is no single standard for a conditional access system throughout Europe, there are effective monopolies in local markets. If for legal or technical reasons a new broadcaster cannot safely make use of the existing decoder population, the barrier to entry into that market is very high. Not only has the new broadcaster got to persuade subscribers to purchase new equipment but he has to work against the marketing and sales initiatives of existing broadcasters.

Both the existing broadcaster making use of another's system and the new broadcaster may suffer from this constraint. Past experience suggests that any new broadcaster should seek to ensure that either it controls the system or is able to migrate to another system, or possibly that it can run more than one system in parallel.

We have included a section on piracy (section 7), both of the decoders and of the overall system. We have examined the way in which piracy has evolved to match the increasing complexity of Pay-TV techniques. Past attitudes to this phenomena have led to significant commercial returns for the pirates, and some of these profits have been ploughed back into further attacks on Pay-TV systems. We believe that any operator should attempt to minimise piracy and fraud and question the acceptance of a certain level of piracy as a necessary evil. Lessons should be learned from previous experience and incorporated into the design and operating procedures for new systems that will support digital television applications.

The introduction of digital broadcast television standards is a major development for the broadcast industry. In section 8 we consider the implications of new digital technology for Pay-TV systems, and for the conditional access element in particular. We examine some of the issues associated with the security of these systems, as well as the possibilities for the operation of multiple conditional access standards within the broadcast environment. When referring to digital broadcast technologies we have used MPEG as an example of such a system, since at the time of writing so much of the effort in Europe is directed at the proposed MPEG standard.

With digitally-based systems the need for scrambling of the video signal no longer exists since the video data can be directly encrypted. If the compression is carried out at the studio the MPEG video may be encrypted prior to distribution, thus providing end to end security. If the data is encrypted after being packetised, the encryption of each MPEG packet can be done in a manner which is independent of the type of data being carried. The synchronisation of the conditional access system can then be carried out at the packet level. We believe that encryption and decryption of the MPEG packets would result in a less complex decoder design than the first option, since the encryption process will be the same for all types of data.

If the decryption of data in an MPEG decoder is to be controlled by passing control words (or keys) across an interface from any conditional access subsystem, then the algorithms for data encryption must be defined within the MPEG standard although, in our opinion, the definition of this interface is not mandatory for a digital broadcast system. The minimum specification for a standard decoder would have to state at which levels the data could be encrypted and how synchronisation should be achieved. If this approach is taken it will be much more difficult, and probably impossible, for a broadcaster to arrange for more than one conditional access system to be used to protect a single component.

Whilst MPEG must define the data structure for the compressed video and audio, it is not clear that it should or need define the data structure for conditional access messages or other data services. The broadcast standard should allow the conditional access system to allocate messages to one or more of the service components. In addition, there may be a requirement for more than one conditional access system to be present in the multiplex; in our opinion the system design should support this.

Our final section looks at the requirements for future Pay-TV systems. The increased functionality available with digital encoding techniques is expected to lead to a corresponding increase in the methods of Pay-TV operation. One example could be the move away from monthly subscriptions for specific channels, to more discriminating arrangements, perhaps based upon thematic arrangements (e.g. a subscription to wildlife programmes, or to sporting events). A further example of future changes could be to supplement pay-per-view services with "near-video-on-demand" operations.

To assist the reader in gaining familiarity with Pay-TV systems we have included a glossary at the end of the report which covers the acronyms and other terms used within the narrative.

## 2. INTRODUCTION TO PAY TELEVISION

In this section we provide an outline review of the origination of Pay-TV systems. We consider a typical Pay-TV network and discuss the technology and the different systems used in its operation. We also note the importance of the interfaces to the viewer, since it is these which determine the ease of use of the system by the final customer.

### 2.1. Overview

Traditionally, television broadcasting has been funded by advertising revenues, licence fees or by direct governmental grants. Pay television (Pay-TV) is a generic term covering alternatives to these, including subscription and pay-per-view systems. Originally the introduction of Pay-TV was associated with the advent of different distribution methods for television services, such as cable and satellite, and reflected the need to seek alternative methods of funding these new types of operation. However, shifting attitudes to state-imposed funding arrangements for television services, and a desire to promote competition and choice, has led to the examination of Pay-TV systems as a method of supporting terrestrial distribution systems as well.

The recent development of digital compression techniques for use with television entertainment systems has dramatically affected perceptions of what is possible with broadcast systems. Digital compression increases the carrying capacity of terrestrial, cable and satellite networks by up to tenfold. This opens up the potential for new types of service, as well as widening the scope for existing ones. However, any increase in the number of television services will have to be funded, which again focuses attention on Pay-TV technology. In this context it should be noted that Pay-TV systems need not necessarily be restricted to pay television services. There is nothing to stop a Pay-TV system supporting a free access television service; the advantages in doing so are the potential for standardisation across the distribution system, and the features such as parental control which are equally valid for free access services.

We review the Pay-TV functions in the remainder of this section, and add more detail on the critical issues later in the report. The successful implementation of a Pay-TV system has to ensure that these individual processes are brought together effectively and securely, whilst remaining as transparent to the viewer as possible.

A Pay-TV system consists of a number of individual processes, some purely technological, others with a strong element of human participation. Fig. 1 illustrates the basic components of a typical system and we shall refer to this when describing the functionality of the Pay-TV system. The illustration assumes the use of digital compression technology and of a geo-stationary satellite for transmission of the service, but the concepts apply equally to analogue formats and other distribution methods.

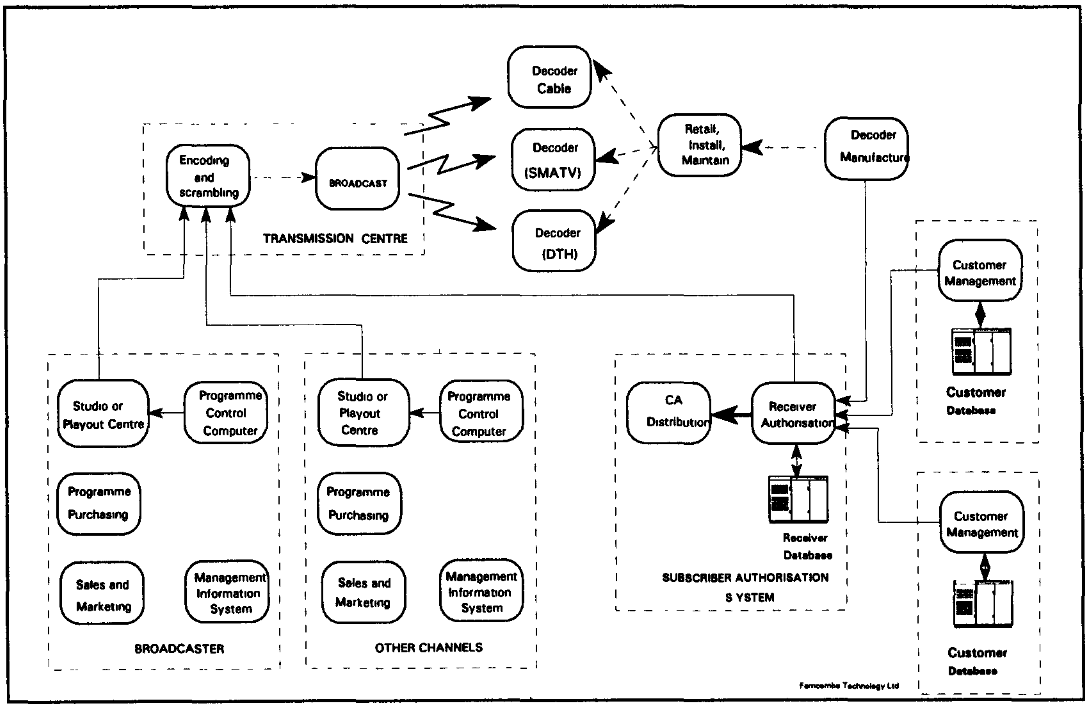


*Figure 1: Pay-TV system overview*

## 2.2. Programme origination and distribution

The programmes that make up a television service originate at a studio (for live items) and/or a playout centre (for pre-recorded items). These facilities may be located at more than one site, and be operated by different organisations. For Pay-TV systems an additional facility is associated with this programme origination process. This is the programme control computer (PCC), and its function is to generate a control channel containing a series of attributes related to each programme item. These attributes characterise the programme in terms of its subscription or pay-per-view value and other Pay-TV functions.

The television services, including the control channel carrying their Pay-TV attributes, are then linked to a transmission centre for distribution to the intended audience. The transmission centre may be co-located with a playout centre or located some distance away.

Any practical form of distribution system has to employ the techniques of encoding and modulation. Examples of encoding systems are the PAL and SECAM formats used in current analogue systems, and the compression and coding algorithms used for the new digital systems. These all exploit frailties in the human optical and aural senses to ensure the efficient usage of the bandwidth occupied by a television channel. Digital compression systems achieve higher efficiencies through the use of signal processing techniques.

Modulation methods include Vestigial Sideband Amplitude Modulation (VSB-AM), used with current terrestrial and cable systems, and Frequency Modulation (FM) as used in satellite distribution systems. Modulation schemes for digital transmission include Quadrature Phase Shift Keying (QPSK) and various forms of Quadrature Amplitude Modulation (QAM). The modulation scheme for a particular application is selected by reference to the parameters of the transmission channel on which it will be used. The trade-offs are typically the bandwidth needed for the modulation scheme versus its susceptibility to potential interference mechanisms.

The selection of a particular Pay-TV system has to take account of the characteristics of the transmission channel, and of the encoding and modulation schemes to be used. We shall see later how characteristics such as noise, multi-path propagation, frequency and phase response have a direct impact upon the implementation of Pay-TV systems.

## 2.3. Scrambling

The objective of a Pay-TV system is to restrict access to services to those viewers fulfilling certain conditions. These are usually financial ones, such as the payment of a monthly subscription fee, but they can also be based upon the geographical location of the viewer, or even his or her age. The first step in implementing such a system must be to process the broadcast signal so as to make it unviewable using standard receiving equipment. This process is called scrambling and it can be applied to either the video or audio signal, or both. The scrambling operation can be carried out at either the studio/playout centre or at the transmission centre, though the former has the advantage of protecting against unauthorised access to the service at the communication path between the two facilities.

It should be noted that scrambling operations are carried out upon the actual picture or sound information. In fact, the scrambling process is often carried out within the signal encoder. The scrambling process has the potential for degrading the quality of the signal, and the scrambling method used has to minimise any such effects.

The main criteria associated with scrambling techniques are their security in the face of attacks upon the system, and the degree of concealment of the picture or sound channel.

## 2.4. Reception of pay-TV services

Viewers to a Pay-TV service will be in possession of a "decoder", which is able to descramble the Pay-TV channel for display on a conventional television set or similar display device. The decoder may be physically separate from a satellite receiver, or fully integrated within it. As systems develop we expect further integration of the decoder into the reception chain, either as an integral part of a single television receiver, or in the development of units similar to the "tower" arrangements found in current audio systems.

The Pay-TV decoder must be provided with the necessary information to descramble the channel. This information originates from a second control system called the subscriber authorisation system (SAS). The SAS contains details of all the legitimate decoders known to exist, which means that there has to be a link between this and the manufacturer(s) of the decoders. The SAS also contains details of the various Pay-TV services to which each decoder is authorised to have access. The SAS issues "entitlements" to each individual decoder which contain the identity of the services that the decoder is permitted to descramble, and the "keys" needed to carry out the descrambling operation. These messages can be sent with the television services, via a second control channel (as illustrated in Figure 1) or distributed by other means, such as the post, using some form of electronically encoded token.

These entitlements have to be protected against fraudulent use or imitation. This is carried out by encrypting the messages with further keys, so that only the intended recipient can make use of them.

## 2.5. Conditional access

The collective term for the scrambling and descrambling process, the encryption and decryption of the entitlement messages, and the hierarchy of keys which link them is "conditional access", and it is this which provides the core functionality of the Pay-TV system. The design of the conditional access (CA) system also includes that of the PCC and SAS functions, since the entire system has to operate in a coherent and secure manner. The prime function of the CA system is to deliver the information needed to allow a legitimate decoder to acquire and access a Pay-TV service.

The part of the decoder that implements the decryption of messages is called the conditional access sub-system, or CASS. The CASS can be totally integrated (or "buried") within the decoder, or designed as a "detachable" part of it. The choice of a buried or detachable implementation forms one of the major characterisations of a Pay-TV system.

An important function often incorporated into the CASS is the provision of information to allow the ordinary viewer to utilise the system. This is usually carried out via the generation of help menus and other information intended for display on the viewer's television screen. This part of the system provides one of the interfaces between the viewer and the Pay-TV system, and its effectiveness determines the ease of use (and thus the acceptability) of the entire operation.

## 2.6. Customer management

The other interface to the viewer, and the final element of the Pay-TV system, is the customer management system (CMS). Here applications to receive Pay-TV services are received from the viewer and processed. Messages containing the identity of the viewer's decoder and its entitlement to various Pay-TV services are then sent to the SAS.

A Pay-TV system can consist of a number of independent channels, each operated by a different organisation and each with its own customer management centre. The entitlements generated by each of these centres are collected by a single SAS and combined to form a single message for each decoder. This simplifies the distribution of authorising messages to a large decoder population.

It should be noted that whilst the SAS is concerned with decoder identities, the CMS is very much orientated towards people. A typical CMS operation will be structured around a comprehensive database of customers and potential customers within the service area of the Pay-TV network. This often contains information on socio-economic groupings and previous purchasing habits, to assist in the targeting of mailshots supporting marketing campaigns.

The CMS will need to deal with a variety of different sales arrangements covering individual viewers, cable and SMATV customers, different levels of service and special offers. Different payment methods are also likely. The CMS will originate billing for customers, and will need to cope with credit transactions, cheque clearances and direct debit arrangements.

Customer management systems have undergone considerable development over the past decade. Modern systems can be integrated into repair operations, running an initial "help" desk to try and resolve problems directly with the viewer, and originating a service visit if these are unsuccessful. A further feature has been to support pay-per-view operations. Here customers wishing to purchase a specific programme item can do so via the CMS. Automated telephone response units have been developed to handle the large peaks of activity occurring prior to the transmission time of the item.

The CMS also has a role to play in the collection of statistics on the performance of the system. As well as information on the growth of the customer base and revenues, the CMS will show activity patterns related to seasonal variations and marketing promotions. Debt and poor payment records can be noted. Finally, the reliability of decoders and other parts of the Pay-TV infrastructure can be monitored, providing invaluable data on how the system is functioning.

This part of the Pay-TV system is distinctly different from those described above, due to the emphasis on interaction with the viewer. The effectiveness of the conditional access and scrambling processes can be precisely quantified by reference to error rates, data speeds and picture impairment. By contrast, the effectiveness of the customer management system has more to do with the friendliness and efficiency of the contact between the service representative and the customer. Many organisations invest significant amounts of training resource into establishing and maintaining a close relationship with the end-user.

## 3. EVOLUTION OF CONDITIONAL ACCESS SYSTEMS

In this section we will briefly describe the evolution of conditional access systems. We will describe the early audio and video scrambling techniques and the development of these techniques to the present state of the art for analogue broadcast systems. In the next section we will discuss the issues associated with digital broadcasting systems.

### 3.1. Criteria for conditional access

Criteria that must be met by a modern broadcast conditional access system include:

- Compatibility with broadcast encoding and modulation standards.

- Rugged data coding, immune to noise and other interference.
- The signal must be capable of being networked via all known media (terrestrial transmission, cable, microwave, optical fibre, satellite).
- The ability to continually update a large population of decoders (i.e. tens of millions).
- Security must be high enough to prevent piracy of signals for longer than the expected system lifetime, at least 10 years.

The need to maintain security over a long period often results in system designers choosing detachable security devices such as smart cards, although not all system designers have come to this conclusion.

In the past, the selection of a conditional access system has not only been concerned with the technology, but also with the commercial relationships between the conditional access system designers, manufacturers and operators. In some cases, such as VideoCrypt and Eurodec, the owners or part owners of the conditional access system are themselves major broadcasters (BSkyB / News International and Canal Plus for VideoCrypt and Eurodec respectively). In all cases the broadcaster is to some extent in the hands of the conditional access system supplier, for the supply of equipment, maintenance of security and sometimes for the operation of the system. This cross dependence of equipment supplier and broadcaster is likely to continue with the development of Pay-TV systems for digital television applications.

### 3.2. Functionality of a conditional access system

As noted in section 2.3, the prime function of the conditional access system is to deliver the information needed to allow a legitimate decoder to acquire a Pay-TV service and also to determine whether it should allow the viewer access to the service. It is the flexibility and features included in these checks that distinguishes the different types of conditional access system and their applications.

In the first generation analogue broadcasting systems there was no provision for scrambling, thus the conditional access system suppliers had to devise their own scrambling techniques. The MAC standard, on the other hand, included the video scrambling and audio encryption (also referred to as scrambling) within the broadcast specification (defined in EBU 3258). Therefore in PAL systems we see a wide range of different scrambling techniques employed, although not all are effective, whilst in MAC systems the same scrambling technique is always used. We can however maintain the distinction between scrambling of the video and audio components and the control over this scrambling in analogue broadcasting systems.

In order to allow access to the service, the conditional access subsystem in the decoder must be able to derive the information required by the scrambling circuits and provide the necessary synchronisation between the scrambling at the encoder and the descrambling at the decoder.

### 3.3. Implementation of a typical conditional access system

In order to provide control of decoders in large broadcast systems, designers make use of cryptographic processes based on a combination of valid keys and algorithms, together with the logical testing of "entitlements". Much of the system design effort goes into the management of the key hierarchy and the secure and efficient addressing of these decoders.

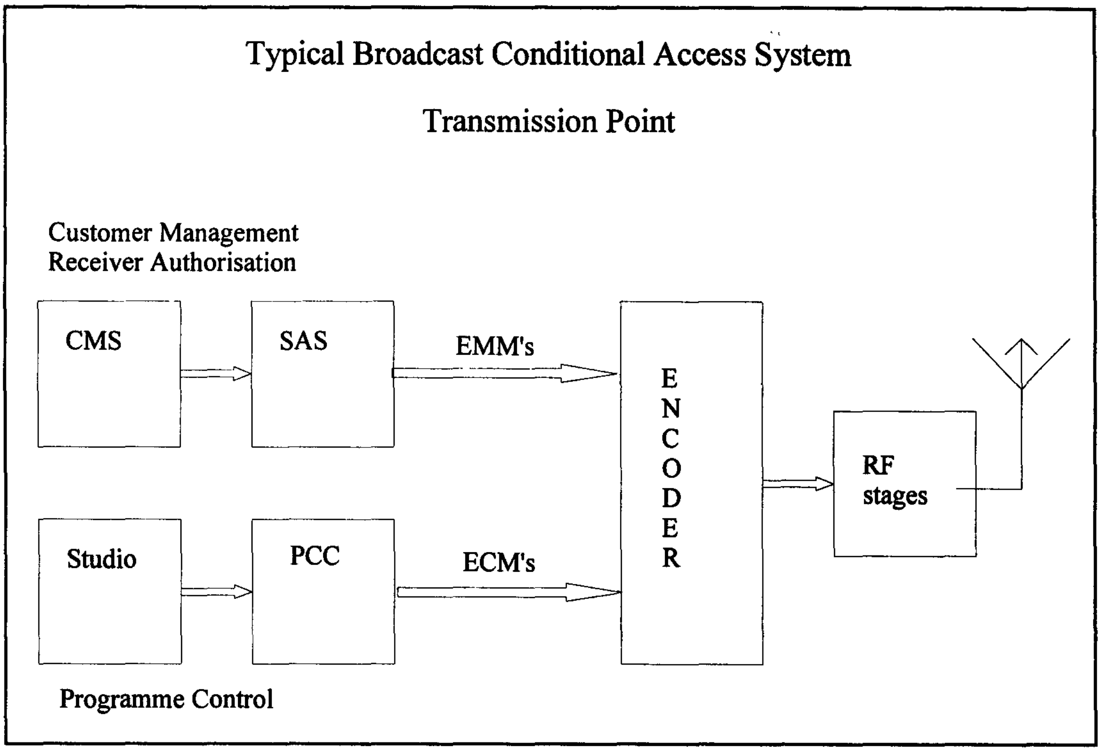


*Figure 2: Broadcast conditional access system*

Figure 2 illustrates a simplified version of the main conditional access system components. We distinguish the two independent control channels in the system, even though these may not be specifically identified in a particular design. The first channel provides the information needed to access the current programme and the synchronising information. This data is generated by the programme control computer, which is usually controlled from the studio or the broadcaster's central control centre. In the conditional access systems proposed for the MAC broadcasting standard this data was said to be carried in entitlement checking messages (ECMs), this terminology has now found a more general application to broadcast conditional access systems and we will use it throughout this document.

The data carried in ECMs includes both the access rights that must be held by decoders before the video may be descrambled, and user features such as programme names. Critical data carried on the ECM channel is usually encrypted and authenticated so that only those decoders with the necessary keys and/or entitlements required to gain access to the programme will function.

The second channel, which typically runs asynchronously from the first, is used to enable and disable decoders. This may be done by delivering entitlements and/or by updating keys to groups of decoders or to individual decoders. These entitlements are stored within each decoder's conditional access subsystem (CASS). "Over air addressing" is the term used to describe systems which are capable of controlling the decoders using a transmitted data channel, which must also be encrypted and authenticated in order to protect the data. In the case of MAC systems these over-air messages are referred to as EMMs, or entitlement management messages. In an over air addressing system, the subscriber authorisation system (SAS) generates the messages required to maintain or delete services from individual, or groups of, decoders. This function is separated from the billing and invoicing of customers which is carried out by the customer management systems (CMS) described in section 2.6. In a small system both the subscriber authorisation system and the customer management system functions shown in may be performed by a single computer or even the encoder itself. Some of the broadcast systems operate a mixture of pre-authorisation and over-air addressing in an attempt to reduce the required control channel data capacity.

Access to a programme is determined by comparing the characteristics of the programme as described by the data in the first channel, the ECM data stream, with the decoder's stored entitlements and of course depends on the decoder holding the correct algorithms and, if necessary, keys. The use of a sophisticated programme control system and subscriber authorisation system allows broadcasters to sell pre-paid subscriptions for a complete service, group of services or individual programmes (referred to as pay-per- view) as well as providing messaging services and information about current and future programmes. Other functions can be included, for instance some systems have been extended to allow for the automatic control of video recorders.

## 3.4. The evolution of video and audio scrambling

### 3.4.1. Analogue video scrambling

Until recently, scrambling of broadcast television signals has been concerned solely with analogue waveforms, e.g. the video and audio components of PAL, SECAM and NTSC encoding systems. Typically, such scrambling has involved the addition of interfering carriers, the modification of the synchronisation or colour burst information, or the delay or inversion of the video signal. There are a number of systems in use today which combine several of these options in an effort to increase security against unauthorised access to the signal.

All of the above scrambling techniques are applied to the signal waveform whilst it is in its analogue form; thus we refer to them here as analogue scrambling methods. They require minimal processing and the decoders are fairly simple in design and cheap to provide. Analogue scrambling systems are thus popular for cable delivery systems, where access to the scrambled signal can be limited by the transmission medium itself. However, we believe that, by themselves, analogue systems do not offer a high degree of security. In some systems the combination of analogue video scrambling and more secure digital encryption of the audio has been used to increase the overall security of the system.

### 3.4.2. Digital scrambling of analogue signals

In a system using a digital scrambling technology the video is digitised at the encoder and decoder and then processed in the digital domain. The number of lines that need to be digitised depends on the scrambling technology. The two most common digital scrambling techniques are "cut and rotate" and "line shuffling", which are described in greater detail later in this section. The most frequently used technology in satellite broadcasting to date has been cut and rotate. It is cheaper to implement than line shuffling since only one line of video needs to be stored in the descrambler; in a line shuffling system the number of lines that need to be stored equals the number of lines over which the shuffling is performed. In both systems the scrambling process is usually seeded from the conditional access subsystem and re-synchronised for each new field, or block of lines, to be scrambled. The subsystem may be connected directly to the scrambling circuits, or indirectly via a control processor. Since the cryptographic computations usually take longer than one field period to generate new seeds for the scrambling process the seeds are sometimes modified by a system variable that is known to both the encoder and decoder, such as a frame counter.

### 3.4.3. Cut and rotate scrambling

In cut and rotate scrambling, sometimes referred to as Active Video Line Rotation, the video is divided into at least two segments, A) and B), as can be seen in Figure 3.. The position of the cut point is different on each line and is usually determined from a pseudo random binary sequence (PRBS) seeded with a key (sometimes called a control word) that has been derived by the conditional access subsystem. The control word may be modified with a frame count before seeding the PRBS, the output of which selects one of many, typically a few hundred, cut points on each video line. In the case of MAC video scrambling the line is cut at two points and both the luminance and chrominance signals are rotated.

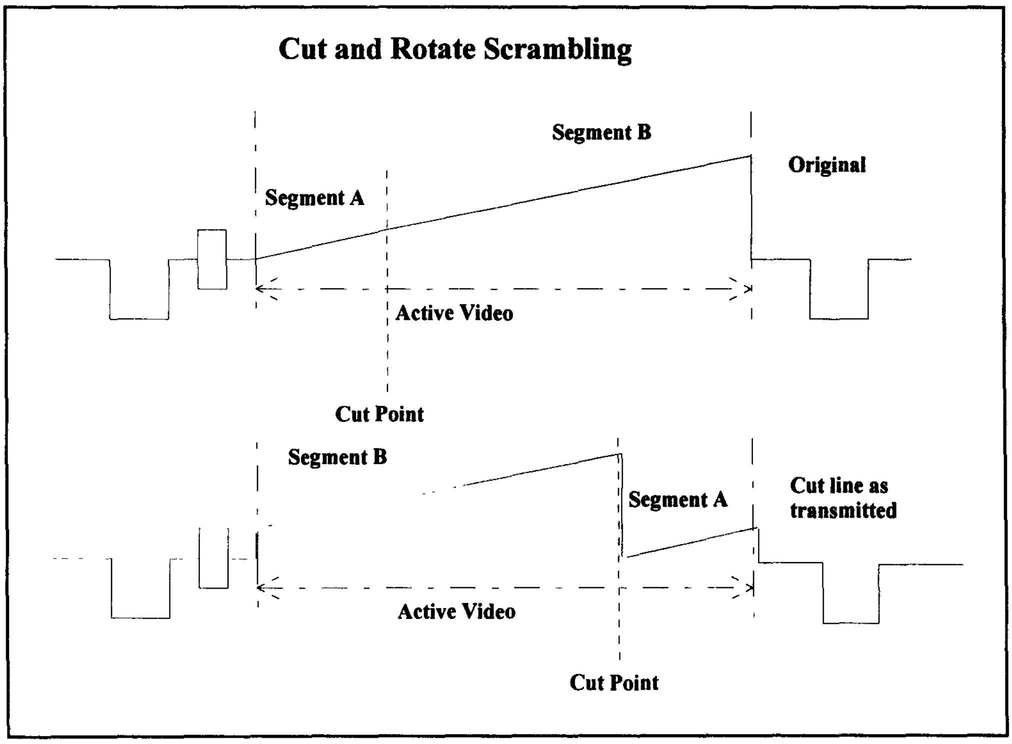


*Figure 3: Cut and rotate scrambling*

### 3.4.4. Line shuffling

In line shuffling systems, a number of consecutive lines are stored in memory and then transmitted in a different order. At the decoder the shuffled lines are again stored in memory, but the conditional access system provides the information to allow them to be rearranged into their correct order.

There are two principal differences in current commercial line shuffling systems. The lines may be shuffled in blocks of lines, or lines may be distributed across the complete field, although the probability of shuffling over a large number of lines may be very low, so that there still appears to be banding in the scrambled video. Usually the colour burst is not shuffled since otherwise the phase of the colour burst could be used to restore the lines to the correct order.

### 3.4.5. Analogue audio scrambling

The most common forms of audio "scrambling" involve frequency shifting or frequency inversion of the audio signal. These techniques can conceal the audio reasonably well, but they can also be defeated without too much difficulty in a pirate decoder. Analogue scrambling of audio signals is usually only found in first-generation systems.

### 3.4.6. Digital audio scrambling/encryption

Digital audio may be found in a subcarrier, such as a NICAM subcarrier, or in some systems the sync pulses have been replaced with a data burst carrying audio data. In the case of case of MAC systems the audio is carried in digitised form in data packets. This digital audio data may be encrypted with a very high degree of security. The encryption of video and audio data is discussed in the following section.

## 3.5. Secure processing of messages

In early conditional access systems the system designers often seemed to rely upon confusion rather than cryptography to protect messages. There was a belief that the protocols used would be sufficiently complex that the hackers would not be able to make pirate decoders. Usually the pirates ignored the messages altogether and simply attacked the scrambling of the video and the audio.

Data Encryption Standard (IBM, US Federal Bureau of Standards)

In the present generation of conditional access systems complex cryptographic algorithms are used; many suppliers suggest that they have used DES[1] or compare their algorithms with DES in publicity literature. Certainly there is no reason not to use a strong algorithm, although national governments may wish to exercise some control over the distribution of systems with strong algorithms that could be used for general purpose encryption of data.

Although good algorithms may be used it is not always true that they are used well. Either there is a weakness in the system design, which allows the processing of messages to be bypassed, or the algorithms may be used to perform an inappropriate function. Whilst it is necessary to encrypt secret data, very little of the data required to control a decoder is secret. It is often more important that the conditional access subsystem can recognise whether the data it is receiving is valid and has not been introduced by a pirate. It is not necessarily true that a message is valid, even if it is addressed to that decoder and has originated from the broadcaster's control system. The replaying of old messages can, for example, provide a very effect means of attacking a system.

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1 Data Encryption Standard (IBM, US Federal Bureau of Standards)

## 3.6. Algorithms

The design aim for a good algorithm is that the plain text data shall be transformed in a way which is sufficiently complex that it is impractical to find a relationship between the resulting cipher text, the plain text and, where applicable, the encipherment key. If we consider a block of data, the plain text, then a single bit change to the plain text should result in each bit of the cipher text changing with a probability of one half.

Most systems today make use of symmetrical algorithms in which the same key is required for encryption and decryption of the data, or in the generation of cryptographic checksums. We discuss the use of asymmetric algorithms in section 5.2.

DES is often proposed as a suitable algorithm for use in conditional access systems. However, it is not difficult to construct a proprietary algorithm for use in a conditional access system and there are good reasons for using different algorithms, or at least variants of the algorithm, at different levels of the key hierarchy. For example, for the encryption of data a fast algorithm, and probably one that can easily be implemented in hardware, is required. For this reason a number of systems make use of a pseudo-random binary sequence (PRBS) for this stage of the process. The encryption of management keys is not so time-critical and may be implemented in software within the conditional access subsystem, which may be a smart card with limited processing power and memory. We do not have any particular preference for or against DES[2], although the system designer should be aware of possible export restrictions.

Cryptographic algorithms can be used in many ways within a system, and simple modifications significantly change their functionality. If we consider using DES, for example, we have to decide whether to use it in its most straightforward mode, the Electronic Code Book Mode (ECM), for ciphering a single block[3] or to use Cipher Block Chaining (CBC), making repeated use of the algorithm to cipher several blocks of data, or in one of its feedback modes, Cipher Feedback (CFB) or Output Feedback (OFB). These modes may be applied to other ciphers and have widespread use in the design of conditional access systems. They are described in detail in a number of publications[4], the precise methods are outside the scope of this document, but we will describe some of their properties as these are relevant to the design of a conditional access system for a digital broadcasting standard.

### 3.6.1. Cipher block chaining

Cipher block chaining enables a series of blocks of data, which may be data packets, to be encrypted. As there is no data prior to the first block, the data stream must start with an "initialising vector". This must be the same in both the encryption and decryption process and is usually kept secret. It is much more important that this initialising vector cannot be modified in the conditional access subsystem since single bit changes to the vector would result in corresponding changes to the first block of

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2 In our opinion, protection against key searching attacks should be considered.

3 This is very unlikely to be suitable for use in a conditional access system.

4 The use of DES is described in detail in "Security for Computer Networks", D. W. Davies and W. L Price.

cipher text, whereas modifying the plain text will have a random effect on the cipher text. This is a good example of needing to ensure the integrity of data rather than its secrecy.

It is necessary to work with an integral numberof 64-bit blocks of data, but padding can be used to complete the last block. For example, in MPEG it has been suggested that packet lengths should be a multiple of 8 bytes. It is not strictly necessary to pad with random data. The main problem with padding is that the presence of padding bytes must be signalled to the decoder.

It is important to consider the effect of errors. A single bit error in the plain text will affect every subsequent block of cipher text, but this is not as dire as it seems since in decryption the plain text will be recovered correctly, except for the single bit error. If an error is introduced in transmission, the block in which the error occurs will be deciphered incorrectly and the feed-forward of the ciphertext with the error will result in a single bit error in the next block. Subsequent blocks will be deciphered correctly. The process does not recover if synchronisation is lost and can, therefore, only be used reliably when data framing is provided.

### 3.6.2. Cipher feedback mode

This mode is suited to a communications environment where the data must not be held up while a complete block of data is received. For example, if data are being transmitted for immediate processing. Cipher feedback operates on characters of fixed length, which may have a length M from 1 to 64 bits. Unlike CBC, above, the ciphering of data is in the feedback path in the encryption process and in the feed-forwards path at decryption. Thus a single bit error at the receiver will propagate through subsequent characters until the bit error has propagated through the feed-forwards path. The bit error will affect M + 1 bits of data.

### 3.6.3. Output feedback

If error propagation were completely unacceptable we could use DES in its output feedback mode. Unfortunately this mode is probably not sufficiently secure for general use in a conditional access system because the effect of key stream repetition, except possibly when M (section ) is equal to 64.

In general the CFB and CBC modes are the most suitable, although in certain circumstances it may be possible to devise a more efficient scheme using one of the other modes of operation, possibly in combination with data such as date and time, or by using sequence numbers in messages.

## 3.7. Key hierarchy

It is a widely held assumption that the algorithms employed in a system will eventually become known. Some system designers have even published the algorithm, so the security depends on the encipherment keys being kept secret. Generally in a secure system we would want to limit the distribution of the secret keys. Unfortunately in a broadcast conditional access system every decoder has to derive the same data from a common control word to seed the encryption/scrambling processes. This implies that the keys and algorithms used, at least for service acquisition, must be present in every decoder. These keys may be referred to as operating or session keys.

Many conditional access system designers try to protect the operating keys by changing them frequently, thus limiting the key lifetime and the risk to the system if the key is found. Clearly, when changing these keys using the over air data channel the new keys must be sent in an encrypted form. The key used to encipher the new service key, which we will refer to as a management key, may be present in a single decoder or in a group of decoders, in which case it may be referred to as a group key. Depending on the complexity of the system there will be number of management keys, which in turn may be loaded by other keys. The top level key would be the unique key. As its name suggests this key would only be present in one decoder and would be associated with the decoder's unique address.

## 3.8. Entitlements

In order to provide control of decoders in large broadcast access control systems, designers make use of cryptographic algorithms which may require the correct key in order to function. However since it would be impractical to distribute a different key for every level of service entitlements are granted to those decoders which hold the necessary keys and algorithms.

Typically an entitlement will allow a decoder to access a service. In some systems, which are referred to as "positive" systems, the entitlement will only be granted for a limited period of time. Thus, even if the decoder still holds the necessary keys and algorithms, the decoder will be able to access the service only if the entitlement is updated.

In a "negative" or "absolute" system the decoder will allow access to the service for as long at it holds the correct keys and algorithms. In this case pirates often attack the system by preventing the reception of messages.

The management of keys and entitlements are independent functions. In a legitimate decoder it is necessary to hold both the valid keys and valid entitlements. In a pirate decoder the testing of entitlements need not be included, thus by subscribing to a free service and obtaining the keys and or algorithms the pirate can make their decoder work on any other service making use of the same keys. It is important therefore to separate low value services and high value services by the distribution of independent keys.

If the use of keys is to be of any value in the defeat of piracy it should be possible to change the keys very rapidly and certainly much more rapidly than a pirate could do so without being detected. If the pirate decoder uses the management keys in order to acquire the operational keys then by acquiring a clone it may be possible to identify the address of the pirate decoder and to disable it by ceasing to distribute keys to that address. It is however outside the scope of this document to provide detailed information on how pirate activities may be countered, so we shall not expand this area further.

## 3.9. Addressing modes

Much of the conditional access system design effort and complexity has gone into the effective addressing and granting of entitlements for decoders. In today's conditional access systems each decoder, or conditional access subsystem, is assigned at least one address. This is often referred to as its unique address. The information bandwidth required by the system will depend on the strategy adopted for addressing subscribers. A number of different strategies are used in existing systems, some of which have proved to be impossible to use operationally. Some of these rely on particular subscriber behaviour to maintain the system efficiency.

Apart from the unique address, which is required for management operations, each conditional access subsystem may be programmed with one or more sub-addresses. Groups of decoders may share a common address and associated key. Entitlements may then be granted to selected members of the group or to the whole group. In this way a large number of decoders may be addressed in a single message. Even taking into account the fact that the groups will include decoders that do not need to be addressed, it is quite possible to increase the number of decoders addressed in a given time by more than an order of magnitude when compared to unique addressing. However, there is inevitably some reduction in the system security.

### 3.9.1. Statistical multiplexing and targeting of entitlement messages

The transmission of access control system data is ideally suited to a system employing statistical multiplexing. Much of the data transmitted is not time critical nor relevant to most decoders and includes a large amount of redundant information. To ensure that all decoders have the necessary entitlement data it is necessary to transmit the data repeatedly at different times of the day in the hope that all the decoders will be switched on and will have received the data on at least one occasion before it is required.

In the simplest case we can expect more decoders to be switched on during the day than during the night. An algorithm designed to cover different viewing habits is clearly very useful when managing large populations. We can improve on the efficiency of the system by targeting the transmission of entitlements according to the current programme schedules.

For example subscribers tuning to a film channel will be most concerned about receiving an entitlement for the current or next movie. Entitlements for programmes to be broadcast in the future are of less importance. Further targeting can be achieved in terrestrial systems since, at least in the case of the fixed decoders, there will be a known population within reach of any one (or two) transmitters. There is clearly no advantage in transmitting entitlements to decoders outside the broadcast area of the transmitter. This reduces the potential population size and consequently the cycle time. Whilst the cycle times are shorter than typical programme lengths there should be few problems with addressing subscribers; if the cycle times are greater than a few hours it becomes necessary to ask viewers to leave their decoders on all the time, which is very undesirable.

## 3.10. Service acquisition times

An important difference between in the clear channels and those controlled by a conditional access system is the noticeable access time when switching between channels. It is not clear what is known about public tolerance to different time delays but existing systems do not all perform well. It is not uncommon for delays of several seconds to occur before access to descrambled signals is provided. On a digital system with many channels to choose from we do not think that subscribers would be happy to have to wait more than half a second per channel as they hop between 60 or more channels.

## 3.11. Features of typical conditional access systems

The following sub-sections indicate typical aspects of the data which is now being, or in the future can reasonably be expected to be, transmitted or made available in a system designed for a large scale Pay-TV channel.

Nationality status

The conditional access subsystem in the decoder may be given a "nationality" by the central control system. Naturally, this can only be as accurate as the central system's records of where each decoder is located. The nationality may then be used to provide automatic selection of preferred language, or control national blackouts. For instance some countries may not permit the viewing of certain material, and companies may wish to limit the reach of advertisements to countries in which the products concerned are marketed.

Decoder status and entitlements

The CA subsystem in the decoder will be told of its status in the central database, i.e. whether or not it has been authorised. This authorisation is usually issued after the viewer establishes subscription arrangements for the Pay-TV services, via the customer management system.

The CA subsystem information is needed to enable it to process other messages needed for its operation. As a security measure this information will change on a regular basis, typically monthly.

It is common for entitlements to be broadcast repeatedly in more than one channel, so that the conditional access system can receive them regardless of the channel to which it is tuned.

Pay-per-view information

Where programmes are offered on a pay-per-view basis, the conditional access system can include an electronic "money box" to which credit can be transferred by messages within the CA control channel. The viewer can then elect to purchase PPV items without having to contact the subscriber management centre. A further feature can be the inclusion of report back facilities, where the CA system will use a modem and telephone link to provide PPV statistics to the central systems, and to receive entitlements.

Geographical location

Some CA systems allow the decoder to be told precisely where it is located (according to the information available to the central system). The conditional access system can then provide either blackout or spot beam facilities. For example the system could refuse access to a pay-per-view sporting event if the viewer lives within the normal catchment area for spectators for that event. Equally, spot beam coverage could be provided for a public service emergency announcement, so as not to concern viewers living outside the area affected.

Parental control facilities

An incidental feature included within many CA systems is the provision of parental control facilities. Here the owner of the decoder can use the CA system to limit access to certain types of programming. Typically this works on a film certificate or similar rating, and prevents sets in homes where there are children from descrambling material above a certain rating unless a release code is entered first. In some systems the rating of a programme and the display of the programme rating can be set according to the decoder's nationality. Effective use of such a system of course relies on the broadcaster transmitting the rating of each programme as part of the ECM data stream.

On screen information

Part of the CA system will be concerned with displaying a variety of messages and other aids to help the viewer operate the system. This is an important part of CA functionality. A typical format for such help is in the generation of a hierarchy of simple menu screens, intended for display on the television receiver. Typical information to be displayed may include instructions and a telephone number for the viewer to contact the customer management centre, or details of the customer's bill and the products (the film and thematic channels) they have purchased.

On-screen displays may also allow for the downloading of programme schedules and channel mapping information. The programme schedule information may take the form of "this programme, next programme" data or may include complete daily or weekly programme schedules for a group of channels. Some systems may then permit the viewer to purchase impulse pay-per-view programmes by reference to and selection from the schedule. Channel mapping information allows the broadcaster to modify the channel number/name relationship without affecting the viewer who is able to work purely with channel names. It may also be possible for viewers to set up their personal preferences via the on screen displays, for instance setting up the order in which channels are scanned when the "next channel" button is pressed on the remote control.

The use of on-screen displays is expected to become more sophisticated with the link up between multimedia and conditional access systems companies. For instance the personal computer software company, Microsoft, has been a partner in a number of initiatives with television broadcasters, telecommunications and cable television companies, and conditional access system suppliers, to incorporate a version of their Windows software into new Pay-TV products. This will be designed to offer and make use of synergy with their video and multimedia computer products, such as Video for Windows (VfW). This drive extends further than just the television brown goods market, with Microsoft also proposing that their software be used to provide standardised operating environments for white goods such as washing machines, dishwashers etc.

## 3.12. Renewable security modules

A system based upon a conditional access subsystem buried within a decoder is likely to be appropriate for small volume applications, where the benefit to the hacker is small and the cost of replacement is not prohibitive. However a large-scale application of a system is likely to attract the attention of the main Pirate or Hacker communities.

Since we believe that all security systems are likely to be broken at some time, the use of detachable security modules, such as smart cards, which can be replaced at relatively low cost, is usually appropriate. Using a smart card as the secure module allows changes to the security system which take any pirate back to the beginning of the attack. It also allows any new methods of physical protection to be adopted. This may also deter a pirate from beginning this kind of attack and is therefore strongly recommended. Care should be taken to ensure that the smart card interface does not limit the security of the system.

The replacement and distribution of smart cards, or other detachable security modules, does pose some additional operational and security problems for the broadcaster, such as control over distribution of the cards (for example in relation to geographic boundaries). These issues are balanced by the logistical complexity of maintaining and controlling a system based on embedded security modules incorporated at the time of manufacture.

Whilst the conditional access subsystem is buried in the decoder, access to the decoder remains under the complete control of the supplier of that subsystem. However if the conditional access subsystem is moved outside the decoder, for example into a smart card, then in the extreme case the decoder could be made to work with any smart card from any conditional access system supplier. If the decoder can be used with a conditional access subsystem from any decoder then it may be described as an "open" decoder, otherwise we will refer to it as a "closed" decoder. See section 6, "Open and Closed Systems".

### 3.12.1. Smart card, form factor and interface

The ISO standard for smart cards specifies the card's physical format, the location of the contacts, the IC and the communications protocol and the card's response to a system reset.

The actual behaviour of a card in a system depends on the application software. Whilst one operator's card may be identified in another's system it will only function if the card and the terminal have compatible application software. Compatibility of the application software is often overlooked by the suppliers of smart card systems and software.

The main purpose of using a smart card is to enable the conditional access system to be upgraded by distributing a device that can be fitted by the subscriber, through the post. There is, however, no need for a smart card intended for use in a Pay-TV application to fit within a wallet or to have the same form factor as an ISO standard card, indeed some systems make use of key-shaped devices. If we were specifying a conditional access system for a digital broadcasting standard we would not restrict ourselves to the ISO standard smart card. We believe that the smart card interface, as currently defined, only just offers the performance required to support the present generation of conditional access systems for analogue broadcasting technologies, even if the smart card is restricted to the cryptographic processing of the messages. It seems unlikely that the card could process more than one television service at a time unless a new and significantly faster interface is adopted. A number of new smart card standards are under consideration at present, hopefully these will provide for greater speed of operation.

## 4. PAY-TV SYSTEMS AND DIFFERENT TRANSMISSION MEDIA

This section reviews how the characteristics of the different transmission media determine the functionality needed from the Pay-TV system. We consider how the potential distortion and interference characteristics can affect the choice of scrambling method and the requirements for transcoding Pay-TV services between different transmission media.

## 4.1. General

The design of a Pay-TV system has to take account of the transmission medium(s) that will be used to carry the services. There are inherent differences between the various alternatives, in the number of channels they can carry, their coverage areas and the interference mechanisms that will be present. These factors will determine the type of scrambling techniques used and the functionality required from the conditional access system.

One important issue that cuts across the various transmission mediums is that of transcoding. It is commonplace for services carried on one medium (e.g. satellite or terrestrial) to be transferred to another (e.g. cable). A well designed Pay-TV system should allow this to be carried out efficiently and cheaply. This requirement will be particularly relevant in the case of digital television, where a number of services might be time division multiplexed together within a single transmitted carrier. Transcoding to a cable medium may require the extraction of individual services from the multiplex, with possible difficulties in processing both the signal control channel and the Pay-TV control channel.

A feature particular to the transcoding of Pay-TV services is the treatment of viewer entitlements. An example is where subscription television services are delivered to some viewers directly from a satellite and to others via a cable system. The satellite channel will contain Pay-TV authorisation information but the cable operator will have his own addressing and enabling requirements. He may also wish to market the channel as part of a "bundle" of other services. A properly designed Pay-TV system should allow simple means to be used for replacing conditional access control channels in such circumstances.

## 4.2. Satellite transmission systems

As far as Pay-TV is concerned, the major characteristics of the satellite transmission system are the extensive nature of the coverage provided by the satellite footprint, the uncontrolled nature of the access to the signal, and (subject to proper planning and co-ordination provisions) the freedom from a number of interference and distortion mechanisms that afflict other systems. The latter characteristic allows the performance of the system to be easily predicted, particularly with reference to error rates for the delivery of the control channels.

### 4.2.1. Transmission characteristics

A satellite-based distribution system is oblivious to international and other regional boundaries. For Pay-TV this has several implications for the way in which the system is operated. Language and cultural differences make it preferable for customer management systems to be operated on a national basis. The central subscriber authorisation system thus has to combine entitlements originating from different centres. This is particularly relevant for digital television, where services aimed at different countries may be included within a single multiplexed signal.

The Pay-TV system must also take account of different languages, currencies and timezones. At the decoder this increases the complexity of the information to be displayed to the viewer. On-screen menus, operating instructions, and pay-per-view prices have to reflect the local language, currency symbols and even prevailing exchange rates.

The advent of digitally-compressed signals has encouraged system developers to explore mechanisms for minimising the bandwidth required to carry a single video channel. This has resulted in a move to 16 QAM (quadrature amplitude modulation) transmission where each symbol represents 4 bits of data and so can have one of 16 possible values. Going from 16 QAM to 32 or 64 QAM would further reduce the bandwidth required but would make the signal excessively susceptible to noise-induced errors.

#### 4.2.2. Security

Since there is no physical connection between the broadcaster and the subscriber's decoder, or pirate decoders, the security of the conditional access system is considered to be of paramount importance. The satellite television market is large enough to provide sufficient revenue to sustain the high development costs required to install and operate secure systems. Nearly all Pay-TV channels are now moving to systems that use "hard" (i.e. digital) scrambling. These systems were developed primarily in Europe for this market, the best known ones being the VideoCrypt system used by BSkyB, the French "Syster" system and the Eurocrypt "M" and "S" standards.

The choice of a "buried" or "detachable" architecture for the conditional access sub-system (CASS) in the decoder has special significance for satellite services. Pay-TV service providers purchase programme rights for their own service areas, which tend to be based upon national or regional boundaries. However, with a satellite distribution system the programme is available over a much wider area, provided viewers can get hold of the appropriate decoder. With a buried solution, it is relatively difficult to organise the large scale illicit shipment of bulky satellite receiver/decoder units, but not so with the very much smaller electronic key or smart card associated with the detachable solution. Large scale activity of this sort (referred to as a "grey market" in cards or decoders) reduces the revenue earning potential of the rights associated with programmes, leading to dissatisfaction from programme makers and distributors.

### 4.3. Cable transmission systems

Cable systems have the potential for carrying a very large number of services, and of offering interactive services. With the advent of digital compression techniques the potential available with cable networks increases dramatically, which means that Pay-TV systems will need to be more flexible and sophisticated.

#### 4.3.1. Transmission characteristics

A properly designed cable network can offer freedom from interference and distortion. One effect which can occur is that of "line tilt", due to imperfections in the frequency response of line amplifiers and other elements in the distribution chain. The effect on the DC level of a video line is shown greatly exaggerated in figure 4.

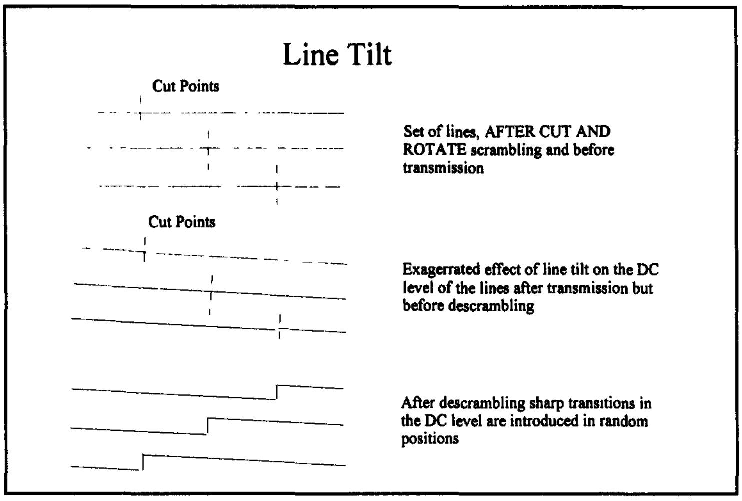


*Figure 4: Effect of line tilt*

The first set of lines shows the video after it has been scrambled, but before transmission. The second set shows the lines after transmission and reception, but before descrambling. Line tilt has occurred (it is shown greatly exaggerated). If the picture was not scrambled the only effect would be a slight and imperceptible change in luminance and hue across the whole picture. The third set of lines shows the effect of descrambling. The resultant effect on actual video is streaky noise, since there are now apparently random changes in the video level across the picture, with abrupt changes at the cut points.

If cut and rotate scrambling systems are to be used on cable systems, the degree of line tilt must be tightly controlled. It is often preferable to move to a different system, such as line shuffling, where this effect is not as prevalent.

In general, the good transmission characteristics of cable networks makes them well suited to digital television formats, and the possibility of using 32 or 64-QAM modulation schemes means that significant amounts of bandwidth compression can be anticipated.

### 4.3.2. Security

The great advantage with cable distribution systems is that access to even the scrambled signal can be controlled. If there is no cable connection into the home there should be no possibility of getting access to the services. In practice, it has been known for illicit taps to be made into the system, either directly or via stray radiation from the distribution network. More common is the technique of paying for access to low-cost tiers to get the cable into the home, and then using various piracy techniques to gain access to premium programming. However, cable operators continue to use less secure systems, based upon balancing the losses from piracy against the increased cost of more-secure decoders.

Future digital systems offer the potential for increasing the security of cable systems, and the presence of a return path via an interactive network could increase this yet further.

### 4.4. Terrestrial transmission systems

Systems developed for terrestrial applications need to be secure for the same reasons as those for satellite television, i.e. the lack of control on who can gain access to the transmitted signal. Unlike satellite television, terrestrial networks introduce a number of transmission impairments, the most common being significant amounts of line tilt (similar to that in cable networks) caused by the transmitters and amplifiers in the distribution chains, and echoes due to multipath effects.

Echoes in terrestrial transmissions are very common, being at their worst in hilly areas or where there are large buildings with reflective cladding, such as in city or town centres. In an unscrambled picture the effect of the echo is to cause a constant banding at the left hand edge of the picture, the sync pulse echo tending to reduce or increase the level of the signal. The echo may be positive or negative. In severe cases "ghosting" can be observed across the whole picture. If cut and rotate scrambling is utilised over the transmission path, since the echo will occur at the start of the scrambled line it will be moved into a random position on the line by the descrambling process. The echo will now be randomly distributed across the field and will appear as noise across the whole picture. This effect has been clearly demonstrated in several trials and makes cut-and-rotate scrambling totally unsuitable for terrestrial networks.

Multipath problems can also affect the digital information carried in the control channels, leading to errors and subsequent deterioration in system performance. In order to overcome this it is necessary to restrict the data rate such that the bit time is greater than the maximum multipath delay which must be tolerated.

Pay-TV systems for terrestrial television have been slow to be adopted in Europe since existing channels have, until now, obtained adequate funding from advertising and government finance. One of the largest and most successful terrestrial subscription television services has been Canal Plus in France. In the UK, the BBC set up night-time subscription services for specialist operations under the brand name BBC Select. However, the introduction of new terrestrial digital services may increase the demand for Pay-TV technology, especially where these carry high value programming, such as premium films and sporting events.

Proposed systems for transmitting digital signals over terrestrial links have implications for Pay-TV systems. Two significant differences from terrestrial distribution systems are the highly regionalised nature of their operation, and the need to cater for the portable television receiver. In addition, the proposed use of Orthogonal Frequency Division Multiplexing (OFDM) techniques and the presence of hierarchical services within the same channel has implications for the way in which control channel data is distributed over the individual carriers in the system.

### 4.4.1. Regionalised systems

A number of different possibilities exist for the distribution of digital television services. In the simplest case a single operator provides a service for the whole regional or national population. However other options may need to be catered for:

- A regional transmission authority could act in the same way as a cable operator, using a single conditional access system applied across all the services carried on the network.
- Scrambled services may be delivered to the regional transmission authority, for transmission without any control being exercised by the transmission authority.

Pay-TV systems for terrestrial applications will need to be developed with these different implementations in mind.

### 4.4.2. Portable receivers

Viewers expect portable receivers to continue to work when transported into different regions. This implies that entitlements for all receivers need to be transmitted across all regions. Taken to extremes, in a fully standardised system the authorisations for a German portable might be transmitted within French and UK control channels. In practice it is expected that some form of re-authorising will be needed, either by contact with the local customer management centre, or by purchase of a locally-valid smart card.

### 4.4.3. Multiple carrier/service operation

The transmission of both high and standard-definition services, utilising different data rates and modulation schemes, within a single channel will have implications for the conditional access system. The scrambling of the data and the synchronisation of the scrambling sequences will need to be co-ordinated. The need to descramble a lower resolution service independently from the high resolution part of the service will require independent hierarchical scrambling sequences. The greater the number of layers in the hierarchy the greater the complexity in the control of the scrambling and descrambling and therefore the greater the decoder costs.

The use of OFDM techniques has the major advantage that signals can be recovered in a multi-path environment. This makes it is particularly suitable for non-line-of-sight paths, such as might be required for mobile or portable TV reception. However individual carriers within the OFDM system will occasionally be subject to partial cancellation. The system must operate with an unpredictable, although fairly small, number of carriers inoperative and the selection of error detection and correction methods must take this into account.

There are several options for the way in which data can be transmitted over the hundreds of individual carriers that make up the OFDM channel, and the design of the conditional access system will need to take account of the relative merits of each of these.

## 4.5. Other transmission media

### 4.5.1. Microwave distribution systems

Microwave video distribution systems (MVDS) are often described as "cableless" cable systems. Typically an individual system will serve a similar sized area and population to that of a cable system. However the services are distributed via microwave channels, typically at about 2GHz, though 12 and 40GHz operations have been proposed.

For Pay-TV operations, the main concern is the freely available nature of the signal within the service area. As for terrestrial and satellite applications this calls for the use of a secure conditional access system. We do not believe that analogue scrambling systems offer sufficient security for Pay-TV systems, although they are still widely employed. Multi-path is an issue with MVDS, so and we would tend to favour systems based upon line shuffling techniques.

### 4.5.2. Master antenna distribution systems (MATV/SMATV)

Master antenna television (MATV) distribution systems and satellite master antenna (SMATV) systems are typically installed in blocks of flats, or hotels. A single antenna is connected to distribution amplifiers. The incoming signal is remodulated and distributed throughout the building, avoiding the need for every flat or hotel room to have an antenna system. The original MATV systems were installed to distribute the terrestrial channels. Increasingly satellite channels, including Pay-TV channels, are being distributed on these systems. In most cases the signals for the Pay-TV services are distributed in their original scrambled form. However in some cases the signals are descrambled and then scrambled using a different system. This allows the SMATV operator to sell a package of services from a single set-top box. In hotels these systems may be used in conjunction with a local playout system to provide additional premium film services.

The market for SMATV systems is very cost sensitive because the revenue available to the system operator is limited. These systems therefore tend to use the lowest cost technology available. Unfortunately these low cost technologies are rather elementary and do not offer much security. As described later, there are very simple means of effecting piracy on SMATV systems and security is dependent on each SMATV operator having a very good knowledge of the subscribers within reach of the SMATV distribution system.

## 5. OPEN AND CLOSED SYSTEMS - TECHNOLOGY

In this section we discuss the issues associated with open and closed systems. Firstly we note that the issue only arises when we consider decoders with detachable conditional access subsystems, such as smart cards. We conclude that there can not be wholly-open conditional access system, although parts of the system may be open, for example the decoders. Unfortunately two open systems may be completely incompatible.

There has been much debate over whether conditional access systems should be "open" or "closed". This results from a number of factors, including a perceived consumer demand for a single set-top box, or television set that can acquire all satellite and terrestrial television services, and from a concern that the technology could be used to create a monopoly.

The issue of open and closed systems only arises when the use of detachable conditional access subsystems is proposed. If the conditional access subsystem is buried in the decoder the issuing organisation will be the only organisation that can address the decoder and the system will be closed. If we consider cable, satellite and terrestrial systems around the world we believe that most existing systems fall into this category.

In the context of decoders with detachable conditional access subsystems we will briefly define what we mean by open and closed systems. If there were an open standard, free from control over who could be granted licences and let us say that the royalty payments were acceptable, anyone would be able to implement a broadcasting system using the standard. In respect of the conditional access system this would imply that they had access to all the cryptographic algorithms, the ability to generate or determine the necessary keys, the means of creating the security modules and all other associated equipment, such as the control systems. Unfortunately, having created this system much of the information would have to be kept secret for security reasons. If this secret information became publicly known anyone, including pirates and hackers, could control the system and it would not be possible for the broadcaster to ensure that only legitimate subscribers could receive their services. The system would be open but unusable.

A single organisation could be created to control the information and operate the system but this is unlikely to meet with the approval of the broadcasters, whose main source of income would be in the hands of another organisation. The same organisation would have to test and approve all decoders and control systems, and through control of technical information and the approvals process it would be able to determine who was able to supply equipment to the market. If this were the only system this organisation could itself be in a monopoly position. The interests of this organisation would not necessarily coincide with those of the broadcaster, for example over the introduction of new features or the replacement of a pirated system. We would not recommend this.

We believe that no advantage can be gained from attempting to control the design of the conditional access system, that this would restrict the future development of systems. It could also facilitate attacks by pirates.

## 5.1. Open decoders

The closest we believe that a system can come to being open is to arrange for all the decoders to be built to the same specification as far as the broadcasting standard is concerned and for a common interface to be defined. To achieve this a single authority[5] would have to be established to control the homologation of decoders. It is not clear how such an authority would be established or controlled.

As suppliers will be able to manufacture decoders with buried conditional access subsystems, or will be able to control access to decoders by other means, an agreement on a standard interface will not automatically result in the creation of an installed base of "open" decoders. Broadcasters and CA system suppliers can also achieve controlled access to an installed base of decoders by providing for the authentication of the detachable conditional

[5] There could be many test houses but ultimately there must be a single point of authority.

access subsystem by the decoders, such that the decoder will only work with one of their own cards. (see below)

If we assume that it is possible to create an "open" decoder then we must define the open standard interface. We have discussed issues relating to the specification of this interface for a digital broadcasting standard in section 8. However if such an interface is to be developed and used, it must be in the commercial interests of the broadcaster and the equipment supplier responsible for the specification of the decoder to do so. We suspect that even if the decoders were to conform to the required specification for an open interface, the broadcasters or suppliers would attempt to arrange for them to be controlled in some way, so as to prevent the use of that decoder by a competing broadcaster or in conjunction with a conditional access subsystem from a competing supplier. On the other hand, in the absence of an open interface and the corresponding hardware, the potential market for pay-TV services would be restricted and potential service-providers might claim that the operators of closed systems were abusing a dominant position.

## 5.2. Authentication of the CASS by the decoder

### 5.2.1. Challenge-response with symmetrical key algorithms

In the case of symmetrical authentication, the same secret key is used by legitimate cards and the card verifier. The card proves its identity by enciphering a random number provided by the verifier and returning the result to the verifier. Because the verifier contains the same secret key as legitimate cards, it can check that the returned enciphered data was created with the same key.

Card personalisation may consist of the issuer generating a card-specific key by enciphering a unique card identification word using the algorithm with the card issuer's master key. The card identification word will typically consist of an issuer identification number, a user identification number (or account number), and the expiry date of the card. The secret key is stored in a secure memory zone on the card. The card identification word (used to produce the secret key) is public and can be stored anywhere on the card and may be the card's unique address.

The verifier transmits to the card a token containing a random number generated by the verifier, and the card responds by transmitting a token containing both its (public) identification word, and the random number enciphered with the secret card key. To remove the necessity of passing the secret key to the verifier, the secret key is derived inside the verifier, using the card issuer's master key (which must be stored in secure memory in the verifier.) This is simply the process used to create the secret key when the card was personalised, i.e. enciphering the card identification word using the issuer's master key.

The derived secret key can now be used to encipher the random number passed to the card, and the result checked against the enciphered number returned from the card.

The main problems with this approach are the security of the secret card key in protected memory on the card and the security of the master key held in the verifier. Where a large number of decoders, and hence verifiers, are available to the public, the master key security is of paramount importance. A pirate gaining access to the single master key of the entire system will have complete access to the system. Certainly the master key cannot be held in an unencrypted form in the verifier memory, and must be protected further. Any compromise of the master key will completely destroy the issuer's investment in the card system.

#### 5.2.2. Asymmetrical challenge-response systems

Use of "trapdoor" public-key methods allows card authentication without derivation of the card's secret key. In such a system, the card is programmed with two related keys, a secret key, and a public one, which are related. As before, the verifier sends a random number to the card, which enciphers it using its secret key, and returns the enciphered result to the verifier. If the verifier can decipher the previously transmitted random number successfully, it will authenticate the card. With its unique secret key, the card is the only one that can possibly have carried out the encipherment.

The best known algorithm is the RSA algorithm[6]. Based on the difficulties of factoring very large products of prime numbers, the RSA algorithm allows for systems requiring a single secret key, held in card memory.

In use, the system could work in the following way:

a. The verifier transmits a token to the card, that contains a random number.

b. The card returns its identification word, and the random number deciphered by the card using its secret key. If the verifier does not hold the related public key, that too can be stored on the card, together with a "certificate" or digital signature calculated over the public key and the card identification word. The "certificate" can be used to verify that the public key and identification word are linked.

c. The verifier enciphers the word returned by the card using the public key, and checks it against the original random number.

d. The "certificate" of the public key should be certified using a system-wide public key.

#### 5.2.3. Asymmetrical "zero-knowledge" techniques

Fiat and Shamir[7] have devised a "keyless" authentication method, where the card issuer, selects two large prime numbers, which are kept secret and only known to the issuer, and generates their product, which can be revealed publicly. The card contains a number of secret values derived from a modular quadratic residue technique, using numbers derived from the card identification word using a cryptographic hash function, also publicly known.

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6 R. Rivest, A. Shamir, L. Adleman;
"A method for Obtaining Digital Signatures and Public Key Cryptosystems";
Comms ACM, pp 120-128, Feb 1978.

7 A. Fiat, A. Shamir;
"How to proof yourself: Practical Solution to Identification and Signature Problem";
Crypto 86 Lecture Notes in Computer Science, Vol 263, pp 186-194, Springer-Verlag.

Authentication starts with the verifier receiving the card's identification word, allowing it to compute a set of numbers derived from the cryptographic function.

A number of challenges and responses are made which involve the square of a random number supplied by the smart card, which is manipulated by the smart card and verifier to produce a unique result if the card is authentic. Repeated a number of times to ensure that the card has used a reasonable mix of its secret keys, this method will give an appropriate verification of a correct card.

Guillou and Quisquater[8] employ a more refined technique than Fiat and Shamir, which only requires one challenge cycle.

El Gamal[9] has proposed another technique based on discrete logarithms, again with a single challenge cycle.

Public key systems are attractive because any terminal can verify authenticity using a public key carried with the card, removing the need for key management. The secret key stored in the card memory may be very long (512 bits is a suggestion we have seen) and difficult to protect from invasive piracy. No item of information stored either on a card or in a verifier will compromise operation of the whole card system. However they require considerable processing. The Fiat Shamir protocol is, however, computationally easier to perform, requiring only multiplications and divisions and not the exponentials of the RSA method.

We expect to see the use of techniques based on public key cryptography to increase in the future and for them to become an integrated part of the conditional access system design in the next generation of systems.

## 6. OPEN AND CLOSED SYSTEMS - COMMERCIAL

### 6.1. Market review

Our experience is that a natural monopoly develops in each market. In the US the satellite broadcasters and the dominant conditional access system supplier are independent companies, although they have formed a joint company to manage the Subscriber Authorisation System. We assume that the broadcasters have agreed to compete on programme quality rather than through the control of technology. In principle it should be relatively easy for a new broadcaster to gain access to the market. However the broadcasters are dependent on the conditional access system supplier and there is little real competition over the supply of decoders.

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8 L. C. Guillou and J. J. Quisquater;
"A Practical Zero-knowledge Protocol Fitted to Security Microprocessors and Minimizing Both Transmission and Memory"; Eurocrypt 88 Lecture Notes in Computer Science, Vol 330, pp 123-128, Springer-Verlag.

9 T. El Gamal;
"A Public Key Cryptosystem and Signature Schema based on Discrete Logarithms"; IEEE Trans Info Theory, Vol IT-31 No 4, pp 468-472, July 1985.

The European experience has been very different. British Satellite Broadcasting (BSB) provides an example of how an open standard can, in practice be closed. BSB used the D-MAC system, which was in principle open, and organised the production of closed decoders through the use of a buried access control system. We believe that BSB hoped to control the market and the installed base of decoders through a complex series of deals with equipment and component suppliers. This was despite the fact that they already had control over all five DBS channels allocated to the UK and thus had full control of the UK satellite television market. In the event, it was competition from "non-DBS" services that resulted in the abandonment of this system.

In attempting to create a closed system the broadcaster could be risking a considerable amount on its relationship with the conditional access system supplier. However it is possible for the broadcaster to arrange for the closed decoder to allow, through a switched external interface, for an alternative conditional access system to be used with the decoders, but only under its control. In the case of the D-MAC system it would have been relatively straightforward to run more than one conditional access system in parallel, thus allowing for competition between conditional access system suppliers but still ensuring that the decoders could not be accessed by other broadcasters trying to enter the market.

The major European Pay-TV broadcasters, BSkyB and Canal Plus, now make use of what are unashamedly closed systems, one of which has aspirations to become a de facto European standard. Past experience is that it is not the drive of a standardisation committee which may result in the imposition of a new standard, but rather the application of market forces - at least in the case of the largely unregulated DTH services in the FSS band.

Unlike the US situation, the key players in the European satellite Pay-TV market are closely associated with conditional access system suppliers. Thus for a new broadcaster to make use of the installed base of decoders, it must negotiate with its main competitor, either directly or indirectly.

Whether or not a broadcaster or equipment supplier will be in favour of an open interface will, we believe, depend on its position in the market. For example, if a broadcaster is established in the market it will not want a competitor to be able to piggy-back on its investment and gain access to all the decoders, which it may have partially funded. If however the broadcaster is second into the market it can be expected to prefer open decoders, until there is significant piracy at which time, depending on the type of attack, both broadcasters may wish that it was not so easy for the pirate to distribute conditional access subsystems, possibly in the form of smart cards. Any equipment manufacturer that does not control a conditional access system will want the decoders to be open, or at least the licences to manufacture the decoders to be freely available. On the other hand an equipment manufacturer that controls a conditional access system may want to limit the suppliers of equipment, at least to those that are prepared to pay a licence fee.

Whilst we can see a fundamental difference between the case of a broadcaster financing the supply of decoders and that in which the subscribers have purchased the decoders, it is hard to argue that a broadcaster should not be able to supply closed system decoders to the market at least as part of a "package" including the supply of programmes. However it is highly undesirable if broadcasters can dictate to the decoder manufacturers that a television or a decoder incorporating their system may not also incorporate another system.

### 6.2. National programme rights

Cross-border control needs to be considered in the distribution of smart cards, since programme rights may only be available for one country. With common receivers available across Europe it would be difficult to control the flow of smart cards across national boundaries, especially in the context of the single European market. Moreover the political changes in Europe will make this situation both more dynamic and more difficult to control. Conditional access systems including a level of geographical control must therefore be made available to broadcasters if this problem is to be addressed by a technical approach instead of a contractual one.

The need to control the cross-border movement of smart cards suggests that the decoders must include something that can be used to make the distribution of the cards as complex as that of the decoders. The techniques proposed in section 8 could be used in this respect.

## 7. SYSTEM PIRACY

In the context of Pay-TV systems, "piracy" consists of systematically obtaining unauthorised access to services, including the case of the unauthorised use of legitimate hardware. In this section we consider why and how Pay-TV systems are attacked. We look at how the piracy market has matched the development of Pay-TV technology, and how lessons from the past should be used to determine Pay-TV requirements for future digital systems.

### 7.1. Introduction

We define "piracy" as obtaining unauthorised access to Pay-TV services. Those seeking to defeat conditional access systems are known as "pirates", or alternatively "hackers"; the habitual usage of the latter term stems from the close relationship between the Pay-TV pirates and those seeking to defeat more generalised computer systems. In both cases the objective may not necessarily be financial gain, but more of an intellectual exercise, coupled with the perception (by the hacker) of striking a blow for freedom against the authoritarianism of society. This explains how sophisticated computer facilities (e.g. at universities and research institutes) can be employed in such attacks. Once a system has been broken financial gain may well be considered, perhaps as another means of demonstrating the success of the individual over society.

This is not to say that commercially orientated piracy does not exist. Recent history has shown that profits can be made from piracy, and then ploughed back into funding further and more sophisticated attacks upon the system. Piracy is a natural adjunct to Pay-TV technology, and it is to be expected that attacks will be made on whatever system is implemented. Common failings in the past have been to underestimate the abilities of the pirate community, and to forego proper security measures in the interests of short term cost and implementation issues.

The history of the piracy of analogue television signals is of relevance to digital television since it illustrates the ways in which the market develops. It also shows how wrong assumptions can allow the pirate to become established within the market place, and once established become very hard to dislodge.

No operational conditional access system can be expected to remain secure for ever, indeed there are few systems in operation today that have not already been compromised. It is essential to consider the response to piracy threats as part of the overall system design. This should take into account the need to migrate to new standards as the technology available to pirates develops during the conditional access system lifetime.

In general, conditional access systems have not been broken through an understanding of the algorithms or key hierarchy employed. Many of the successful attacks have been made through the system features, access to programmes through free subscriptions, extension of free preview periods or very often by preventing the receipt of messages by a decoder. In some cases a leak from the design laboratory or the system operator has enabled the pirate to develop their "product".

Although systems will be attacked even when there is no commercial gain to be made from such an attack, the real problems come when that attack threatens the business operation of the broadcasters using the systems. If there are many different systems either planned or in operation the threat from the breaking of one system is limited. If there is a single system in use the opportunities for pirates are greater, so too is the overall threat to the system and the cost of implementing anti-piracy measures. Failure at one point in the system, by one broadcaster or card manufacturer, could affect all the system users. Flexibility in response to as yet unforeseen piracy attacks is essential.

It is important, too, to consider the different viewpoints of the broadcaster and the equipment supplier. To the broadcaster piracy represents at worst lost revenue, whilst piracy may actually lead to increased equipment sales. For instance, if a pirate smart card becomes available this may lead to increased sales of the decoders in which the cards can be used.

## 7.2. The history of pay-TV piracy

As noted above, piracy is as old as Pay-TV itself, with most of the early developments in both occurring in the USA. Early cable distribution systems utilised simple interfering carriers superimposed in less critical areas of the video signal spectrum. Authorised subscribers were provided with simple notch filters which removed the interfering carrier. Needless to say any technician or college student with access to a spectrum analyser could easily construct an identical filter. Others of the "analogue" scrambling methods described previously were then tried and a war of attrition would occur as the cable company changed the scrambling technique whenever the number of pirate decoders started to impact upon revenues.

The real growth in piracy occurred when the size and price of TVRO satellite terminals fell to the point at which they began to appear in domestic gardens. Owners of this equipment could receive the Pay-TV channels being distributed to cable head-ends via satellite links. Originally, these channels were not protected by any form of conditional access, as it was not thought necessary. Eventually the loss in revenues began to hurt the Pay-TV operators to the point at which they had to invest in scrambling equipment. However, by that time the Direct-to-Home market had become well established and there was considerable public backlash against the loss of these "free" services. This established a willing market for any illegitimate decoders, but also ensured that DTH became an established method of broadcasting.

As with their cable counterparts, the early satellite conditional access systems did not pose too much of a problem to the dedicated hacker. The sync and video-modification techniques (inversion, removal, delay etc.) were soon overcome. Service providers then moved to more complex combinations of techniques. One of the most notorious compromises of an established Pay-TV system was on the General Instrument "Videocipher II" product. This represented a new level of sophistication on the part of the pirate community since it involved the analysis of the control code stored in the decoder memory. The code was then altered so that an authorisation for one channel effectively authorised all the channels. This gave rise to the name of "musketeer" for the hack (all for one and one for all). The situation was eventually retrieved by the manufacturer, but the success of the attack spawned a number of imitations over the following years.

## 7.3. Current piracy activities

Europe has quickly caught up with the USA in piracy activity. With many of the early analogue-based systems totally compromised, attention has switched to the newer digital systems introduced by operators such as BSkyB. These have the attraction for the pirate of offering a highly lucrative market if the system can be compromised and pirate devices produced at economic prices.

The following are all attacks that we are aware of at present and which appear to have had some degree of success in compromising a Pay-TV system. We are wary of making any firm pronouncements, since operators may be able to nullify the operation of some of these compromises in the future. Nevertheless, the piracy community will argue that even a short-lived compromise counts as a success, since it generates a considerable amount of publicity, has the potential for earning some revenues, and encourages further endeavours.

### 7.3.1. Unauthorised card exchanges

Organisations exist which co-ordinate the exchange of smart cards outside of the geographical regions to which they are supposed to be limited. This bypasses the arrangements for selling programme rights for specific countries, allowing a film or similar item to be seen in wider areas than those to which the programme distributor agreed. If this trade becomes extensive the programme and film industry will begin to suffer losses in revenue, and will be less willing to enter into arrangements allowing other programme material to be broadcast.

### 7.3.2. Illicit "smart cards"

Crude (but apparently effective) replacements for smart cards are now said to be available for some of the main satellite broadcasters in Europe. We assume that the sale of such devices would be illegal within the "home territory" of the broadcaster concerned. However, it is easy to distribute something that is similar in size to a normal smart card, and it is not yet obvious how effective any legal constraints on such trade may be.

### 7.3.3. Card disablement

A number of other attacks have taken place which interfere with parts of the smart card operation or with the exchange of messages between the card and the main part of the decoder. These have involved the introduction of diodes and replacement of interface components. The attacks show a clear appreciation of the working of smart cards and similar devices, and emphasise the care needed when adopting the "negative system" approach of C A operation.

#### 7.3.4. System "highjacking"

One of the more intriguing developments at the moment is the apparent highjacking of a complete Pay-TV system. A manufacturer is marketing a Pay-TV encoder that is claimed to be able to deliver programming that can be decoded by the VideoCrypt decoders already installed in the UK and Europe. This undermines the investment made in developing the original Pay-TV system and in installing the decoders. It is not unusual for service-providers to subsidise decoders, either directly or indirectly, to maximise the take up of services, and such highjacking could represent a considerable financial loss.

#### 7.3.5. SMATV piracy

SMATV piracy is an example of how the system security may be bypassed. In systems which rely upon the distribution of scrambled signals throughout apartment buildings it is possible for the system to be pirated for the benefit of the whole community within the apartment building without any understanding of the system security. By inserting an authorised descrambler into the distribution system, and feeding the output into the apartment's network the descrambled signal is made universally available. Whilst this may not be practised on a large scale, prevention relies on a good understanding and auditing of such networks.

### 7.4. Issues related to open and closed systems

It is often said by conditional access system suppliers that the smart cards can be replaced in the event of a break, whereas in a system using a buried security module the whole decoder must be replaced. This is a significant advantage but it is gained at some cost. Taking advantage of this would require time and effort, and the card issuer and the programme provider(s) will need to have a mechanism for reaching a mutual understanding about the need, timing and extent of card replacement in the event of system piracy to all or part of the system. In a conditional access system making use of buried conditional access sub-systems, the pirate must modify or distribute new decoders, which is a major task.

In a smart card system the pirate is able to distribute pirate cards with ease. There are a number of reported attacks on smart cards and we expect to see more. If a break is successful the broadcaster may have to replace all the smart cards in circulation, the cost of such an exercise could run into many millions of ECUs for a large population of subscribers.

A pirate system which replaces the smart card can be made both more flexible than the original card, and user configurable. A home PC could easily perform the functions of the current generation of smart cards once the critical data is known. In addition a change in smart card design that involves no more than a simple program or data alteration could have significant cost and lead time implications for the CA system supplier and broadcaster, yet leave the card open to a new attack by the pirate. If the compromise results from a failure of the overall system and not the card itself, replacement of cards will not have the desired effect. The economics of card replacement could easily be stacked against the broadcaster so that it is not a viable option.

This situation is improved if the pirate cards can be disabled by broadcast messages, but this assumes that the cards have an identity and that the broadcaster can prevent the card from receiving the necessary cryptographic keys required to continue operating.

#### 7.4.1. Smart cards in a closed decoder - card authentication

The decoder may include a buried security device. This could be used for two purposes, firstly to verify that the smart card has been issued by the legitimate system operator and secondly to verify that the smart card has been authorised to work with that decoder. In the first case the purpose is to develop a closed decoder, in the second the aim is to prevent the distribution of cloned cards since the cloned smart card will only work in the decoder in which it is authorised. These requirements have been widely discussed, a number of systems have attempted to address one or both of these, for example the NR-MSK[10] system proposes the use of a buried conditional access subsystem (CASS) or adaptor, which includes a unique key and a secret/public key pair. VideoCrypt decoders make use of the Fiat Shamir protocol.

### 7.5. Attacks on the system operation or design

Decoder piracy is only one mode of piracy that needs to be considered in the implementation and operation of a system. It is however one of the most important modes, since sales of pirate decoders can be very profitable and very visibly undermine the broadcaster's business.

A conditional access system is vulnerable both in its design, implementation and operation. Wherever secret data is held, whether it is at the design centre or the operations centre or in the decoder, the system is potentially vulnerable to attack. For example, at the customer management centre the operators may illegally authorise their friends, or the more senior managers may run batch transactions and sell the service to external organisations. Even in the most sophisticated systems, using complex key hierarchies, the initial data must be created and programmed into decoders (or detachable security modules). Security must always be enforced by a combination of good system design, management practises, personnel policies and auditing procedures. Whilst the system can not be protected against all forms of attack, it is of great value to know that an attack has taken place. (The use of a safe may not prevent bank robberies, but hopefully the robbery is self-evident).

### 7.6. Commercial response to piracy

It is often suggested that piracy can be managed by allowing for the loss of revenue as part of the overall business plan. Whilst it is true that not all pirates will subscribe to the service if they are switched off we believe that following this approach can create significant business problems.

Whilst it is unrealistic to assume that piracy can be eliminated altogether it is, we believe, essential to keep the level of piracy to a minimum. If this form of fraud is allowed to build up to a budgeted level the following consequences should be considered:

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10 NR-MSK access control system, proposal for a new part 6 of the EBU specification for the MAC/packet family, issue 2, July 20th, 1989, section 4.5.6.

- The pirate community will be building up a business with an increasing value as the legitimate business grows.
- The revenue captured by the pirate community can be invested in new developments to overcome any anti-piracy operations by the legitimate broadcaster.
- An expectation of piracy builds up amongst the population (in the US denial of a service has even been challenged, unsuccessfully we believe, as an infringement of civil rights). In extreme cases piracy can lead to resistance to taking out a subscription to a service (after all why subscribe when your friends get the service for nothing).
- It can be very difficult to reduce the level of piracy. Once the pirate communities have built up a business, with reserves from the good times, they become formidable operators. If the piracy level goes above that which is planned it can be very expensive to recover the situation.
- The level of system piracy may affect the cost and availability of programme material. This is particularly true with newly released films, and pay-per-view programming (where copyright fees may eventually be related to audience sizes).

A successful piracy attack not only represents a loss of revenue to the legitimate broadcaster but the replacement of pirated decoders with a new ones can represent a substantial cost burden on legitimate subscribers and/or the broadcaster. The manufacturer of the new or upgraded system, on the other hand, may profit from these increased sales. It is in the interest of the broadcasters, more than any other organisation, to maintain the security of the system.

## 8. DIGITAL TELEVISION

The introduction of digital broadcast television standards is a major development for the broadcast industry. In this section we consider the implications of new digital technology for Pay-TV systems, and for the conditional access element in particular. We examine some of the issues associated with the security of these systems, as well as the possibilities for the operation of multiple conditional access standards within the broadcast environment.

When referring to digital broadcast technologies we will use MPEG as an example of such a system since at the time of writing so much of the effort in Europe is directed at the proposed MPEG standard. We have, however, discussed the issues in the context of a general digital broadcast standard, so that some of the issues may not apply to MPEG itself.

### 8.1. Options for data encryption

It is not possible to encrypt an analogue video signal, even if it is processed in the digital domain in the encoder and the decoder. Instead, a number of schemes have been devised for manipulating the analogue video waveform. Examples of these are cut and rotate scrambling and line shuffling. In these schemes there is a clear boundary between the derivation of the control words required to initialise these processes and the processes themselves. In a digital broadcasting standard there is no need to manipulate the video signal since the video data may be encrypted. This significantly increases the freedom of the system designer, but complicates the task of the standardisation bodies.

For example, in MPEG the video signal will be encoded into a digital data stream. We will refer to this as the application layer[11]. This data stream will be carried in the MPEG multiplex in data packets, which form the transport layer[12]. The data could be encrypted prior to being packetised or it may be packetised in the clear and encrypted at the transport layer, or both.

### 8.1.1. Encryption at the application layer

If the compression is carried out at the studio the MPEG video may be encrypted prior to distribution, thus providing end-to-end security. However the decryption process in the decoder is dependent on the type of data being carried and could be different for the video and audio components, or for any independent data services. Therefore, if the data is encrypted prior to being packetised, the conditional access system would have to have some knowledge of the structure of the MPEG video data in order to synchronise the encryption and decryption processes at the encoder and the decoder.

### 8.1.2. Encryption at the transport or packetised layer

If the data is encrypted after being packetised, the encryption of each MPEG packet can be done in a manner which is independent of the type of data being carried. The synchronisation of the conditional access system can then be carried out at the packet level.

Encryption at the transport or packetised layer will still, however, need to be done for different component types. These may be identified by the programme identifier (PID) in the MPEG multiplex. This implies that the decoder must have a separate decryption process for each component of a service that it needs to acquire.

We believe that encryption and decryption of the MPEG packets would result in a less complex decoder design than the first option since the encryption process will be the same for all types of data.

11 We assume that MPEG compressed video is self synchronising and may be carried on any transport layer.

12 We assume that MPEG packets may be carried over any network.

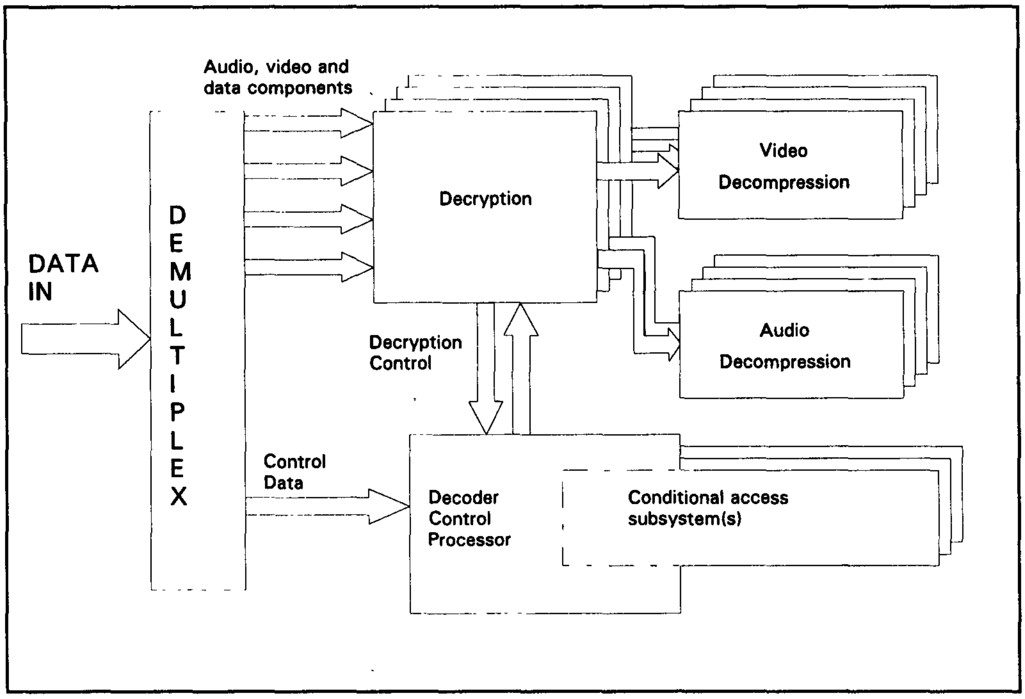


*Figure 5: Digital decoder with buried or detachable subsystem*

We note that the encryption of the video data must take place after all the compression stages. Provided the data is decrypted in the reverse order to its encryption, the data may be encrypted many times and then successfully decrypted. In this way a conditional access system supplier could arrange for their data streams to conform to the MPEG standard, whilst remaining proprietary by superimposing an additional layer of encryption.

We have shown a possible decoder architecture in Figure 5. In this case the decryption of the data takes place after de-multiplexing. If the MPEG packet structure is removed in the demultiplexer then we have shown the case of a decoder working on the application layer. If the packet structure is intact after de-multiplexing the encryption process may be working on the transport layer. If we are decrypting at the transport layer the decryption process could be shown prior to the de-multiplexer. However even in this case, because of the data rate and the amount of processing required and the need to decrypt components independently, we expect that more than one decryption engine would be required.

In Figure 6 we have shown an alternative decoder architecture which could allow for the decryption of the data to take place inside the conditional access subsystem, which could be detachable.

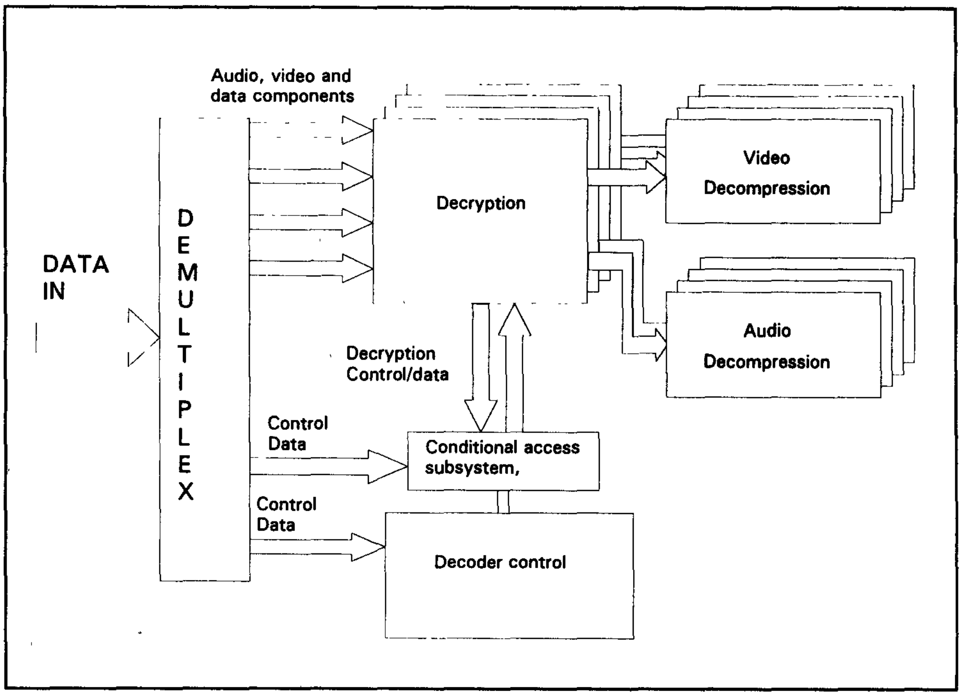


*Figure 6: Alternative decoder architecture*

### 8.2. Data encryption and control

Many of the conditional access systems used in conjunction with analogue broadcast standards use a control word to control the scrambling sequences. In MAC these scrambling sequences were defined as part of the broadcast standard and this could also be done for MPEG.

In many analogue systems this control word is made available on an open interface between the conditional access subsystem and the scrambling circuits. We note that in a truly secure system these control words should not be transferred across an open interface, unless the data rate is sufficiently high that the threat from someone re-broadcasting control words can be ignored[13].

### 8.3. Encryption control interface

In this section we discuss the encryption interface requirements. If interworking is required between conditional access systems a common interface must be defined. However in a digital system such an interface is not essential.

If the decryption of data in an MPEG decoder is to be controlled by passing control words (or keys) across an interface from any conditional access subsystem, then the algorithms for data encryption must be defined within the MPEG standard.

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13 This would certainly be true if the data capacity required to transmit control words was of the order of that required to re-broadcast the television service in the clear.

the data to be encrypted or decrypted in an exclusive-OR (XOR) operation, or alternatively this data is used to control the video scrambling process. However this is only one means of encrypting data. One alternative is for the data to be passed through a suitable cryptographic algorithm. The algorithm may act on a stream of data or on blocks of data.

The publication and use of PRBSs for the encryption of data has been accepted in many broadcast conditional access systems and we are not aware of any published attacks on the PRBSs of systems such as Videocrypt and the MAC. We expect that it would be easier to gain general approval from conditional access system suppliers for the use of a PRBS than it will be for the use of DES or a similar proprietary algorithms, regardless of the technical merits of either approach.

The optimum solution depends on a number of factors, including the required channel acquisition time, the probability of errors in the data to be decrypted and the nature of the data. Not surprisingly there is no one solution upon which experts are likely to agree.

It is not necessary for a digital broadcast standard to specify how the data will be encrypted. The minimum specification for a standard decoder would have to state at which levels the data could be encrypted and how synchronisation should be achieved. If this approach is taken it will be much more difficult, and probably impossible, for a broadcaster to arrange for more than one conditional access system to be used to protect a single component. We have discussed this further in the next section.

### 8.4. Support for multiple conditional access systems

Consider the case of a conditional access system with a specified algorithm for the encryption of data. To again access to a given component of a service we would have to provide the correct key (or control word) to that algorithm. All decoders, whether they are open or closed, would operate correctly provided the same key was made available to them at the right time. The key need not be published in the clear in any decoder, that would be a matter for the decoder designer. Provided there was some degree of co-operation we could use any conditional access system to deliver the control words to the required set of decoders.

Allowing more than one type of decoder to run in parallel would enable a broadcaster to migrate over a long period of time from one conditional access system to another, without having to replace the installed base of decoders. Alternatively the broadcaster could deliberately choose to operate more than one system in parallel, thus enabling competition between conditional access system suppliers but we should not underestimate the complexity of running more than one system in parallel before considering this as a viable option.

### 8.5. Transmission of control words

It is a widely held view that the transmission of encrypted control words provides the best means of initialising the scrambling sequences. However if the control word duration is very short it may be difficult to repeat the control word frequently enough to ensure that it will be acquired by all decoders. We must be certain that there can be no confusion about the validity of a control word queued for transmission at the end of one period, but not received and processed in the decoder until the start of the next period. This can be achieved by identifying control words with a sequence number, which may only need to be a few bits in length. Depending on the chosen key hierarchy and the algorithms employed it may not be necessary to transmit encrypted control words to the decoder, although we must be certain that only legitimate decoders can make use the data to allow a subscriber to gain access to services to which they have subscribed. Certainly not all systems in use today require the transmission of encrypted control words. It is not, however, possible to describe the operation of these systems in a document of this type.

### 8.6. Mapping of components and keys (control words)

We suggest that the mapping of components to keys (control words) is a function of the conditional access system and should be determined by the conditional access system. The access control data, for example the ECM and EMM data, has to be recovered from the multiplex. In MPEG, one or more PIDs could be allocated[14] to the access control data.

There may, however, be more than one conditional access system present on the channel (see section ). In this case the control data for different conditional access systems should not be required to share the same PIDs. The system design should allow for more than one conditional access system to be present in the MPEG multiplex.

### 8.7. Component identification

Whilst MPEG must define the data structure for the compressed video and audio, it is not clear that it should or needs to define the data structure for conditional access messages or other data services. Again, even if the data structure for the conditional access messages is defined there is no reason why this data should not be carried as an independent data service.

In a typical analogue television system there is only one television service on each RF channel. As a consequence, existing system and decoder designs typically only ever provide for accessing a single television service (although this may comprise a number of independently scrambled components, for example video, audio and teletext components). We have encountered other systems which support picture-in-picture facilities, or for video-recording one channel whilst watching another. However none of these really address the issue of acquiring multiple independently scrambled components in an efficient manner.

In a digital broadcasting system many television services and their associated components will be available on a single RF channel. There will, therefore, be little reason for not allowing picture in picture or the recording of one service whilst watching another. Certainly the conditional access system design should allow for this. Whilst it would be possible to design a system for MPEG by considering a straightforward extension of existing systems, so that the functions of the existing systems were duplicated according to the number of television services to be acquired, this could well result in an inefficient design.

In table 1 below, we show an example of a channel carrying two television services within the same multiplex. We have given the service components IDs, which could be the MPEG

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[14] There is no need to create the distinction between different types of access control data such as ECMs and EMMs in the broadcast system.

PIDs (or CID). The conditional access system data relating to each component will also be carried within the multiplex under its own PID or set of PIDs. Although the allocation of PIDs to the conditional access system data will depend on the conditional access system (since it may require one or more PIDs, this can not be specified within MPEG) we believe that it should be possible to determine the allocation of all the PIDs from MPEG control data, even if for practical reasons the decoder might obtain the same information from the conditional access system control data.

| Television Service 1 | | | Television Service 2 | | |
|---|---|---|---|---|---|
| Component description | Component PID | Conditional access system ID, PID | Component description | Component PID | Conditional access system ID, PID |
| Video 1 | 1 | A,100 | Video 2 | 7 | B,200 |
| Audio 1 | 2 | A,100 | Audio 3 | 8 | B,200 |
| Audio 2 | 3 | A,100 | Audio 4 | 9 | B,201 |
| Teletext 1 | 4 | A,100 | Teletext 3 | 10 | B,240 |
| Teletext 2 | 5 | A,100 & B,250 | Teletext 2 | 5 | A,100 & B,250 |

*Table 1: Television services and components*

| Television Service 1<br>Conditional access system A | | Television Service 2<br>Conditional access system B | |
|---|---|---|---|
| Component description | CA - Key stream/CW | Component description | CA - Key Stream/CW |
| Video 1 | A1 | Video 2 | B1 |
| Audio 1 | A1 | Audio 1 | B2 |
| Audio 2 | A2 | Audio 2 | B3 |
| Teletext 1 | A3 | Teletext 1 | B4 |
| Teletext 2 | A4 | Teletext 2 | B5 |

*Table 2: Mapping of the key streams to components*

Table 2 shows the mapping of the service components to the CA key streams, and relates to the television services in table 1. In our example of two television services, each controlled by different conditional access systems, we can see from table 2 that the access rights for the video and the first audio component in television service 1 are the same. The second audio component and the teletext components have been given different access rights, even though the access control data is can be found under the same PID (100) in the multiplex. It is not be possible from within the MPEG multiplex to determine this, since a single PID has been allocated to the conditional access data for all the components of the service. In the second television service the access control data for the video and the first audio component share the same PID, the remainder are on different PIDs, yet the access rights for all the components are different.

In Table 1 we have also shown the case of a single component, the second teletext component, being part of more than one service and being accessible by more than one conditional access system. In this case each conditional access system provides the necessary control data from its own key stream.

In the same way the access control system will need to place messages carrying entitlement data in the multiplex. These are frequently referred to as Entitlement Management Messages (EMMs). The distinction between an ECM and an EMM is, to some extent, arbitrary. They are both conditional access system messages and should be identified as such within the multiplex. An examination of some access control systems shows that the same type of data, for example text or configuration messages may be carried in either ECMS or EMMs. The classification of the messages is, in any case, independent of the broadcast standard, thus we see no need for such a distinction to be made. The broadcast standard should, therefore, allow the conditional access system to allocate messages to one or more component IDs, (PIDs) which may for convenience be labelled as ECMs or EMMs.

## 9. FUTURE PAY-TV REQUIREMENTS

### 9.1. Thematic sales

Current CA and CMS operations are geared to selling a customer a continuous subscription to a service, or else to selling the rights to watch a single premium event. An alternative approach which could be offered would be to be able to purchase the rights to watch all programmes on one or a group of channels on a specific theme. For example a viewer might opt to be able to watch all wildlife programmes transmitted on a group of channels, some satellite and some terrestrial. The provision of such a service would rely on the ECM data stream carrying information on the thematic content of the current programme (as is already catered for in the design of some CA systems, but is not yet, we believe, being used on a programme-by-programme basis by any broadcasters).

The advent of digital television services, leading to increased numbers of channels, is likely to make indiscriminate subscription to all programmes on groups of channels less attractive, and thematic sales may offer an alternative way for the broadcaster to obtain revenue. It would also offer them immediate feedback on the cost-effectiveness of different programme strands.

The provision of such services, whilst not posing technical problems in the CA systems, will result in the need to enhance and expand the operation of customer management systems and centres.

### 9.2. Pay-per-view

With the introduction first of advance and then impulse pay-per-view it will become possible for broadcasters to strike different forms of rights purchase deals with the film companies and sports events organisers. It will be possible to pay a royalty based on the exact number of viewers (assuming there is no piracy of the signal). For some events or premium films the broadcaster may wish to restrict the numbers who can watch, and charge a correspondingly high fee for those who do. For example there are suggestions that the film companies might agree to the pre-theatrical release screening of a new film provided that the numbers of viewers allowed to see it was restricted to some preset limit. The film companies' view is that this pre-release showing could be used as sales publicity for the film on its theatrical release. However for this proposal to be considered seriously, a foolproof method of preventing the unauthorised taping of the signal would have to be associated with the CA system.

### 9.3. Video on demand

One of the new types of service likely to be implemented with digital television systems is that of Video on Demand. In purist terms this envisages a scenario where a viewer can choose a film from a catalogue of titles and commence watching it at any time. This degree of functionality could be made available on a cable distribution system, though it is better suited to telephone based compression systems, such as the Asymmetrical Digital Subscriber Loop (ADSL) technology currently being developed by some telephone companies in the USA and Europe. However a variant of this which may be implemented on satellite and cable systems is Near Video on Demand. A limited number of recently released titles are made available in adjacent channels. Each channel contains a multiplex of services, each of which carries the same film but starting at different times, perhaps fifteen minutes apart. The viewer pays a premium price to watch a particular film, but has a choice of start times.

### 9.4. Interactive television

The synergy between television and computer multimedia products is such that it seems certain that there will now be a concerted effort to introduce multimedia products linking locally computer generated graphics with broadcast signals. An extension of this is to provide feedback, either to a broadcasting centre or to a local controller such as a PC, so that the viewer's relationship to the signal becomes interactive rather than a passive recipient. This is likely to lead to a wide range of leisure and commercial activities, the majority of which are almost certainly not yet foreseen, or if foreseen have not been considered to have the potential to become major ingredients of new products or activities. An example of this effect is the spin-off of leisure activities from the development of the personal computer, the sheer scale of which was not foreseen. In a similar way it may be that interactive video or other forms of multimedia become the technology drivers of future television broadcasting technology, rather than, for example, high definition television.

## 10. GLOSSARY

| | |
|---|---|
| ADSL | Asymmetrical digital subscriber loop |
| AM | Amplitude modulation |
| APPV | Advance pay-per-view; the ability to purchase the right to view a specific programme by making an advance purchase via a CMS, or ARU/VRU. |
| ARU | Automated response unit (to allow a consumer to interface with a CMS by means of a touch-tone telephone) |
| Attributes | Pay-TV characteristics of a programme item |
| CA | Conditional access |
| CASS | Conditional access subsystem , e.g. a smart card |
| CATV | Cable television distribution system (community access TV) |
| CBC | Cipher block chaining (of DES) |
| CD-I | Compact disk - interactive (a multimedia system) |
| CD-ROM | Compact disk - read only memory (a multimedia storage system) |
| CFB | Cipher feedback (of DES) |
| CMS | Customer management system |
| Cut and rotate | Digital scrambling technology, cutting television lines into two or more segments and reversing their order |
| CW | Control word |
| DES | Date encryption standard (IBM, US Federal Bureau of Standards) |
| DTH | Direct to home television broadcasting from satellites |
| DVI | Digital video interactive (a multimedia system) |
| ECM | Entitlement checking message |
| ECM | Electronic code book mode (of DES) |
| EMM | Entitlement management message |
| EEPROM | Electrically erasable programmable read only memory |
| Entitlement | A right to descramble (watch) the category of programme described |
| EPROM | Erasable programmable read only memory |
| FM | Frequency modulation |

| | |
|---|---|
| FSK | Frequency shift keying |
| Hacker | Similar to Pirate, sometimes distinguished by motive (fun rather than financial reward) |
| IC | Integrated circuit |
| IEC | International Electrotechnical Commission (also CEI - Commission Electrotechnique Internationale) |
| IPPV | Impulse pay-per-view; the ability to purchase the right to view a specific programme without needing to make advance arrangements |
| ISO | International Standards Organisation |
| JPEG | Joint photographic experts group (of ISO / IEC) |
| Line shuffling | Digital scrambling technology, re-ordering lines in a field |
| LNB | Low noise block |
| MAC | Multiplexed analogue components (one television system used for DTH) |
| MATV | Master antenna television (distribution system) |
| MMDS | Microwave multipoint distribution system |
| MPEG | Motion picture experts group (of ISO / IEC) |
| MVDS | Microwave video distribution system, same as MMDS |
| NTSC | National television system committee |
| NVOD | Near video on demand; the purchase of one of a limited set of programme items, with a choice of defined start times |
| OFB | Output feedback (of DES) |
| OFDM | Orthogonal frequency division multiplexing |
| PAL | Phase alternate line (one analogue colour TV system) |
| PCC | Programme control computer |
| PCS | Programme control system |
| Pirate | Designer, manufacturer or user of unauthorised decoders |
| Pirate decoder | Decoder that gains access to a scrambled service without a legitimate entitlement |
| PPV | Pay-per-view; the purchase of the right to view specific programme items |
| PRBS | Pseudo random binary sequence |

| | |
|---|---|
| PSK | Phase shift keying |
| QAM | Quadrature amplitude modulation |
| QPSK | Quadrature phase shift keying |
| RAM | Random access memory |
| ROM | Read only memory |
| RSA | RSA public key cipher (Rivest, Shamir and Adleman MIT 1977) |
| SAS | Subscriber authorisation system |
| SECAM | Séquence couleur à mémoire (one analogue colour TV system) |
| Soft encryption | The system may be broken without resorting to a complex expert attack |
| Soft scrambling | The system may be broken without resorting to a complex expert attack |
| Smart card | IC mounted inside a credit card sized laminate |
| SMATV | Satellite master antenna television (distribution through hotels, blocks of flats) |
| SMS | Subscriber management system (another term for CMS) |
| Subscription | Purchase of the right to view a Pay-TV service for a particular period of time (typically a month or a year) |
| Sync pulse | Horizontal or vertical synchronisation pulses |
| Tier | One of a group of independent access rights |
| TVRO | TV receive only |
| UHF | Ultra high frequency band in the radio spectrum |
| VBI | Vertical Blanking Interval |
| VCR | Video cassette recorder |
| VfW | Video for Windows (multimedia extension for Microsoft Windows) |
| VOD | Video on demand; the purchase of a viewer specified programme to begin at a time specified by the viewer |
| VRU | Voice response unit (to allow a consumer to interface with a CMS by means of a touch-tone telephone) |
| VSB-AM | Vestigial sideband - amplitude modulation |

Chapter 3

# Channel coding and modulation

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# Table of contents

## EXECUTIVE SUMMARY

The significant progress achieved in the use of digital techniques in production, transmission and emission of radio and television services is rapidly changing the established concepts of broadcasting. The availability of digital techniques at low cost is the key for future introduction of multi-programme TV services at various quality levels: EDTV (Enhanced Definition: 625 lines with 16:9 aspect ratio) and STDV (Standard Definition: equivalent to PAL or SECAM), as well as the evolution to HDTV.

There are three media that must be considered in connection with the introduction of digital TV: satellite, cable and terrestrial.

It is thought that satellite will offer the first opportunity for the introduction of digital multi-programme television services in the near future. However, although primarily focused on direct-to-home reception, satellite broadcasting also requires signal distribution via large cable networks and community receiving installations serving single buildings.

Terrestrial digital TV broadcasting has also to be considered, although it represents a more difficult transmission medium.

This report is intended to give general information on the different aspects of channel coding and modulation. References are listed at the end of each chapter.

*Chapter 1* gives the main characteristics of the various transmission media and outlines the major differences concerning digital TV broadcasting.

*Chapter 2* describes the types of modulation suitable for satellite broadcasting. After an overview of satellite characteristics, the constraints applicable to digital modulation are discussed. It is shown that the best modulation choice is a multi-level phase modulation (M-PSK) in conjunction with a powerful code for error control. Two types of demodulation are described: QPSK associated with a convolutional code and Trellis-coded 8 PSK.

A possible solution for the short-term introduction of satellite multi-programme digital television is presented: it is based on QSPK modulation associated with an inner convolutional code of rate 3/4 with Viterbi decoding. The best outer code is the (255,239) Reed-Solomon code or a shortened version to cope with the packet length defined in the MPEG-2 multiplex. A bit-rate of about 34 Mbit/s could be adopted This would enable 3 EDTV services in 16:9 or 5 SDTV services compatible with small-antenna receivers (50cm or less) to be broadcast.

*Chapter 3* discusses the introduction of digital television in cable networks. The constraints imposed by cable networks are presented. Three types of modulation suitable for cable distribution are described and compared: QAM, VSB-AM and COFDM. It is shown that single-carrier VSB-AM and COFDM satisfy the requirements of the cable networks.

However, it seems that 40 Mbit/s is the maximum bit-rate transmissible in a 8 MHz cable network channel, simultaneously with the existing analogue signals. A good compromise between satellite and cable could be found at around the level of 34MBit/s.

*Chapter 4* presents the constraints imposed by terrestrial TV broadcasting on the channel coding and modulation schemes suitable for that transmission media. The emphasis is put on the constraints of spectrum management below 1 GHz, where the RF channels usable for future digital TV broadcasting are almost all occupied by existing analogue TV signals. To these constraints of sharing the spectrum, must be added the difficulties inherent to terrestrial transmission: echoes, shadowing and interference.

Today's analogue systems have various kinds of limitations:

- intrinsic quality of the received signal (noise, ghosts);
- inadequate services (no possible extension to HDTV on the one hand, nor to plug-free SDTV on the other hand);
- high cost of the broadcasting infrastructure and poor utilisation of the spectrum resource.

It is then shown that the problems with regards to echoes (ghosts) and spectrum management can be considerably reduced by the use of Single Frequency Networks, which enable a given service to be broadcast on the same RF channel throughout a wide area, while dramatically saving frequency resources. In the opinion of the authors, the modulation scheme which is most efficient in terms of complexity and performance is the COFDM scheme (Coded Orthogonal Frequency Division Multiplex). The report presents in details the characteristics and performances of the COFDM scheme, as a function of the desired transmission bit-rates and reception ruggedness. An overview of other types of modulation is also given.

The conclusion of chapter 4 lists the possible options of network design as a function of the scale of coverage area.

*Chapter 5* deals with the prospects for inter-operability among all broadcasting delivery media. It is shown that the adoption of a unique modulation scheme optimised for the three transmission media: satellite, cable and terrestrial is practically impossible. In fact, the satellite channel is basically non-linear and power-limited, but does not suffer from stringent bandwidth limitations; terrestrial and cable channels are linear and allow relatively high S/N ratios, but are band-limited and currently affected by echoes (in terrestrial reception) and other distortions. However, a common modulation scheme could probably be adopted for terrestrial broadcasting channels and for cable networks.

The commonality in the user receiver for all delivery media should be found at the level of the multiplex.

In any case, technical constraints do not contribute alone to this question of inter-operability.

The conclusion given by Mr. Pommier at the Montreux '93 TV Symposium[1] gives a wider view on this question:

> "... The repetition of the scenario of the MAC systems can be perhaps avoided. To achieve that, it is of course necessary to reduce the importance and the number of factors which are divisive. Amongst the previously-described factors, most can be reduced greatly by the judicious choice of technique and technology. Others depend only on the will of people to implement open and standardised systems instead of closed systems, the development and the use of which they are able to control. Finally, if only the differences related to the physical characteristics of the broadcasting media should remain, the technicians would succeed without any doubt in proposing generic solutions covering most parts of the chain. The experience acquired from previous attempts to unify broadcasting techniques, which

1 D. Pommier: "Digital - the synergistic link between terrestrial and satellite broadcasting", Montreux International Televison Symposium, 1993.

has at times been disappointing, might help everybody better to understand each other's wishes. In this sense, the effort of concentration which is being developed at European level should continue with the aim that a coherent introduction policy for digital television is put forward. In that process, the service providers will, without any doubts, have the final word but they cannot act alone".

## 1. MAIN CHARACTERISTICS OF THE VARIOUS TRANSMISSION MEDIA

This introduction outlines the major differences between the transmission media for digital TV broadcasting. It is extracted from the paper: "Digital: the synergistic link between terrestrial and satellite broadcasting" by D. Pommier of the CCETT, presented at the 18th International Television Symposium of Montreux, June 1993.

The main differences between cable, satellite and terrestrial transmission channels lie in:

- the noise and interfering signal levels,
- the channel bandwidth,
- the presence or the absence of significant multipath signals.

These differences are roughly summed up in the table below:

| | Satellite | Cable | Terrestrial |
|---|---|---|---|
| Bandwith | 26 to 36 MHz | 7 to 8 MHz | 7 to 8 MHz |
| Carrier to noise and interfering signals | $10 \le \frac{C}{N+1} \le 20dB$ [1] | $25 \le \frac{C}{N+1} \le 30dB$ [2] | $8 \le \frac{C}{N+1} \le 30dB$ [3] |
| Multipath signals | no | some limited in delay and level | numerous[4] and large in delay and level |

*Table 1.1*

*Notes:* *1.* Depends on the size of antenna and the atmospheric attenuation
*2.* Will depend on the injection level of the digital signals in the cable media
*3.* Depends on the network topology, i.e. single frequency network or conventional network and the reception mode (mobile, portable, fixed).
*4.* In the case of digital terrestrial broadcasting, multipath signals may be not only the result of the reflections on various obstacles, but may also be due to the use of several transmitters or gap-fillers transmitting the same signal at the same frequency in the same area.

The details of the characteristics of the satellite, cable and terrestrial channels are given in Sections 2, 3 and 4.

Those differences impose inter-operability constraints between the different transmission media, which are discussed in Section 5.

## 2. MODULATION FOR SATELLITE BROADCASTING

### 2.1. Frequency bands for digital TV introduction

The Broadcasting Satellite Service (BSS) frequency band, between 11.7 and 12.5 GHz, has been allocated by the WARC-77 for Direct-to-the-Home (DTH) television with high power satellites operating with an EIRP greater than 60 dBW. Nevertheless, these expensive high-power satellites have shown low reliability, and the number of available channels (5 per country) seems too low to be economically viable. In fact, DTH services have been introduced mainly in the Fixed Satellite Service (FSS) band which was originally intended for professional point-to-point or point-to-multipoint applications. In the FSS band, the satellite power is restricted to around 52 dBW by the Radio Regulations to minimise the probability of interference into terrestrial microwave links (the satellite power can be up to 12 dB higher in the BSS band). Nevertheless, DTH services in the FSS band, no longer penalised by this relatively low EIRP (owing to the drastically improved receiver sensitivity), have succeeded in attracting encouraging audiences because of the large number of channels available from a single satellite and because of the extensive, often Europe-wide, coverage.

So it is thought that digital TV will also be introduced in the FSS bands, at least at the beginning.

It is becoming increasingly evident that the WARC 77 plan requires a general revision to take into consideration the recent advances in receiver performance and the need to improve the planning flexibility so as to allow supra-national (linguistic area) coverages in Europe.

More recently, WARC-92 allocated the 21.4 - 22.0 GHz frequency band for digital HDTV satellite broadcasting. Research work is underway within the framework of the European RACE project HDSAT to develop for long-term application what is called a "digital studio quality HDTV system".

Table 2.1 gives the main characteristics of the bands in which digital TV will be introduced.

| Frequency band | Channel bandwidth | EIRP in the coverage area | Examples of satellites | Comments |
|---|---|---|---|---|
| 10.9-11.7 GHz | 26 MHz<br>36 MHz | 51 dBW<br>50 dBW | ASTRA<br>Eutelsat | |
| 11.7-12.5 GHz (BSS) | 27 MHz | > 60 dBW | TDF 1/2<br>TVSAT 2 | Replanning |
| 12.5-12.75 GHz | 36 MHz | 52 dBW | Télécom 2 | |
| 21.4-22 GHz (BSS) | | | | Studied by the RACE project HDSAT |

*Table 2.1*

## 2.2. Channel characteristics and consequences on the modulation choice

The main characteristics of a satellite channel are as follows:

- Non-linear transmission characteristics due to the travelling-wave-tube amplifier (AM/AM and AM/PM conversion).
- Propagation attenuation very great, more than 200 dB.
- Additional atmospheric loss, due to rainfall. So the service availability cannot be guaranteed for 100% of the time.

So, the modulation must comply with the following constraints:

- Constant or quasi-constant envelope modulation. Indeed, the TWT must be operated at or near saturation in order to maximise the transmitted power. This excludes amplitude modulation.
- Ruggedness against noise and interference, in order to minimise the receiver antenna sizes.
- Good spectrum efficiency.
- Simplicity of the receiver.

The modulation selected must offer a compromise between these requirements. Antenna size is a key factor for DTH services and must be kept at a reasonable level (60 cm is an acceptable target).

The interference problem can be critical for digital TV. Interference is mainly due to emissions from adjacent channels of the same satellite. Small receiving antennas may also pick up interference from adjacent satellites, especially in the FSS band where the orbital spacing is only 3 degrees on the geostationary orbit.

For a digital signal, interference has the same effect as noise in reducing the bit-error rate (BER). For the link budget calculation, it is possible to add the interferer power and the noise power.

Possible types of modulation which comply with the above constraints are phase or frequency modulation. Some years ago, Continuous Phase Modulation schemes (CPM) were widely studied, especially MSK (Minimum Shift Keying) or TFM (Tamed Frequency Modulation) *[1]*. Nevertheless, these modulations are no longer considered for digital satellite broadcasting. Despite the advantage from the constant envelope characteristic of CPM signals, they cannot easily achieve a bandwidth efficiency better than 1 bit/s/Hz. In fact, when looking for bandwidth and energy-efficient schemes requiring relatively simple demodulator structures, it is natural to think of M-PSK in conjunction with a powerful code for error control.

At the time being, it seems that the best choices for digital satellite broadcasting are as follows *[2, 3]*:

- QPSK associated with a convolutional code
- 8PSK associated with a trellis-code (trellis coded modulation: TCM).

These schemes offer a wide range of spectrum and power efficiencies. The transmitted signal bandwidth (null-to-null) is given by the equation (for a linear channel):

$$B_w = \frac{(1+\alpha)D}{r \log_2 M}$$

where $\alpha$ is a roll-off factor of the Nyquist filters, M is the number of modulation levels (typically 4 or 8), r is the coding ratio of the convolutional or trellis code, D is the useful bit rate.

Inversely, when the bandwidth is fixed, the bit rate D which can be transmitted in $B_W$ is given by the equation:

$$D = \frac{r \log_2 M}{1+\alpha} B_w$$

Table 2.2 gives the possible bit-rates applicable to a 36 MHz satellite transponder ($B_W = 36$ MHz).

| Modulation | M | Coding ratio r | Roll-off factor $\alpha$ | Bit-rate D |
|---|---|---|---|---|
| QPSK | 4 | 3/4 | 0.5 | 36 Mbit/s |
| | | | 0.2 | 45 Mbits |
| 8-PSK | 8 | 2/3 | 0.5 | 48 Mbits |
| | | | 0.2 | 60 Mbits |

*Table 2.2*

## 2.3. QPSK modulation associated with a convolutional code

### 2.3.1. Channel coding and modulation

Convolutional encoding and Viterbi decoding are used to improve the bit error rate performance of the QPSK modulation. Encoding consists in introducing a correlation between the information bits. The Viterbi decoder performs a maximum likelihood estimation over the received data on the basis of quantized samples of the demodulated signal (soft decision) */4/*.

Since the early seventies, the convolutional code of constraint length K = 7 and rate 1/2, optimum in the sense of maximum free distance and minimum number of bit errors caused by remerging paths at the free distance, has become the *de facto* standard for coded digital communication.

A technique known as "puncturing" permits the use of a modified version of the basic rate 1/2 code for any rate (n - 1) / n that is only moderately less efficient than the optimum code for that rate but has a better spectrum efficiency */5/*. This technique deletes a fraction of the symbols generated by the rate 1/2 code, but utilizes the same decoder as the latter, with these deleted symbols replaced by erasures.

Very recently, a new class of convolutional codes called "turbo codes" was found by Claude Berrou (Ecole Nationale Supérieure des Télécommunications de Bretagne). Their performance in terms of BER is close to the Shannon Limit *[6]*.

The "turbo code" encoder is built from a parallel concatenation of two recursive systematic convolutional codes and the associated decoder is made of P identical pipeline decoders using a feedback decoding rule. The performance improves dramatically with the number of iterations. A good performance / complexity trade-off was found to be 3 iterations. It is worth mentioning that each iteration involves exactly the same operations: N iterations can be obtained by cascading N identical chips or by including N identical structures in one chip. Thus each receiver can have its own performance / complexity trade-off: this versatility is obviously an interesting feature.

The typical code rate is 1/2 but this code can also be punctured to provide code rates 2/3 or 3/4.

### 2.3.2. BER versus Eb/No

Figure 2.1 gives the BER versus Eb/No curves for the three following modulation schemes:

— uncoded QPSK (curve 1)

— QPSK with rate 1/2 convolutional code (curve 2)

— QPSK with rate 3/4 punctured convolutional code (curve 3).

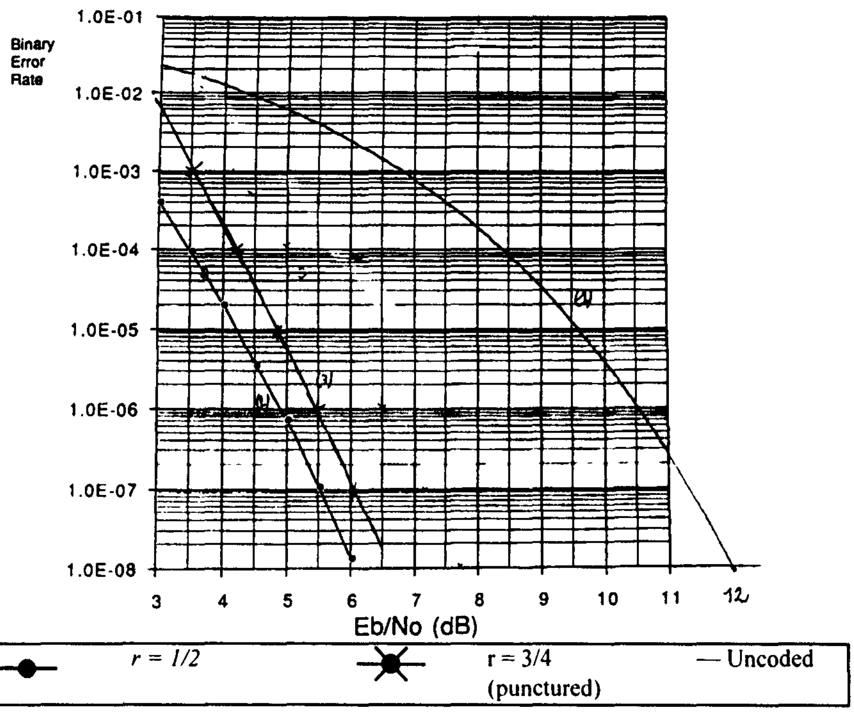



*Figure 2.1: BER versus E/No for QPSK modulation associated with a convolutional code.*

It can be seen that the 1/2 rate code offers a gain of 5 dB at BER = $10^{-4}$. The 3/4 rate punctured code has a gain of 4.2 dB. This code is very attractive: it represents a good compromise between noise performance and spectrum efficiency.

Figure 2.2 gives the results obtained with a "turbo-code" of rate 1/2 after n = 1, 3 and 5 iterations.

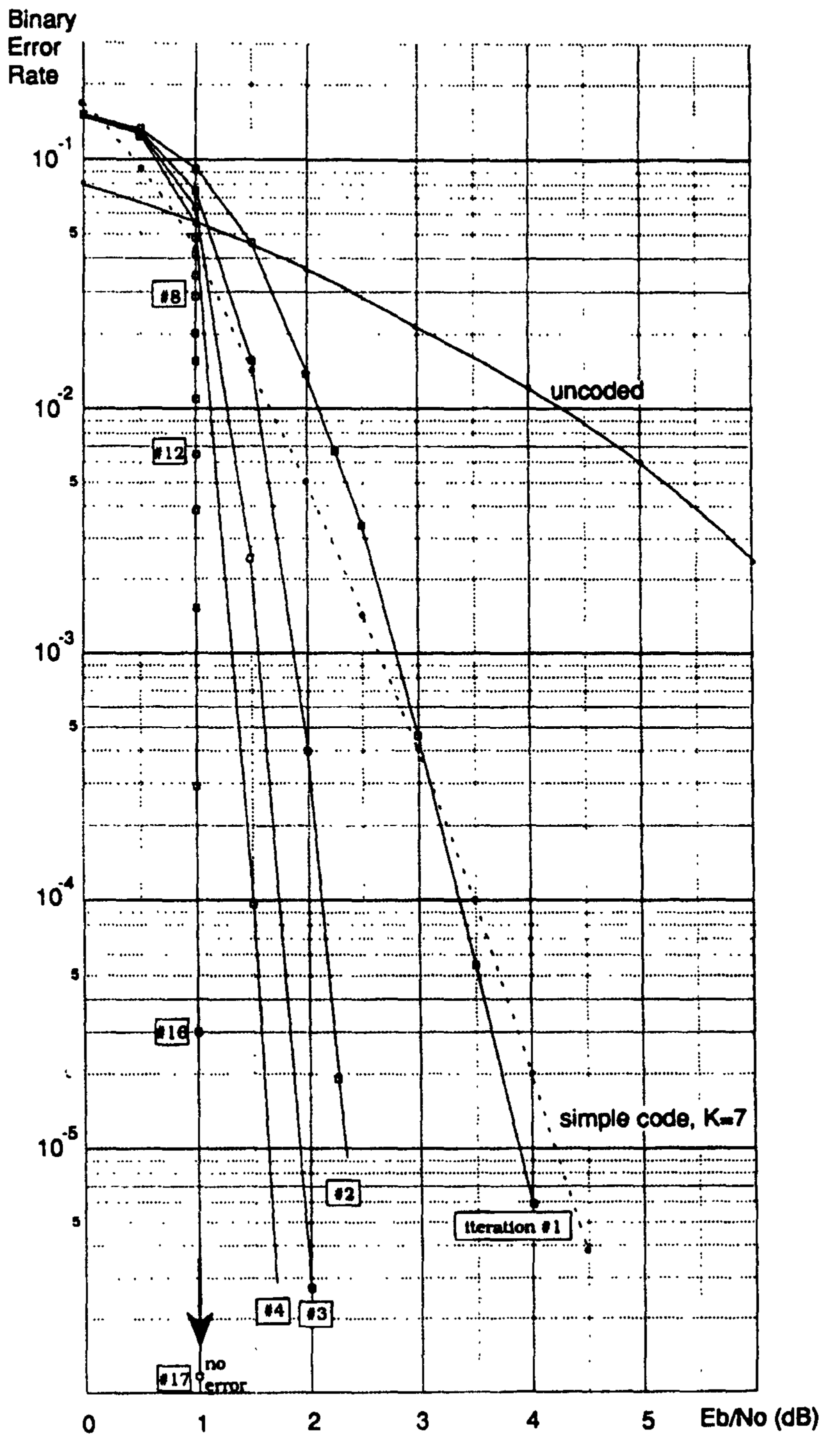



Figure 2.2: *BER versus E/No for QPSK modulation associated with a rate 1/2 "turbo-code" as a function of the number of iterations.*

Figure 2.3 shows the BER versus E/No for a "turbo-code" of rate 3/4 after 3 iterations.

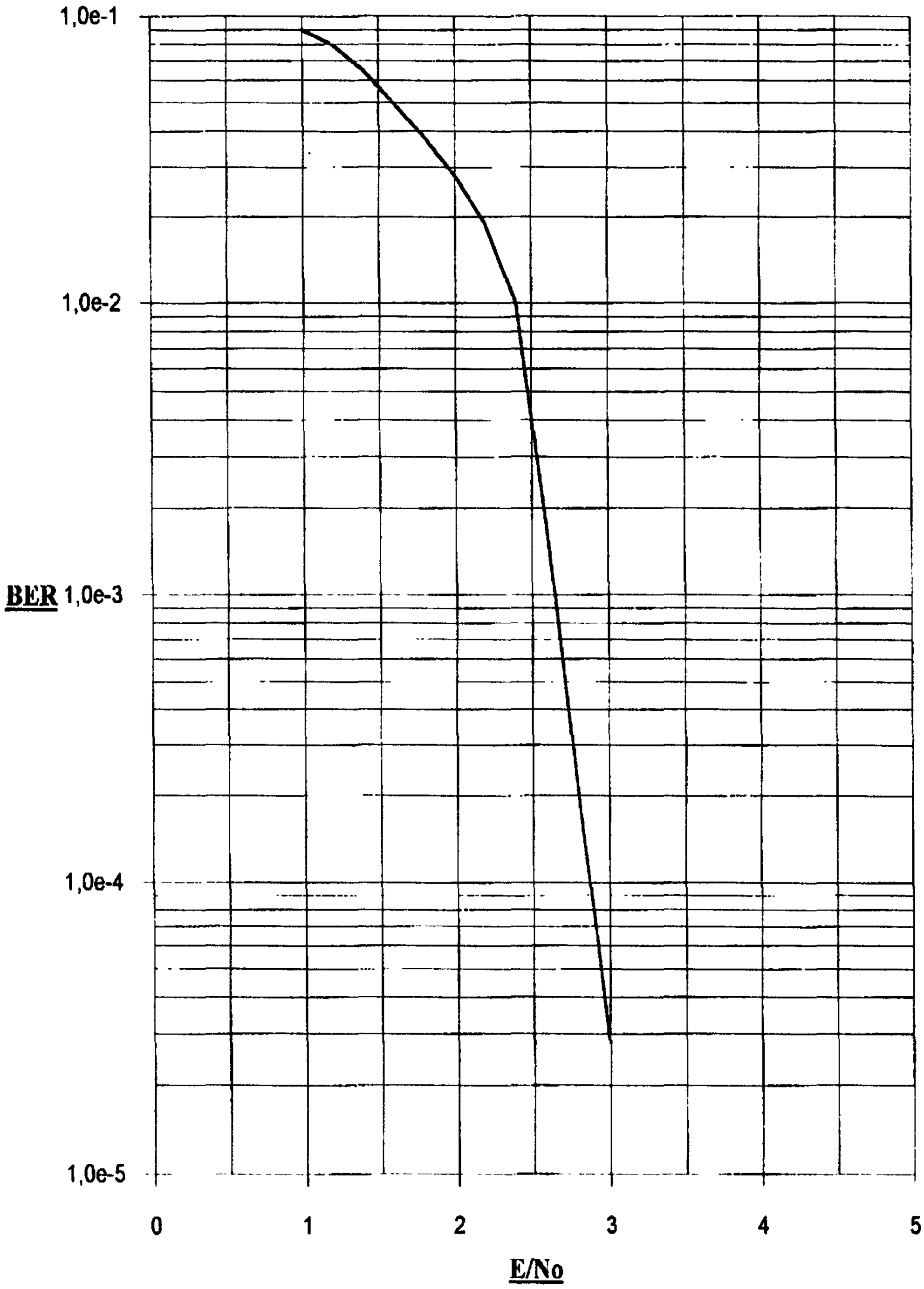


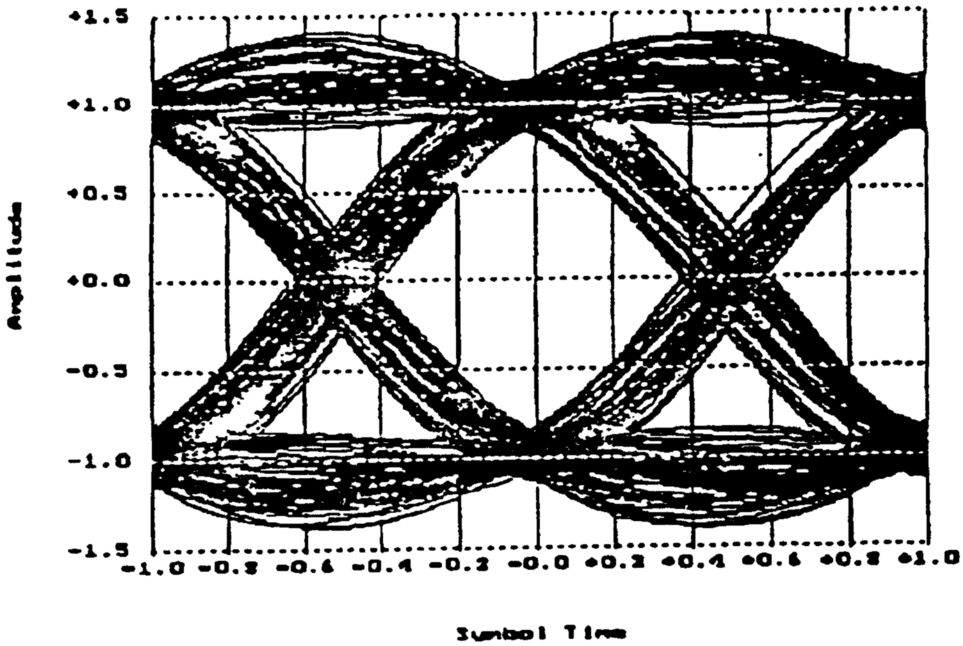

Figure 2.3: *BER versus E/No for QPSK modulation associated with a rate 3/4 "turbo-code" (3 iterations).*

Compared to the best convolutional 64-state binary code, the gain of the "turbo-code" is very impressive.

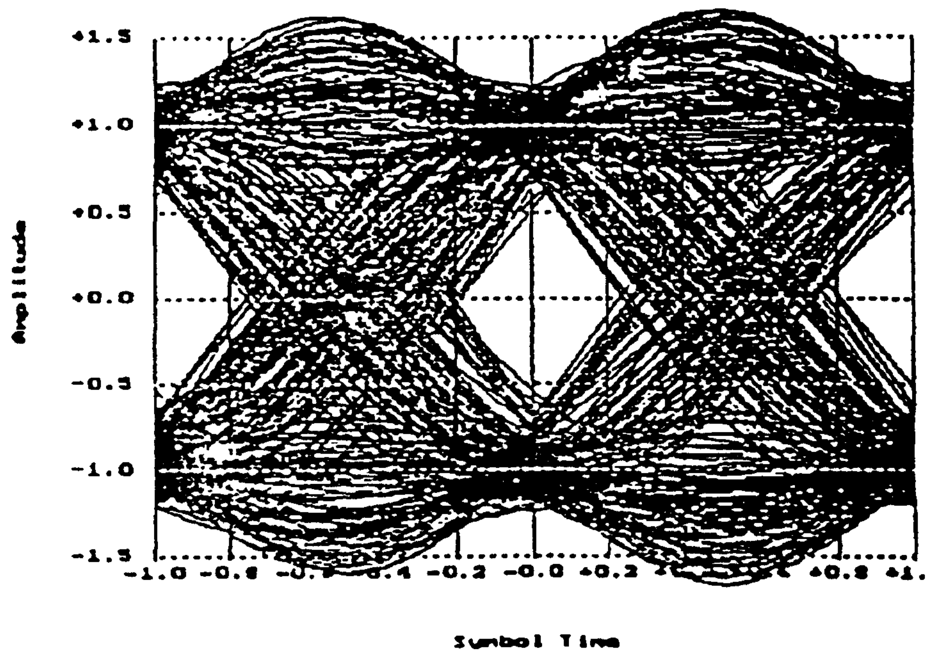
### 2.3.3. Selection of the Nyquist filter roll-off

The impairment due to the TWT's non-linearities depends on the Nyquist filter roll-off. When the roll-off decreases, the amplitude modulation between phase states increases and also the distortions due to AM/AM and AM/PM conversion in the TWT.

Figure 2.4 shows the eye diagrams obtained at the demodulator output for two roll-offs, in the case of a linear channel and that of a satellite channel whose TWT is operated at saturation.




*Eye diagram for $\alpha = 0.67$ rate including satellite transponder nonlinearities and filtering*




*Eye diagram for $\alpha = 0.25$ rate including satellite transponder nonlinearities and filtering*

*Figure 2.4: Eye diagrams for QPSK demodulation in a satellite channel.*

Computer simulations have been done in CCETT with the following hypothesis:

- Modulation: QPSK associated with a rate 3/4 convolutional code.
- TWT operated at saturation.
- Nyquist filtering: it was assumed that a raised cosine filter with a variable roll-off factor, equally split between modulator and demodulator, is used.
- The IMUX filter is an 8 pole constant group delay Butterworth filter.
- The Viterbi decoder has a truncation length of 80 bits.

The BER was estimated by the Monte-Carlo estimation procedure.

Table 2.3 gives the impairment D E/No for BER = $10^{-4}$ obtained with roll-off 0.5 and 0.2. D E/No is defined as the difference in E/No for BER = $10^{-4}$ between a linear and a satellite channel.

| Roll-off | Δ E/No for uncoded QPSK | Δ E/No for rate 3/4 coded QPSK |
|---|---|---|
| 0.5 | 0.7 dB | 0.4 dB |
| 0.2 | 3 dB | 1 db |

*Table 2.3*

It can be seen that the BER suffers degradation when the roll-off factor decreases from 0.5 to 0.2 but this degradation is considerably reduced with coded modulation. So it seems possible to use a roll-off factor of only 0.2 in conjunction with a rate 3/4 coded QPSK modulation. In this case, the spectrum efficiency of the modulation is 1.25 bit/s/Hz.

### 2.3.5. Channel equalisation

In a satellite transmission chain, some impairments are caused by amplitude or group delay variations in the filters (filters at the satellite input and output, and the channel selection filter in the receiver) and also by the short echoes due to a possible mismatching between the LNB and the indoor unit. These impairments can be considerably reduced if an equaliser is used in the receiver. Equalisation can also be used to reduce the effects of non-linearities induced by the TWT.

A state of the art of the equalisation techniques can be found in Deliverable N° 3 of the HDSAT project. It was shown that equalisation can also be combined with clock recovery *[7]*.

### 2.3.6. Relation between C/N and E/No

Generally, the performance of a satellite transmission system is given as a function of the C/N ratio defined as follows: C is the mean power of the digital transmitted signal and N is the noise power measured in the IF receiver bandwidth $B_W$.

The relation between C/N and E/No is given by the formula (in dB):

$$(C/N) in B_w = E/No + 10 log(\frac{D}{B_w})$$

where D is the useful transmitted bit rate and $B_W$ is the IF filter bandwidth.

### 2.3.7. Examples of theoretical performance

Table 2.4 gives, for a 36 MHz transponder bandwidth ($B_W$), the theoretical C/N required to obtain a BER of 2. $10^{-3}$ and $10^{-4}$ as a function of the useful bit rate. The considered modulation scheme is QPSK associated with a 3/4 rate convolutional or turbo-code (3 iterations).

| Useful bit rate D(Mbit/s) | 27 | 40 | 45 |
|---|---|---|---|
| C/N in 36 MHz for BER = $10^{-4}$ convolutional r = 3/4 code | 3.2 dB | 5.0 dB | 5.5 dB |
| C/N in 36 MHz for BER = $2.10^{-3}$ convolutional r: 3/4 code | 1.9 dB | 3.7 dB | 4.2 dB |
| C/N in 36 MHz for BER = $10^{-4}$ turbo-coder r = 3/4 | 1.6 dB | 3.3 dB | 3.9 dB |
| C/N in 36 MHz for BER = $2.10^{-3}$ turbo-coder r = 3/4 | 1.3 dB | 3.1 dB | 3.6 dB |

*Table 2.4*

## 2.4. Trellis coded modulations

### 2.4.1. Principles

With the growth of digital satellite communications, transponder power grew but information rate requirements grew even faster, so that bandwidth limitations began to be felt. With QPSK modulation, even with code rates approaching 1, bandwidth efficiency is limited to less than 2 bit/s/Hz.

To attain bandwidth efficiencies in excess of 2 bit/s/Hz, it is necessary to couple coding with symbols of a higher level than the four phase levels provided by QPSK. Thus if quasi-constant envelope modulations are required (as for operation over a satellite channel), the choice must be Multiple Phase Shift Keying (MPSK) with 8, 16 or even more levels.

Optimum trellis codes for a given complexity (constraint length or number of states) were found by Ungerboeck *[8]*, who provided the impetus for coding of bandwidth-limited channels though a technique called "set partitioning". While he also showed that all such codes can be implemented using a binary convolutional encoder coupled with appropriate mapping of the binary code symbols on to the signal constellation, the resulting best codes for a given complexity (constraint length) were quite different from the classical binary convolutional codes used with QPSK.

More recently, Viterbi *[9]* showed that it was possible to use the K = 7, rate 1/2 convolutional code with signal constellations of 8-PSK and 16-PSK while providing power efficiency that in most cases is virtually equivalent to that of the best Ungerboeck codes for constraint length 7 or 64 states. This approach, called "pragmatic", permits the use of a single basic coder and decoder to achieve respectable coding gains for bandwidth efficiencies greater than 2 bit/s/Hz.

### 2.4.2. BER versus E/No

Figure 2.5 compares the best 64-state codes as found by Ungerboeck for 8-PSK and 16-PSK with those generated by the Viterbi pragmatic coder. The conclusion is that, for almost all purposes, the two sets of performance are equivalent, but the implementation of the latter is far simpler and more flexible.

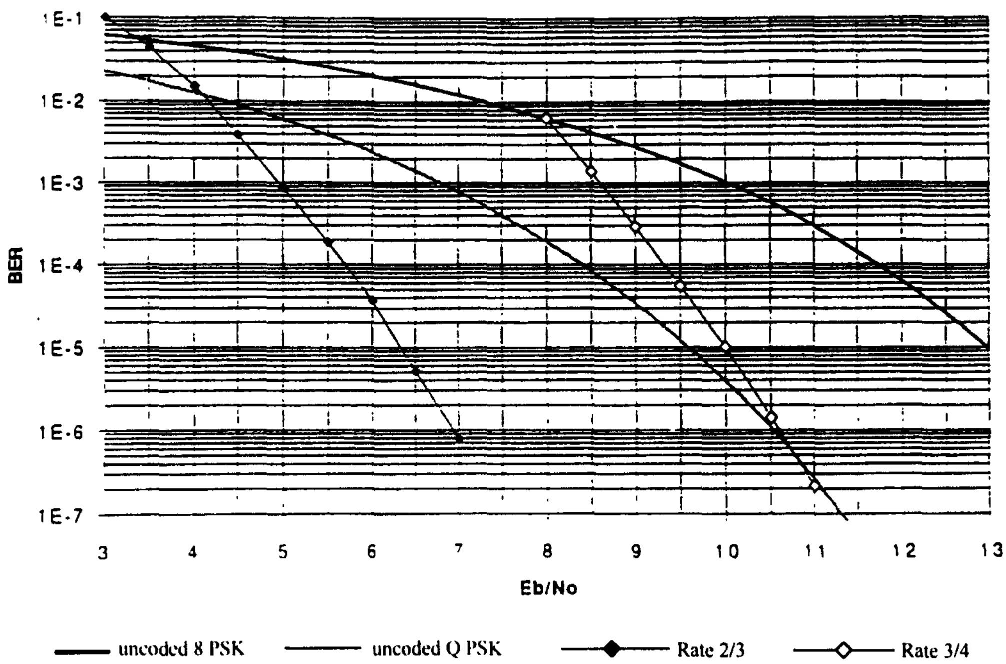



Figure 2.5: *BER versus E/No for 8-PSK modulation associated with a 2/3 rate trellis coder.*

### 2.4.3. Examples of theoretical performances

Table 2.5 gives for a 36 MHz transponder bandwidth the theoretical C/N required to obtain a BER of $2.10^{-3}$ and $10^{-4}$ as a function of the useful bit-rate.

The considered modulation scheme is 8-PSK associated with a pragmatic trellis code.

| Useful bit rate D (Mbit/s/s) | 48 | 60 |
|---|---|---|
| C/N in 36 MHz for BER = $10^{-4}$ | 6.9 dB | 7.9 dB |
| C/N in 36 MHz for BER = $2.10^{-3}$ | 5.9 dB | 6.9 dB |

*Table 2.5*

## 2.5. COFDM modulation

COFDM (Coherent Orthogonal Frequency Division Multiplex) is a multi-carrier modulation method *[10]* particularly suitable for terrestrial broadcasting, because of its inherent ruggedness against linear distortions originated by multipath propagation. This modulation has been adopted for digital audio broadcasting (DAB project). A full description of this modulation is given in section 4.

However, the COFDM signal, independently of the modulation adopted on each carrier (QPSK, 8PSK, etc...) presents a variable envelope distribution, of Rayleigh type, which has the following consequences when the signal is transmitted in a non-linear channel (satellite):

- spectrum spreading due to the inter-modulation between the carriers
- distortions on the demodulated signal.

So the TWT cannot be operated at saturation. An output back-off of at least 3 dB is required to obtain acceptable performance. Studies are currently underway within the HDSAT project in order to reduce the effects of non-linearities.

Nevertheless, in the light of these investigations, it can be concluded that, particularly on low/medium power satellites, the COFDM approach does not seem to allow satisfactory solutions.

## 2.6. Relation between BER and picture quality

### 2.6.1. Transmission system modelling

The transmission system considered is shown in figure 2.6. It consists of the following elements:

- picture and sound coders
- service and programme multiplexer
- outer error correcting code (e.g. Reed-Solomon Code)
- interleaving process
- inner error correcting code associated with the modulator
- RF modulator (QPSK or 8-PSK)
- satellite channel
- RF demodulator
- Viterbi decoder
- interleaving process
- outer error correcting decoder
- demultiplexer
- picture and sound decoders.

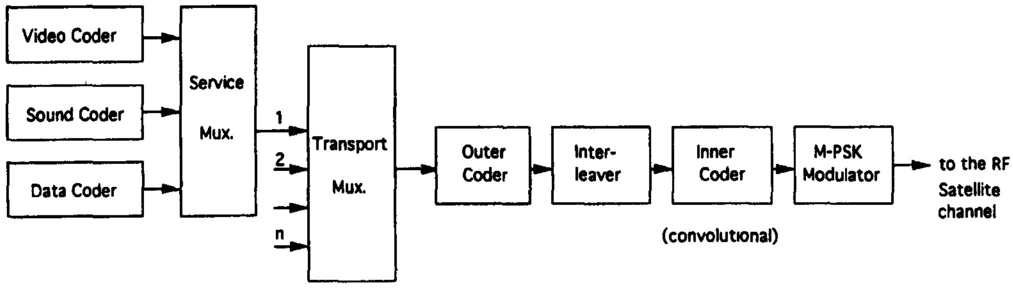



*Figure 2.6: Satellite transmission system model (transmission side)*

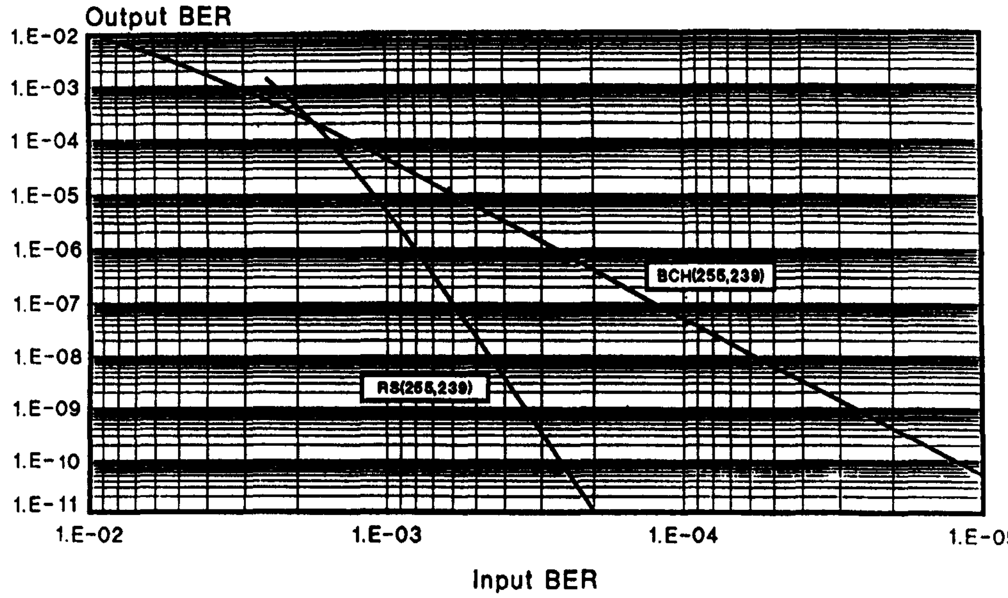
In this transmission model, error protection is obtained with two codes combined together to achieve the performance level of a long code without the corresponding increase in hardware complexity. The choice of the so-called inner code (channel code) and the so-called outer code (baseband code) has to be done with care. Typically an inner convolutional code (or a "turbo-code") is combined with an outer Reed-Solomon code. The Viterbi decoder (for the inner code) deals with the channel noise. The Reed-Solomon code corrects the remaining burst errors. It can be associated with the demodulator in order to protect the total data stream entering the demultiplexer or after the demultiplexer, at the level of each individual service component.

It is very important that the outer code is sufficiently interleaved in order not to exceed the burst-error capabilities of the Reed-Solomon code.

The video compression systems based on the use of Hybrid DCT, motion compensation and entropy coding are very sensitive to transmission errors. A BER of about $10^{-11}$ is generally required to obtain a subjective quality virtually transparent to the studio standard. The service failure, determined by the loss of synchronization, corresponds to a BER of about $4.10^{-4}$.

We can then define, for a QPSK modulation associated with a 3/4 rate convolutional code (inner code) and with a (255, 239) Reed-Solomon Code (outer code) the values of E/No corresponding to the limit of perfect picture quality and to service failure.

Figure 2.7 gives the relation between the BER at the input and at the output of a (255, 239) Reed Solomon decoder.




- BCH(255,239) and RS(255,239) codes performance

*Figure 2.7: Relation between input and output BER of a (255/239) Reed-Solomon decoder*

### 2.6.2. Relation between E/No and picture quality

a) E/No corresponding to the limit of perfect picture quality

BER after Reed-Solomon decoding : $10^{-11}$

BER after Viterbi decoding : $2.10^{-4}$

Corresponding E/No ratio : 4 dB

b) E/No corresponding to the service failure:

BER after Reed-Solomon decoding : $4.10^{-4}$

BER after Viterbi decoding : $2.10^{-3}$

Corresponding E/No ratio : 3.2 dB

It can be seen that for an E/No decrease of only 1 dB, the picture quality decreases from excellent to unusable. Digital techniques exhibit abrupt breakdown failure characteristics. This is very different from analogue systems which have generally a slow picture degradation when the C/N decreases. In order to illustrate this characteristic, figure 2.8 shows the relation between picture quality and C/N for analogue and digital systems.

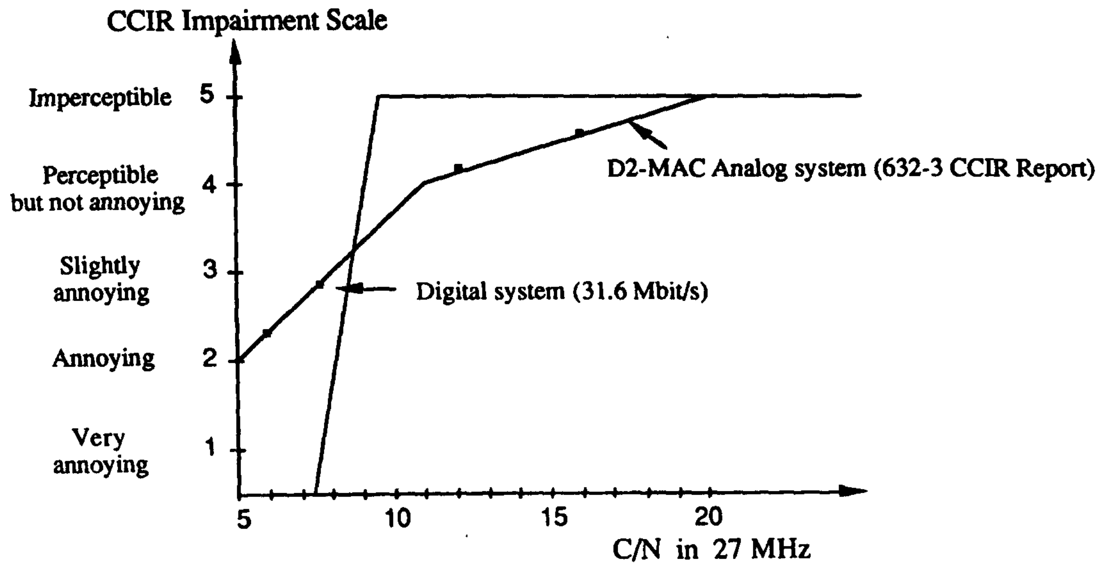



*Figure 2.8: Relation between C/N and picture quality for analogue and digital system*

It is supposed that the channel bandwidth is 27 MHz for both systems. The main characteristics of the two systems are as follows:

a) analogue system (D2-MAC)

modulation: frequency modulation

frequency deviation: 13.5 MHz/V.

b) digital system

- modulation: QPSK
- Nyquist roll-off: 0.2
- inner code:(255, 239) Reed-Solomon code
- margin: 3 dB (including TWT non-linearities, interferers, implementation margin)
- useful bit rate: 31.6 Mbit/s (this bit rate allows to broadcast in a 27 MHz channel 3 EDTV services with a quality similar to MAC).

It can be seen that the digital system guarantees a good picture quality at a C/N significantly lower than the analogue system. However, if we compare the C/N corresponding to service failure, the advantage lies with the analogue system. It can be concluded that, for a digital system, service availability is the key factor. It is difficult to accept performance inferior to an analogue system.

## 2.7. Service availability in relation with atmospheric attenuation

### 2.7.1. Service availability and quality objective

Two quality factors are considered necessary for digital satellite broadcasting:

- good quality must normally be available
- outage times should be severely limited.

Depending upon the coding and modulation scheme quality is related to the received carrier-to-noise ratio. Rain is considered to be the most important source of attenuation on the space-earth links. Attenuation caused by rain is heavily dependent on the frequency band and the intensity of precipitation at the specific location.

The effect of rain is to decrease of the received signal power and to increase the receiver noise temperature. Both these phenomena combine to reduce the C/N ratio.

### 2.7.2. Atmospheric attenuation at 12 GHz

At 12 GHz, the figures in table 2.6 may be given as estimates for attenuation caused by rain for an average temperate climate (Europe) as a function of the percentage of the worst month. The corresponding C/N decrease is also given for a receiver with a noise figure F = 1.2 dB, which is currently achieved by consumer receivers.

| Percentage of worst month | 99% | 99.9% |
|---|---|---|
| Atmospheric attenuation (dB) | 1.2 dB | 4 dB |
| Receiver noise temperature increase | 1.5 dB | 3 dB |
| C/N decrease (compared with clear sky reception) | 2.7 dB | 7 dB |

*Table 2.6*

### 2.7.3. Atmospheric attenuation at 20 GHz

Table 2.7 gives the results obtained in Paris with a receiver having a noise figure F = 1.7 dB. It seems that this value could be obtained with existing mass-production techniques *[11]*.

| | | |
|---|---|---|
| Percentage of worst month | 99% | 99.9% |
| Atmospheric attenuation (dB) | 5.2 dB | 11.2 dB |
| Receiver noise temperature increase | 1.9 dB | 2.5 dB |
| C/N decrease (compared with clear sky reception) | 7.1 dB | 13.7 dB |

*Table 2.7*

### 2.7.4. Consequences for the service availability

The ideal service continuity target (outage time) for satellite digital TV systems would be 99.9% of the worst month (corresponding to about 40 minutes of outage time). This is currently achievable by conventional FM/TV systems. It seems that this target can be obtained at 11-12 GHz with relatively small receiving antennas. An example of a link budget at 12 GHz is given in the next section.

However, for the 20 GHz frequency band, conventional digital techniques exhibit abrupt breakdown failure characteristics and will not be able to provide the required service availability without a penalty in respect of the satellite transmitted power. It is necessary to use techniques which achieve a graceful degradation of HDTV services during high fades. A possible approach is described in reference *[12]*. This approach uses a concept of layered modulation in conjunction with layered picture coding and channel coding. By means of this technique, the service continuity can be extended without the need to increase the satellite transmitted power, the service quality under severe atmospheric attenuation being reduced from high definition to standard definition.

## 2.8. Examples of relations between useful bit rate and antenna size for digital satellite broadcasting with Telecom 2 and Europesat

Telecom-2 satellite is an example of an FSS satellite currently used for DTH services in France.

Europesat is a proposal from Eutelsat for introducing digital TV in the BSS band. Its characteristics have been determined on the assumption that the WARC-77- Plan will be replaced by a new plan based on the following hypothesis: 33 MHz channel bandwidth and allocation of the 40 channels at the same orbital position with supra-national coverage.

### 2.8.1. Hypothesis

*Service availability*

For digital satellite broadcasting, the antenna size is determined by service continuity. The following hypothesis has been taken into account: service continuity is assumed for 99.9% of the worst month (this corresponds to a service interruption of 90 seconds per day). The corresponding BER is $2.10^{-3}$.

*Margin in comparison with theoretical results.*

A 3 dB margin has been taken into account. It corresponds to the following impairments:

- satellite non-linearities: 1 dB
- interferers (co-channel and adjacent channels): 0.5 dB
- demodulator implementation: 1.5 dB.

*Modulation schemes*

The following modulations are considered:

- QPSK associated with a rate 1/2 or 3/4 convolutional code (or turbo code)
- 8 PSK associated with a pragmatic rate 2/3 trellis code.

*Parameters of the receiving station*

- noise figure: 1.2 dB
- antenna efficiency: 0.70
- pointing error: 0.5 dB.

*Link budget*

For link budget calculations, the following hypothesis has been adopted:

- atmospheric loss for 99.9% of the worst month: 4 dB
- corresponding C/N decrease for F = 1.2 dB: 7 dB.

*Satellite characteristics*

Telecom 2

Transponder bandwidth: 36 MHz

EIRP (primary coverage area): 52 dBW

Up-link C/N: 26 dB

Europesat

Channel bandwidth: 33 MHz

EIRP (primary coverage area): 56 dBW

Up-link C/N: 30 dB

### 2.8.2. Antenna sizes for DTH reception

Telecom-2 (Modulation: QPSK)

| Useful transmitted bit-rate (Mbit/s) (before channel coding) | 27 | 34 | 45 | 45 |
|---|---|---|---|---|
| Channel coding | Convolutional K = 7 | Convolutional punctured K = 7 | Convolutional punctured K = 7 | Turbo-code (3 iterations) |
| Coding rate | 1/2 | 3/4 | 3/4 | 3/4 |
| E/No for BER = $2.10^{-3}$ | 2.3 | 3.2 | 3.2 | 2.6 |
| (C/N) in 36 MHz for BER = $2.10^{-3}$ | 1.1 | 3.0 | 4.2 | 3.6 |
| (C/N) in 36 MHz for BER = $2.10^{-3}$ with 3dB margin | 4.1 | 6.0 | 7.2 | 6.6 |
| Clear sky C/N in 36 MHz | 11.1 | 13.0 | 14.2 | 13.6 |
| G/T at 12.5 GHz (dBK) | 13.4 | 15.4 | 16.7 | 16.0 |
| Antenna size (cm) | 51 | 64 | 75 | 69 |

*Table 2.8*

Europesat

| | | | |
|---|---|---|---|
| Useful bit-rate (Mbit/s) | 34 | 45 | 54 |
| Modulation | QPSK | QPSK | 8-QPSK |
| Channel coding | Convolutional punctured K = 7 | Convolutional punctured K = 7 | Trellis (pragmatic) |
| Coding rate | 3/4 | 5/6 | 2/3 |
| E/No for BER = $2.10^{-3}$ (dB) | 3.2 | 3.9 | 4.7 |
| (C/N) in 33 MHz for BER = $2.10^{-3}$ (dB) | 3.3 | 5.2 | 6.8 |
| Clear sky C/N in 33 MHz (with 3dB margin) (dB) | 13.3 | 15.2 | 16.8 |
| G/T at 12 GHz (dB/K) | 10.8 | 12.8 | 14.5 |
| Antenna size (cm) | 40 | 50 | 60 |

*Table 2.9*

## 2.9. Possible solution for a short-term introduction of satellite multi-programme digital television

The availability of digital techniques at low cost is the key for future introduction of satellite multi-programme TV services at various quality levels: EDTV and SDTV, including the evolution to HDTV. Broadcasters, satellite operators and receiver manufacturers are actively developing common plans in this direction in order to obtain an ETSI standard as soon as possible.

The main European bodies involved in this standardisation are the ELG (European Launching Group), WGDTB (Working Group on Digital Television), EBU (sub-group V4/MOD) and ETSI. The HDSAT project is also an active contributor to the modulation studies.

In the light of the above investigations, it seems that a suitable transmission technique to cope with the power constraints of the 12 GHz satellite channels is based on QPSK modulation associated with an inner code of 3/4 rate convolutional type with Viterbi decoding. The best outer code is the (255, 239) Reed-Solomon code or a shortened version to cope with the packet length defined in the MPEG-2 multiplex.

This technique has minimal demodulation performance degradations and reasonable receiver complexity for the consumer market. Medium power satellites are the ideal channels for rapid introduction in Europe of these new services.

The standardisation of the bit-rate is a difficult task due to the various transponder bandwidth of the satellites (26, 27, 33, 36 MHz) and also due to the constraints related to retransmission in the cable networks (see section 3).

A possible compromise could be obtained at about 34 Mbit/s. This bit-rate is well adapted to a 33 MHz transponder. It complies with WARC-77 and could also be accepted for wide-band FSS satellites, even if it leads to a slight under-use of the transponder capacity. This bit-rate permits a multiplex of three EDTV services with 16:9 format, or 5 SDTV services compatible with small antennas receivers (60 cm or less), to be broadcast

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# 3. INTRODUCTION OF DIGITAL TELEVISION IN CABLE NETWORKS SELECTION CRITERIA FOR FEASIBLE MODULATIONS AND BIT RATES

## 3.1. The context - satellite/cable compatibility

Compatibility between satellite and cable is one of the essential requirements for any definition of a digital TV broadcasting system.

Transmission through a cable network must comply with the following constraints:

- Head-end processing must be kept to a minimum, and there must be transparency between satellite and cable as regards bit rates.
- Digital broadcasting must not replace existing analogue channels. It must be able to co--exist with these channels, if possible without causing any modification of the existing infrastructure, at least as regards coaxial-cable distribution to viewers.
- Every effort must be made to ensure compatibility with existing network equipment in order to minimise extra costs incurred by the introduction of digital broadcasting, especially as regards user terminals. This constraint is particularly important for VSB-AM tuners and demodulators.

## 3.2. Constraints imposed by cable networks

The transmission of digital signals through large collective antennae and cable networks requires separate processing for each channel, with a modification of the modulation between satellite and cable. The frequency bands used at the present time for analogue transmission in cable networks are as follows:

- the VHF band between 47-68 and 111-300 MHz with channel spacing of 7 MHz (Germany, etc.) and 8 MHz (France);
- the hyperband between 300 and 450 MHz with an 8 or 12 MHz step;
- the UHF band between 470 and 862 MHz with a 8 MHz step.

Figure 3.1 represents the frequency plan used in Germany on the DBP Telekom networks.

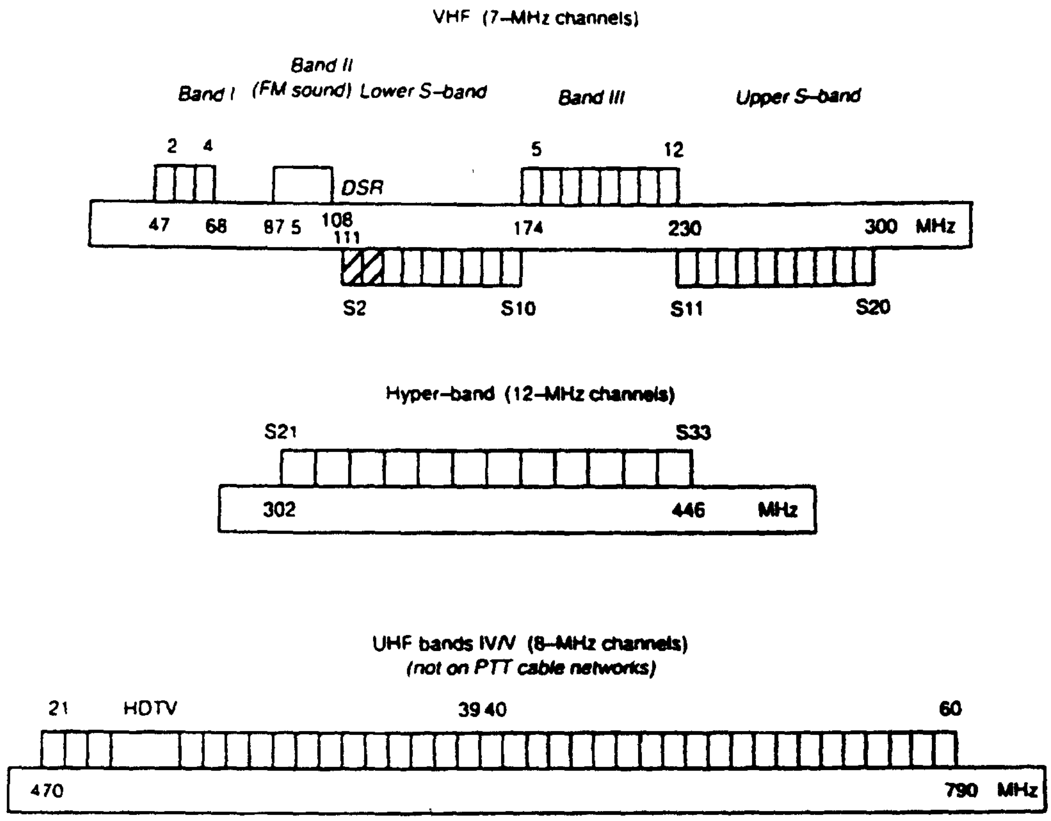



*Figure 3.1: Channel arrangement on German cable networks*

Figure 3.2 shows France Télécom's frequency plan in VHF.

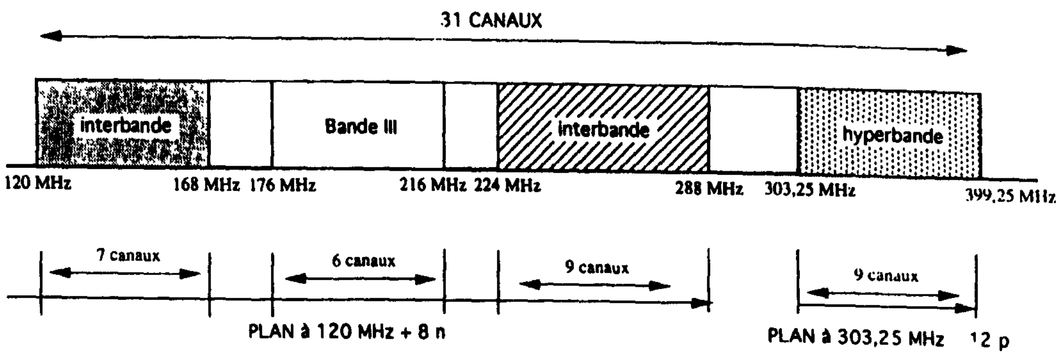



*Figure 3.2: Channel arrangement in the VHF band on French cable networks (France Télécom)*

The distribution of digital signals in a cable network must take account of these bandwidths (8 and 12 MHz channels in France).

Network planning for the transmission of analogue systems must comply with the following constraints:

- the frequencies used for terrestrial broadcasting are not implemented in cable networks over the same coverage area in order to avoid co-channel interference;
- channel allocation is carried out so as to minimise inter-modulation.

Generally speaking, digital signals are more resistant to interference and it should be possible to use "taboo" channels for analogue transmission in networks. This is a strong incitement to use the same bandwidth in digital and analogue transmission, for example by inserting one digital channel between two analogue channels.

Moreover, it is preferable for the output power from a digital channel to be significantly lower than for an analogue channel (at least 10 dB). This will allow for the introduction of a large number of digital carriers without any modification of the present networks.

Finally, account must be taken of certain networks that have a star configuration. Service selection is carried out in a distribution centre and each link between the distribution centre and the user transmits only the selected service (with possibly a second, optional service). This is the case in some of France Télécom's "1G" networks in which services are carried to users on optical fibres, and in a number of private networks (e.g. in Disneyland) which use "twisted pair" wires to carry signals.

The above considerations result in the following constraints when considering the type of digital modulation suitable for the cable distribution of signals received from a satellite:

- 8 MHz bandwidth
- C/N ratio at least 10 dB lower than that of an analogue signal;
- maximum compatibility with existing equipment (in particular, re-utilisation of the tuner-demodulator used for analogue transmission);
- existence of a baseband representation of the digital signal.

## 3.3. Feasible digital modulations

### 3.3.1. Description of the modulation

There are three types of modulation that could be used for the introduction of digital TV through cable networks:

- single-carrier modulation of the QAM type,
- single-carrier modulation of the VSB-AM type,
- multi-carrier modulation of the COFDM type.

#### 3.3.1.1. Single-carrier modulation of the QAM type

This is a well-known type of modulation and has the greatest spectrum efficiency. However, it cannot be introduced in networks without the development of specific modulation and demodulation equipment. There is no compatibility with existing equipment. The combination of convolutional coding and this type of modulation (trellis coding) improves performance as regards noise but leads to increased complexity in the demodulator.

The maximum transmissible output in an 8 MHz channel depends on the number of states in the modulation and the roll-off of the Nyquist filters, as shown in the following equation:

$$D(\,Mbit\,/\,s\,) = \frac{8\,log_2 M}{1+r}$$

where

M = Number of modulation states

r = roll-off from the Nyquist filtering (between 0 and 1).

| Number of modulation states (M) | 4 | 16 | 32 | 64 | 256 |
|---|---|---|---|---|---|
| Max. transmissible rate D(Mbit/s) | 14.5 | 29.1 | 36 | 43.6 | 58.2 |

*Table 3.1*

### 3.3.1.2. Single-carrier modulation of the VSB-AM type

This type of modulation is currently used in networks to transmit analogue television signals (SECAM and D2MAC). It can also be used for the transmission of digital signals in cable networks. It has the advantage of being able to re-use at least some of the modulation and demodulation equipment developed for analogue signals. Spectrum efficiency is slightly lower than that obtained with QAM modulation due to the presence of a vestigial bandwidth. The draft layout for the implementation of this type of modulation is shown in Figure 3.3.

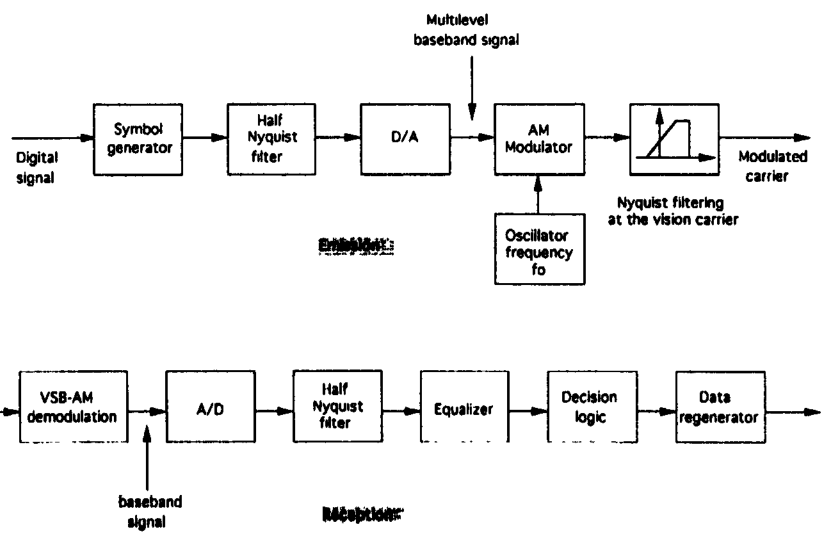



*Figure 3.3: Conceptual block diagram of a VSB-AM modem*

The main characteristics of this type of modulation are as follows:

- width of the residual side-board: 0.75 MHz,
- Nyquist filtering at the vision carrier implemented on transmission,
- residue of the carrier transmitted: to be defined,
- Nyquist filtering in baseband divided equally between transmission and reception.

Under these conditions the maximum rate transmissible in an 8 MHz channel is given in the following equation:

$$D(\,Mbit\,/\,s\,) = \frac{14.5 log_2 N}{1+r}$$

Where

r = roll-off from the Nyquist filtering in the baseband

N = number of states of the signal in the baseband.

The following table gives the maximum rate for a roll-off of 0.07:

| Number of states of the signal in the baseband | 2 | 4 | 8 |
|---|---|---|---|
| Max. transmissible bit rate (Mbit/s) | 13.5 | 27 | 40.5 |

*Table 3.2*

### 3.3.1.3. Multi-carrier modulation of the COFDM type

This type of modulation is envisaged at the present time for the terrestrial broadcasting of digital TV but it could also be used in cable networks. Its spectrum efficiency is comparable to the levels achieved with single-carrier modulation of the QAM-N type, if the type of modulation per carrier of an COFDM signal is identical to the single-carrier modulation. The difference in spectrum efficiency comes from the fact that, in COFDM, there must be a guard interval to absorb echoes and this reduces the duration of the useful symbol. However, the maximum duration of an echo in a cable network is limited to a few microseconds (2 to 4) and it is possible to reduce the loss of spectrum efficiency by increasing the duration of symbols for each carrier (128 or 256 msec is a feasible figure).

This type of modulation is very sensitive to phase and frequency instability in transposition oscillators. However, the problem can be overcome by combining a pilot signal with the COFDM signal. At the same time, this provides compatibility with existing analogue tuners and demodulators. The spectrum obtained in this way is shown in Figure 3.4. It consists of the following signals:

- a pilot signal of frequency of equal to the vision frequency of an analogue VSB-AM signal;
- a blank frequency band of between *f0* and *f0* + P/NT (P integer > 1; NT is the duration of a symbol excluding the guard interval);
- all the COFDM modulated carriers between *f0* + P/NT and *f0* + Q/NT.

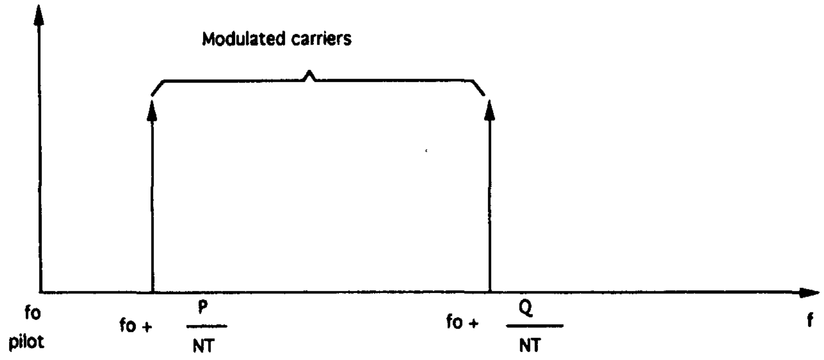



*Figure 3.4: Spectrum of the COFDM signal with a pilot*

The pilot signal of frequency *f0* can be modulated in amplitude at a low index by a reference signal that transmits the symbol frequency and, where appropriate, the field sync, which facilitates the building of the receiver. Figure 3.5 shows the draft layout of a demodulator for a cable network. It consists of the following units:

- a VSB-AM tuner and demodulator which outputs the reference signal and the COFDM signal. This type of demodulation overcomes the problem of phase and frequency noise in the oscillators. The noise is suppressed during demodulation.
- the COFDM signal processing unit. The signal is sampled at frequency 2/T. A simple form of digital processing is used to output the samples I(nT) and Q(nT) applied to the FFT.
- the processing of the reference signal which synchronises a time base that supplies clock signals at 1/T and 2/T.

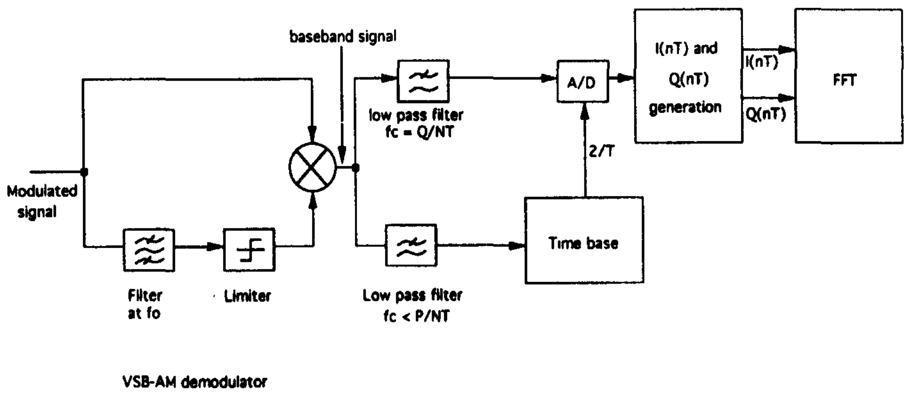



*Figure 3.5: Conceptual block diagram of a COFDM demodulator for cable reception*

The following table shows the transmissible rate in an 8 MHz channel, based on the following parameters:

- guard interval (Δ): 2 μ
- duration of useful symbol (NT + Δ): 130 μ
- width of useful COFDM spectrum: 7.1 MHz (P/NT = 0.1 MHz; Q/NT = 7.2 MHz).

| Modulation per carrier | QPSK | 16-QAM | 64-QAM |
|---|---|---|---|
| Useful rate (Mbit/s) | 14 | 28 | 42 |

*Table 3.3*

### 3.3.2. Theoretical performance as regards noise

For each modulation, the E/No and C/N ratios are given, measured in 8 MHz and corresponding to a bit error rate of $10^{-5}$ (C is the mean power transmitted).

The relationship between C/N and E/No is as follows:

$$(C/N) \text{ in } W = E/No + 10 \log \frac{D \quad \text{---> Mbit/s}}{W \quad \text{---> Mbit/s}} \qquad (W = 8 \text{ MHz})$$

#### 3.3.2.1. Single-carrier modulations of the QAM type

It is presumed that the Nyquist filtering is divided equally between transmission and reception, with a roll-off of 10%.

The theoretical results are given in the following table:

| Number of modulation states | 16 | 64 | 64 | 256 |
|---|---|---|---|---|
| Channel coding | none | trellis<br>r = 2/3 | none | trellis<br>r = 3/4 |
| Max. transmissible rate (Mbit/s) | 29 | 29 | 43.6 | 43.6 |
| E/No for BER = $10^{-5}$ | 13.4 | 10.4 | 17.8 | 14.8 |
| C/N in 8 MHz for BER = $10^{-5}$ | 19 | 16 | 25.2 | 22.2 |

*Table 3.4*

#### 3.3.2.2. VSB-AM modulation

For a given error rate, the transmission of the residual carrier leads to an increase in the mean power transmitted, depending on the modulation index and number of states, as shown below:

| (m) | Number of modulation states | Increase in output $\Delta C/N$ (dB) |
|---|---|---|
| 1 | 4<br>8 | 2.8 dB<br>3.4 dB |
| 1.3 | 4<br>8 | 1 dB<br>1.3 dB |

*Table 3.5*

Under these conditions, the theoretical results in the presence of noise are given in the table below. They are based on the following presumptions:

- residual bandwidth: 0.75 MHz
- width of Nyquist filter in baseband: 6.75 MHz.

| Number of modulation states | 4 | 8 | 8 | 16 |
|---|---|---|---|---|
| Channel coding | none | trellis r = 2/3 | none | trellis r = 3/4 |
| Max. transmissible rate (Mbit/s) | 27 | 27 | 40.5 | 40.5 |
| E/No for BER = $10^{-5}$ | 22.3 | 19.3 | 28.3 | 25.3 |
| C/N in 8 MHz for BER = $10^{-5}$ | 20.5 | 17.2 | 26.2 | 23.2 |

*Table 3.6*

### 3.3.2.3. COFDM modulation with pilot

The following table indicates the theoretical results for a BER of $10^{-5}$, supposing that the output power from the pilot signal is 5 dB less than the total output power from the COFDM carriers.

| Modulation per carrier | 16-QAM | 64-QAM | 64-QAM | 256-QAM |
|---|---|---|---|---|
| Channel coding | none | trellis r = 2/3 | none | trellis r = 3/4 |
| Max. rate (Mbit/s) | 28 | 28 | 42 | 42 |
| C/N in 8 MHz for BER = $10^{-5}$ | 20 | 17 | 26.2 | 23.2 |

*Table 3.7*

### 3.4. Advantages and disadvantages of the three types of modulation

The main advantages and disadvantages of the three types of modulation described above as applied to digital transmission in a cable network are shown in the following table.

| Modulation | Advantages | Disadvantages |
|---|---|---|
| Single-carrier QAM | - Well-known modulation<br>- Good spectrum efficiency<br>- Good performance as regards noise | - Not compatible with analogue tuners and demodulators<br>- No signal representation in baseband<br>- Difficult to recover the carrier in a consumer electronics environment |
| Single-carrier VSB-AM | - Compatible with tuners and demodulators used for analogue signals<br><br>- Signal representation in baseband<br><br>- Digital processing on one channel only (1 single CAN) | - Spectrum efficiency lower than the QAM (vestigial bandwidth)<br><br>- Impairment with regard to noise because of carrier residue |
| COFDM with pilot | - Compatible with tuners and demodulators used for analogue signals<br><br>- Signal representation in baseband<br><br>- Unaffected by oscillator phase noise<br><br>- Compatible with terrestrial broadcasting (allows for microwave re-transmission within an apartment building) | - Spectrum efficiency lower than with QAM<br><br>- Impairment with regard to noise because of pilot |

*Table 3.8*

## 3.5. Inter-modulation (analogue channel interference from digital channels)

### 3.5.1. Simulation model

Simulations were based on the network in Paris which is representative of major cable networks. In 1994, this network will transmit 40 analogue channels (SECAM and D2-MAC), 21 in VHF and 19 in UHF. In certain geographical sectors, VHF operation is limited to 340 MHz and it will not be possible to insert digital channels in this band. Research has therefore been carried out based on the assumption that digital channels will be introduced in UHF only. The UHF band allows for the distribution of 48 channels (channels 21 to 68). Given the 19 analogue channels, this leaves open the possibility of inserting 29 digital channels.

For the purposes of simulation, each digital channel is represented by a multiplex of 4 unmodulated carriers such that the total output power is equal to the output power of the digital signal. Research has shown that the total number of carriers must be greater than 100 in order to provide a realistic representation of the interference induced on an analogue signal by a digital signal, and this is the case here (4 x 29 > 100). The carrier frequencies are selected to take account of the subjective impairment in the analogue signal. The scale of the impairment depends on the distance of the interference source from the vision carrier.

The calculations were also based on the hypothesis that non-linear coefficients of order 2 and order 3 are constant throughout the band. This approximation is particularly true when a large number of rays are generated. In fact, for a high number of rays, the variations in the non-linear coefficients become negligible compared to the number of beats generated.

### 3.5.2. Results

The network specifications guarantee a C/ (N + I) ≥ 41.5 dB at user terminal level for a SECAM signal transmitted in VSB-AM (C is the peak carrier output. N + I is the sum of the noise and interference measured in an 8 MHz band).

The simulation program made it possible to determine, for each of the 40 analogue channels (21 VHF and 19 UHF), the level of impairment introduced by the insertion of 29 digital channels. The results obtained are shown in the table below.

| | | S/IM2 | S/IM3 |
|---|---|---|---|
| Impairment introduced by 29 digital channels on 40 analogue channels with a 10 dB back-off | m<br>σ | 1.9 dB<br>1.4 dB | 1.8 dB<br>0.4 dB |
| Impairment introduced by 29 digital channels on 40 analogue channels with a 13 dB back-off | m<br>σ | 1 dB<br>0.9 dB | 0.9 dB<br>0.2 dB |
| Impairment introduced by 29 digital channels on 40 analogue channels with a 15 dB back-off | m<br>σ | 0.7 dB<br>0.6 dB | 0.6 dB<br>0.2 dB |

*Table 3.9*

The back-off R is defined as the difference between the peak output power of the analogue carrier and the mean output power of the digital signal. As regards the digital signal, the C/N + I ratio in 8 MHz is given by the equation:

$$C/(N + I) = 41.5 - R(dB)$$

m and $\sigma$ represent respectively the mean impairment and the standard deviation obtained for all the 40 analogue carriers. It is noticeable that a back-off of 10 dB results in a mean impairment of approximately 2 dB with a high level of dispersion over the order 2. For SECAM signals, this corresponds to a loss of approximately 1 grade on the 5-grade impairment scale and an impairment of this level is not acceptable.

A back-off of 13 dB is necessary before the mean impairment is equal to 1 dB, still with major dispersion over the order 2. However, this level of impairment can be acceptable. In most cases, it is equal to less than one half-grade on the 5-grade impairment scale.

This being so, the following condition must be complied with for the digital signal:

$$C/(N+I) \leq 28.5 \text{ dB}$$

This value can correspond to a BER of $10^{-5}$ at the output to the demodulator if the entire multiplex transmitted is protected by a powerful error-correcting code, e.g. a Reed-Solomon. However, for this BER, digital modulation must guarantee a sufficiently large margin to take account of any network defects, in particular defects in the user terminal (ghosts, oscillator phase and frequency noise etc.).

With a back-off of 13 dB, the margin M is given in dB by the following equation (for BER = $10^{-5}$)

$M(dB) = 28.5 - (C/N + I)$ theoretical

The following table gives the technological margin for various rates and modulations as described above, for a back-off of 13 dB. Only modulations producing a margin of more than 3 dB have been retained.

| Modulation | Rate (Mbit/s) | Margin with a 13 dB back-off |
|---|---|---|
| VSB-4AM (m = 1) | 27 | 6.2 |
| VSB-16AM (m = 1.3) + trellis coding r = 3/4 | 40 | 5.3 |
| COFDM with QAM-256 + trellis coding r = 3/4 + pilot at -5 dB | 40 | 5.3 |
| QAM-256 with trellis coding r = 3/4 (20 % roll-of) | 40 | 6.7 |
| QAM-256 with trellis coding r = 3/4 (10 % roll-off) | 43.6 | 6.3 |

*Table 3.10*

### 3.6. Preliminary conclusions

In the light of this preliminary study realised by computer simulation, it seems that 40 Mbit/s is the maximum bit rate transmissible in an 8 MHz cable network channel, simultaneously with the existing analogue channels. A good compromise between satellite and cable could be found at around 34 Mbit/s. This bit rate may also be acceptable for satellite broadcasting, even if it leads to a slight under-use of the capacity of a 36 MHz transponder. It is the price that has to be paid in order to ensure transparency between satellite and cable as regards bit rates.

However, it is difficult to know if the implementation margins are sufficient in order to take into account simultaneously the impairments due to cable networks and user terminals. It is the industry's responsibility to define these margins.

Any attempt to exceed 40 Mbit/s for cable would constitute a technical challenge that would be difficult to achieve in the near future. The transmission, for example, of a bit rate of 45 Mbit/s in an 8 MHz channel requires 64-QAM modulation with a 6% roll-off and this would be difficult to achieve in receivers manufactured for use by the general public.

The choice of modulation is particularly difficult. It depends on the feasibility of a suitable demodulator in a "consumer electronics" environment and it is evident that a great deal of research remains to be carried out, in conjunction with industry, in order to determine the best possible compromise between price and performance. The research undertaken at the CCETT, for example, has shown that phase noise from transposition oscillators used in a cable network and, more especially in the user terminal, is a critical factor in the design of a digital terminal. As far as this problem is concerned, COFDM modulation with a pilot provides a particularly attractive solution.

# 4. TERRESTRIAL DIGITAL TV BROADCASTING

## 4.1. Overview of the system requirements in the context of spectrum congestion

In the process of defining a new system for digital terrestrial TV broadcasting, operational constraints related to frequency management, as well as service concepts in agreement with current and future trends, constitute the original requirements on which the broadcasters have to base their developments.

Below one Gigahertz in particular, RF spectrum has a limited capacity and is greatly sought after. Analogue TV broadcasting already occupies a very large part of that frequency region and any new non-compatible digital TV system will have to share that band with existing services, while being attractive to the public in terms of the increased number of services.

The introduction of a digital service will initially be subject to the current constraints inherent in network frequency planning, a prerequisite for the launch of a digital simulcast of services currently broadcast on the analogue terrestrial network. In other words, in the short term digital television services will have to be introduced in "taboo channels" - these gaps in the spectrum currently unusable for analogue broadcasting - within a more or less extensive service area; this implies development a system of that is not only robust enough to withstand interference from analogue TV services, but also will not disturb them.

To these constraints of spare spectrum capacities, we must add the difficulties inherent to the reception of terrestrial television broadcasting (echoes, shadow areas, etc). Moreover the masking conditions for reception at roof-level, in an apartment or even two metres from the ground are so many elements which, in terms of liaison efficiency, affect fixed reception and portable reception very differently.

Consequently, digital techniques will have to provide high ruggedness to multipath transmission (ghosting), power efficiency to deal with spectrum interference (industrial and thermal noises), weaker protection ratios (to and from analogue channel interference), high spectrum efficiency to offer high data rate or high protection, and last but not least, Single Frequency Network options.

## 4.2. Limitations of today's analogue systems

### 4.2.1. Intrinsic quality

Analogue transmission of audio-visual signals is affected mainly by thermal and industrial noise ("snow" on pictures and "crackles" or hiss on sound) and by unsatisfactory propagation of electromagnetic waves, i.e. reflections and diffraction, which result in "ghosting" over the main picture, or a fluctuating sound quality, depending on the environment of the receiver.

### 4.2.2. Inadequate services

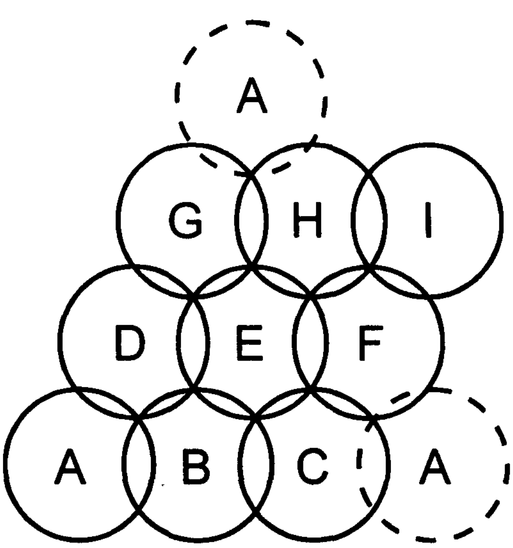
The lack of flexibility in the analogue signal quality that can be obtained with today's analogue systems impedes the development of more demanding services (e.g. high-definition TV). At the same time, the introduction of new services is hindered by the congestion of the terrestrial spectrum and the required economic viability cannot be achieved due to the difficulty of defining a method of conditional access which is totally transparent with regard to the quality of the unscrambled analogue signal.

Terrestrial television has not yet been able to dispense with the need for connection to a fixed aerial. In fact, television takes advantage of neither the specific character of terrestrial broadcasting nor the convenience which a plug-free receiver would offer.

### 4.2.3. The high costs of network engineering

Since the only way round the problem of co-channel interference is by prohibiting the re-use of the same channel over considerable distances, in some instances up to several hundred kilometres from the service area, the result is extremely inefficient in terms of frequency planning (cf. figure 4.1) as, in conventional 9-frequency layouts, each channel is unusable over approximately 89% of the land area!

Moreover, the low immunity to noise shown by analogue signals requires high levels of transmitted power, which taken in conjunction with the ever-present problems of shadowing and the consequent implementation of significant numbers of relay stations, results in a broadcasting infrastructure that is extremely costly.




The present situation
1 service = 9 frequencies

*Figure 4.1: Frequency planning for conventional analogue TV systems*

## 4.3. Service constraints and implication on network structure

It goes without saying that any new system should overcome all the restrictions of conventional systems listed in the previous paragraphs! In other words, its spectral and power efficiencies must be extremely high. Similarly, it should be capable of functioning correctly despite echoes and, therefore, any form of linear distortion, including those which are time-variable.

Above all, a new system must be introduced within the current analogue context, and absolutely needs to be able to cope with the existing PAL/ SECAM/ NTSC environment.

Finally, economic criteria, notably receiver costs, are vital factors in the acceptability of any new system designed for the general public.

### 4.3.1. Principles with regard to echoes

The two major difficulties in terrestrial broadcasting i.e. the intrinsic quality of the signal and frequency planning, are often considered to be mutually exclusive. However, interference caused by remote transmitters broadcasting the same service as a local transmitter may be considered as equivalent to artificial echoes and, in effect, these two essential problems merge into one. In this situation, it is easy to understand why the potential of any system is first and foremost dependent on its strategy towards echoes! Traditional analogue systems are content to ignore the problem. Conventional digital systems, e.g. single-carrier with equaliser, try to eliminate echoes in order to provide good intrinsic reception quality in a large number of cases, on condition that reception is by means of fixed antennas. This, though, does not provide a satisfactory long-term solution to the problem of frequency planning and network costs.

It seems that only a radically innovative attitude towards echoes would be capable of resolving all the problems of terrestrial hertzian broadcasting and introducing revolutionary new approaches.

### 4.3.2. Single Frequency Networks

It is believed that a major feature of a new digital terrestrial TV system should be the possibility of operating Single Frequency Networks.

A Single Frequency Network (SFN) is a set of transmitters spread throughout a given territory and transmitting the same signal (synchronised precisely) at the same frequency. This allows for variable coverage area (whole country or parts of country). However, this technique can also be used locally to provide coverage in a shadowed area by direct re-amplification using a co-channel retransmitter (often called "gap filler").

As soon as gap filling or Single Frequency Networking are used, space diversity becomes available on the network side. This means that spatially distributed transmitters contribute by adding power to the received signal. As generally the position of these transmitters is not concentrated in any one direction, this feature is very useful to fill in a complete shadow when an obstacle (building, hill) is masking a given direction in the horizontal plane.

On the other hand, when the receiver is close to a given transmitter of the network, this space diversity is no longer available; however, it is not needed in this case because the close proximity of the transmitter implies a high signal-to-noise ratio.

The design of a SFN structure is of course strongly related to the capabilities of the broadcasting system in terms of acceptable relative echo powers and delays. The transmitter separation distance is a major economic factor, and for large area coverage, the larger it can be, the better.

However, depending on service requirements, it is possible to design SFNs at a smaller scale, for instance at the scale of a large city. This is generally called "dense network" operation. This is discussed in detail in paragraph 4.7.

## 4.4. COFDM principles

This paragraph only gives an overview of the principles of COFDM. For more detailed description, please refer to the following articles:

1. Pommier, Wu: Interleaving or spectrum spreading in digital radio intended for vehicles. EBU Review n° 217, June 1986.
2. Einstein, Ebert: Data transmission by frequency division multiplexing using the discrete Fourier Transform - IEEE Transactions on Communication Technology, Vol. COM, 19, n° 15.
3. Lard, Halbert: Principes de modulation et de codage canal en radiodiffusion numérique vers les mobiles - EBU Review n° 224, August 1987.
4. Le Floch, Halbert, Castelain: Digital sound broadcasting to mobile receivers - IEEE Transactions on Consumer Electronics, Vol. 35, n° 3, August 89.
5. Le Floch, Helard, Castelain, Rivière: Démodulation cohérente du système COFDM dans un canal de radio mobile - 13th GRETSI Conference, September 1991.
6. Mason, Dury, Lodge: Digital TV to the home, when will it come? Proceedings IBC '90.
7. Pommier, Ratliff, Meier-Engelen: A hybrid satellite/terrestrial approach for digital audio broadcasting. NAB, Atlanta, 30 March - 4 April 1990.

The transmission channel can be described by its impulse response, which characterises the various signal paths in terms of delay and attenuation. When the product of the delay spread by the signal bandwidth increases, frequency selectivity affects the transmission.

For reception in a Single Frequency Network, the channel impulse response usually extends over a few tens of microseconds, and in some cases as much as 100 µs or more. This being so, frequency-non-selectivity concerns only low bit-rate transmissions (a few kbit/s) and can under no circumstances constitute a valid hypothesis for TV broadcasting.

Moreover, the diversity provided by the use of wide-band transmission must be considered as an advantage if the communication system is designed to make use of multipath signals rather than be restricted by their presence. Because of the spread of the channel response, it is highly unlikely that fading would affect a whole frequency band covering several Megahertz.

The COFDM system relies on the following principles (figure 4.2):

- The first principle consists of splitting the information to be transmitted into a given number of modulated carriers, each with a low bit-rate (2), so that the corresponding symbol time becomes larger than the delay spread of the channel. Then, provided that a guard interval is inserted between successive symbols, the channel frequency selectivity no longer generates inter-symbol interference. Nevertheless, some of the carriers are enhanced by constructive interference, while others are suffering from destructive interference. Therefore, if only this principle was applied, some information would be well received, and some would be lost through the "local fading".

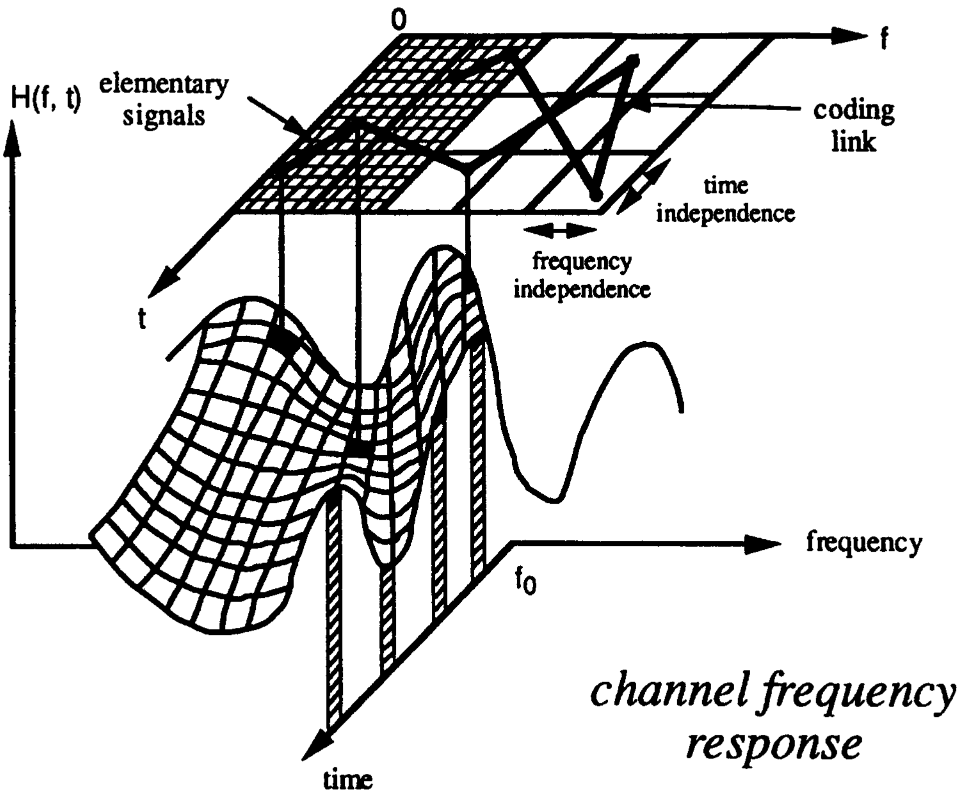



*Figure 4.2: Principles of COFDM*

- The second principle systematically exploits the multipath signals between the transmitter and the receiver, by using the fact that signals sufficiently separated in frequency and time cannot be affected identically by impairments in the channel. Therefore, the COFDM system includes linking of the elementary signals (information modulating a given carrier during a given symbol time) transmitted at distant locations of the time-frequency domain. This is achieved by convolutional coding associated soft-decision Viterbi decoding, in conjunction with frequency and time interleaving (3).

The time-frequency diversity obtained by such interleaving allows the receiver to integrate local fading phenomena over the whole signal bandwidth and over the time interleaving period: the system performance is then related to an "average signal-to-noise ratio" criterion.

This signal-to-noise ratio will increase as soon as the received signal power is augmented by echoes that cannot combine destructively: this is the case when the echoes are separated by a minimal delay equal to the inverse of the signal bandwidth. Therefore, the COFDM system combines the multipath echoes constructively (power sum).

The OFDM technique, known since 1970 in the scientific literature, had not been implemented before 1988 for broadcasting purposes. The first known implementation was realised in the context of the Digital Audio Broadcasting (DAB) project, in conjunction with a powerful channel coding technique (the C of COFDM), to allow strongly-distorted signals to be received satisfactorily, as it is the case in mobile reception under multipath conditions.

When considering the reception of digital TV signals, the two following considerations in comparison with DAB have to be taken into account:

- the required spectrum efficiency must be higher for TV, typically 4 bit/s/Hz compared to 1 bit/s/Hz for DAB;
- the receiving conditions should be less critical with TV.

These considerations led to the fact that some researchers thinking that the "C" of COFDM could be completely omitted, because severe multipath conditions would not affect the transmission anymore.

However, in a Single Frequency Network, it is quite possible that two transmitter contributions would be received with similar levels, thus inducing 0 dB echoes.

The use of coding is therefore believed to be necessary, even under good signal-to-noise ratios, in order to be able to achieve good reception when deep notches affect the TV signal spectrum.

## 4.5. COFDM system performance

### 4.5.1. Spectral efficiency and power efficiency

It goes without saying that COFDM allows the most recent techniques as regards block or trellis-coded modulation to be used. Indeed, selecting the type of modulation applied to each sub-carrier depends solely on the compromise between the rate transmitted and the robustness required.

Figure 4.3 gives the performance of the COFDM system in terms of possible trade-offs between the available bit rate in an 8 MHz channel, and the carrier-to-noise ratio necessary to obtain an unimpaired picture quality.

The Shannon limit is the theoretical limit for error-free digital transmission over a Gaussian channel (a channel affected only by additive white Gaussian noise).

The performances are given for two types[2] of coded modulation:

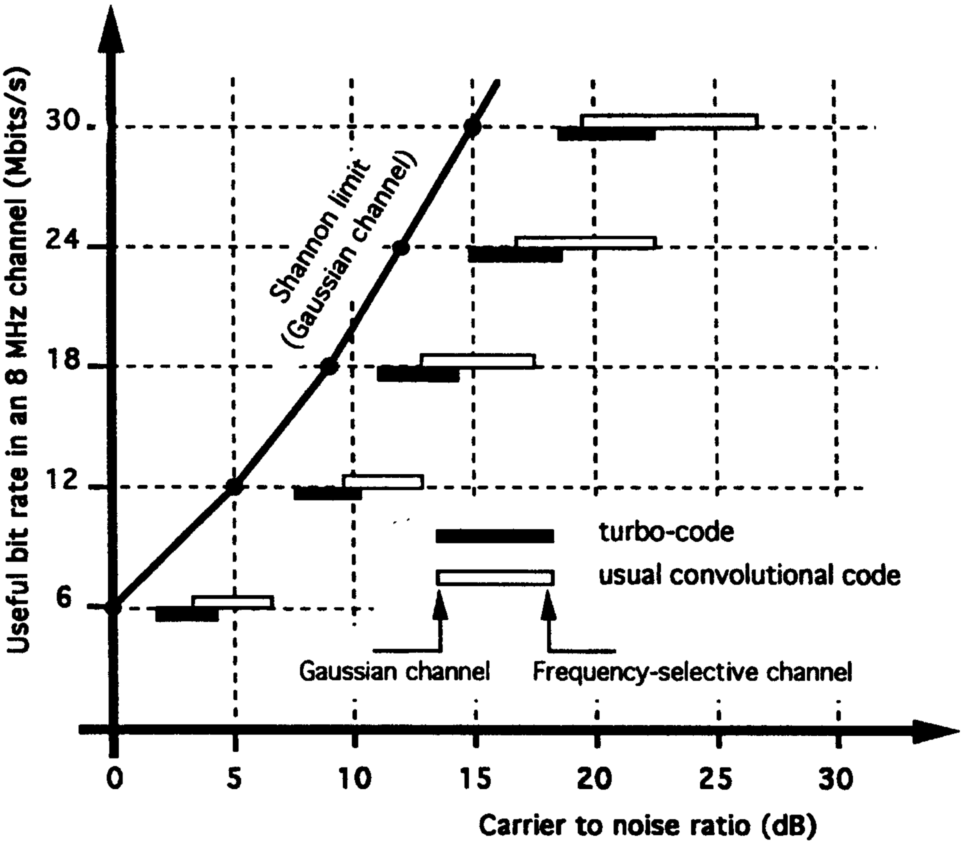
- usual convolutional codes,
- turbo-codes, which are codes in a new family, derived from the usual convolutional codes and showing better performance.

Figure 4.3 has to be understood in the following way:

The left end of the segment gives the performance of the coded modulation in the Gaussian channel; the right end of the segment gives the performance in the most

[2] These are given as examples, others being also under consideration.

difficult channel (the frequency-selective channel), in which deep notches may appear due to echoes as strong as the main signal.




| Useful bit rate(Mbit/s/s) | Code rate | Modulation associated to each carrier |
|---|---|---|
| 6 | 1/2 | 4-PSK |
| 12 | 2/4 | 16-QAM |
| 18 | 3/4 | 16-QAM |
| 24 | 4/6 | 64-QAM |
| 30 | 5/6 | 64-QAM |

*Figure 4.3: COFDM performance*

### 4.5.2. Flexibility of service multiplexing

Any digital system can provide all the usual advantages associated with this type of transmission, particularly as regards flexibility (broadcast bits can convey any type of service) and ease of scrambling.

As far as the multiplexing of various services is concerned, COFDM also offers a number of unusual, and COFDM-specific, characteristics:

- because the same frequency band may be shared between several services and COFDM systematically exploits the range of frequencies available, multiplexed services are more robust than services transmitted separately. For example, where the system is applied to a sound broadcasting service (DAB), the multiplexing of 6 high-quality stereo services in a 1.5 MHz band makes them almost invulnerable to flat fading.

- in addition to the conventional opportunities for time and frequency multiplexing, COFDM also offers the possibility of selecting different types of modulation for different sub-carriers. "Multi-resolution"-type modulation, i.e. modulation interpreted differently depending on the quality of the channel or receiver, is also possible on all or only some of the sub-carriers. The flexibility resulting from these combinations makes it feasible to believe that a hierarchical television service which would be capable of adapting to different levels of receiver mobility or channel quality might be devised.

### 4.5.3. Advantages of COFDM as regards the introduction of new digital broadcasting services

In digital television broadcasting, one of the strategies envisaged for the introduction of new services is based on the hypothesis that they will be allocated the so-called "taboo" frequency bands (cf. figure 4.6 b).

There are two main families of "taboo" channels. The first is defined as the two channels immediately above and below the channel being used; these channels are not used because of the inadequate selectivity of television tuners. The second includes the so-called "co-channel locations" which are impaired by a signal broadcast from a transmitter which is geographically too distant to give a picture without noise but which is close enough to disturb another service. Although the use of these normally prohibited channels is a tempting idea, it creates two restrictions which would appear, at the outset, to be incompatible. The digital signal must be extremely resistant to PAL/SECAM/NTSC co-channel interference, and, inversely, cause very little interference to existing conventional service signals.

<u>Ruggedness of COFDM signals to co-channel analogue interferers</u>

The vision carrier of a conventional analogue signals, together with the chrominance and sound sub-carriers, may be considered as narrow-band interferers. This being so, they affect only a limited section of the COFDM signal spectrum, and the signal then automatically behaves as if the few sub-carriers concerned were affected by local fading. The remarkable ruggedness of COFDM to narrow-band fading also provides a considerable degree of immunity to narrow-band interferers.

<u>Ruggedness of conventional analogue signals to COFDM-induced interference</u>

Because it consists of a number of independent carriers with random phases, the COFDM signal has a Gaussian amplitude distribution, and is therefore seen by a PAL, SECAM or NTSC receiver as white noise. This is ideal as it minimises the subjective annoyance suffered by viewers of conventional television services.

This remarkable characteristic is further emphasised by the high level of power efficiency obtained from this system.

Finally, COFDM offers the new possibility of synthesising a tailored spectrum: at the cost of a slight reduction in the bit-rate, it is possible to cut out those COFDM carriers which correspond to particularly sensitive frequencies of an analogue co-channel signal. It is possible, for example, to create a tailored spectrum (or even a comb spectrum !) in order to avoid any risk of interference on the PAL/SECAM/NTSC vision carrier and sound sub-carrier.

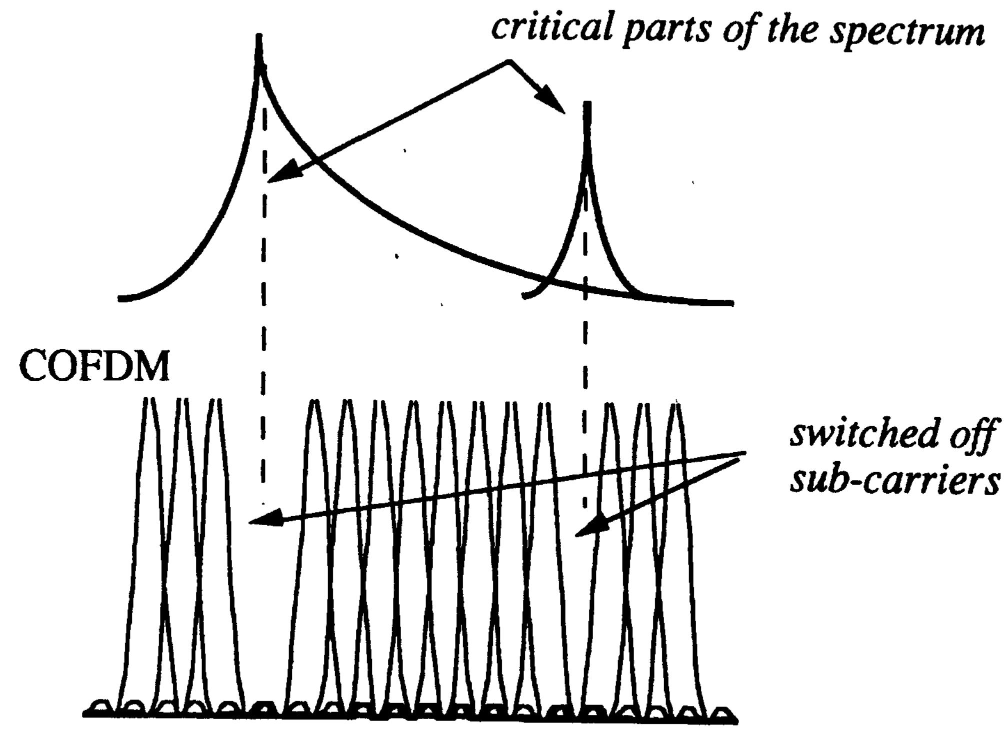



*Figure 4.4: Formatting the COFDM spectrum to optimise sharing with existing signals*

## 4.6. Modulation schemes other than COFDM

Alternative solutions to the COFDM technique are known as single carrier modulation techniques. The main problem of single-carrier modulation is the handling of echoes and the resulting inter-symbol interference (ISI): unlike COFDM, the symbol length is very small (typically 0.15μs), and echoes may occur, that are much longer than a symbol (up to tens of μs in an urban area, up to more than 100 μs in a Single Frequency Network).

This means that the received signal samples are actually a combination of many consecutive symbols. The coefficients of this combination reflect the channel's impulse response.

Equalisation roughly consists in inverting this combination by combining several consecutive received samples with the proper coefficients. These coefficients depend on the channel response, which must hence be evaluated. This may be achieved through a training sequence (start-up), followed by a blind adaptation of the coefficients (tracking). Wholly blind equalisation is also possible: it eliminates the training sequence but needs some time for convergence. Blind processes take advantage of the fact that ISI-corrupted- and ISI-free signals show different statistical properties.

In all cases the number of required coefficients (hence the complexity) increases with the length and the strength of echoes: long echoes mix up more symbols together and combinations with more than one large coefficient (generated by strong echoes) are difficult to invert. Known equalisation algorithms of reasonable complexity cannot cope with echoes longer than a few symbol lengths, nor with "0 dB echoes" (echoes that are as strong as the main ray). This means that they cannot work in a Single Frequency Network, and that even their use in a conventional network raises a lot of complexity problems as soon as the channel response spreads over more than 10 µs.

However, some work is going on to try and simplify equalisers, knowing that the spread of one echo is much smaller than its delay: this reduces the actual number of interfering symbols, hence the minimal complexity of the equaliser. But Single Frequency Network operation still seems out of reach for equalised single-carrier systems.

In Europe, Single Frequency Networks were first seen as a way to broadcast over large areas using one frequency. In America (and in some European countries as well) such large Single Frequency Networks are not being considered because few, if any, nation-wide services are envisaged: all American proposed systems are close to an equalised single-carrier system.

However, it should not be forgotten that local Single Frequency Networks are also very attractive: they allow a much sharper decrease of the received power outside the service area; constraints on frequency re-use are loosened and spectrum management is a lot more efficient.

## 4.7. Network design possibilities in digital TV

The following paragraphs examine COFDM in terms of a "broadcasting system" and show that the new horizons opened up by this technique are even more attractive when looking at the implementation of a broadcasting system as a whole.

### 4.7.1. Gap-fillers

Since the system has been designed to take advantage of echoes, it is possible to create them deliberately.

The initial application of this idea provides the means of eliminating residual areas of shadow by using passive reflectors or small active relays WITHOUT HAVING TO CHANGE THE CARRIER FREQUENCY for the reflected or relayed signal. The signal is picked up at a location in which reception conditions are satisfactory. It is then re-amplified and rebroadcast at the same frequency towards the shadow area that is to be covered. These "gap-fillers" do not require any additional frequency spectrum and their basic simplicity ensures very low costs. If we consider that the total number of relays necessary to provide national coverage in a country like France amounts to several thousand units, it is easy to see the benefits of such an approach.

Since 1988, full-scale tests at around 900 MHz have been carried out in Rennes (France) and have successfully shown the feasibility of such "gap-fillers".

### 4.7.2. Dense networks and power efficiency of the system

If one imagines a network of terrestrial transmitters distributed over a given territory, all of them time-synchronised and broadcasting the same signal on the same frequency, then the useful power received at the input to the receiver is the sum of the incoming powers from all the transmitters. The various incoming signals are seen as echoes of the same signal and combine positively if their temporal spread is in such that it is compatible with the selected duration of the guard interval. In other words, COFDM enables constructive overlapping of the various areas of transmitter coverage. The linking, by this technique, of the largest possible number of interdependently operating transmitters offers many advantages:

- The broadcasting infrastructure is less expensive since it avoids the need for excessively powerful transmitters whose cost increases in relation to the area of coverage.
- It enables the launch of a new service with minimum initial investment, then a gradual expansion in the area served.
- It makes more efficient use of the power transmitted as shown qualitatively in the diagram below (figure 4.5).
- Unlike analogue systems which cater for extremely variable signal-to-noise ratios, digital systems are very sensitive to a threshold effect within which there is marked quality loss and beyond which the quality is no longer improved. By enabling a more accurate designation of the area of coverage, and greater uniformity in respect of incoming power, dense networks can considerably reduce this drawback, which is specific to digital systems. Indeed, the COFDM concept could be summarised as: "the right power, at the right place".
- It improves control of co-channel interference, and dramatically reduces the frequency re-use distances. This is considered to be an essential point for the possible use of COFDM for HDTV in America; it is one of the principal arguments advocated by American COFDM supporters, as a possible solution to the problem of spectrum sharing in dense urban areas.

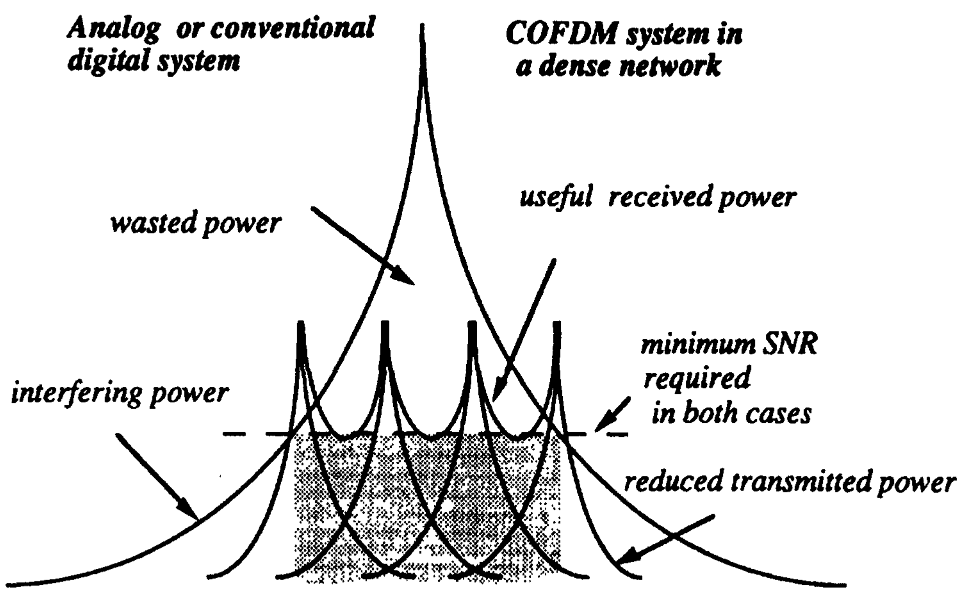



*Figure 4.5: Power efficiency of dense networks*

- In addition to the time and frequency variations of COFDM, it brings spatial diversity into transmission.

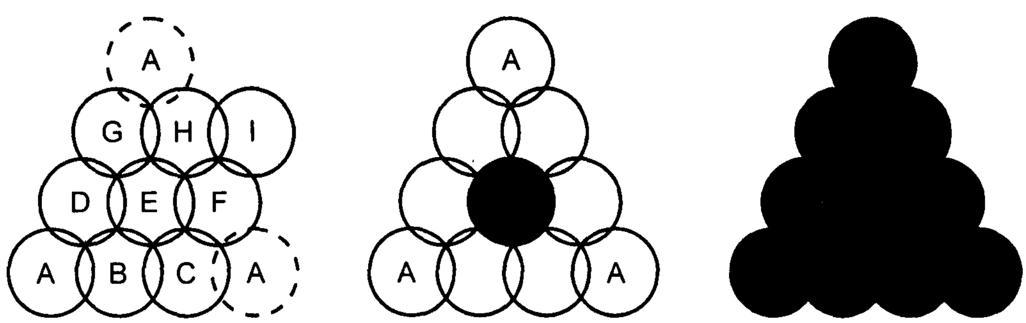
An experimental network comprising two transmitters operating at 60 MHz, which was brought into service to cover the town of Rennes, confirmed the fact that the overall coverage area served by two transmitters operating simultaneously is considerably greater than the sum of the coverage areas for each of the transmitters operating singly.

### 4.7.3. Single frequency networks and spectral efficiency of the system

The concept of a network of transmitters operating synchronously on the same frequency conflicts with conventional research and development techniques which require geographically-adjacent transmitters to operate on two distinct frequencies, whether or not they are broadcasting the same service.

If we consider that the forty-five 8 MHz channels in the UHF band allocated to television in France are totally saturated by five national services, COFDM can provide the means of increasing this number nine-fold. If we also consider a hypothetical rate of 24 Mbit/s in each channel, then the combination of COFDM and source coding provides a gain in terms of spectral occupation in a ratio of 9 x 4 = 36 !

The following diagram shows the dramatic overall spectral efficiency achieved by a system based on COFDM.




The present situation
1 service = 9 frequencies!

In the near future

In the long term
1 service = 1 frequency

*Figure 4.6 a: Frequency planning for conventional analogue or digitalsystems*
*Figure 4.6 b: Introduction of a digital service in an analogue context*
*Figure 4.6 c: Frequency efficiency of a single-frequency network*

## 5. INTER-OPERABILITY BETWEEN ALL BROADCASTING DELIVERY MEDIA: THE COMMON-RECEIVER CONCEPT

Ideally, a digital receiver should accept input signals from all TV/HDTV broadcasting delivery means such as satellite and terrestrial transmitters, coaxial and optical cables (B-ISDN), video cassette-recorders and laser-disc players.

However, as it has been shown in this document, satellite, cable and terrestrial channels have extremely different transmission characteristics and hence different modulation schemes tailored to these transmission characteristics have to be used.

A common modulation scheme could be considered for terrestrial broadcasting and cable distribution (COFDM is the only possible candidate we know of at present).

Commonality in receivers intended for all delivery media should be found at the level of the multiplex.

A digital TV receiver for conventional 625-line television services with 4:3 or 16:9 .aspect ratios should incorporate the following basic functions, identical for all the broadcasting media:

- programme demultiplexer operating at different bit-rates;
- conditional access system serving a number of competing service-providers;
- picture decoder suitable for various picture qualities from LDTV to EDTV;
- sound decoders.

The tuners and demodulators which are specific to the broadcasting medium could be inserted in the TV receiver or available as a separate unit, as it is currently the case for satellite or cable reception.

In order to make provision for the future introduction of digital HDTV, possibly in association with hierarchical coding and modulation, it is desirable to have a flexible multiplex capable, for example, of extracting a conventional 625-line picture from the 1250-line HDTV signal. This is a difficult task because these constraints must not delay the short-term introduction of digital television in Europe.

The answer to the question "What is the optimum extent of inter-operability among all broadcasting media?" has not been fully solved so far, probably because it is related to various aspects of digital TV broadcasting: technical constraints, of course, but also the (public and private) operators' and service-providers' policies, regulation, consumer electronic manufacturers' constraints, and introduction-calendar perspectives.

Some helpful suggestions related to this problem are given in the above-mentioned paper that Mr. Pommier presented at the Montreux '93 International TV Symposium.

> "... The repetition of the scenario of the MAC systems can perhaps be avoided. To achieve that, it is of course necessary to reduce the importance and the number of factors which are divisive. Amongst these factors previously described, most can be greatly reduced by the judicious choice of technique and technology. Others only depend on the will of the people to implement open and standardised systems instead of closed systems, the development and the use of which they are able to control. Finally, if only the relative differences to the physical characteristics of the broadcasting media should remain, the technicians would succeed without any doubt in proposing generic solutions covering most part of the chain. The experience acquired from previous attempts to unify broadcasting techniques, at times disappointing, might allow a better understanding of the wishes of everyone. In this sense, the effort of concentration which is being developed at the European level should continue with the aim that a coherent introduction policy to digital television is put forward. In that process, the service providers will, without any doubt, have the final word but they cannot act alone."

# *PART 2 - THE SERVICE AND IMPLEMENTATION PERSPECTIVE*

Chapter 4

# Implementation issues for digital television services

Terence Long, Terence Long Consultants.

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APPENDIX A

## SUMMARY & CONCLUSIONS

## 1. INTRODUCTION

Advances in digital processing techniques and technology have now reached the point where it is recognised not only that the introduction of digital broadcasting services is, eminently feasible, but that it is also possible to envisage a total conversion of the mixed analogue/digital broadcasting system of today to one that is "all-digital" in its operations in future. It is also recognised that the long-term economic and cultural benefits of such a conversion could be very considerable, viz taking the totality of possibilities of all the media together:

- a very considerable increase in the efficiency of spectrum usage (possibly releasing spectrum for non-broadcast use);
- a vast increase in the quantity and technical quality of services available to the public;
- a very considerable improvement in services to portable receivers;
- new services dedicated to mobile (in-car) reception;
- a considerable reduction in the per-programme transmission costs;
- a considerable increase in flexibility of operations.

While the potential long-term benefits are clear, the difficulties in achieving satisfactory service introduction followed by a smooth transition to "all-digital" operations in each of the television distribution media should not be underestimated. In considering the new service possibilities for the different media summarised below, it would be only prudent to bear in mind the constraints imposed by, for example:

- spectrum availability;
- constraints on transmitted power (EIRP) imposed by mutual interference considerations;
- the cost of transmission and the transmission infrastructure;
- the cost and availability of programme material;
- the cost, availability and desirability of the end user equipment;
- the problems of starting new services using new standards in competition with the established services.

From the discussion below, it will be apparent that the technical and marketing problems to be faced in the terrestrial sector are far more complex than those to be faced in the satellite case. Today the technical problems of digital satellite broadcasting are largely solved and attention is naturally focused more on the marketing of the initial digital services, as well as ways and means of assuring their further evolution. It should be borne in mind therefore, that the degree of attention devoted to terrestrial broadcasting relates to the particular problems of this media rather than reflecting any judgement about its relative importance. The fact is that much of the discussion of terrestrial systems is inevitably somewhat speculative, because in many important technical areas key problems are still under study.

## 2. TECHNOLOGICAL DEVELOPMENTS AND SERVICE POTENTIAL

Recent advances in the techniques of video and audio compression, modulation/channel coding and VLSI production have laid the foundation for the establishment of digital television broadcast services to the home. While the problems of service introduction and evolution are rather different in each of the media, the advances in compression techniques and VLSI are of common benefit in all areas.

### 2.1. Digital video compression

Systems able to reduce the video source bit-rate by 20 times (with no perceptible picture degradation), or by 40 times (with some perceptible degradation on critical picture sequences) have been developed within the ISO/IEC MPEG programme, and these systems are now in an advanced stage of standardisation.

While there remains the need to develop a precise methodology for subjective picture quality assessment following picture compression/decompression, a consensus is emerging to support the view that 40:1 compression will be acceptable for most DTH broadcasting applications involving the normal range of broadcast material viewed in the home environment. This provisional judgement leads to estimates of the digital bit-rates needed to support differing picture quality levels as follows:

| | | |
|---|---|---|
| Limited Definition TV | (LDTV) | ~1.25Mbit/s |
| Standard Definition TV | (SDTV) | ~5Mbit/s |
| Enhanced Definition TV | (EDTV) | ~10Mbit/s |
| High Definition TV | (HDTV) | 18-30Mbit/s |

where LDTV is approximately "home" video recorder quality; SDTV is approximately PAL quality; EDTV is approximately CCIR Rec.601 quality. The HDTV requirement is more complex because it is argued in some quarters that the HDTV 30Mbit/s level may be required for very large home displays of full HDTV quality, should these become available in the more distant future. It should be noted however that, by implication, the 'FCC special panel on Advanced Television' in the USA have accepted a video bit-rate of around 18Mbit/s as suitable for HDTV, and that there will almost certainly be commercial pressures in favour of additional programmes rather than the utmost in HDTV quality. These pressures are likely to consolidate a bit-rate of around 20Mbit/s as the accepted HDTV standard for over-air (DTH) broadcasting.

Commensurate progress has been made on the compression of the associated audio signals with the standardisation, within MPEG, of the Musicam and multi-channel Musicam systems. With Musicam, for example, compact disc quality stereo is maintained at a compressed bit-rate of 256kbit/s.

## 2.2. VLSI

The trends in VLSI production are well established and a generally accepted "rule of thumb" is that "cost of silicon" falls by 30 times every 10 years. Thus flexible performance MAC receivers using highly sophisticated digital processing were produced in volume in 1990 at prices not too dissimilar to their equivalent in PAL. Current estimates for the volume production of MPEG - 2 main profile decoders and associated memory, within MPEG, suggest an initial volume production cost of around 40 ECUS for these components. In the light of these developments Hughes has suggested a "target" price of around 200 ECUS for a volume produced 'receiver' for their planned "movie on demand" digital broadcasting service (other estimates suggest that around 300 ECUS may be more realistic).

Current thinking in Europe on these issues suggests that the target retail price of a complete DTH package for multi-programme SDTV (consumer dish and receiver/decoder), and of the corresponding consumer equipment for cable or SMATV applications should, be less than 600 ECUS at time of introduction, and should be less than 350 ECUS within two years of introduction. Applying the same technological trends to the more complex processing required for HDTV suggests that a similar price structure could be targeted for HDTV receiver/decoders by 1998/99.

## 2.3. Digital satellite television broadcasting

Advances in satellite broadcast systems technology have considerably reduced the transmit power required to achieve service area coverage to small receiving antennas (dish diameter ~60cm.). Today medium power satellites (~50dBW EIRP) operating in the FSS band are providing a range of PAL and MAC services suitable for reception on small receiving antennas. Using similar satellites, digital systems can be engineered to provide useful bit-rates of around 36Mbit/s ruggedly receivable on small antennas throughout the service area -i.e. sufficient bit-rate to support seven SDTV programmes per satellite channel instead of the one obtainable with PAL today. There is no problem of spectrum availability in the FSS band and several medium power satellite systems are planned for introduction in the near future including:

- Astra (1D) & (1E) : each with 18 transponders
- Eutelsat F 1-2-3-4 : each with 9 transponders
- Eutelsat F 60 : with 18 transponders

Each of these satellite systems would be suited to provide a range of digital television satellite services. With suitable multiplexing and signalling the services could include:

- 2 HDTV (18Mbit/s) services per transponder
- re-configurable to 7 SDTV services/transponder
- re-configurable to around 20 LDTV services /transponder
- all channels with digital stereo sound, service identification, conditional access and general purpose data capability

In principle digital television systems engineered to operate with the high power satellites of the BSS band could support higher digital transmission capacity with rugged reception on antennas considerably smaller than for the FSS band. There is however the question of the higher transmission cost per satellite channel to consider, a question that has led to the suggestion that the BSS band should be re-planned in Europe to take advantage of the technological advances made since the BSS band was agreed internationally in 1977 (the WARC '77 plan). As things stand the band is (almost) fully allocated in Europe for national BSS services but the "take up" in terms of operational or planned services is exceedingly low.

## 2.4. Digital terrestrial television broadcasting - the "all-digital" scenario

The development of experimental Coded Orthogonal Frequency Division Modulation (COFDM) systems for Digital Terrestrial Television Broadcasting (DTTB) has demonstrated a potential for future services that extends enormously beyond that obtainable with the PAL and SECAM systems. While studies of DTTB systems are far from complete, there is sufficient evidence to suggest that in an "all-digital" scenario each main transmitter in the terrestrial network, using transmit power levels 10-15 dB down on PAL, could in principle provide service area coverage with a useful bit-rate of 30-45Mbit/s. Thus in the "all digital" scenario DTTB networks should be able to support a range of services to roof-level antennas similar to those potentially available in the satellite sector. Each service would have the additional advantage that a degree of portable reception would be possible and some services could be engineered specifically for reception on portable or mobile (in-car) reception.

Furthermore the inherent properties of COFDM (relaxed protection ratios) together with a judicious use of "dual frequency" and "single frequency" concepts in network planning should allow a four or fivefold increase in the number of channels available for use at each transmitter. Thus with (say) 16 channels/transmitter available it is possible to envisage a mixed package of service types ranging from HDTV services to roof-level antennas (with good portable reception at lower resolution), through SDTV services dedicated primarily to portable reception (but receivable on the main set in the home), to LDTV services dedicated exclusively to mobile reception.

Unfortunately the UHF/VHF spectrum allocated to terrestrial broadcasting is heavily used in most countries in Europe and there is little prospect that clear spectra for "all-digital" operation can be found. It seems almost certain therefore that the considerable potential of the terrestrial system will only be realisable if credible scenarios for the phasing out of PAL and SECAM are identified. Such scenarios might be based upon the natural growth of DTTB, following its introduction in the related ("taboo") channels of the PAL and SECAM networks, or, alternatively, on free market developments, within a framework of agreed technical standards, which involve parallel digital transmission of the main terrestrial programmes in the satellite sector.

## 2.5. DTTB introduction – service area coverage issues

The feasibility of introducing low power DTTB services in the so called "taboo" channels of the PAL/SECAM networks in Europe is under intensive study in a number of European national and international R&D programmes. Given the need to constrain the DTTB transmitted so as to limit perceptible interference into the existing services, the key technical questions concern:-

a) the degree of impairment to the PAL/SECAM services to be allowed (the COFDM to PAL/SECAM protection ratios - especially the co-channel protection ratios)

b) the service area coverage obtainable (w.r.t. PAL/SECAM) for different modulation levels (QPSK, 8PSK, 16QAM, etc.) and their associated error management schemes.

While current judgements on a) vary to some degree from country to country, the more serious disagreement relates to b) where some experts argue that the large variations in received signal strength associated with the precise locations of roof-level receive antennas (the location factor) are such as to severely reduce the DTTB coverage predicted in some projects (e.g. the SPECTRE and DIVINE projects) owing to the sharp failure characteristics of digital systems and the difficulty therefore of defining a precise area boundary. While it is accepted that there is a need for a new definition of what is meant by edge - of - area coverage (with analogue as well as digital systems), the arguments presented in the Annex to this report (including the option of "step-wise graceful degradation"), suggest that coverage predictions based upon detailed terrain data, median field strengths and a location factor of 50% (as for SPECTRE and DIVINE) are realistic.

## 2.6. DTTB introduction – spectrum availability and service quality limitations

Studies to ascertain the channel availability and likely population coverage for DTTB in each European country are at a relatively early stage, and few results have been published to date. The situation varies considerably from country to country, being perhaps most difficult in Italy and easiest in Scandinavia owing to the differing degrees of low power PAL/SECAM relay stations in use in each region. However reports from studies that have been published to date suggest that there is reason to be optimistic about the technical possibilities. For example:-

- The results of studies in the UK suggest that there are sufficient usable main transmitter locations to provide four or five channels of DTTB to 80% of the UK population. Each channel could be of EDTV quality with coverage matching that of the PAL service. Coverage would be patchy however with parts of the south coast receiving little or no coverage.
- Results of studies in France (using an ultra conservative location factor of 90%) suggest that population coverage for four EDTV services would vary from 52% to 69% respectively across the four national networks.
- One or two HDTV channels of wide area coverage are reported as feasible in Sweden. (with somewhat reduced PAL protection ratios).
- Good coverage predictions would also be expected in Germany provided uhf channels 61 to 69 (currently used for other purposes) were to be made available for DTTB. The relative freedom from interference constraints that these channels could provide might enable some services to be of HDTV quality.

In summary, while there are constraints on the quality and number of services possible during the DTTB introduction phase in Europe, there is likely to be sufficient spectrum available for the start of technically feasible DTTB services in many countries using the existing main transmitters alone. In most countries the services could be of EDTV quality and there is potential in some countries for HDTV services, although these might be of rather limited area coverage.

### 2.7. Cable television

Noting the continuing up-grading of network performance and capacity, and the possibilities for high data rate transmission in the 7 and 8MHz cable channels, the potential performance of the European Cable Networks may be summarised:-

In principle, the Cable Television Networks have the potential to be able to distribute the full range of digital television services deliverable, in future, by satellite and terrestrial broadcasting.

However a number of practical problems will need to be urgently resolved if the networks are to be made ready for the possible launch of satellite services by mid 1995 (as discussed in 3.3 below)

## 3. IMPLEMENTATION – PROBLEMS OF SERVICE INTRODUCTION AND EVOLUTION

### 3.1. Digital satellite television broadcasting

It is evident that, from a technical perspective, the way is now clear for the implementation of digital satellite services in the FSS band if the market so demands. The digital systems have the potential to significantly reduce the transmission costs of conventional quality television by offering more services per satellite channel or, alternatively, they can support higher - definition services. There will initially, of course, be few digital receivers in the field and it could take a long time to build up a new audience. Currently satellite coding/modulation equipment of broadcast quality offering a range of service options is being marketed, and the European satellite service providers have made it clear that they wish to be in a position to launch new digital services by mid-1995. As reported, the initial markets will be for multiple SDTV or LDTV services such as near "video on demand" - that is services requiring receivers with a 36Mbit/s demodulation and demultiplexing capability but with subsequent processing limited to 5Mbit/s or less.

While it will clearly be of considerable importance that receivers/decoders for the new services are very affordable, there will be trade-offs to consider which might discourage opting for the cheapest possible price. The initial, and later costs of sophisticated FSS band DTH receivers of different types will almost certainly depend considerably upon the view that manufacturers take of initial market size and of its potential growth. Related to these considerations are judgements about the importance of harmonising (where sensible) with standards to be used in related media (for digital cable and terrestrial television distribution, for example) Of particular importance perhaps is to try to ensure that first - generation receivers in the field will be compatible with future service upgrades and do not create barriers to the launch of new higher quality services such as HDTV. In considering these matters in this report it is concluded that very simple, low cost, additions can be made to first generation receivers that will ensure that there will be no serious barriers to the evolution to digital satellite HDTV services. There are however questions related to the choice of the HDTV standard (hierarchical or fixed format) that warrant detailed study if the best long - term arrangements are to prevail.

It remains to be seen of course whether the new digital satellite services can be successfully launched in the face of commercial competition, and whether they will advance sufficiently to influence developments in other areas.

In summary, there are no significant technical barriers to the launch and future evolution of digital satellite services in the FSS band. Services at serval different quality levels including HDTV, can be supported. Spectrum is available and several medium power satellite systems suitable for digital television broadcasting are planned to come into operation in the near future The per - programme transmission costs will be relatively low, but the success, or otherwise of the service will critically depend upon the attractiveness or not of the programmes on offer, receiver/decoder sales and audience size.

### 3.2. Digital terrestrial television broadcasting – the complex problems of transition to "all-digital" operation

The characteristics of UHF/VHF terrestrial broadcasting make the DTTB situation, in general, far more complex than for the satellite sector. In addition to the vagaries of propagation (especially in the edge of area refraction zone) and the shortage of usable spectrum, there are also the problems of regional services, portable reception and international interference/frequency planning co-ordination to take into account. However where DTTB services can feasibly be initiated in parallel with PAL and SECAM the technical costs of "start-up" should be relatively low, especially if sharing the existing transmit antenna with the conventional services proves feasible. The new transmitters will be relatively low powered (25 to 30 dB down on PAL/SECAM), and much of the conventional service infrastructure could, in principle, be used to support the new digital service. This would be particularly appropriate in a "simulcast" scenario whereby the main PAL/SECAM services are duplicated in (say) digital EDTV to make major savings in the all-important area of programming costs. There would remain the additional costs of re-mastering material for the 16:9 EDTV picture format, and the cost of additional programmes when the system is re-configured to provide additional channels (though this could be largely time shifted "repeats" and archive material).

Assuming that the long-term aim, at least of administrations and the public, is eventually to realise the considerable benefits of an all-digital system, the key questions raised in this scenario (and in alternative scenarios discussed in the report) are:-

- how will the "take-up" of the digital services grow to the point where they can be considered as the primary terrestrial services to allow a firm date to be announced for the switching off of PAL/SECAM and the "switch-over" to an "all-digital" system?
- what constraints might usefully be applied to the definition of an "introduction" DTTB receiver standard to ease the transition to the final system standard(s)?
- who will pay for the considerable engineering costs that will be needed in preparation for, and following, the "switch-over" to "all-digital" operations?
- how will a possibly growing digital satellite sector influence the attitudes and decisions of the terrestrial broadcasters?
- how will a smooth transition to "all-digital" be achieved without a major effort in international co-ordination, bearing in mind that co-channel interference into PAL/SECAM will remain as a problem till all of Europe has changed over?

Given certain assumptions, partial answers to some of these questions can be suggested. If for example the programming strength of the traditional broadcasters kept them in an essentially dominant position in the broadcast market, this might allow the DTTB market to grow slowly but steadily in spite of growing competition from the satellite sector. In this scenario:

- growing DTTB receiver sales in the PAL/SECAM main-set "replacement market" would begin to erode PAL/SECAM as the predominant system for viewing on the mainset in the home. The prospective purchaser would be somebody wishing to continue viewing the high-production-value "favourite" programmes only available terrestrially, but with the added advantage of higher quality and a modest increase in programme choice.
- as the DTTB market grew, measures could be taken to expand DTTB coverage w.r.t. PAL/SECAM and new receiver features might be introduced.
- in this scenario the choice of an "introduction" receiver standard based upon 16QAM rather than QPSK would certainly assist the transition and prepare the ground for the "switch-over" to new standards (if that point were ever to be reached!).

However, even within this scenario which many would argue is most problematic, a totally satisfactory answer to the many complex questions posed seems unlikely without a great deal of further study and some rather imaginative new ideas. Nevertheless the strength of the traditional broadcasters, hold on the market should not be underestimated.

At first sight an alternative strategy of encouraging the main terrestrial broadcasters to simulcast their programmes by satellite seems most unsuitable, because it is hardly to be expected that they would wish to encourage a development that would enhance the attractiveness of the satellite sector and by doing so assist their "competition". However, if counter to the earlier assumption the digital satellite television sector develops successfully and begins to build audiences for "higher definition" movies and sports, as well as "video on demand", the challenge to the terrestrial broadcasters might be such as to require a relatively fast and vigorous response. In this scenario it would not be surprising if the terrestrial companies countered by launching their own digital satellite services including in a multi-programme package higher-definition versions of their best terrestrial programmes, as well as programmes from their extensive archive stores. It is scenario-scripting along these lines that has given some support to the idea that the terrestrial sector might decline in relation to the satellite sector, over time, and that due weight should therefore be given to dedicated portable services in the current DTTB debate.

In summary, there are no major problems to be overcome for the introduction of DTTB services in parallel with the conventional services where suitable spectrum is available. Market-led scenarios that might lead to the eventual "switch-over" to "all-digital" operation are, however, highly speculative and very uncertain. Whatever the value of these speculations it clear that if the long term potential of DTTB is to be realised in one form or another a great deal of further thinking, discussion and study will be required in the years ahead.

### 3.3. Digital cable television distribution

While no problems of a fundamental nature have been identified in preparing the cable networks for the digital distribution of potential satellite and terrestrial broadcast services, there are a number of practical matters that warrant serious attention. Of particular importance is the probable need for a new, cable-TV-specific, high-bit-rate digital demodulator for the home receiver/decoder. The specification for such a receiver will depend upon trade-offs between the cost of the demodulator (16, 32 or 64QAM), and costs in network transmission capacity (7/8MHz or higher channel separations), trade-offs yet to be examined closely and therefore far from resolution (at least in any of the standards groups considering these matters.) In summary there is a good deal of experimental and field trials work needed in the cable television area before the costs involved in preparing them for digital operation can properly be assessed. At present it would appear that insufficient attention is being given to this important area in those groups dealing with standardisation and the organisation of European R&D.

### 3.4. Standardisation and the technical feasibility of service introduction

Currently major efforts are underway, in ISO-IEC MPEG and in EBU/ETSI JTC, to achieve a standardisation framework for the future development of European digital television broadcast systems. While actual service start dates cannot be accurately predicted, estimates of when services might be technically feasible (based upon the trends in technological development) can reasonably be made. The figure below provides a summary of what is likely to be feasible, together with an indication of the key specification milestones involved.

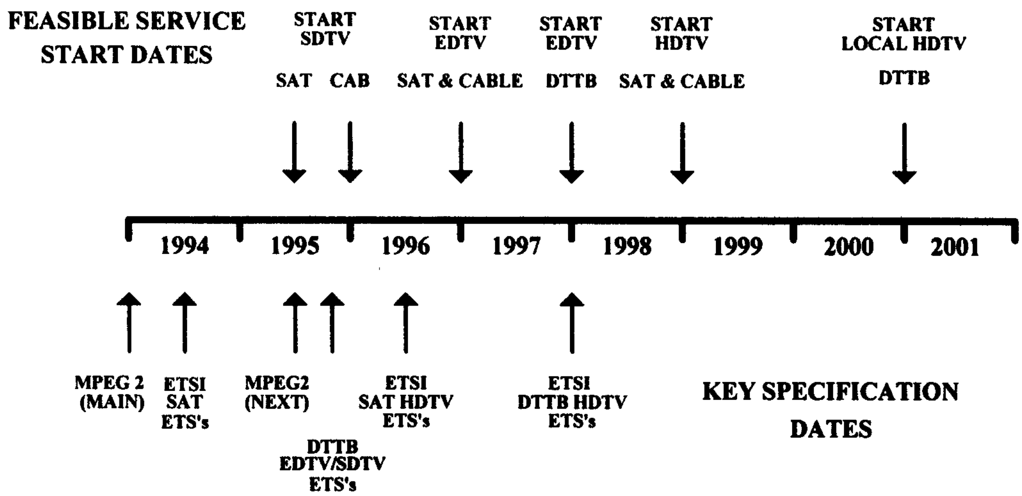



*Figure 1.1: Key specifications and service feasibility timescales*

## 4. CONCLUSIONS

1. In technical terms, the use of digital television broadcasting techniques offers enormous advantages for each of the broadcast media when comparisons are made with the analogue PAL and SECAM systems in use today.

2. In the satellite broadcasting sector the processes of technical development and standardisation have reached a sufficiently advanced stage to make the launch of new digital television services, by mid 1995, technically and economically feasible. While the initial system specification may be limited to the purposes of multi-programming, there would appear to be no technical barrier of consequence to prevent such services developing to the point of being able to offer the full range of technical quality options.

3. Digital cable television systems have not yet been developed to the point where it is clear that they will be able to be adapted economically to distribute the new digital satellite services in the mid 1990s. While there would appear to be no fundamental problems to resolve a good deal of experimental/field trials work remains to be done. Of particular

importance is the need to define specifications which will enable the production of economically viable end-users equipment. Given the possible importance of cable distribution to the success, or otherwise, of the new satellite services a more urgent approach to these problems is indicated.

4. While the potential gains to be had from converting the terrestrial television system to "all-digital" operation are very great indeed, scenarios that might lead naturally to the necessary phasing out of PAL and SECAM seem very problematic. A good deal of further thought and discussion is therefore indicated if it is to be assumed that national and European strategies will be required to encourage progress in this most important area. The need to find solutions which will enable the full potential of the terrestrial networks to be "unlocked", over time, remains therefore the greatest single challenge in the digital television field.

## REPORT

## 1. INTRODUCTION

It is very clear that the world of telecommunications is in the middle of a revolution that will eventually lead from the mixed analogue/digital situation of today to "all-digital" operation in the future. In broadcasting, digital processing is used increasingly in television studios, for link applications and in the domestic receiver. It is clear also that developments in digital television broadcasting technology now underway could lead, eventually, to the conversion of the entire broadcasting system to "all-digital" operation. Currently a high proportion of broadcasting R&D in Europe and the USA, and most of the long/medium term planning term planning for new services is being based upon an all-digital approach.

In the USA for example, 4 of the 5 proponent systems for "Advanced Television" before the Federal Communications Commission (FCC) are digital. These systems have been extensively tested and evaluated, and the FCC Special Panel has recommended that further development of systems should be based upon the digital systems only. Subsequently the digital proponents have agreed to merge their development efforts and create a single system for further evaluation by the FCC in the spring of 1994. The new system proposal will be entitled the HDTV " Grand Alliance " Proposal *[1]*.

In Europe a number of major digital television R&D projects are well underway including for satellite transmission:

- RACE 2064 (FLASH TV)
- RACE 2075 (HDSAT)

and for terrestrial broadcasting:

- HDTV-T (the German National Project)
- HD DIVINE (the Swedish National Project)
- Vidinet (FI/DBP Telecom)
- Diamond (Thomson CSF/LER)
- Sterne (CCETT)
- Spectre (ITC/NTL)
- RACE 2082 Digital Terrestrial Television Broadcasting (dTTb) (European Consortium)

Additionally a number of private ventures involving digital satellite television broadcasting have been reported, including:

- The Hughes "DirectTV" project, aimed at delivering more than 100 separate programme channels simultaneously to the home and cable head-ends.
- The development of "System 2000" for digital satellite television broadcasting by NTL (UK)
- The NTL link-up with Scientific Atlanta to market digital broadcasting compression systems world-wide

- The collaboration between News Datacom and CANAL PLUS to develop multi-channel digital satellite television (similar to Hughes) in Europe over the next few years (a possible service launch with ASTRA 1-E).

Most of these ventures envisage using source coding/decoding technology being developed under the ISO-IEC MPEG group, which has now reached an advanced stage in the standardisation process.

In considering the service potential in the different media, a fundamental distinction should be made between the satellite and terrestrial broadcast situations. In the satellite case, spectrum is available for new digital services in both the Broadcasting Satellite Service (BSS) and the Fixed Satellite Service (FSS) bands, and digital broadcast systems, able to offer a wide range of services (including HDTV), are in an advanced stage of development.

In the terrestrial case it is most unlikely that new spectrum will be made available for DTTB, and services will have to start-up by sharing the allocated spectrum with the existing PAL and SECAM services. However if the systems employed in the introduction phase of DTTB are flexible and "upwards compatible" it is possible to imagine an evolving situation which allows today's PAL and SECAM systems to be phased out.

For the terrestrial case then it is useful to define two phases:

i) A long-term scenario in which, following the phasing-out of PAL and SECAM, "all-digital" operation will be possible and the full range of DTTB services can be realised.

ii) The medium term transition phase, in which DTTB services are introduced and operated in parallel with PAL and SECAM and the service possibilities are more constrained.

Against this background it is the purpose of this note to summarise the potential of digital broadcasting techniques, and review the current state of development. The summary of potential is given in Sections 2 to 4 of the report under the headings:

- The potential for digital television broadcasting – the service options
- Implementation – the problems of service introduction and evolution
- International standardisation

The review of the state of the art in digital television development for the different media is given in the Annex under the general heading "System Characteristics".

## 2. DIGITAL TELEVISION BROADCASTING - THE SERVICE OPTIONS

### 2.1. General considerations

The use of digital compression in video source coding, combined with appropriately chosen channel coding modulation and multiplexing schemes, will provide the broadcaster with a range of service options that vastly exceeds those available today using the PAL and SECAM systems - whether the broadcast medium is satellite, cable or terrestrial.

In considering these new service possibilities, however, it is always necessary to bear in mind the constraints imposed by, for example:

- spectrum availability;

- mutual interference in an efficient frequency plan;
- limitations on transmitter power (EIRP) imposed by mutual interference considerations;
- the cost of transmission and the transmission infrastructure;
- the cost and availability of programme material;
- the cost, availability and desirability of the end-users' equipment;
- the problems of starting new services using new standards in competition with the established services.

The technical trade-offs required to optimise the systems in the light of these considerations are numerous and transmission-media-dependent. Of primary importance, however, is the trade-off between service coverage and service quality - a trade-off that, in digital systems particularly, depends upon the transmission power available to the channel. (In general, the higher the EIRP available to serve the service area, the higher the useful bit-rate).

In over-the-air systems, next in importance are the assumptions to be made with respect to the planned reception conditions - the receiver antenna characteristics, noise figure, linearity, etc. which will qualify the quality and availability of the service under various propagation conditions.

In terrestrial over-the-air systems, where portable and mobile services are feasible. and where received field strengths are highly variable, there are further questions to be answered before full assessment of service potential can be made; questions such as:

a) Is the primary service aimed at:
   - Fixed roof-level antennas of standard design?
   - Portable receivers with low-performance antennas?
   - Mobile (in-car) receivers?

b) Is there a secondary service to take into account? For example, portable reception of reduced coverage in parallel with a "full" coverage service for fixed roof-level antennas.

c) Is there the need for a graceful failure characteristic to take account of the vagaries of propagation at the edge of service area ?

In proceeding to an assessment of the service potential of digital television services, we must define not only the picture and sound quality in the centre of the coverage area but also the "quality of service", over time at the edge of the service area, in terms of percentages of reduced quality (graceful degradation) and loss of picture (outage times or service "availability" percentages).

## 2.2. Compressed digital video and audio quality - the subjective assessment of digitally compressed picture quality

Following bit-rate compression, picture quality is highly dependent upon the information content of the scene. In a properly designed system, almost perfect pictures are obtained from most scenes, but a softening of resolution occurs on scenes whose "information content" calls for bit-rates exceeding the capacity of the channel.

The frequency and duration of the periods of reduced resolution will depend on the maximum resolution of the source and reproduction system, the efficiency of the compression algorithm and the "statistics" of the varying information content of the picture sequence.

At present we have no scientific assessment of what the viewer will make of this situation. If, for example, 96 per cent of scenes are reproduced with full high definition resolution and 4 per cent with reduced resolution, will the viewers' enjoyment be unacceptably disturbed? If so, would it not be better to provide a service of enhanced definition which only suffers a reduction of resolution in 1 per cent of scenes?

What surely matters is the overall impression the viewer will have gained at the end of a typical programme, and it can be argued that the assessment methodology should be based upon this principle. It is also fundamentally important that the information statistics of the picture sequences used in the compression codec assessment are typical of the programme type, as established by an extensive analysis of over-the-air programming as transmitted today.

A practicable method of low-bit-rate codec assessment that would appear to meet the above requirements described by Lodge *[2]* is being studied within RACE dTTb project. Until the results of this study are complete, however, and a possible new methodology has been agreed, we must rely upon the assessment of engineers who are experienced in the digital television art, derived from earlier subjective methodologies. Here, although there is some divergence of meaning and definition, there is, nevertheless, sufficient common ground for it to be reasonably clear what is meant by the different quality levels available from todays' state-of-the-art compression technology. For example, based upon an assessment of the ETSI 34Mbit/s codec and MPEG results, the European Working Group on Digital Terrestrial Broadcasting (WGDTB) [3] has defined four quality levels (and by implication the compressed digital bit-rates needed to support each of these levels) as follows:

- HDTV Quality, where the potential exists for the delivery of a picture which is subjectively identical with the interlaced HDTV studio standard. Quality shall remain consistent with this for a given proportion of television programme material (where this is a percentage in the high nineties, but is yet to be identified).
- EDTV Quality, where the potential exists for the delivery of a picture which is subjectively indistinguishable from the 4:2:2 level of CCIR Rec. 601. This quality shall be maintained for a given proportion of programme material (which will be a percentage in the high nineties, but has yet to be identified).
- SDTV Quality, where the quality is approximately equivalent to that of current PAL or SECAM. This equivalent quality may be achieved from pictures sourced at the 4:2:2 level of CCIR Rec. 601 and subjected to processing as part of the bit-rate compression. The result should be such that, when judged across a representative sample of programme material, subjective equivalence with PAL and SECAM is achieved.
- LDTV Quality, where the quality is equivalent to that obtainable from the MPEG-1 system, which operates at a source resolution approximately 1/4 of the 4:2:2 level of CCIR Rec. 601. This quality is considered by some to resemble that of VHS (albeit over a relatively low proportion of programme material.

The compressed video bit-rates needed to support these quality levels can also be derived from *[3]* to be:

For:

| | | |
|---|---|---|
| HDTV | ~ | 40Mbits/s |
| EDTV | ~ | 10Mbits/s |
| SDTV | ~ | 5Mbits/s |
| LDTV | ~ | 1.25 - 2.5Mbit/s |

Using a similar basis for its assessment of the performance of state-of-the-art digital compression, the Independent Television Commission *[4]* suggests that the following guide to compressed bit-rate/quality level relationships:

| | |
|---|---|
| Studio high definition (HDTV) quality (1,250 lines) with no perceptible degradations | = 40Mbit/s |
| HDTV quality with some distortion on critical scenes | = 20Mbit/s |
| 625 line studio-quality with no perceptible degradation | = 10Mbit/s |
| 625 line quality with some distortion on critical scenes | = 5Mbit/s |
| Reduced quality (312 line) with no perceptible degradations | = 2.5Mbit/s |
| Reduced quality (312 line) with some distortion on critical scenes | = 1.2Mbit/s |

In relation to this table, the ITC further suggests:

> "As a guide for comparison, the fourth example here (5Mbit/s) is often considered as roughly equivalent in quality to PAL and the sixth (1.2Mbit/s) as roughly equivalent to home video (VHS) quality (although the nature of the distortions which occur in the digital and the analogue systems is very different in both cases). As an indication of the amount of compression which is implied in the above figures, an initial representation of 625 line studio-quality pictures in digital form takes up around 200Mbit/s and so with no picture degradation the compression factor is around 20:1, while with some picture distortion on critical scenes a compression factor of around 40:1 can be achieved. On the sound side, the capacity required for a compressed high-quality digital stereo signal is around 0.25Mbit/s".

Taking the two assessments together and noting that

a) the maximum requirement for HDTV viewing at the minimum distance of 3H is about 1000 lines *[5]*

b) a degree of impairment to the HDTV picture in the home will be acceptable, and that an optimum design would aim to achieve the agreed degree of impairment

c) as seen today, prospects for the development of a "true" HDTV home display in the near future are rather poor (see Table 2.1) *[6]*

| | LCD | PDP | EL | Flat CRT |
|---|---|---|---|---|
| Display quality | Already solved | Needs more R&D | Already solved | Already solved |
| Response speed | Needs more R&D | Already solved | Needs more R&D | Already solved |
| Large size | Difficult | Already solved | Difficult | Very difficult |
| Life time | Already solved | Needs more R&D | Already solved | Already solved |
| Contrast | Already solved | Already solved | Difficult | Already solved |
| Luminance | Already solved | Already solved | Difficult | Already solved |
| Colour | Already solved | Needs more R&D | Difficult | Difficult |

Key: Already solved — Needs more R&D — Difficult — Very difficult

*Table 2.1: Potential evolution of flat panel displays for HDTV*

It would seem useful to define 40Mbit/s as a studio level. For the purposes of simplicity in this Technical Annex then, the working assumptions for quality levels to be delivered to the home are as shown in Table 2.2.

| DTH Broadcast Quality Level | Compressed Digital Video Bit-Rate |
|---|---|
| HDTV | ~ 20Mbit/s |
| EDTV | ~ 10Mbit/s |
| SDTV | ~ 5Mbit/s |
| LDTV | ~ 1.25Mbit/s |

*Table 2.2: Working assumptions quality level versus compressed video bit-rate requirement*

### 2.3. Satellite broadcasting - service potential

The characteristics of digital satellite broadcasting systems are reviewed in Section A1.4 of the Annex. Based upon the discussion in A1.4, the useful bit-rates obtainable from existing and planned satellite systems in digital operation may be summarised as follows:

| Satellite Category | Useful Bit-Rate | 'Dish' Diameter (cm) | |
|---|---|---|---|
| | | Beam Centre | - 3dB. Contour |
| High Power (BSS) | 45Mbit/s | < 40 to 45 | < 40 to 65 |
| Medium Power (FSS) | (41 to 45)Mbit/s | 65 to 110 | 90 to 150 |
| | ~ 36Mbit/s | < 60 | < 85 |

*Table 2.3: Available video bit-rates - digital satellite systems*

Thus, allowing a total of ~0.5Mbit/s for stereo sound, SI, CA and general purpose data, a system operating in the BSS band could support:

- 2 HDTV services, or
- 4 EDTV services, or

- 8 SDTV services, or
- 25 LDTV services

With suitable service multiplexing and signalling arrangements, the service package can be configured and re-configured to any combination of HDTV, EDTV, SDTV and LDTV with as many sound and data channels from the 45Mbit/s MUX, as may be needed.

For example, a service operator might operate 7 SDTV channels plus 3 LDTV channels as standard, but could switch to 2 EDTV; 3 SDTV; 5 LDTV services for peak evening viewing where the EDTV channels carry the prime time movies and sports events.

At other parts of the day, during normal business hours for example, there might be a market for high bit-rate data accompanying certain LDTV business channels. Dynamic switching to re-configure the programme service is easily accomplished with properly designed digital systems and can be achieved on a programme-by-programme basis or, if required, within programme transmission - from full picture display to text with a "windowed" quarter sized picture for example. Such facilities will be particularly advantageous with certain types of low-cost programming - Open University lectures and business presentations being only two such examples.

A similar range of services will also be available from medium-power satellites in the FSS band, albeit with a requirement for larger diameter receiving antennas.

Assuming that the public satellite operators and the "environment authorities" alike will all prefer the receiving "dish" diameter not to exceed about 60 cm in the central area, then the somewhat reduced useful bit-rate of around 36Mbit/s will mean either a small reduction in the range of services or, as seems more likely, some reduction in the quality level for each service, i.e. 18Mbit/s for each HDTV channel rather than 20Mbit/s.

## 2.4. Digital terrestrial broadcasting - service potential

The characteristics of a digital terrestrial television broadcasting (DTTB) are reviewed in Section A1.5 of the Annex below. The situation here is far more complex than that for the satellite broadcasting case because:

a) While in an "all-digital" scenario the potential of DTTB is similar to that described for satellite systems, currently the introduction of the highest bit-rate DTTB services is constrained by the presence of comprehensive PAL/SECAM services in the allocated uhf/vhf bands. Thus a distinction needs to be made therefore between the introduction, transition and final phases of DTTB when describing the potential features of the system.

b) There is the need to consider how systems can make the evolutionary step between the constrained performance of the transition phase and the full performance of the final phase of DTTB.

c) There is the probable need to provide for a graceful failure characteristic.

d) Unlike satellite television services, terrestrial television services are receivable on portable and in-car receivers, either as secondary services (the norm for portables today) or as primary services where the digital system is tailored to suit either portable or mobile reception requirements.

e) The complex problems posed by portable and mobile reception are in the relatively early stages of study in the R & D projects, and it is therefore difficult to predict the relationship between performance and coverage at the present time.

It is nevertheless possible to make a reasonable assessment of the service potential for the different types of DTTB service on the basis of the discussion of Section A1.5 as follows:

### 2.4.1. DTTB in the longer term: service potential in the "all-digital" scenario

In this scenario it is assumed that the new digital network is essentially based upon the old PAL/SECAM main transmitter network and that, following the phasing out of the PAL/SECAM services, the main transmitters have been re-engineered to suit digital requirements, possibly in the form of mixed "single" and "dual" frequency networks allowing (say) ten to twenty channels at each main site, each of sufficient high transmit EIRP to support, at the highest level, 64QAM modulation receivable on standard roof-level antennas throughout the transmitter service area. (i.e. transmit power about 10dB lower than is used for PAL or SECAM).

At these power levels the DTTB system may be engineered for the different types of primary and secondary service shown in Table 2.4

| Primary Service | Useful Bit - Rate | Coverage | |
|---|---|---|---|
| | | primary service | secondary service |
| Fixed Reception (roof-level antenna) | 30 - 45Mbit/s* | Full area (HDTV → LDTV) | Reduced area to portables |
| Portable Reception (stub antenna) | ~ 15Mbit/s | Approaching Full area (LDTV → EDTV) | Full area (Fixed antennas) |
| Mobile Reception (in-car antenna) | ~5Mbit/s | Similar to DAB | —— |

*Depends upon details of the configuration - see below

*Table 2.4: DTTB coverage – fixed, portable and mobile reception*

The range of services potentially available at these bit-rates depend to some extent on the balance of requirements seen by the operator. The possibilities discussed in the Annex are briefly summarised below.

<u>Services for fixed reception (roof level antennas)</u>

Configuration A): Fixed Format Coding and Modulation (64QAM)

With around 45Mbit/s of useful bit-rate each transmit channel could support a range of primary services identical to those obtainable from the BSS satellite systems.

The system would be reconfigurable as in the satellite case.

A range of portable receivers would be available but portable reception would be somewhat "patchy", as is the situation today.

There might be a considerable number of locations where no reception is possible or the "outage" time is high.

Configuration B): Hierarchical Source and Channel Coding with nested lower Level Modulation (64 and 16QAM)

A somewhat reduced range of services to roof-level antennas compared with A), depending primarily on the "weighting" given to the lower modulation levels intended for portable reception.

For example:

- 1 HDTV and 1 EDTV service per channel
- Dramatically improved coverage for portable reception at the SDTV and LDTV quality levels
- Very good graceful degradation characteristics
- Re-configurable to the same modes as configuration A

A feature of the system is that it supports a range of receiver options for fixed and portable reception and thus provides the consumer with a wider range of choice in terms of quality and price.

Services for portable reception

With 15Mbit/s useful bit-rate, the system could support:

- 3 SDTV services per channel or
- 8 LDTV services per channel

It would also be re-configurable to suit changing business needs.

The effects of multipath would mean reception would remain somewhat "patchy" - especially towards the edge of service area.

Services for mobile reception

With 5Mbit/s bit-rate the system would almost certainly be configured to provide LDTV services, possibly with mono sound only as the standard mode. Hence 3 or 4 LDTV services per channel.

Possible Service Mixes

| Channels | Possible Services |
| --- | --- |
| 1 – 8 | 8 HDTV + 8 EDTV (Fixed reception)<br>(with good SDTV portable reception) |
| 9 – 12 | 12 SDTV portable services<br>(Full reception on Fixed antennas) |
| 13 – 16 | up to 16 LDTV Mobile Services |

*Table 2.5: Illustration of possible service mixes*

With 10 to 20 channels per transmitter potentially available, it is possible to imagine that each of the above types of service will find its place in the mix of services on offer. For example, if 16 channels per transmitter were available, and the market so demanded, they could be organised as shown (purely for illustration) in Table 2.5.

Thus, the potential for a dramatic increase in the number and range of services in the long term "all digital" scenario is enormous. While a good deal further work is yet to be done to fully prove the feasibility of the techniques involved (especially in the area of frequency planning), the greater task is that of dealing with the problems of DTTB start-up in the presence of PAL and SECAM, where the problems are more complex and performance considerably constrained.

### 2.4.2. Service potential in the introduction and transition phases of DTTB

In this scenario it is again assumed that the digital transmission network will be based upon the PAL/SECAM main transmitter network with transmission from the same masts using the same antennas, or antennas mounted in close proximity. It is also implicit that the usable frequency channels will be based upon the "related" frequencies that are unusable at a particular transmitter in the particular PAL/SECAM network. In this situation the maximum EIRP is limited to a value typically 24 - 30dB below the analogue service EIRP (so as to avoid significant interference into the existing services). At the lower power level more typically available (-30dB), wide area coverage can only be obtained with QPSK or 16QAM with substantial forward error correction (FEC). Where -24dB is usable, 16QAM with lighter error protection can give higher quality results. Several examples which give estimates of coverage for different schemes are worked out in Section A1.5.3 of the Annex.

The following summary of potential performance for roof-level services is based upon the examples of A1.5.3:

a) Transmit Power -30dB w.r.t. PAL/SECAM: QPSK, or 16QAM with 1 in 2 FEC
   - 1 EDTV Service per channel with graceful failure to LDTV
   - Re-configurable to 2 SDTV Services or several LDTV Services
   - Considerably improved portable reception at the LDTV level

b) Transmit Power -24dB w.r.t. PAL/SECAM: 16QAM Fixed Format

- 1 HDTV Service per channel (no graceful failure characteristic)
- Re-configurable to provide 4 SDTV or a greater number of LDTV Services
- Poor portable reception

c) Transmit Power -24dB w.r.t. PAL/SECAM: 16QAM Hierarchical Format

- 1 HDTV Service of reduced coverage per channel (~70% service area radius)
- EDTV to full area coverage
- SDTV to well beyond service area
- Good portable reception
- Re-configurable as for b) above

For Services dedicated to Portable Reception:

d) Transmit Power -30dB w.r.t. PAL/SECAM: QPSK with 1 in 3 FEC

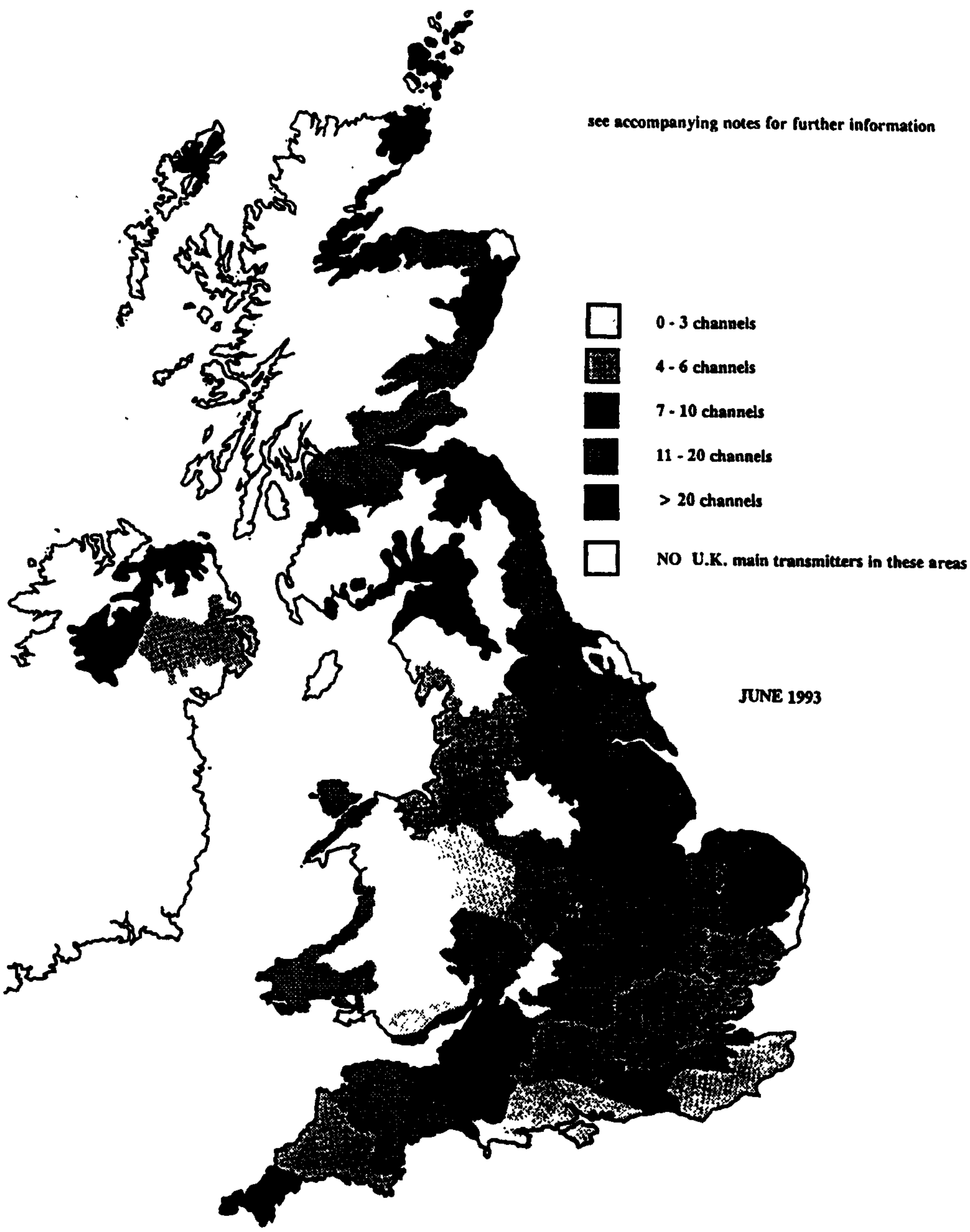
- 1 SDTV Service per channel with relatively good performance in the inner service area
- Re-configurable to 4 LDTV Services
- Reception remains "patchy" but considerably better than PAL

Channel availability

The situation in Europe varies from country to country, spectrum occupation for PAL being most intense in Italy and least perhaps in Scandinavia. Studies of DTTB feasibility are at an early stage in most countries with few results published. However, studies aimed at forming a first assessment of the area and population coverage possible with DTTB, EDTV and HDTV services have recently been reported from France *[7]* Sweden *[8]* and the UK *[4]*. While the assumptions, methods and criteria are somewhat different in each of these studies, (especially regarding the question of location factor) and noting that there is as yet no agreed international set of criteria to apply, the conclusions from the studies nevertheless provide a reasonable picture of the likely situation in the introductory phase of DTTB.

Summary of the situations in the UK, France, Sweden and Germany

The UK - The most extensive studies to date have been carried out in the UK under the SPECTRE project where, using sophisticated terrain-based computer techniques, supported by field trials, a fairly comprehensive picture of DTTB coverage possibilities has been built up over the past eighteen months. The results obtained suggest that there are sufficient usable main transmitter locations to provide four or five channels (supporting EDTV services of full coverage) to around 80% of the UK population. The general coverage situation, however, is "patchy", as illustrated in Figure 2.1.




Source: Independent Television Commission

*Figure 2.1: Potential DTTB coverage from transmitters at the main sites in the UK*

France - Tables 2.6 and 2.7 taken from *[7]* provide a first assessment of likely population coverage for a) a stand-alone EDTV system and b) a stand-alone HDTV system respectively using the main stations in each of the four UHF networks in France. It should be noted that the location factor used in these studies was more severe than that used in the UK studies (90% compared with 50%).

| FRANCE: | Network 1 | Network 2 | Network 3 | Network 4 |
|---|---|---|---|---|
| EDTV population coverage - Fixed reception | 69% | 64% | 60% | 52% |
| EDTV population coverage - Portable reception | 25% | 22% | 20% | 19% |

*Table 2.6: Population coverage – EDTV services (France)*

| FRANCE: | Network 1 | Network 2 | Network 3 | Network 4 |
|---|---|---|---|---|
| Coverage - Fixed reception | 51% | 41% | 34% | 30% |

*Table 2.7: Population coverage HDTV services (France)*

Sweden - Studies in Sweden are primarily aimed at achieving HDTV coverage using 64QAM. While no details of transmit power/channel use are available, the feasibility of one or two HDTV channels is reported.

Germany -While the channels currently allocated for the PAL services are heavily used and the DTTB possibilities therefore more constrained than in the UK, France or Scandinavia, there is the possibility of obtaining UHF channels 61 - 69 (currently used for other purposes) in the medium term. Somewhat higher transmit powers will be permissible on average than is the case elsewhere, because the interference-into-PAL criteria relates to more-remote services in neighbouring countries.

In summary, while there are constraints on the quality and the number of services likely to be possible in the DTTB introduction phase in Europe, there is likely to be sufficient spectrum available for the of technically feasible DTTB services in many countries using the existing main transmitters alone. With further engineering and changes in emphasis during transition phase the of DTTB it should be possible, in principle, to create the framework for the eventual phasing-out of the PAL/SECAM networks. There are however numerous implementation problems to be considered before clear national or European strategies can be established for DTTB, given an assessment of likely developments in the satellite and cable sectors.

## 2.5. Digital television distribution by cable – service potential

The characteristics of typical existing cable systems are described in Section A1.6 of the Annex. On the basis of the theory and experimental investigations reviewed in Section A1.6, there would appear to be no fundamental problems to be overcome to obtain the high bit-rates per 7 or 8MHz cable channel needed to carry broadcast material from the satellite or terrestrial sectors. Using 16QAM, bit-rates of around 30Mbit/s should provide rugged performance in the 8MHz channels of modern broadcast networks. Higher bit-rates per channel should also be possible with higher-level modulations such as 32 or 64QAM in new, or up-graded, networks with adaptive equalisers (or up-graded "echo" performance), provided care is taken with the system design. On this basis the service potential for the cable distribution of digital television may be summarised as:-

In principle, cable television networks have the potential to be able to support the full range of digital television services deliverable in future by the terrestrial or satellite media.

## 3. IMPLEMENTATION – PROBLEMS OF SERVICE INTRODUCTION AND EVOLUTION

### 3.1. General considerations

At present, in Europe the factors conditioning service introduction vary for each of the television broadcast/distribution media, not only in terms of spectrum availability and development time-scales, but also in terms of the political and regulatory frameworks that apply (both at the national and at the European levels). While use of the limited resources of the BSS and terrestrial television bands is rather heavily regulated, broadcasting in the FSS bands is relatively unconstrained by regulation. Thus satellite operators/service providers can, in principle, choose whatever transmission standards suit their business needs. Additionally, the implementation problems in the FSS band appear to be slight compared with the terrestrial case where service quality is constrained and transmission costs can be relatively high.

Digital satellite broadcasting

It is clear at present, for example, that the development of broadcast quality digital video compression in MPEG has set the scene for the early introduction of digital satellite services in the FSS band if the market so demands. QPSK/MPEG equipment of satellite broadcast quality offering a range of service options is being marketed, and the satellite service providers operating in the FSS bands have made it clear that they wish to be in a position to launch new digital services by 1995. While it is reported that the primary target for the new services will be the DTH market, there is, of course, the important secondary market of cable distribution to also take into account. Thus attention must be given to the technical standards to be employed for cable television to enable the new markets to develop and (it is hoped) expand. Of primary concern, however, must be likely attractiveness of the new services to the potential DTH viewer/subscriber and the related question of the likely cost of DTH receivers for the different types of service - at service start, and subsequently.

Likely receiver costs - technical standards

The initial, and later, costs of sophisticated digital DTH satellite receivers of different types will almost certainly depend considerably upon the view that manufacturers take of initial market size and of its rate of growth. The cost question can broaden to take account of developments in related fields including the possibility of harmonising (where sensible) the standards to be employed for terrestrial television broadcasting with those to be used for satellite broadcasting in the FSS and BSS bands (as well as with the future digital standards for secondary distribution in cable networks). There is the need then to try to create a "standards" structure for the introduction and further development of digital television services that embraces the foreseen requirements for all the broadcast media; recognising, of course, that important elements of the system(s) specification will be "applications specific". (Harmonisation issues also, of course, involve the consumers' general interest - these matters are discussed further in Section 4).

Terrestrial television

Unlike satellite television, which can be national or international in its application (but not yet regional or local) terrestrial television is, and is likely to remain, primarily a medium for the delivery of national, regional or local television programmes to audiences contained within the national boundaries. There is then, in principle, the possibility that DTTB will take on somewhat different characteristics in each European country to suit differing national situations or differing political/regulatory regimes (as happened with SECAM and the variants of PAL). Whatever the national priorities during service introduction, or for the further development of DTTB, it is the assumption in the discussion of DTTB system in this section that all national administrations would wish to keep the door open to the eventual phasing-out of the conventional analogue services to achieve some of the benefits offered in the "all-digital" phase of operations. In any event, if the introduction phase for DTTB comes several years after that for the digital satellite services, as seems most likely, then the decisions regarding the system standards to be adopted for DTTB will almost certainly be conditioned by the availability and price of ICs developed for digital DTH satellite receivers; (as well as by the manufacturers, and the publics, interest in achieving commonality in the technical standards and interfaces to be employed). In discussing the introduction and evolution of DTTB services then, a degree of common interest and purpose will be assumed. It is accepted that there may be "national" variations to take into account, but it is believed that these can be based upon the development of a common technology and a flexible system standard.

Cable television

Traditionally, cable television has played a complementary role to that of the terrestrial and satellite services, providing an alternative means of distributing off air programme services to the public without competing directly in the domains of programme production or programme service provision. While this situation could change in future, at present there is little sign that the European cable operators have plans to emulate their cousins in North America where cable television has grown to the point of being the dominant force i n the television market place. There are however issues relating to the share-out of subscription revenues between the cable companies and the satellite service providers that might influence decisions on future cable standards, especially in the areas of Service Identification (SI) and Conditional Access (CA)

However, in the main, new cable system standards have been devised in response to developments in the broadcast media and this pattern will almost certainly be followed for digital television where the main concern will be to adapt cable operations to developments in the digital satellite and terrestrial television broadcasting media, if and when these occur.

## 3.2. Satellite television

### 3.2.1. FSS bands

The initial commercial advantages of digital transmission are clear:

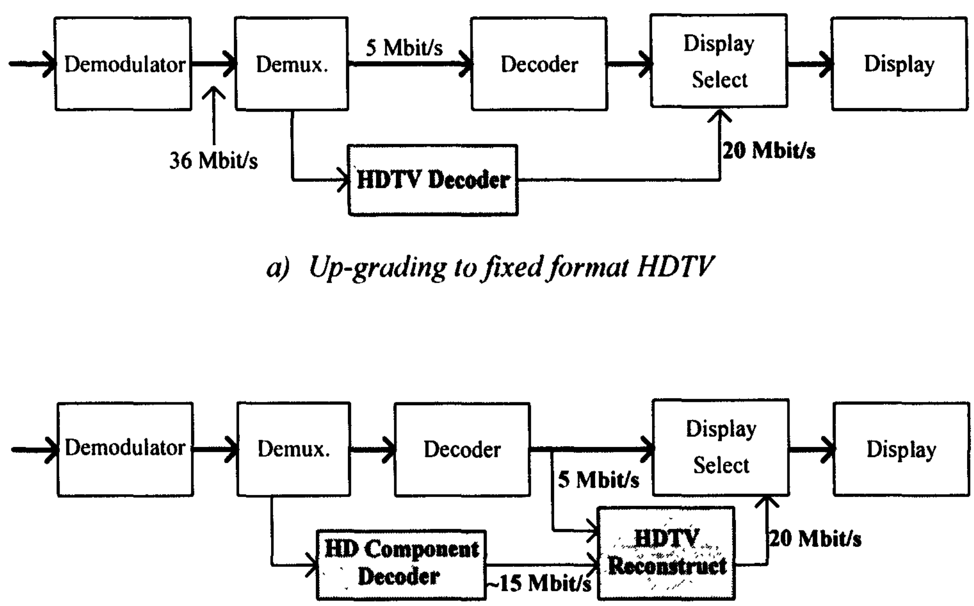
- to reduce the transmission cost per conventional TV channel;
- to offer more programming choice to the viewer;

- to increase and protect revenues by means of greater security in the conditional access system;
- to have the option to be able to introduce higher quality services such as HDTV if and when the market so demands.

As reported, one application for the new services could be "video on demand"; that is by splitting a 40Mbit/s channel into 8 x 5Mbit programme channels (say) and transmitting the same 2-hour movie through each programme channel with start times staggered by 15 minutes per channel. As a single satellite of the Astra type can accommodate 18 digital channels, it could easily support several subscription "video on demand" packages each with a choice of five movies (say) and an average "access-time" of 7.5 minutes as well as other channels devoted to news, documentaries, general entertainment, etc.

Given the low programme channel bit-rate, demultiplexing and processing should be relatively straightforward and low-cost to implement. The ability to reconfigure the system to provide addition programme streams of LDTV quality could also be easily incorporated to provide a package with a number of SDTV and LDTV market options. There remains, however, the question of how first-generation receivers (with limited decoding options) can be given the capability to deal with an evolving market that has an increasing element of EDTV and HDTV transmission.

One first-generation receiver option is to simply pass the undecodable data stream to a demultiplexer output port, allowing an EDTV or HDTV decoder to be added at a later date as shown in Figure 3.1 a).




*a) Up-grading to fixed format HDTV*



*b) Up-grading to hierarchical HDTV*

*Figure 3.1: Migration routes from 1st generation systems to HDTV*

A second option occurs if the HDTV programme is hierarchically coded, with the first generation SDTV or LDTV formats preserved as a nested sub-set of the HDTV transmission data. In this option, the packets carrying the "core" SDTV/LDTV picture information accompanied by stereo sound, SI and CA data can be decoded in the normal way, while packets carrying higher resolution data only are "ported-out" to enable the receiver to be converted to HDTV capability at a later date. This option (shown in Figure 3.1 b)) is somewhat more complex than the first because of the need (when converting to HDTV capability) to reconstruct the HDTV signal from its component parts. There will also be the need to build in somewhat more complex switching arrangements than option 1 into the first-generation receivers than option 1. It does, however, provide the basis for a much smoother transition to the higher-definition systems, or the obvious reason that when these HDTV services are introduced by the operator (possibly using a second transponder or satellite), there will be the opportunity to "sell-into" the established market of "first generation receiver" homes. As a secondary benefit of this approach, it does, of course, provide the basis for graceful failure for the HDTV service, which although of less significance than in the terrestrial case is, nevertheless, of importance for out-of area viewing.

The market advantage of the second option is not important if the new HDTV transmissions are using the same programme material - to provide an HDTV up-grade to a established video-on-demand service for example. However, it is extremely difficult to forecast what future market demand may reveal and it would be dangerous to block future options by over-confident predictions today. It would seem important then to understand, in some detail, the cost factors involved in the options as summarised in Table 3.1.

| Choice of Digital Satellite HDTV Format | Effect on Future Services | First Generation Receiver Cost(s)/Considerations |
|---|---|---|
| Fixed - not backwards compatible | • Receiver mod. needed for HDTV programme view<br>• Up-grade to HDTV cheaper<br>• Non Graceful Failure | • Demultiplex Port<br>• Display Select circuit. |
| Hierarchical<br>(or nested Simulcast)<br>- backwards compatible | • HDTV programmes viewable in SDTV<br>• HDTV up-grade more expensive<br>• Graceful Failure | • Demultiplex Port<br>• Decode Port<br>• Display Select circuit. |

*Table 3.1: HDTV format options and implications Re. First-generation and HDTV receiver costs*

<u>BSS bands</u>

In principle, the technical arrangements for the BSS band parallel those for the FSS band.

Either A) The European Component of the WARC '77 plan remains in place to provide:

- higher transmitter power/45Mbit/very small "dishes";
- a limited number of national channels

B) The band is re-planned on a medium-power/European basis to provide system characteristics similar to those for medium power FSS services

In either case, the problem of up-grading first-generation receivers is essentially the same as for the FSS band.

### 3.3. Terrestrial television

In considering a European framework for the market-led introduction of DTTB, account must be taken of the interests of the main players:

- The Broadcasters, whose investments in programmes, studios, and transmission infrastructure must be justified;
- The Manufactures, where the large R&D and pre-production costs must be recovered from product sales;
- Those in National Administrations who are responsible for spectrum allocation and utilisation, who will wish to see the very considerable potential of DTTB realised as soon as is reasonable.

The technical considerations that might be relevant to future European or national policy decisions are discussed in this sub-section under the headings:

1) Regulated Introduction with simulcasting as the priority service
2) Unregulated Introduction - New Services on Demand
3) Simulcasting as the main element of mixed new service/simulcast regulation strategy.

#### 3.3.1. Simulcasting

<u>The introduction phase</u>

In this scenario, it is assumed that the purpose of the regulation is two-fold:

a) to create the conditions in the DTTB market place that allows a firm timescale to be set for the termination (or phasing out) of the PAL/SECAM services to enable a switch-over to an "all digital" mode of operation.
b) to allocate to all existing PAL (or SECAM) broadcasters an equitable share of the spectrum available for DTTB (as is being done in the USA for the existing NTSC broadcasters for example).

The first point to note is that in those countries where the spectrum for DTTB is limited by large numbers of local services operating in parallel with national services, the additional technical constraints imposed by (b) may be such as to threaten the viability of objective (a). In such a situation, a concentration of the limited spectral resources on the main national/regional networks may be more effective. The second point is that inherent within this strategy is the notion that the existing broadcasters will take the lead in creating the DTTB market place. To do that they must be able to offer the public something better than the current PAL services; i.e. higher picture quality, more programme choice, or both.

Existing broadcasters' requirements

For existing broadcasters, the emphasis is on higher quality because their programming costs far exceed the technical costs of transmission. This salient fact has led almost all existing broadcasters to support introductory phase system requirements which provide for:

- Very High Quality EDTV with Built-in Extendibility to HDTV
- because this is the quality that the existing broadcasters need to be able to offer the public, so as to compete with alternative means of delivering HDTV, and is therefore one of the main justifications for their investment in simulcasting. High quality is also one measure of the increase in spectrum utilisation supported by DTTB.
- Hierarchical Channel Coding/Error Protection
- -because this (along with a matching hierarchy in source coding) will help to extend the main transmitter coverage economically to "match" that of the existing analogue transmissions, provide rugged reception on home-portables, and a graceful failure characteristic.
- Reconfigurability to Multi-Programme Operation because this will allow a variety of (low-cost) programming and "repeats" to be delivered to the public during off-peak hours. This should make the overall service more attractive; it would also boost receiver sales; and could generate new sources of income.

Additionally, the broadcasters would wish to see the system able to support the SFN concept so that they will be able to expand coverage, in due course, by the use of local SFN relay networks. (In this they are, of course, supported by the administrations' long-term interests).

Start-up costs

With these broad system requirements, the service introduction costs should be relatively low. They would include (perhaps in decreasing order of importance):

- the additional programme production and film processing costs for the higher-definition-peak-hour services;
- additional "off-peak" programme costs (the use of repeats and archive material could help to keep this down);
- the provision of new transmit antennas or, alternatively of new channel-combining arrangements to enable the existing antennas with the continuing PAL services to be fed in parallel;
- the provision of new (low power) transmitters and standby arrangements;
- additional feeds and switching arrangements;
- additional infrastructure, maintenance, monitoring, etc..

The all-important question of receiver costs is likely to depend to a significant extent, upon the availability of ICs developed for the digital satellite services. If the satellite-receiver demultiplexing and decoding ICs are based upon a flexible MPEG-2 design, whose key parameters could be reset in response to over-the- air signals, then the basis of a low-cost first generation DTTB receiver could be ensured (especially if the satellite decoder structure supports a hierarchical approach). In this case, the major manufacturing investment required for DTTB should be limited to demodulation and front-end (tuner) development and the plant needed for the high-volume production of these components. In any event the experience gained in the production of digital satellite DTH receivers should assist greatly in reducing the costs of first generation receivers to acceptable levels in the DTTB case.

More fundamentally, receiver costs will be determined by the particular choice of COFDM system parameters to be used in the introduction phase. In considering the likely performance/benefit cost equation at service introduction, it is noted that:

i) a choice of symbol period ($T_s$) greater than 64µS (to 128µS or greater) might involve significant additional complexity and cost not only in terms of demodulator processing and memory but also in the receivers tuning arrangements (owing to problems of increasing phase noise as $T_s$ is increased and carrier separation reduced);

ii) using main transmitters only, there is no immediate need for anything more than a small guard interval ($T_g$).

iii) QPSK demodulation is less complex than 16QAM (16QAM and 64QAM also need special synchronising and equalisation signals which add to receiver complexity and reduce the available useful bit-rate);

iv) initially it might be expensive to build adaptive CCI filters into the receiver in order to save the loss of useful data capacity in transmission;

Taken together, these points suggest that, in certain market situations, there will be pressure to opt for a restricted receiver specification to keep costs down, and the question arises, therefore, as to whether this possible development will inhibit the future evolution of the system.

<u>The transition phase(s) of DTTB</u>

<u>Market development</u>

The obvious objective, following the introduction of DTTB services, is for the DTTB service take-up to grow to the point where it can be seen as the **primary** [future] service. At this point measures can be taken to speed-up further growth in the DTTB market to hasten preparation for the phasing out of PAL or SECAM. It is likely that the first phase of growth will depend upon the attractions of EDTV pictures and/or the diversity in programming available in the re-configured mode. The initial sales market may be primarily "replacement" and, therefore, fairly slow unless receivers are very affordable, or the new "programming" quite different from the old. As the market develops and receiver prices fall, additional sources of income might be developed either from subscription services or from increasingly-targeted advertising revenues based upon time-shifted programmes from the same programme production base. Thus, an increasing number of well-financed programme services together with falling receiver prices will justify an expansion in DTTB coverage and possibly some changes in the system to take account of progress in receiver design.

## Interim technical developments

If and when DTTB services have grown to the point that they are seen clearly to be able to replace the PAL/SECAM services within an EC-member-state, a number of evolutionary changes might be considered:

- first, measures to increase coverage/service availability;
- second, ways and means of improving receiver/system performance.

Measure to increase DTTB coverage might include:

a) In areas where the DTTB receiver population was approaching that of the PAL/SECAM-only receiver population, it would be possible to envisage increasing the DTTB transmission power on an annual basis (1dB/annum say). This would, of course, lead to a gradual reduction in the quality of the PAL/SECAM services but it would not be perceived (at least initially) by the public and might only be of real consequence during periods of atmospheric interference ( 5% of time).

b) The re-engineering of certain PAL relay stations to allow some increase in the Regions' DTTB EIRP. (Again, almost certainly to the detriment of certain "pockets" of PAL reception).

c) DTTB relays could be introduced where they would significantly increase the coverage. This would almost certainly be done by the use of SFNs and a 4-10μS (say) guard-band. Thus there would be an "in-area" loss of useful DTTB data unless this requirement (the guard band) had been built-in to the introduction phase system specification. Alternatively, if receivers with adaptive CCI filtering capability had been introduced as "second generation" receivers, it might be possible to engineer a simultaneous change in the system's transmission format to compensate all, (except those with first generation receivers) for this loss.

## Receiver developments

Whilst there might be considerable improvement in the cost/performance of 16 and 64QAM receiver implementations, and receivers able to operate with long symbol periods, it is difficult to see how these developments can be taken advantage of until the network is re-engineered for all-digital operation (unless the introduction-phase system is based on 16QAM). Improvements in receiver front-end and tuner designs, together with improvements to the graceful failure control characteristics will be of benefit to the consumer but will not be of major consequence.

If a QPSK first-generation receiver implementation is assumed, the major potential for interim improvements remains in the areas of adaptive CCI filtering and receiver cost-reduction through the greater integration of receiver functions.

In the last period of transition, however, it will be necessary to consider the need to market receivers that are prepared for the switch-over to all-digital operation. At switch-over, the system changes could be radical in several respects and, in a QPSK introduction phase scenario, these receivers will need to be "dual standard" in performance as discussed below.

The switch-over to all-digital operation

Whatever the final outcome from continuing studies on Single and Dual-Frequency Networks (DFNs), it is clear that major re-engineering works will be involved, and that following switch-over, there will almost certainly be a further stage of transition in operations before the full capability of the new digital network can be realised.

Continuing constraints

The first constraint on the "all-digital" network operation concerns the continued presence of PAL/SECAM - almost certainly in some neighbouring countries, and possibly in "pockets" of the nation (because it may be impossible, in practical engineering terms, that every village or valley continues to receive a service in any other way). Thus the initial "all digital" frequency plan must be adapted to suit a further period of development, involving most importantly, co-ordination on a European level.

A second constraint concerns the number of DTTB receivers in the field that are not fully prepared to work with the final system specification. If, for example, the final DTTB system specification for roof-level services were based upon the following parameter set:

- Fixed Format and hierarchical coding options (with 64 and 16QAM elements);
- A long symbol period with $T_s \geq 256\mu S$
- A guard interval $T_g \geq 25\mu S$
- Reconfigurable;

whereas the introduction phase receiver specification was based upon a symbol period of 64µS and:

a) QPSK only or

b) a mix of 16QAM and QPSK

then the QPSK receiver could not receive the service and none of the earlier receivers would be able to continue working to their full specification. In this scenario "dual standard" receivers would be needed to bridge the gap between the pre and post "switch-over" standards. Thus, depending upon the number of receivers of early design remaining in the field, some of the post "switch-over" channels would have to be configured to provide a continuity of service until the old DTTB receiver population dies away (2 or 4 out of 10-20 channels per region say).

Alternatively if the digital networks were predominantly dual-frequency (DFN), the need for large guard intervals would be avoided. In this case, the choice of an introduction phase specification based on 16QAM could assist the transition scenario as illustrated in Figure 3.2.

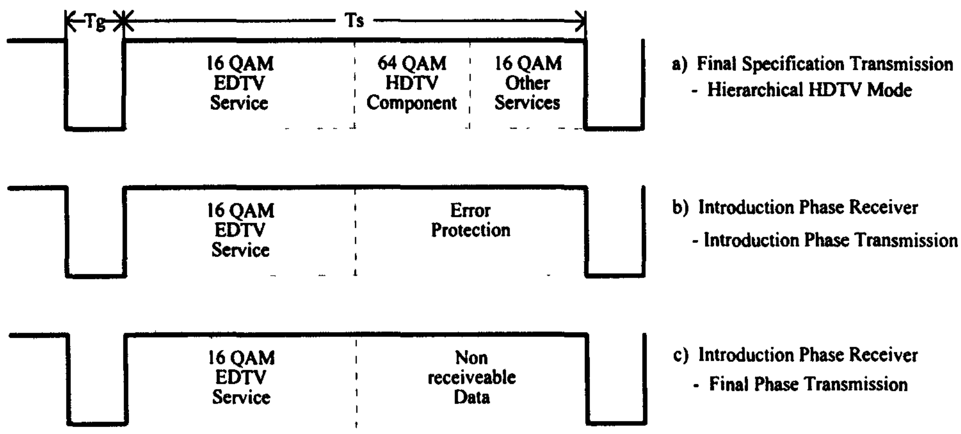



*Figure 3.2: Guard band for transition and final phase systems*

Where, referring to Figure 3.2 above:

a) Final system specification

Configured to provide an HDTV hierarchical service with a nested EDTV service:

- $T_s = 64\mu S$
- $T_g = 5\mu S$
- EDTV service: 50% of $T_s$ with 16QAM to give ~ 13.5Mbit/s
- Added HDTV component 25% of $T_s$ with 64QAM to give ~ 11.25Mbit/s
- Final 25% of $T_s$ with 16QAM for other services
- Overall useful bit-rate 31Mbit/s

b) Introduction Phase Specification

Configured to provide EDTV with graceful failure

- $T_s = 4\mu S$
- $T_g = 5\mu S$
- 50% of $T_s$ with 16QAM signal data to give ~ 13.5Mbit/s
- 50% of $T_s$ with 16QAM error protection data
- Useful bit-rate ~ 13.5Mbit/s
- Video bit-rate ~ 10Mbit/s

Then, given the right SI and receiver structure, an Introduction Phase Receiver working to the Final Phase Transmission Specification (but with signals of considerably higher power) could adapt to receive the full 16QAM component in the new transmission format - that is the full EDTV service as shown in c) of the Figure 3.1.

A further problem concerns polarisation. If the greatest efficiency is to be gained from the new network, advantage will need to be taken of the polarisation discrimination in the plan (especially in DFN's). This implies that, in many areas, new roof-level receiving antennas will almost certainly be needed to receive the highest quality services.

In summary, the new network plan at "switch-over" will be constrained in several areas. First in the network planning area where, in addition to the polarisation problem, the most important constraint will be the continuing PAL/SECAM interference from neighbouring countries. There will be the need, therefore, for agreement in Europe to co-ordinate the timing and extent of the national "switch-over" plans. This could be a particularly difficult exercise bearing in mind the differing situations in each country regarding the development of services in the satellite, terrestrial and cable television media. Second in the choice of the Introduction Phase System Specification where, in certain scenarios, a choice for QPSK causes certain problems and 16QAM has considerable advantages.

Re-engineering considerations

The cost of re-engineering the network for "switch-over" should not be overlooked. There will be numerous problems to solve, not the least of which may concern:

- the need to combine 10-20 programme feeds for transmission
- the new transmit antennas that will be required where polarisation changes are envisaged
- problems of "mast loading" when a new antenna must be installed in parallel with the PAL antenna prior to "switch-over"

  the costs associated with the installation of a large number of new transmitters and their supporting infrastructure.

### 3.3.2. Unregulated DTTB introduction

Introduction

Because there are so many difficult problems to be solved in the preparations for a planned "switch-over" to DTTB, the idea of "unregulated DTTB introduction" may seem attractive. However, this approach would not necessarily avoid the problems, for the following reasons.

The outcome of an auction for the limited spectrum resources available for DTTB would, of course, be quite unpredictable. Assuming that four channels (A, B, C, D) are available and the bidding limited to the four Primary Service configurations:

| | | |
|---|---|---|
| a) | Simulcast | (roof-level/standard RF system) |
| b) | Non-Simulcast | ( " " " " " ) |
| c) | Portable | (harmonised i.e. receivable with service (a) or (b) equipment) |
| d) | Mobile/portable | (non-harmonised) |

Then of the numerous possible outcomes we have:

(i) At one extreme {a (A+B+C+D)}, all channels are simulcast and the situation is the same as that already discussed above in Section 3.3.1.

(ii) At the other extreme {d (A + B + C + D)}, all channels are non-harmonised mobile/portable services which implies no growth of a DTTB receiver base in any way compatible with the final system specification or of the necessary manufacturing technology.

(iii) In an intermediate case – that is widely thought to be most likely outcome – {b (A + B + C + D)}, all channels are non-simulcast but the only difference from situation (i) concerns programme content and the impact this might have on the growth of a DTTB receiver base.

Thus from a technical point of view (with certain qualifications) outcomes (i) and (iii) are precisely the same and the discussion of Section 3.3.1. applies. This is because, provided a standardised introduction phase system is employed (primary service for roof level antennas), the main differences from the simulcast scenario concern the programme service quality and the effect of this has on the rate of growth of the DTTB market and its manufacturing base.

There are, however, certain (perhaps minor) differences from a technical/administrative perspective. These concern the technical compromises with the PAL service quality needed in the introduction and transition phases to achieve increasingly satisfactory performance for DTTB. The advantages to be gained by increasing DTTB coverage at the expense of the PAL service discussed in Section 3.1.1 above include:

- "tailoring" PAL coverage
- "re-engineering" certain PAL relays
- adding DTTB relays
- "slowly" increasing DTTB transmit power

In the non-simulcast scenario, actions such as these might well be contested by the incumbent PAL broadcasters in a situation where they had nothing to gain and everything to lose.

More importantly, the motivation for purchasing DTTB receivers will be quite different in the simulcast and non-simulcast scenarios. In the simulcast scenario, a large percentage of potential purchasers will be in the PAL/SECAM receiver "replacement" market, and it is implicit that they wish to continue to view their favourite programmes. And, as soon as all PAL/SECAM programmes are simulcast digitally, the market for combined PAL/SECAM-and-DTTB sets will disappear. By contrast the non-simulcast DTTB market will inevitably be much more like the satellite market of today (although probably more complex with greater choice available). The decision whether to buy an additional receiver/decoder (or replacement combined PAL/SECAM-and-DTTB set) here will depend primarily upon the new programmes on offer which themselves will be in direct competition with those of the satellite services.

While it is clearly impossible to make any sensible forecast of the possible future growth of DTTB in the various scenarios, an attempt to suggest the possible patterns of growth if DTTB proves successful and the associated technical ramificat ions is shown in Figures A for the simulcast scenario.

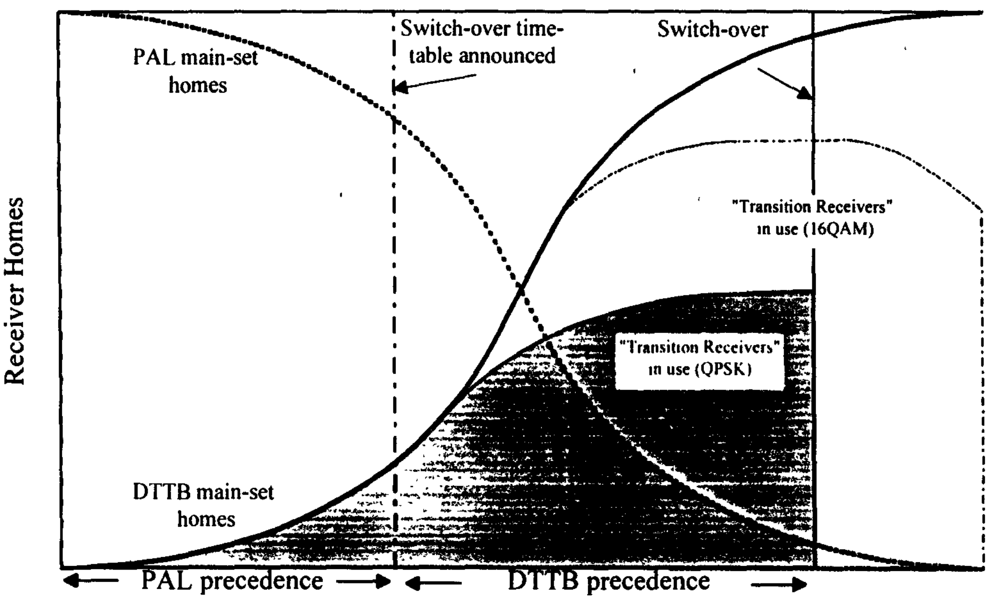



*Figure 3.3: Possible DTTB growth (simulcast scenario)*

Note: the difference between the number of Transition Receiver's in use and the total of DTTB main sets in the home is accounted for by the purchase of dual standard (Final and Transition Standard) receivers. (i.e. more in the QPSK case than in the 16QAM case).

In the non-simulcast case, growth might be similar to Figure 3.3 but it should be noted that:

a) There will be no equivalent reduction in the viewing of PAL services on PAL sets (as the main home receiver)

b) It will therefore be less clear when to designate DTTB as the preferred service and announce a switch-over time-table.

## 3.4. Cable television

The initial problem for cable operators will be to adapt their networks in preparation for the launch of the new digital satellite television services. If, as is reported, these services are to be based upon the "near video on demand" concept, then the number of conventional 7 or 8 MHz channels required in the cable system will be very high unless the network is adapted to provide for digital transmission in some of its bands. Thus there will be the need for "Transmodulation" at the cable head-end to convert the received satellite signals for distribution of the satellite programme package in 16, 32 or 64QAM, so as to provide 30-40Mbit/s per (2x8), 8 or 7MHz cable channel.

Depending upon the choice of modulation system (COFDM or single-carrier) there may also be the need to "up-grade" network performance to reduce the effects of echoes. If high bit-rate transmission in the narrow-band 7/8MHz channels does prove feasible then it may be possible to utilise areas of network capacity left vacant today owing to various practical "taboo" situations in cable networks analogous to those that apply in terrestrial broadcasting – using low-power digital transmission in channels adjacent to higher power PAL/SECAM transmission or in other channels left vacant owing to local oscillator or image channel considerations for example. There would be the need, however, to evaluate the effect of multiple digital channels (albeit at lower average power than for PAL) on the quality of the conventional services – inter-modulation being of considerable concern in cable networks employing long chains of cascaded amplifiers. If, following further investigation, the feasibility or cost of implementing 32/64QAM in the network, or in the necessary receiver decoders, cannot be justified, an alternative arrangement could involve the use of 12MHz or 16MHz channels, spacing to reduce the bit-rate per MHz (although this would be of course expensive in terms of network capacity). While the network constraints require further investigation, it is possible to imagine that even if 8MHz channel working is feasible demodulators employing 32 and 64QAM may prove too expensive for the initial digital cable television market. In this scenario attention would focus on the use of 16QAM in 16MHz channels and ways and means of migrating to the higher modulation levels of 32 and 64QAM over time would come into consideration. In any event, the cable operators will be faced with some difficult trade-offs, and decisions, if they are to be able to relay all the new digital services – especially those operators with limited network capacity.

While the use of 16QAM- COFDM could have some initial advantage by harmonising cable requirements with DTTB, the disadvantages of 16QAM would appear to be quite significant:

a) 2:1 loss in digital distribution capacity

b) loss of the possibility of using standard 8MHz tuners

For these reasons it is believed that priority should be given to solving the problems of high bit-rate transmission in 7/8MHz channels.

## 3.5. Summary of main points - service introduction and evolution

A) Digital satellite television broadcasting

Systems for the delivery of digital satellite services to the home are in an advanced stage of development and standardisation.

Service introduction with multi-channel low bit-rate programme services is expected to occur in 1995.

Given the right multiplexing structure in introduction-phase receivers, there should be no technical/commercial barrier to the further evolution of the system to support full HDTV quality.

B) Digital terrestrial television broadcasting

The objective of converting terrestrial television broadcasting to all-digital operation can be achieved in the following ways:

a) by "simulcasting" the current terrestrial programmes as part of a new digital satellite service; a service involving perhaps a multiplicity of other channels and "higher definition" service options.

If the new digital satellite television services expected to be launched in the mid 1990s prove to be successful, a significant proportion of potential audience will have acquired suitable digital TV sets , and the costs and difficulty involved in launching the new "package" should be relatively small. Consumers' reactions, in terms of the receiver replacement and/or additional receiver market, would almost certainly be very positive -

retaining all of their favourite programmes (perhaps in higher quality) plus numerous other programme choices including time-shifted "repeats". There remain, of course, a number of questions concerning commercial motivation and regulation in addition to the basic assumption of the successful technical development of the digital satellite television field.

b) by "simulcasting" in the related (taboo) channels of the terrestrial PAL and SECAM networks. Here the service options are considerably more constrained than in the satellite case, both in terms of quality and number of services possible during the introduction and transition phases. The primary service will be to roof-top, but the secondary services to portable receivers must also be taken into account. While studies are still in progress and in several areas there are important issues unresolved, the picture that is emerging suggests:

  - It will be technically practical to *introduce* simulcast EDTV services in a number of countries although the service coverage might be patchy and therefore it would be difficult for potential viewers of digital programmes to know whether they would actually be able to obtain satisfactory reception in their homes before acquiring a digital receiver. This raises the issue of consumer confidence.
  - The systems employed are somewhat more complex than in the satellite case, and even the simplest receiver implementation (QPSK only) could be (initially at least) more expensive than in the satellite case.
  - There are technical arguments that suggest that a truly satisfactory transition to "all-digital" networks can be achieved to be fed in only if the "introduction phase" transmission system is based upon the use of 16QAM. Unfortunately, 16QAM receiver implementations are likely to be (initially at least) a good deal more expensive than those based upon QPSK only. There would also be a major difficulty if large symbol periods and guard intervals were used in the "final systems" specification to support the Single Frequency Network (SFN) concept.
  - While Single Frequency Networks (SFNs) are only in the early stage of study and evaluation, there are technical arguments that suggest that reliance on large guard intervals for main network operations is impractical, and that the guard interval should be conditioned only by the requirements of local relay SFNs.
  - Associated with the study of possible all-digital network implementations, the concept of a dual-frequency network (DFN) has been introduced. In this a single programme channel could achieve national coverage by the use of two frequencies together with two (orthogonal) polarisations. In this arrangement, there is no requirement for the longer symbol periods and guard intervals SFNs. However many homes would need to change their antennas.
  - In this simulcast scenario, in many countries, the terrestrial, receiver replacement market will continue to be driven by the consumers' desire to continue to watch "favourite programmes" that are available only from the high production-value PAL/SECAM services (provided, of course, these are not available from the satellite sector).
  - In the absence of a rapidly growing digital satellite market, the growth of the field of DTTB receivers to the point of deciding the DTTB service has preference over the PAL/SECAM services could be steady (and hence the timing of the decision to announce a schedule for the "switch-over" to all-digital predictable). On the other hand, if the growth of digital satellite services able to offer a wide range of services and quality levels was rapid, growth in the "simulcast" DTTB market would be problematic.

c) by non-simulcast DTTB services in the PAL/SECAM related channels. Provided these services are intended as primary services to roof-level antennas and the transmission system employed is standardised as for the simulcast case, then there is no technical difference of consequence between the simulcast and the non-simulcast scenarios and the points laid out in b) above apply. The essential differences relate to consumer preferences in the PAL receiver replacement market.

   In the full simulcast case, the "replacement" consumer is purchasing primarily to continue to view the PAL or SECAM programmes. His choice is simply between another PAL set or a new DTTB set (offering higher quality pictures and an additional variety and diversity in programming); there is little, if any demand for PAL/SECAM and DTTB sets. As DTTB sales increase, the PAL/SECAM receiver population will decline proportionately

   In the non-simulcast case, the purchaser buys an additional receiver/decoder as a means to access new programmes (much as he does today if he buys a satellite receiving system primarily). In this case, the impact of purchases of DTTB receivers on the population of PAL receivers is likely to be minimal, and there will be a significant demant for PAL/SECAM and DTTB sets.

d) By simulcast or non-simulcast services where the primary service is to portables with "harmonised" standards and receiver growth is such to support switch-over at some point in time.

e) By a simple edict that sets a "switch-over" time scale and relies on the development and sale "dual PAL/DTTB receivers", before the switch-over, to viewers who wish to be able to receive the same terrestrial programmes afterwards. DTTB transmissions have begun.

   Clearly d) and e) are the least attractive of the above paths to "all-digital". The situation regarding a), b) and c) is less clear.

C). <u>Cable television</u>

The many complex problems to be faced by the cable operators with the advent of digital satellite television broadcasting are in the early stages of investigation. While there would appear to be nothing of a fundamental nature to prevent the introduction of relayed digital satellite television services in the cable networks, there are a number of important practical problems to resolve. Decisions regarding the specification of performance of limited introduction-phase digital decoders could necessitate consideration of migration strategies that would enable the system to evolve to its final, more efficient, specifications.

## 4. INTERNATIONAL STANDARDISATION

Currently major international efforts are underway to achieve a framework for the future development of digital television broadcast systems. Central to this activity is the work of ISO/TEC MPEG concerning Video Source Coding and Multiplexing standards. Second in importance, from a European point of view, is the work of the EBU-ETSI JTC (working in conjunction with the "European Launch Group on DTTB") which is planning to produce European Technical Standards (ETSs) covering all aspects of digital broadcast systems.

## 4.1. ISO/IEC/MPEG

MPEG is developing standards for digital video and audio source coding and multiplexing, suitable for broadcast applications, within the framework illustrated in Table 4.1.

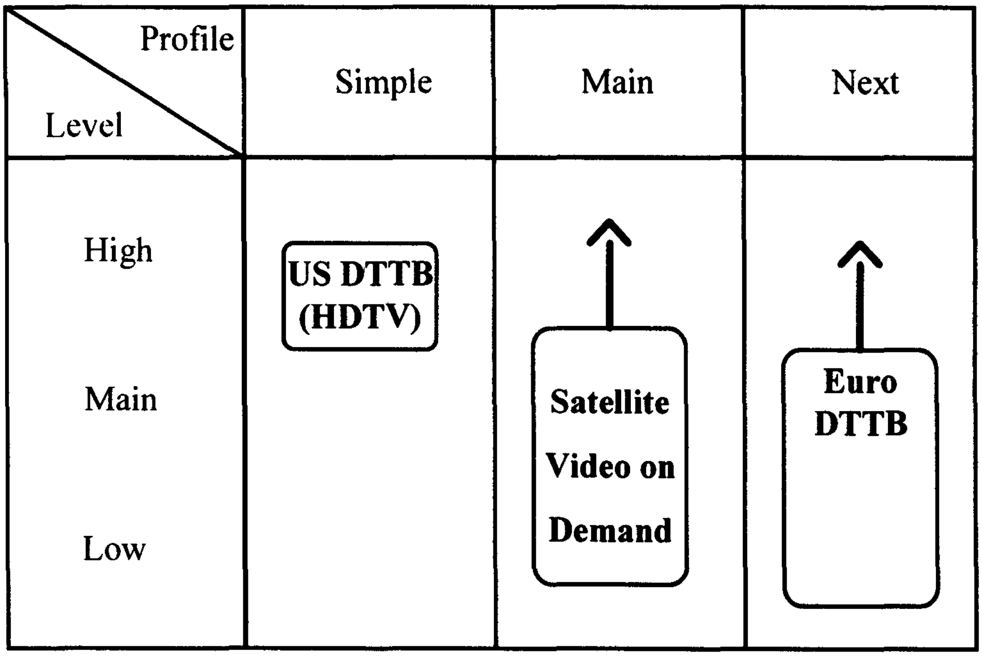



*Table 4.1: MPEG 2 family of standards*

Notes:
1. The MPEG standards are "generic" rather than "application specific" and the choice of parameters employed with the standard can be varied according to the application (e.g. bit-rate, scalability levels, etc.)
2. While the terms "Profile" and "Level" are not rigorously defined within MPEG, in broadcasting terms:
   "Profile" describes features such as:
   - backwards or forwards compatibility
   - scalability (next Profile only)
   - etc.
   "Level" describes performance characteristics such as:
   - maximum bit-rate
   - bits/sample
   - etc.

Within this framework, the main profile/main level standards are already frozen for video and audio and the multiplexing standard will be frozen in the second half of 1993. CCIR requirements for "scalability" (to allow for hierarchical coding) have been included in the MPEG "requirements" document under the "Next" profile label, and the stated US requirements for terrestrial HDTV under the high level of the "simple" profile (no "B" frames). The key issue from a European point of view is to ensure that the Main Profile Multiplexing Standard is capable of supporting reconfigurability and an evolution of standards to higher quality levels in future (by the nested hierarchical approach and/or by the simulcasting of "core" video information). The European inputs to MPEG are being co-ordinated by the ELG in co-ordination with EBU-ETSI JTC.

## 4.2. EBU-ETSI JTC

In co-ordination with the ELG, the "JTC" has taken the responsibility to ensure, as far as is possible, that the ETSs needed for the launch of satellite services are produced, and ratified, within the envisaged timescales of potential service providers. Urgent pre-standardisation work (generating input for MPEG-2) is being dealt with in EBU V4/mod. augmented by experts from ETSI (shown here as "V4/mod").

Groups have been set up to produce ETSs in the following areas:

| | |
|---|---|
| - Video Source Coding | - ETSI group NA5 (in liaison with "V4/mod") |
| - Modulation & Channel Coding | - ETSI sub-group SES4 E (in liaison with "V4/mod") |
| - Multiplexing | - ETSI Project Team (in liaison with "V4/mod") |
| - Conditional Access | - The ETS will be produced by CENELEC |

It is implicit in these activities that the future evolution of satellite systems and a sensible degree of "harmonisation" with future DTTB standards are important issues - especially in the definition of the multiplexing structure. However given the MPEG-2 timescales and the lack of a definitive DTTB profile, the best that is likely to be achieved, at this stage, is a "core" multiplexing structure that keeps the door open to a number of the more obvious future options. Indeed the MPEG-2 agreement will define the system at only a syntax and protocol level, and it is recognised that the higher "application level" definition lies outside MPEG-2 considerations. Thus the opportunity to achieve a broadcast multiplexing standard covering the future requirements of all the broadcast media should be retained.

It has also been recognised by some members of the "JTC" sub-group responsible for co-ordinating the production of ETCs (JTCDT) that there are a number of other issues that will need future attention. These should include:

- conditional access (CA) system philosophy
- common service information (SI)
- system synchronisation
- satellite RF parameters
- cable and MATV

<u>Discussion</u>

Considering the importance of defining clear strategies for the introduction and evolution of digital broadcasting systems, and the key roles that the SI and CA systems might play in bringing such strategies to realisation, a stronger focus on this system area seems justified. (Indeed it is worth noting that the schisms that developed around the original MAC specification were largely to do with SI implementation - the Blue Book, etc.).

Service identification

The steps that may be necessary to assist systems to migrate from an "introductory phase" to a "final-phase" specification are discussed in Section 3. Further consideration of this discussion could provide a basis for a more rigorous definition of the SI and multiplexing system requirements, and harmonisation, for each of the broadcast media. This in turn could provide the basis, in due course, for a significant rationalisation in component supply leading to reduced receiver manufacturing costs.

Conditional access

While digital systems with SI are easily adapted to support highly secure CA systems, it is important that a balance be struck between the interests of operators and the manufacturers and the public interest. Experience gained through the development of the MAC system(s) suggests that a common system satisfying all the operators' technical/security requirements could readily be achieved (indeed the two Eurocrypt systems emerged as almost identical). However, differences arose regarding the CA systems "philosophy" which in turn led to different receiver implementations and arrangements that would be extremely "unfriendly" to the user who wished to subscribe to a number of different services.

It is important, therefore, to resolve these differences in the operators' "systems philosophies" if progress is to be made on the standardisation of a common CA system with user-friendly characteristics. If these differences continue to remain unresolved, the proliferation of "smart cards" and their card readers could be a significant factor inhibiting the growth of service and receiver sales.

Finally, it should be emphasised that the secondary distribution of the satellite digital services in cable networks does not appear to be receiving the attention warranted. Properly-prepared cable networks could play a key role in determining the economic viability of new satellite service launches. It would seem sensible therefore to progress standards in this area in parallel with those for the satellite services themselves.

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# APPENDIX A

## SYSTEM CHARACTERISTICS

### 1.1. System(s) overview

All digital television systems, whether for satellite or terrestrial broadcasting, or for secondary distribution over cable networks, can be represented by the blocks shown in Figure A1.1 (noting that the channel coding/modulation function will be tailored to the characteristic of the broadcast channel - the systems employed here being, in principle, quite different for the satellite, cable and terrestrial applications).

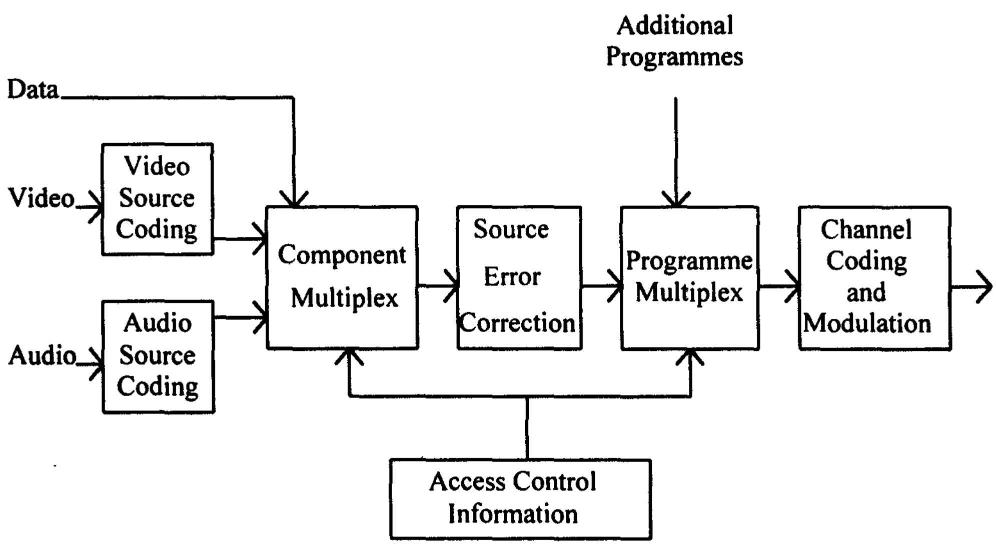


*Figure A1.1: Structure of a digital television system*

At first sight it would appear that all the other blocks could be generic; that is that the functional requirements are sufficiently close to enable the satellite, cable and terrestrial systems to be supported by a common system design and common inserted circuit (IC) implementations. However, when system optimisation is considered, differences emerge which can conflict to some extent with the objective of harmonising requirements and system components among applications.

The first trade-off to be considered is that between quality of service and flexibility in operation, where a system is able to support a number of quality levels. In general, video source coding algorithms are optimised for a given bit-rate (quality level), and compromises will need to be made to achieve an agreed best over-all performance.

Secondly, and of more importance, is the probable requirement for hierarchical coding in terrestrial systems to improve coverage and failure characteristics (and to provide for improved portable reception). Here the differences in coding requirements between the terrestrial and satellite/cable systems could mean that the compromises needed to achieve a "harmonised" performance in a single system may be difficult to justify.

## 1.2. Video source coding

Enormous progress has been made in the art of video compression in recent years. Currently a wide range of techniques are under study and certain techniques (motion-compensated DCT) have been developed to the point of achieving broadcast quality today. Of particular importance are the systems for generic source coding (and multiplexing) being developed within ISO-IEC MPEG. The MPEG-2 system is now in an advanced stage of standardisation and the emerging family of standards is receiving widespread support from broadcasters, service operators and industry alike *[3]*.

The state of the art in digital video coding has been reviewed in a number of recent papers *[9] [10] [11]*.

The main themes are covered by the headings:

- Motion compensated Model Coding (MPEG/DCT)
- Sub-band Coding
- Multi-resolution coding.

<u>Motion-compensated DCT</u>

The generalised functions of the source coder are shown in Figure A1.2

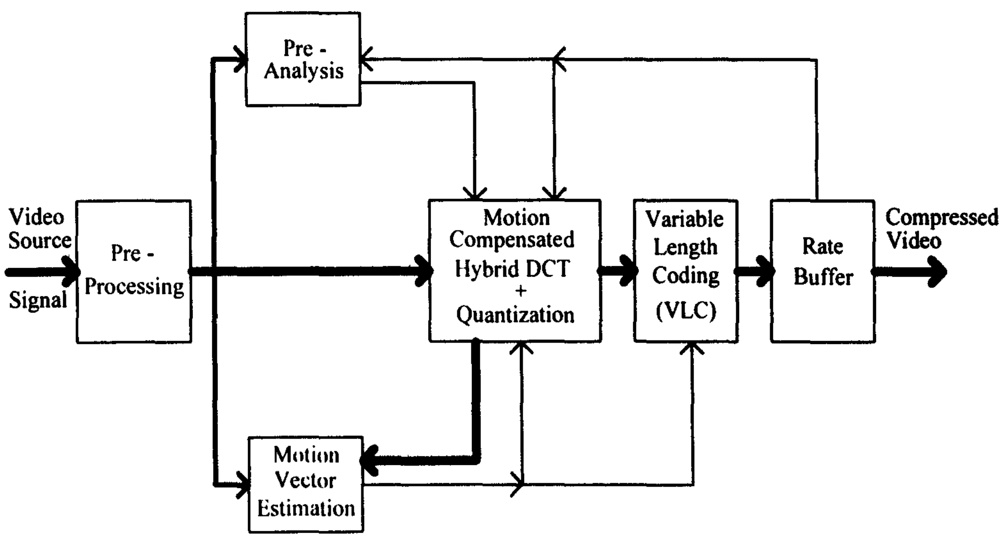


*Figure A1.2: Video source coder*

A description of the evolving MPEG implementation of the motion compensated DCT is given in *[12]* together with the summary of bit-rate requirements to achieve different levels of video quality given in Table A1.1. (in a US television standards context, and assuming CCIR Rec 601 as the highest video source standard).

| Transmitted Quality Level | Source 60 Hz Video | Source 24 Hz Movie |
|---|---|---|
| VHS | 2.0 – 2.5Mbit/s | 1.5 – 2.0Mbit/s |
| NTSC | 2.5 – 6.5Mbit/s | 2.5 – 5.0Mbit/s |
| CCIR Rec 601 | 3.5 – 9.5Mbit/s | 3.0 – 7.5Mbit/s |

*Table A1.1: Quality v bit-rate (MPEG)*

The trend in the relationship between compressed bit-rate and achievable conventional television transmitted picture quality level was summarised in a presentation at the 1993 Montreux Television Symposium by Dr M. D. Windram *[13]*.

The graphs presented are reproduced below as Figure A1.3.

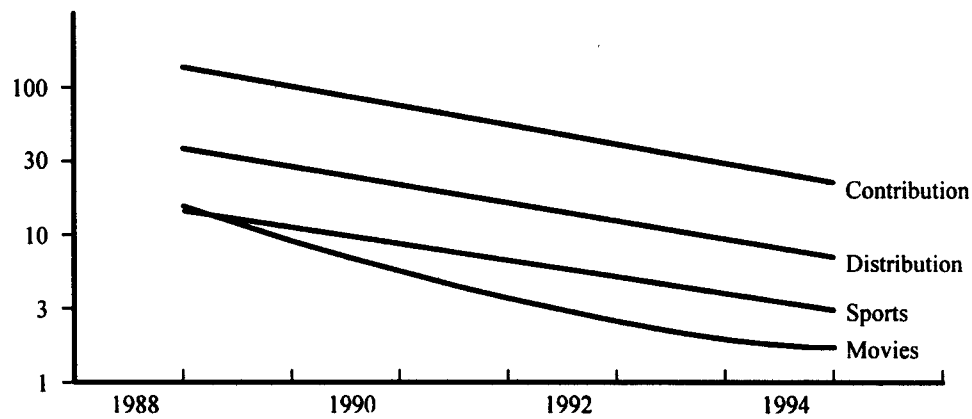


*Figure A1.3: The changing performance of video compression techniques*

<u>Sub-band coding</u>

The possible application of sub-band coding to broadcasting applications has been described by a number of authors and was reviewed at the 1993 Montreux Symposium *[14]*.

In sub-band coding the two dimensional picture resolution is split by the filters into a number of bands and each is individually coded as illustrated in Figure A1.4.

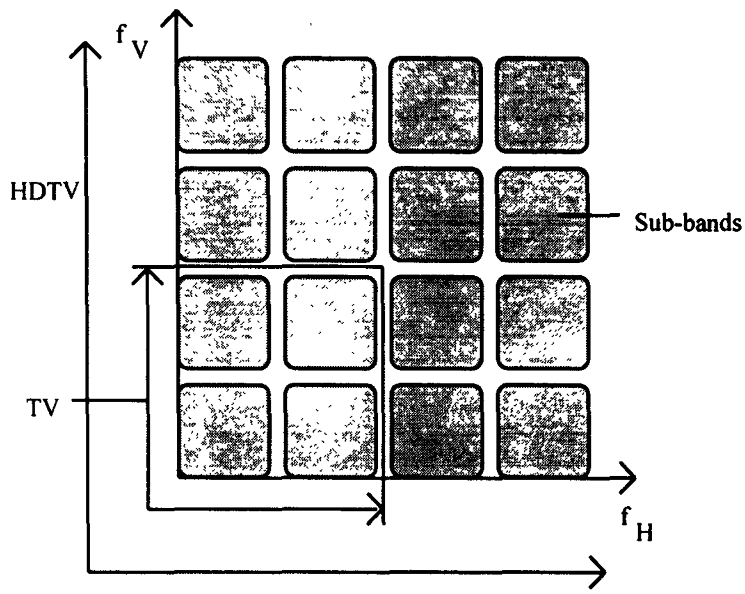


*Figure A1.4: Sub-band coding*

Multi-resolution coding

The CMTT Group of Experts (CMTT/2) are studying coding systems for the secondary distribution of TV, EDTV and HDTV which allow for "hierarchical" and "compatible" distribution of pictures of various resolutions *[15]*. That is, coding systems in which different picture resolutions can be abstracted from a common transmission format, as is illustrated in Figure A1.4 for the general case of sub-band coding.

The compatible schemes under study include:

- Perfect Reconstruction Modulated Filter (PRMF) Systems (where the two-dimensional video spectrum is split into 64 sub-bands using separate banks of PRMF filters for the luminance and chrominance.
- Pyramidal DCT
- Hierarchical sub-band DCT

In the context of compatible/hierarchical coding, the CMTT/2 Group has defined four levels of compatibility as follows:

A video transmission system employing two or more standards is:

- forward compatible if a new standard decoder is able to decode pictures from the signal or part of the signal produced by an existing standard encoder;
- backward compatible if an existing standard decoder is able to decode pictures from the signal or part of the signal produced by a new standard encoder;
- upward compatible if a higher-resolution receiver is able to decode pictures from a signal transmitted by a lower-resolution encoder;

- downward compatible if a lower-resolution receiver is able to decode pictures from the signal or part of the signal transmitted by a higher-resolution encoder;

Possible compatibility methods for obtaining forward or backward compatibility are:

- simulcasting (backward, forward (optional));
- embedded bit stream (forward and backward);
- syntactic extension (forward);
- switchable encoder (forward and backward).

Hierarchical coding is also under study within the ISO-IEC MPEG 2 programme *[16]*, where it has been recognised that at the source coding level, resolution and bit stream scalability are required for a hierarchical system. While resolution scalability can be achieved with either block or frequency-scanning techniques, bit stream scalability requires the use of frequency-scanning techniques.

The results of recent studies have been reviewed in the European Working Group on Digital Terrestrial Broadcasting which has stated that these "investigations have shown that frequency scanning is better than block scanning (2dB S/N ratio at 8Mbit/s). On the other hand, the cost of resolution scalability can be roughly evaluated at less than 1dB in S/N ratio lost in quality for the same bit-rate (including the overhead specific to resolution scalability) compared with a stand-alone scheme. In other words, the overcost of scalability means a bit-rate increased by approximately 12% for the same quality level".

## 1.3. Audio source coding

Interest in Europe is focused on the ISO/EIC MPEG MUSICAM and multi-channel MUSICAM systems. With MUSICAM, compact disc stereo quality is maintained at a bit-rate of 256kbit/s.

Multichannel coding is required for surround sound, enhanced sound effects with HDTV, and for multi-lingual applications (e.g. multi-lingual sound dubbing for "international" films). Not all of these applications require the same bandwidth and it is therefore appropriate to define a hierarchy of transmission bit-rates within the specification of multichannel audio source coding. The MUSICAM system provides such a hierarchy with defined bit-rates of:

256kbit/s (approximately CD stereo quality)

128kbit/s (approximately FM stereo quality)

as well as for a number of lower quality levels.

## 1.4. Satellite systems

<u>The current solution and trends</u>

Satellite Broadcasting is typified by the use of relatively low transmitter powers (50-250W) using spot-beam antennas emitting from specified positions in the geostationary orbit. Operations in the Broadcasting Satellite Service (BSS) band are constrained by the WARC '77 plan which assumes relatively high powers (≥200W) and spot beams engineered to provide national coverage. Operations in the Fixed Satellite Service (FSS) band are typified by low or medium powers (typically ≤50W) with beams that can be tailored for national services or, more typically, for wider European coverage.

The following review of digital satellite broadcasting characteristics draws heavily on the work of the EBU and on recent review papers by Cominetti *[17]* Lothian *[18]* Windram and Drury *[13]*.

High power transmissions - The WARC 77 plan

The WARC '77 plan was established for the use of BSS in the band 11.7-12.5GHz, for Regions 1 and 3. The plan assigned five channels to each country, with a channel spacing of 19.18MHz. It was based on FM/TV systems (PAL, SECAM, NTSC) with a receiver bandwidth of 27MHz. The protection ratios are: 31dB (co-channel, CCI) and 15dB (adjacent channel, ACI)

The plan was devised to ensure a carrier to noise ratio (C/N) of 14dB (in 27MHz BW) at the -3dB area contour for 99% of the worst month assuming a receiving system with a 90cm antenna and a figure of merit of 6dB/°K. The planned EIRP requirements assumed the use of high power (~230W) TWTAs.

Medium power transmission - Astra and Eutelsat in the FSS band

Progress in technology (especially improved receiver noise figures) have enabled satellites operating in the FSS bands to provide DTH services with considerably reduced transmitter power (~50W). Such services which typically use PAL/FM with somewhat increased frequency deviation, are receivable over wide areas of Europe with 60cm receiving antennas.

Technology trends

Current receiving systems (NF=1.5dB and 60cm antenna, 70% efficiency), available on the market, offer significantly better performances than the WARC '77 assumptions, providing for a G/T of 13.5dB/°K. Further improvements are expected with the new receivers (NF=0.8dB), soon to be available on the market.

Very promising is the technology progress achieved in receiving antennas. Various types of small size planar antennas have been developed for DBS reception in the 12GHz band because of their advantages over parabolic antennas: these include ease of installation, less degradation of performance in the presence of wind, rain and snow, and ease of installation. Furthermore, these antennas have potential to be able to control the beam direction electronically, to shape different radiation patterns, to select the desired satellite and to track it without mechanical control.

Significant improvements have also been made in satellite technology-improvements which reduce the need for orbit position selection for eclipse protection. Shaped-beam technology can be employed to cover the service area efficiently with perhaps only 1dB variation in EIRP and to provide the geographical isolation capability that is not feasible with a simple beam.

The progress in receiving systems and satellite technology has enabled satisfactory results to be obtained with satellite power that is much lower than was assumed in the WARC '77. This is the key of the success of DTH (Direct-to-Home) television services via medium-power pan-European telecommunication satellites (e.g. Eutelsat II and Astra) in the FSS band, which are no longer penalised by their relatively low EIRPs.

WARC' 92

The WARC '92 has allocated the frequency bands 21.4-22GHz in Regions 1 and 3 to the broadcasting satellite service for wideband HDTV (W-HDTV) on a primary basis from 1 April 2007. By Res. 525, a set of interim procedures to allow W-HDTV BSS to be introduced from April 1992 has been adopted. This has stimulated several European organisations to join in a project, called HD-SAT, as part of the European Communities RACE II programme (Research and Development in advanced Communication Technologies in Europe). HD-SAT began in 1992, for a duration of 3 to 4 years, and intends to prove the technical feasibility of bandwidth-efficient coding and digital modulation systems for W-HDTV satellite broadcasting in Ka band (30/20GHz) with picture quality virtually transparent to the HDTV studio production system. Other key elements are: compatibility and interworking with terrestrial infrastructure including cable, MMDS, and ATM networks.

The receiver technology at 22GHz is rapidly improving. Low-noise down-converters (LNC) at 21.4-22GHz have been developed for the consumer market, using low-cost packaged components based on GaAs HEMT. A noise figure of 1.6-1.7dB is achievable, with a conversion gain higher than 60dB.

The overall performance of the satellite chain can be affected by high rain attenuation and depolarisation, particularly on the up-link at 30GHz. In order to avoid reduced up-link availability, various techniques can be adopted, e.g.: drive limiter amplifiers on the stations, onboard demodulation and regeneration of the digital signal.

### 1.4.1. Modulation and channel coding - general considerations

In systems employing the high digital video compression ratios needed for efficient DTH broadcasting (such as motion-compensated DCT) the sensitivity of picture quality to bit errors is very high and typically, following error correction, a Bit Error Rate (BER) of between $10^{-8}$ and $10^{-10}$ is required for subjectively perfect pictures. This means that the performance of those systems in a noisy channel is critically dependent on the error protection used.

The low bit errorr rate is needed for high quality pictures may be obtained by using a forward error correction (FEC) scheme with high coding gain. The FEC is typically implemented by a convolutional code in combination with a block code. The block code may be, for example, a BCH (255, 239) code, or a Reed-Solomon (255, 239) code. The advantage of the Reed-Solomon over the BCH is a reduced interleaving depth (range 2 to 4 rather than 16 to 32) which results in a smaller memory requirement for the FEC implementation.

The convolutional code used has a low rate (typically 1/2 or 3/4) giving high redundancy and hence forming the major part of the overall coding gain.

If 8PSK is used, rather than 4PSK, then the convolutional code is replaced by a Trellis code (forming an optimal sequence coder and decoder) to obtain a better optimal coding/modulation scheme. In this case, the block code would be associated with the video encoder and the Trellis code with the modulator.

Figure A1.5 shows the block diagram of a digital transmission chain and figure A1.6 shows the probability of error versus carrier-to-noise ratio for various modulation systems, ignoring the effect of the satellite transponder and other implementation margins. In each case, the carrier-to-noise ratios are referred to the symbol rate of the modulation system. 8PSK gives a C/N penalty of about 3dB relative to 4PSK measured in the same bandwidth. However, the use of 8PSK with a 2/3 Trellis code may be better than 4PSK in certain applications.

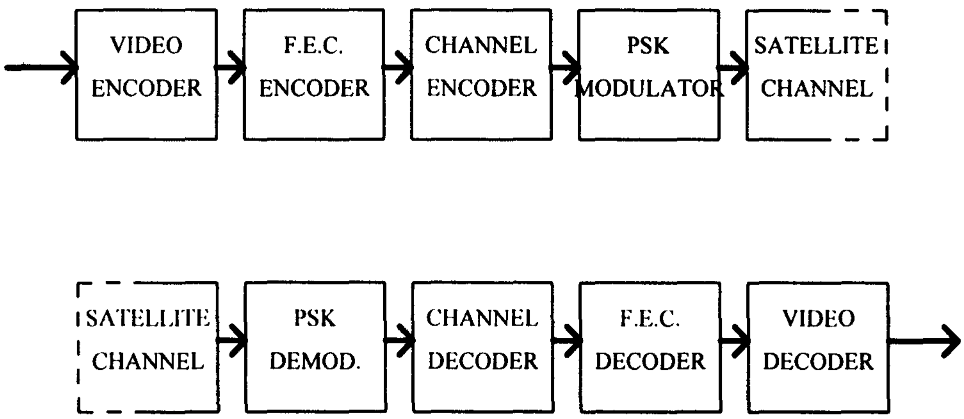


*Figure A1.5: Block diagram of digital transmission chain*

<u>C/N Requirements for digital HDTV (and comparisons with HD-MAC)</u>

Compared with analogue transmission, the failure characteristics of digital systems is very steep. It is convenient to define two threshold values of C/N for a satellite broadcasting system:

| | | | |
|---|---|---|---|
| i) | "Service quality" threshold | = | the C/N corresponding to signal quality required for, say, 99% of the worst month |
| ii) | "Service availability" threshold | = | the C/N below which signal becomes unusable and is exceeded for, say, 99% of the worst month. |

The difference between these two thresholds is about 1dB for the digital system, compared with about 6dB for HDMAC (including Enhanced E7) The implication for digital TV systems is that the service availability criterion becomes the dominant constraint when planning services. Table A1.2 below illustrates the typical and worst-case rain fade statistics for Europe.

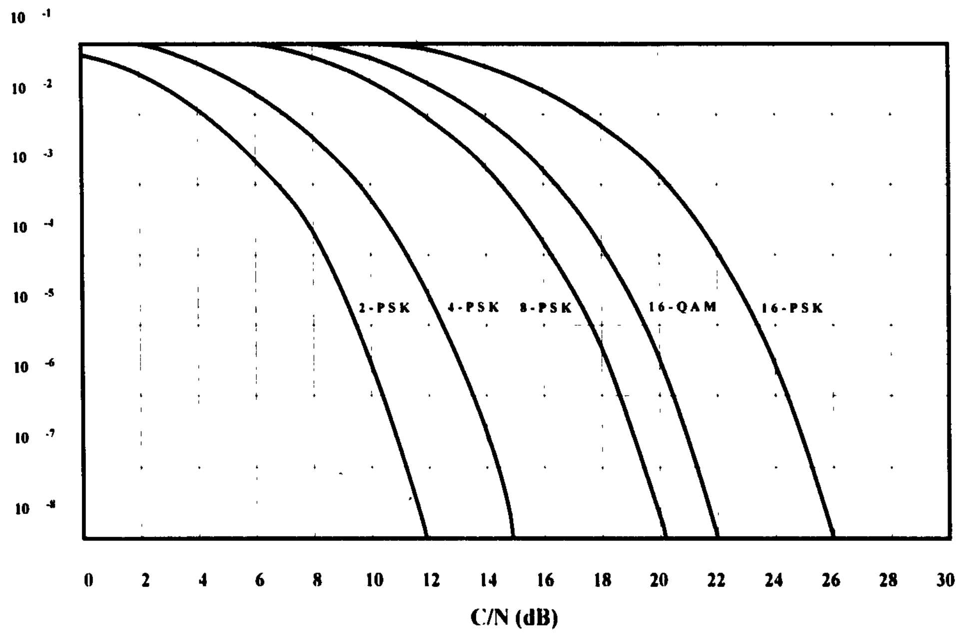


*Figure A1.6: Probability of bit-error versus C/N for various digital modulation systems*

| Rain Region | Typical Area | Atmospheric Loss (dB) | |
|---|---|---|---|
| | | 99% w.m. | 99.9% w.m. |
| H | Central Europe | 1.4 | 4.7 |
| L | Monaco | 3.1 | 10.3 |

*Table A1.2: Rain fade statistics for Europe*

The parameters of two examples of digital systems are summarised in Table A1.3.

| System | Example 1 | Example 2 |
|---|---|---|
| Video bit-rate (Mbit/s) | 35 | 45 |
| Error Protection | RS (239/255) | RSV (239/255) |
| Modulation | 4 PSK | 8 PSK |
| Channel coding | 3/4 Conv. | 2/3 Trellis |
| Total bit-rate (Mbit/s) | 50 | 72 |
| Symbol rate (MS/s) | 25 | 24 |
| RF bandwidth (MHz) | 32.5 | 31 |
| C/N 27MHz[1] BER $10^{-8}$ | 8.4 | 10.3 |
| Margin[2] (dB) | 2 | 2 |
| C/N (dB) 'Quality' | 10.4 | 12.3 |
| C/N (dB) "Availability" | 9.4 | 11.3 |

*Table A1.3: Digital broadcast system examples*

(Note 1) A reference noise bandwidth of 27MHz has been used for comparison with existing DBS systems.

(Note 2) This figure includes implementation margin (1dB), effect of AM-PM conversion (0.5dB) and interference (0.5dB).

Clearly the required service quality will be achieved at a C/N of 10 - 12dB which is lower than for an HDMAC system. However the service availability threshold is comparable with the FM threshold of 10dB for HDMAC. Thus when rain fading is taken into account, the apparent power advantage of the digital system becomes rather small.

### 1.4.2. Modulation and channel coding - practical considerations

In a practical DTH system, a prime requirement is to achieve satisfactory reception in the primary coverage by receivers with small antennas (≤60cm) whilst at the same time minimising costly transmitter power requirements. Thus there is a trade-off in the design between the quality achievable (useful bit-rate), the quality and availability of service outage-time, and required transmitter power - the trade-off needing to be evaluated against the background of the service-relevant propagation statistics (e.g. the fading effects shown in Table A1.2).

In this context the higher-order modulations, such as 16QAM and 32QAM (4 and 5bits/Hz), proposed for digital terrestrial TV broadcasting in the USA, are not power-efficient because they require the TWTA to operate significantly below its nominal power (i.e., 5 to 6dB output back-off), in a quasi-linear condition *[19]*. These constraints also apply to OFDM (Orthogonal Frequency Division Modulation), currently considered for terrestrial television in Europe, because of its high sensitivity to non-linearities.

However, as discussed in the general considerations above, channel coding approaches based on the use of QPSK rate 3/4 (System A) or TC-8PSK rate 2/3 (System B) are suitable for satellite digital transmissions *[19]*. In both systems powerful error protection is provided by concatenating the RS(255, 239) code of the codec with the convolutional code, or the trellis code, associated with the digital modem. Soft-decision Viterbi decoding is adopted in the receiver. These techniques achieve significant coding gain over uncoded QPSK or 8PSK.

| Modulation System | Spectrum Efficiency % | Eb/No (dB) @ BER = $2x10^{-4}$ | |
|---|---|---|---|
| | | AWGN | Satellite |
| QPSK | 100 | 8.0 | 9.3 |
| QPSK 3/4 | 75 | 4.3 | 5.3 |
| TC – 8 PSK 2/3 | 100 | 5.4 | 6.9 |

*Table A1.4: Performance of modulation and channel coding systems*

Table A1.4 compares the performance of the two systems on a linear channel, with additive white gaussian noise (AWGN), and on the satellite channel, in terms of Eb/No at BER = $2x10^{-4}$ at the Viterbi decoder output. With the current picture coding algorithms, this BER provides quality pictures, after error correction by RS(255,239) code, with a residual BER of about $1.1x10^{-4}$. On the satellite channel, the optimum system in terms of power efficiency is QPSK 3/4, requiring 5.3dB Eb/No with 75% spectral efficiency. TC-8PSK requires 6.9dB Eb/No and offers 100% spectral efficiency.

The implementation of the complex algorithms being considered for modulation and channel coding requires an extensive use of VLSI technologies at the demodulator. Systems based on QPSK modulation are easier to implement than systems based on trellis coded 8PSK which, in addition, require a larger implementation margin. VLSI single-chip soft-decision Viterbi decoders for rates 1/2 and 3/4 convolutional codes are already available on the market for a maximum bit-rate of 45MBit/s. Trellis coded 8PSK rate 2/3, at a maximum bit-rate of 50MBit/s, have also been developed recently. Single-chip RS(255, 239) coder/decoders are also available for bit-rates in excess of 160MBit/s. The next step could be the development of fully digital modems for TV/HDTV applications based on the advanced solutions currently adopted in digital transmission at the Intermediate Data Rate (IDR) on communications satellites.

Interference considerations and transmission capacity

In Europe, Eutelsat II telecommunications satellites carry 36/72MHz transponders, in Ku-band (14/11GHz), suitable for digital transmission at 60/120MBit/s with QPSK modulation. On the 72MHz transponders, TDMA telecommunication services at 120MBit/s are in regular operation. The 36MHz transponders are particularly suitable for the distribution of digital television on supra-national coverages. A transmission capacity of 45MBit/s or 60MBit/s is usable with QPSK 3/4 (System A) or TC-8PSK 2/3 (System B), respectively.

The Astra satellites 1A,1B,1C introduced by the Société Européenne des Satellites (SES), carry transponders with 26MHz bandwidth. The usable transmission capacity is then about 34MBit/s and 44MBit/s, depending on the transmission system (A or B). The capacity will be further increased to about 41MBit/s and 55MBit/s in the new generation Astra satellites (1D and 1E), operating in the 11.7-12.5GHz band (19.2° E), which will carry 33MHz bandwidth transponders.

In the case of BSS satellites at 12GHz (WARC '77), the channel bandwidth is practically defined by the receiver since the five channels provided by the satellite are separated from each other by about 77MHz (4x19.18MHz). A fundamental requirement for the introduction of digital television is the need to comply with the WARC '77 protection ratios (31dB CCI, 15dB ACI) in order to ensure coexistence with the analogue services (e.g. in PAL, D2-MAC) already into operation. Results of the RAI studies *[20]* indicate that a maximum symbol rate of 30 MBaud is usable in the WARC '77 channels. The corresponding useful bit-rates are 45MBit/s and 60MBit/s for channel coding systems A and B, respectively.

It is important to note that, in a fully-digital scenario, it is possible to operate without any modification to the WARC '77 Plan with significant reduction of the satellite EIRPs (of 7 to 10dB), thanks to the very low CCI protection ratios (13 to 16dB) required by the digital systems. This would potentially allow great flexibility of the coverage-area design to be achieved.

On the basis of these considerations, the European Working Group on Digital Terrestrial Broadcasting (WGDTB), in considering the satellite broadcast possibilities, has taken a useful bit-rate of 45MBit/s as the basis for further discussion of standards applicable to the multiplexing of video, sound and data services in digital satellite broadcasting. *[3]*.

Examples of possible System A (QPSK: 3/4) implementations providing useful bit-rates of 41 to 45Mbit/s are given in Table A1.5 below.

| Satellite | Orbital Position | EIRP (dBW) | Number of Channels | Satellite Channel Bandwidth (MHz) | Receiving diameter (cm) for 99.7% w.m. high quality service | |
|---|---|---|---|---|---|---|
| | | | | | Beam Center | - 3bB contour |
| High Power | | | | | | |
| TDF 1.2; TV-SAT 2 | 19° W | 64 - 65 | 5 | 40 | < 40 | < 40 |
| TELEX-X | 5° W | 63 | 2 | 42 | < 40 | < 40 |
| HISPASAT | 31° W | 58 | 5 | 42 | 40 | 55 |
| EUROPESAT 1 | 19° W | 57 | 14 | 42 | 45 | 65 |
| Medium Power | | | | | | |
| ASTRA*- (1D), (1E) | 19.2° E | 53.5 | 36 (2x18) | 33 | 65 | 90 |
| EUTELSAT II F1 - 2 - 3 - 4 | 13 - 10 - 16 -7° E | 51.5 | 36 (4x9) | 36 | 85 | 120 |
| EUTELSAT II F - 6 (Hot Bird) | 13° E | 49.5 | 18 | 36 | 110 | 150 |

*The receiving antenna diameters refer to 41Mbit/s; QPSK 3/4 signals transmitted on 33MHz channels.

*Table A1.5: Characteristics of high and medium power digital satellite systems*

The table shows the characteristics and performance of high power and medium power satellites, currently in operation or planned for future introduction, in terms of capacity (number of channels) and on-axis EIRPs. The last two columns give the antenna diameter for high quality reception (99.7%) at the beam centre and at the -3dB contour. The receiving antennas are very small (40 TO 65cm) for high power satellites at 12GHz. Europesat 1 at 19 W should provide 14 channels, potentially usable for direct-to-home broadcasting of up to 56 (14x4) EDTV programmes or 14 HDTV. In the case of medium power satellites, Eutelsat II F-6 (Hot Bird) will provide up to 72 (18x4) EDTV programmes or 18 HDTV, receivable with 110 to 150 cm antennas. Eutelsat II F1, 2, 3, 4, currently in operation, could provide a total of 144 (4x9x4) EDTV programmes (or 36 HDTV), but from four different orbital positions. A similar impressive transmission capacity will be made available by the future Astra 1D, 1E satellites, from the same orbital position (19.2°E) in the BSS band (11.7-12.5GHz). The required receiving antennas will range from 65 to 90 cm.

This evolutionary scenario for satellite television, stimulated by the introduction of digital techniques, is expected to be a reality before the end of the century. In a longer-term perspective, digital W-HDTV should be introduced in the 21.4-22GHz band, with the same system concepts as those proposed for use in the 11-12GHz band. However, due to severe abrupt failure characteristics, in the event of rainfall this band may not be able to provide the required service availability without a penalty on the satellite transmit power. A method to extend service continuity up to about 99.9% of the worst month, without increasing the satellite power, has been developed *[21]*. This advanced system is based on the adoption of layered modulation in conjunction with layered picture coding and layered channel coding. It provides graceful degradation from HDTV quality, achievable for most of the time, to conventional TV quality during high rain fades.

Hardware availability and performance

A flexible digital codec using MPEG digital compression is manufactured and marketed by National Transcommunications Limited (NTL) in the UK. The codec is suited to cable head-end, point-to-point and DTH applications. In a paper summarising the performance of the system presented at the 1993 Montreux Television Symposium *[13]*, it is suggested that for medium power (33-36MHz bandwidth) satellites it is appropriate to try to achieve satisfactory satellite "quality" and "availability" times for transmitter powers of 50dBW together with receive antenna sizes of 60cm or less. In this case the optimum gross bit-rate with a QPSK 3/4 system and a receive noise figure of less than 1.5dB reduces to around 50Mbit/s, and the useful bit-rate to around 36Mbit/s.

Clearly the trade-off between receiver dish size, transmitter power, and the fine-tuning of number and quality of services available from each satellite channel will be decided by the broadcasters in light of their developing needs.

## 1.5. Terrestrial systems

### 1.5.1. General considerations

The Digital Terrestrial Television Broadcasting (DTTB) situation is a much more complex one than for the satellite case because:

a) While in an "all digital" scenario the potential of DTTB is similar to that described for satellite systems, currently the introduction of the highest bit-rate DTTB services is constrained by the presence of comprehensive analogue services in the allocated UHF/VHF bands. Thus a distinction needs to be made between the introduction, transition and final phases of DTTB when describing the potential features of the system.

b) There is the need to consider how systems can make the evolutionary step between the constrained performance of the transition phase and the full performance of the final phase of DTTB.

c) There is the probable need to provide for a graceful failure characteristic.

d) Unlike satellite television services, terrestrial television services are receivable on portable and in-car receivers, either as secondary services (the norm for portables today) or as primary services where the digital system is tailored to suit either portable or mobile reception requirements.

e) The complex problems posed by portable and mobile reception are in the relatively early stages of study in the R & D projects.

Thus before assessing the long term potential of DTTB it will be necessary to review the situation existing today in the allocated terrestrial television bands.

### 1.5.2. The European television network infrastructure

With the PAL and SECAM systems, the analogue nature of the transmission makes the received signals highly susceptible to perturbation by receiver noise and very low levels of co-channel interference (CCI) - the effects of which are commonly visible on even the best of domestic receivers. Thus to obtain good coverage from a single PAL transmitter, relatively high powered transmissions are required to reduce the effects of receiver noise (EIRPs typically 100kW).

In a terrestrial broadcasting network, coupling the PAL high power requirement with its susceptibility to CCI means that main transmitters using the same frequency must be positioned at considerable distances apart, and that nearby main transmitters must of necessity use different frequencies. In European networks typically a high percentage coverage is obtained for a number of national and regional networks by the use of a "skeleton" network of main transmitters radiating high EIRPs from high masts augmented by large numbers of low powers relay transmitter to fill in "shadow" areas resulting from the regional topology. These arrangements mean large numbers of frequencies are needed to support a single national programme channel. Thus the system is extremely inefficient in terms of spectrum utilisation. In the UK, for example, 44 RF television channels are needed to support the four "national" programme channels, and similar ratios are common elsewhere. Figure A1.7 shows the number of transmitters using each of the UHF channels in Europe overall (actually in mid-1990, although the numbers have not changed much since then). In many countries, there are restrictions on channels 61 to 69 (they are used by other services) and this causes the discontinuities which can be seen. There are also restrictions for channels 34 to 37, and channel 38 is used by Radio Astronomers.

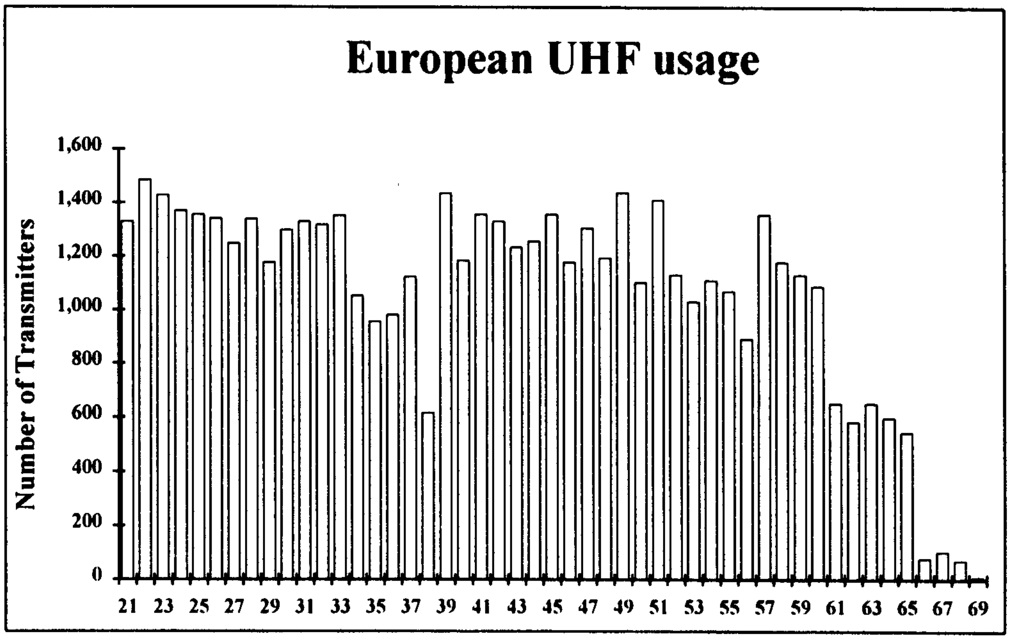


*Figure A1.7: Number of transmitters for each UHF channel*

Service protection

The primary source of interference in any European country is co-channel emissions. One primary aim of planning is to keep the impairment caused by co-channel interference at:

- no worse than CCIR Impairment Grade 4 for any interference which is continuous;

- no worse than Grade 3 for interference which is present for only 1% of the time (5% in a few countries).

The latter requirement is usually the more difficult to meet. Few problems are experienced with other interference mechanisms and they now offer few constraints on planning, either for the choice of channels at a given station, or much more importantly, for the choice of channels for overlapping coverages from adjacent stations. The last point is particularly important. Within Europe, there are extensive coverage overlaps from adjacent stations in most countries and, indeed, any channel relationship in overlap areas can be found, except for co-channel.

Although the directional properties of a domestic receiving antenna help to reduce the levels of unwanted signals, it is very important to note that where the programmes from adjacent stations are different, viewers will probably use both, usually with separate antennas (but not always!). In the case where the overlapping services are in adjacent channels, the field strength of the wanted signal can be more than 20dB below that of the unwanted signal. This has important implications in the case where additional (for example, digital) adjacent channel transmissions are proposed from a given transmitting station.

There have traditionally been some planning constraints regarding the channels to be used at a given station. For example, adjacent channels, channels subject to local oscillator interference, or image channels have tended to be avoided[1] in the past (although not in all countries). After the performance of television receivers has improved, these constraints have been removed.

The last constraints to go has been the use of adjacent-channels at a given site (although this is now happening), partly because of the difficulties of obtaining: enough suppression of spurious components in the lower adjacent channel (a major problem with high efficiency klystron transmitters), and enough isolation between transmitters to avoid inter-modulation problems.

However, the use of separate, but nearby, sites for adjacent channel transmissions has proved successful. The effect for the viewer is that he can use a single receiving antenna, provided that the transmitting sites are fairly close together, and that he experiences little or no picture impairment so long as the field strengths of the signals are within a few dB of one another.

In most countries in Europe, the PAL/SECAM broadcast networks have evolved to maximise the number of programme channels obtainable from the VHF/UHF allocations and to maximise the "national" coverage. The exact situation varies from country to another, as the following examples illustrate.

Examples of existing situations

In the United Kingdom, planning for analogue television in Bands IV and V has provided four services, each covering 99.3% of the UK population. Generally all four services are radiated from each transmitter site. The channels are grouped in the form

[1] These channels to be avoided are usually called "taboo channels" or "related channels". Note that the "taboo" classification of these channels refers only to their non-use at the same transmitting site. They may be more appropriately referred to as "related channels".

N, N+3, N+6, N+10, for Band IV and lower Band V, and N, N+4, N+7, N+10 for upper Band V. These groupings were chosen to avoid reception problems from adjacent channel, image channel, and local oscillator interference.

In Germany, the VHF/UHF bands, (I, III and IV/V) are used by analogue television services. Full area coverage for the three public networks is provided, which requires the operation of roughly 290 high power transmitters and in addition 8000 fill-in stations. Moreover about 200 private stations are operated in areas with a high population density. Channels 61-69 are not used by analogue television services. If this part of the spectrum can be made available for broadcasters, it may offer a good chance for a start with digital terrestrial television.

In Scandinavia, the introduction of digital terrestrial services seems to be viable without changing current PAL broadcast network because this now provides nationwide coverage for only two services. The presumption made in the preliminary studies is that each of the current networks should have the possibility to be "simulcast" with the same coverage area in a digital format. The single frequency option is interesting, particularly the concept of Local Single Frequency Networks giving the possibility of rather sharply defined regions for regional broadcasts. In the short term, nationwide SFNs will be difficult, due to the existing PAL transmitter networks.

In France, bands I, III, IV and V (up to 830MHz) are presently used to broadcast 6 networks. Band III is shared with a land mobile system (radiocom 2000) and no possibility exists in that band at present. The networks have 112 main stations and 3290 re-broadcast stations. The percentages of coverage are respectively 99.9% for networks 1,2, 3 and 87, 77 and 66% for networks 4, 5 and 6. Studies on the possibility of introducing digital television in the UHF band are now being carried out on the adjacent channels of the three first networks.

In Spain, five programmes (2 public and 3 private) with national coverage, and 1 or 2 programmes with regional coverage, are at present provided using bands 1, III, IV and V (channels 2 to 4, 5 to 11 and 21 to 65). The two public networks reach 98 and 96% coverage, and the three private 80%. Although all television stations at present allocated in bands I and III must migrate to the UHF bands before the beginning of 2000, leaving them free for other services, studies have confirmed that there will be a certain number of free and taboo channels available for the start of digital broadcasting, giving the possibility to simulcast the 5 current PAL programmes with national coverage.

In Italy, the channel occupancy in the VHF/UHF bands is very close to spectrum saturation. Planning criteria, derived from the Stockholm Plan, allow the use of taboo channels with protection ratios updated to the improved performance of new-generation receivers. It is unlikely to be possible to introduce Single Frequency Networks (SFN) on a nationwide basis, at least in the initial phase. Instead, there are opportunities of finding frequency assignments on a local basis in urban areas. By careful selection of the transmitter locations, a service level of about 50% of the population resident in towns with more than 50,000 inhabitants seems possible, for at least one digital Television/HDTV network.

Channel availability for DTTB

New techniques of channel coding like COFDM could allow the implementation of single frequency networks and permit much more efficient use of the frequency bands than the existing analogue system. However, such networks can only be placed in a band reserved solely for this service and there is, at present, no broadcasting band (or other band) clear of any occupation.

If we look below 3GHz, which is a reasonable limit for good terrestrial transmissions, frequency bands are used by numerous services and notably by land mobile systems which have, and will grow in, importance. Moreover, other new broadcasting systems are searching for new allocations, and WARC 92 has just allocated a band for Digital Audio Broadcasting between 1452 and 1492MHz.

In this context, it is not likely to be possible to obtain a new band for digital terrestrial television broadcasting. It is thus necessary to make the new broadcasts in the bands currently allocated for television broadcasting, which seems to be logical, since the new service could take the place of the analogue PAL and SECAM services in the long term.

One possibility is to use the "related" channels; that is, those which have not so far been used at each transmitting site. This would require the radiated power to be kept low. This is necessary to avoid co-channel interference to service areas of other nearby analogue transmitters, which operate in its related channels.

The maximum effective radiated power of a digital service will, therefore, almost certainly be limited by the constraint of not causing co-channel interference to existing analogue services. This may lead to digital transmitter radiated powers of the order of 20 to 30dB less than the existing analogue transmissions at the same site.

The exact level of coverage will depend on the modulation method used for the digital service. It will, for example, be less for an HDTV service using 64QAM than for a standard-definition service based on QPSK. Potential coverage will be difficult to quantify with any precision until full agreement has been reached the protection ratios and minimum field strength figures for the digital system.

### 1.5.3. The characteristics of DTTB systems

Network economics

In terrestrial broadcasting networks designed to provide "national-wide" coverage, either as a single programming chain or as a set of regional programmes, the most economical arrangement, whether analogue or digital techniques are employed, is that of a "skeleton" network of main transmitters using relatively high EIRPs augmented by lower-power relays to "fill-in" shadow areas. The distance between the main transmitters in today's analogue networks based upon this arrangement is of the order of 70 - 80km, with the main transmitter coverage area being, in the most simple case (nominally to the horizon) within a circle some 40 - 50km in radius.

Transmitter power requirements

In simple theoretical terms the main difference between analogue AM and digital television broadcasting is that much less transmitter power is required in the digital case to achieve the same coverage area. The C/N requirements to achieve error-free pictures with various digital modulation levels in the PSK/QAM family are compared with the PAL C/N requirement for grade 4.5 pictures in Table A1.6. Also shown in Table A1.6 are the relevant transmit power requirement assuming the same transmitter/receiver antennas and locations.

| | PAL | 2PSK | 4PSK | 16QAM | 64QAM |
|---|---|---|---|---|---|
| C/N required for - grade 4.5 (PAL) - error- free digital | 45dB | 11dB | 14dB | 21dB | 27dB |
| Theoretical EIRP w.r.t. PAL | — | - 34dB | - 31dB | - 24dB | - 18dB |
| Assumed Digital Additional Implementation Margins | — | — | — | 1dB | 2dB |
| Amended EIRP required w.r.t. PAL | — | - 34dB | - 31dB | - 23dB | - 16dB |

*Table A1.6: System C/N requirements*

Digital reception characteristics

As the experience with Teletext has amply demonstrated, compared with PAL, digital broadcast systems are relatively susceptible to echoes due to multi-path reception, especially to multiple low level echoes of short delay time, which reduce eye-height in a notably digital system but cause only a degree of picture "softening" with PAL. The second main characteristic of digital reception is that as the C/N is reduced, at a certain threshold value, the transition from almost perfect, error-free, pictures to total picture failure is typically very rapid. This is especially true when sophisticated error protection schemes are employed. Thus digital systems do not intrinsically exhibit the desirable graceful failure characteristic of the PAL and SECAM systems.

Characteristics of terrestrial UHF channels

The signal-to-noise ratio (and also the signal-to-interference ratio) in a terrestrial UHF channel is highly variable because of factors which include:

Localised variations in field-strengths: due to screening and reflection by buildings and hills, fading throughout a wide range of frequencies occurs due to short-delay multipath efforts or signal attenuation. These effects are highly variable in the short term (e.g. as the wind blows and moves the transmitting and receiving antennas and any reflecting or attenuating object in the direct or reflected signal path), the medium term (e.g. with tidal fading or increased reflections from wet surfaces after rainfall) and the long term (e.g. as new buildings are constructed or trees grow and come into leaf). Thus deficiencies in signal-level will occur, not only at the fringes of the designated service areas, but also in "pockets" well within the coverage area; the size and location of the pockets will vary with time. If the new digital services share the existing UHF bands with analogue services, the co-channel transmitters could be carrying PAL, (or SECAM) signals of analogue services in the same or different countries. Even if new channels within or outside the existing bands are used for digital broadcasting, CCI may occur from other transmitters carrying the same or different digital services, although CCI from digital services would be noise-like and therefore be of a lower order than PAL/SECAM transmissions. For some viewers, the CCI source will be on the same bearing as the wanted signal; in such cases improvements in antenna directivity will not help to reduce the effects of CCI.

In an attempt to take account of these variations, some authors have used a simplified approach based upon the "location factors" derived from CCIR Rec. 370. Considering the abrupt failure characteristic of the (assumed) DTTB system in light of the (assumed) probability of achieving, or not, the average predicted wanted field strength at a particular location, it has been argued (based on CCIR Rec. 370) that to ensure satisfactory reception of 90% of locations an increase in EIRP of around 17dB is required. (Addendum to Technical Report of the IRT, No. B 126/92 - R. Bruggar and G. Petke) *[22]*.

In network planning for conventional analogue broadcast services using the simplified CCIR Rec. 370 statistical prediction method, location variation values are derived from a log/normal distribution graph which is based upon an amalgam of many measurements of TV carriers and VHF/FM signals. In effect the distribution of Rec. 370 applies only to narrow band signals and includes the effects of major and minor features of local topography. It is not therefore directly applicable to wider band signals such as COFDM whether used for DAB or DTTB, or where more sophisticated terrain-based planning techniques are employed.

Investigations into DAB systems reported in the CCIR in December 1992 *[23]* suggest that the field strength probability distribution of a 1.75MHz COFDM signal is significantly less than the narrow band measurement would suggest (at least 3dB reduction in standard deviation).

Investigations into the location factors applicable to COFDM/DTTB systems of 8MHz bandwidth with fixed roof-level reception are planned under the ITC's SPECTRE programme for 1993. While not yet certain, it seems likely that a further reduction in the standard deviation of usable signal strength will be obtained with the yet wider-band COFDM signal.

There remains, however, the question of what location factor to use in planning DTTB services. Noting that, in many cases, DTTB coverage will be limited not by noise but by CCI (PAL-COFDM) during the introduction and transition phases of DTTB, it could be argued that there are a number of factors in favour of the use of a location factor of 50%. These include, for example:

- The possibility of adaptive CCI filtering in receivers;
- The possibility, in many locations, of improved antenna discrimination against the source of CCI;
- The possibility of introducing a (step-wise) graceful failure characteristic.

Furthermore, the variability of the signal level with local location will mean that locations extending well beyond the edge of nominal service area will be able to receive error-free pictures. Thus, as today, in many locations the best signals will not necessarily come from the nearest main transmitter.

It can be argued therefore that the essential problem is not that of population coverage but of how an individual will be able to know whether his particular location and antenna arrangements are satisfactory for DTTB reception when considering the purchase of a new DTTB receiver. Possibly, experience will show that when in-area PAL or SECAM transmissions are received from the nearest, main station at high quality across all PAL or SECAM channels the probability of satisfactorily receiving DTTB will also be high, thus reducing the need for a special test. Alternatively, greater reliance may be placed upon the planning figures to ascertain where pockets of dubious reception are - again reducing the problem to manageable proportions. Either way, this is an important topic for further study, even when systems and receivers have designed-in step-wise graceful failure characteristics.

### 1.5.3.1. DTTB systems under study in the USA

In 1990, the Federal Communications Commission (FCC) ruled that only with NTSC non-compatible systems would be considered for adoption as the standard for terrestrial transmission of HDTV. Transmission was to take place in taboo channels, simulcast with existing NTSC services. Four different digital systems were developed, all of them based on digital video and audio compression, channel coding, and modulation techniques:

Advanced digital television (ADTV)

- Advanced Television Research Consortium (ATRC = Philips NA, Thomson, D. Sarnoff Research Center, NBC, Compression Labs)

Digital spectrum compatible HDTV (DSC)

- Zenith/AT&T

Digicipher

- General Instrument Corporation

Channel compatible digicipher (CCDC)

- Massachusetts Institute of Technology (MIT)/General Instrument Corporation

A test facility (the Advanced Television Test Centre (ATTC)) was created to assist the FCC Advisory Committee in its choice of system. Testing procedures were defined and prepared by sub-committees of the Advisory Committee for Advanced Television Systems (ACATS). Hardware prototypes of the four digital systems were tested one after the other.

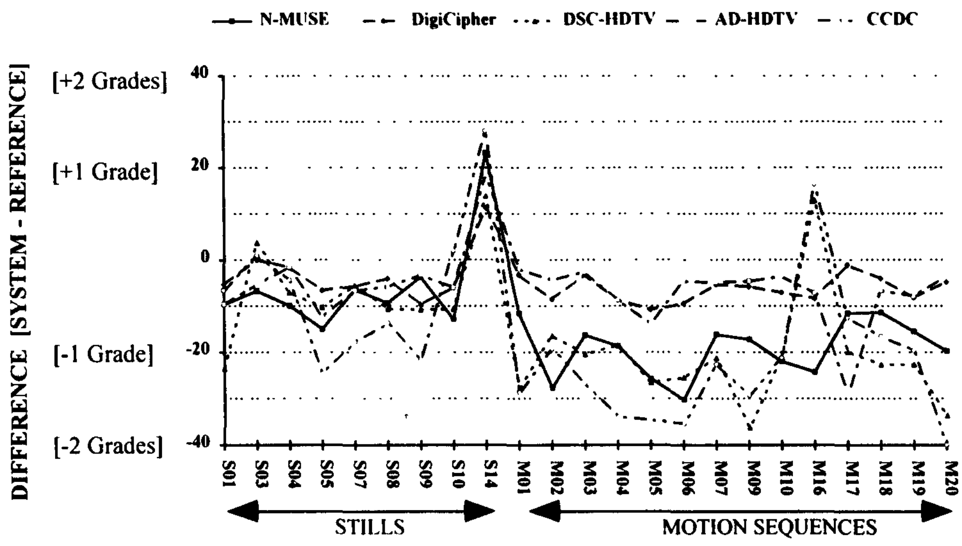


*Figure A1.8: Average differences between quality judgements for the 1125-line studio quality reference and for each of the proposed ATV systems*

The results obtained so far in the US test and evaluation programme were reviewed in The Report of the FCC Special Panel *[24]*. Some of the more important findings may be summarised as follows:

- recognition that enormous progress has been made in the development of (digital) ATV systems for the United States.;
- assignment plans have been prepared for simulcasting by all the existing NTSC broadcast stations;
- proposals achieve good coverage in almost all areas but concern is expressed regarding the degree of interference from ATV-into-NTSC;
- on picture quality, 2 systems performed well, 2 systems less well (see Figure A1.8);
- because all four systems would benefit significantly from further development the Special Panel did not recommend any one of the four excellent systems for adoption at that time;
- the Special Panel recommended that the four proponents be authorised to implement their improvements (as submitted and approved);
- improved systems improvements to be ready for testing not later than March, 15th 1993.

Subsequently, in May 1993 agreement was reached on the HDTV "Grand Alliance" proposal. The proposed system will be reviewed by the FCC's Advisory Committee on Advanced Television Service.

In outline, the Grand Alliance Proposal will be based upon:-

- A multi-format system:

  1050/1:1/30, 24 square pixels

787.5/1:1/30, 24 " "

1050/2:1/60 " "

- Video compression:

  using MPEG-2 syntax (but no B Frames)

- Modulation using one of:

  4 VSB/6VSB/32QAM/32SS-QAM

  (6VSB is a trellis-coded version of 4VSB)

- Audio coding using one of:

  Dolby AC-3

  Multi-channel Musicam

  MIT-AC

- Transport format:

  a packetised, prioritised data format

### 1.5.3.2. DTTB systems under study in Europe

In Europe almost all DTTB studies are based upon the (Coded) Orthogonal Frequency Division Modulation (COFDM) approach, successfully demonstrated for Digital Audio Broadcasting (DAB).

The advantages of this approach are well documented *[25] [26] [27]*.

The following summary of COFDM and OFDM design properties has been largely abstracted from these references.

<u>OFDM system characteristics</u>

The OFDM multiplex consists of a large number of carriers equally spaced in frequency, where each carrier is modulated by some digital modulation method, e.g. QPSK or 16QAM. The spectrum of each modulated carrier is arranged to overlap the spectrum of its neighbouring carrier in such a way that the information content of each carrier is mutually orthogonal. It is in this way that the OFDM modulation scheme achieves its high spectral efficiency.

This orthogonality criterion is satisfied if the carriers form a set of orthogonal functions. In the case of sine and cosine waves this condition is met if the carrier frequencies are harmonically related. Under these conditions the orthogonal carrier set is a Fourier series. The OFDM signal is constructed by means of the Inverse Discrete Fourier Transform and demultiplexed by means of the Discrete Fourier Transform (DFT) function. Other requirements of a practical receiving system are carrier recovery, and synchronisation to the DFT blocks, the details of these are described in reference *[28]*.

OFDM TV interference rejection

The OFDM spectrum is ideally suited for use in a hostile interference environment where the interference can be approximated as single tones or continuous waves (CW) at known positions in the spectrum. This property of OFDM arises because information need not be sent on the carriers affected by the CW interferers. Hence, provided the interference can be well defined, portions of the transmitted spectrum can be cut out, leaving a spectral template where valid information is transmitted. The receiver would naturally only look for information within the given template - hence it would ignore the interference.

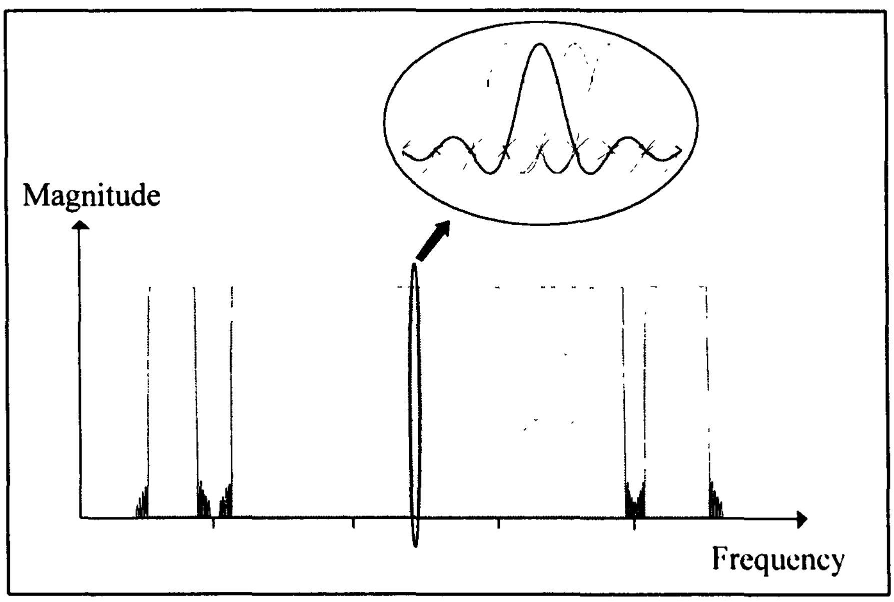


*Figure A1.9: The conditioned OFDM spectrum*

Figure A1.9 shows how the OFDM spectrum is tailored to reject the vision and sound carriers from co-channel interfering (CCI) analogue television signals. Furthermore, since the spectrum is rectangular with steep sides it just fits the channel making it spectrally very efficient. The exact width of the spectrum can also be adjusted by cutting out carriers at the edge of the channel, thereby optimising OFDM to give best adjacent channel interference (ACI) rejection as well. To achieve the very high levels of ACI rejection required (some 70dB), a good quality SAW filter is necessary at IF, with further filtering at both RF and baseband.

Alternatively, it has been proposed that interference detection and processing in the receiver could be used adaptively to "null-out" the interfering signals only when significant CCI was present (typically up to 5% of time) *[29]*. This would allow the full OFDM spectrum to be used (saving a loss of up to 20% in useful transmission capacity). Thus only suitably-positioned low-priority data (the highest frequency components) would be lost to receivers towards the edge of area, and normally only for a small proportion of the time.

Reception in multipath - fixed, mobile and portable reception

The OFDM signal gives excellent performance in the presence of multipath propagation conditions because the symbol period of each carrier can be made much longer than the delay spread of typical multipath reflections. In this way the eye-height of received symbols is not significantly reduced.

For normal fixed reception using directional receiving aerials, the delay spread of typical multipath is short. Hence, the standard OFDM system, without special error correction or special measures taken to combat long delay echoes, should give excellent performance for fixed reception of digital television. OFDM, operated in this way, is at its most efficient since nearly all the available data capacity is used for information bits, and coherent demodulation of the carriers is possible.

However when single frequency networks (SFNs - see below) are considered for relay operations, it is desirable to include a guard interval to overcome the effects of significant echoes of longer delay. The required guard interval is approximately equal to that of the echo delay time. Hence significant multipath signals via a path of around 3km would require a guard interval of around 10μS and longer multipaths proportionally longer intervals. As inclusion of a 10μS guard interval in an OFDM system employing a 64μS symbol period implies a loss of data capacity of around 13.5%, systems with longer symbol periods are being studied.

For mobile reception with non-directional receiving aerials the multipath requirement is very severe indeed. Often in built-up areas only reflected signals are present and no direct path to the transmitter exists. The receiver is constantly moving through a changing multipath environment and adaptive equalisation of digital signals is virtually impossible. However, OFDM, if it is, correctly tailored to this environment, can be made to work well. This has been demonstrated many times in the Digital Audio Broadcasting (DAB) application *[30]*.

For mobile television reception to be possible, much heavier levels of error correction are required, and coherent demodulation then becomes less feasible. The additional overhead of the error correction, perhaps a factor or two, coupled with the loss of data capacity due to the guard intervals, will restrict the resolution of the picture obtainable. Enhancing the modulation from 4-level (such as QPSK) to 16-level (such as 16QAM or 16 PSK), will probably not provide a workable means for achieving greater resolution. The 16-level modulation schemes have greater sensitivity to channel distortions, and schemes such as 16QAM will require coherent demodulation of the carrier, which is difficult to perform in a mobile environment because reflection paths are continuously changing.

Portable reception

For portable reception with stub antennas there is a mean loss of signal level w.r.t. roof-level antennas theoretically estimated to be:

| | |
|---|---|
| loss of antenna gain | 10dB |
| loss of height factor | 14dB |
| | 24dB |

Because building attenuation can also be very severe (tens of dBs), these severe losses have led many experts to doubt the feasibility of designing for portable reception. However experience with PAL/SECAM suggests that the variations of signal level with location are such that areas of satisfactory reception can be found in many locations in the inner service area. This would suggest that if special measures were taken to improve the capability of digital television reception at the SDTV and LDTV quality level there could be a very significant improvement compared with PAL:

a) in the number of locations where portable reception is possible;

b) in the reception of pictures that are noise and "ghost" free.

Symbol period

To date, most experimental OFDM/DTTB system designs have been based upon a symbol period of Ts=64μS with very small guard intervals to preserve transmission capacity. As it is clearly advantageous to employ time-domain guard intervals and long Symbol Periods ($T_s$) when significant long delay echoes are a problem, the question arises as to the practical problems of increasing $T_s$ from 64μS to a much higher value. Studies to date suggest that if T is increased by N frames:

- signal processing complexity increases by log N times;
- signal storage requirements increase by N times (Thus cost factor alone might limit the maximum $T_s$ to around 512μS in a practical system).

More seriously perhaps is that as $T_s$ is increased the carrier separation in the 8MHz bandwidth reduces ( $F=1/T_s$). This reduction can constrain the phase-noise-suppressing quality of the receivers' carrier recovery circuits which can in turn significantly increase the C/N implementation margin of systems - especially systems employing higher order modulation such as 16 and 64QAM.

If these considerations lead to a maximum value of $T_s$ in the region (64-256μ S), then the maximum practical value for guard intervals ($T_g$) would also likely be limited to (10-20%) of these values, as shown in Table A1.7.

| $T_s$ | $T_g$ (max) for a loss in transmission capacity of:- | |
|---|---|---|
| | 10% | 20% |
| 64μS | 6.4μS | 12.8μS |
| 128μS | 12.8μS | 25.6μS |
| 256μS | 25.6μS | 51.2μS |

*Table A1.7: Transmission capacity loss due to guard intervals*

Single frequency networks

As OFDM has excellent performance in the presence of interference and multipath it may be used in a single frequency network (SFN). This allows complete coverage of a country having a large number of terrestrial transmitters all on the same frequency. There are two ways OFDM can be used with an SFN and these have different properties.

A first method, when using directional receiving aerials, is to make use of the fact that QPSK-OFDM has a protection ratio for OFDM interfering into itself, which is lower than the front-to-back ratio of the receiving aerial. This means that an interfering signal arriving from behind the aerial will not be harmful so long as it is not stronger than the wanted signal arriving at the front of the aerial. The same argument also applies to signals received at the sides of the aerial. Moreover, the path loss across the service area is also greater than this co-channel protection ratio, so an interfering signal arriving at the front of the aerial from the next transmitter in line will also be sufficiently attenuated. Hence, in principle, each transmitter may use the same frequency without interference. Note also that each transmitter may also radiate different programme material as the condition is independent of the data stream. In practice such networks must take account of the vagaries of propagation and of regional/local topography and a good deal of experiment and computer modelling is required to establish where, and to what extent, SFNs will be effective.

This SFN method is less useful when higher-order modulation is used for HDTV, because the OFDM-to-OFDM protection ratios are likely to exceed the front-to-back ratios of the receiving aerials. In practice, even with QPSK modulation, some outage due to interference can be expected at the edges of the service areas. This, however, may not be too much of a problem if the viewing population is distributed well within the service area.

A second method of producing a single frequency network with OFDM is to make use of its enhanced multipath properties. Provided that all transmitters in the SFN are radiating the same programme material, and this is a constraint, the adjacent transmitter signal can be considered as a delayed version of the wanted transmitter signal. The interfering transmitter resembles an active multipath reflection of the wanted signal-which is rejected by the OFDM system. As a typical distance between main transmitters leads to echo delays of the order of 300μS and therefore impractically long Guard Intervals ($T_g$) and Symbol Periods ($T_s$), this technique in its simple form is considered impractical. A variant that reduces the requirements on $T_g$ and may prove to have some merit involves feeding all the main transmitters with synchronously fixed signals to create a zone mid-way between transmitters where the received signals add constructively. In this arrangement destructive interference occurs only much closer the ratio of the "wanted" transmitter where the ratio of the "wanted" signal-to-destructive signal is favourable. The performance of such systems when employing a guard interval of the order of 30μS is under study. Use of an SFN in this way may suit a national broadcaster, but the constraint of having to have the same, synchronously fed, programme material at each transmitter is likely to be unacceptable in commercial broadcasting which today is regionally based. Provided the loss of data capacity caused by the guard intervals is acceptable, this second type of SFN may be useful for designing relays to fill-in gaps in the main transmitter coverage area. Here the constraint on programme material does not present a problem because the same digital service is required in the relay coverage areas. Furthermore, this method of providing relay coverage is very efficient, because it would not need to use additional RF spectrum. The main practical problem likely to be encountered with an SFN relay provided by this second method, is the requirement for adequate site shielding. Since the relay transmitter is basically a high gain amplifier, receiving and transmitting on the same frequency, there will be a tendency for the amplifier to oscillate. In order to prevent oscillation, it is likely to be necessary to position the receiving aerial well away from the relay transmitting aerial and also provide site shielding between the two. This practical constraint may require the two halves of the relay to be sited either side of a hill.

Dual frequency networks

In areas where SFNs prove impractical, an alternative scheme under study involves the use of dual frequencies and dual polarisation as illustrated in Figure A1.10 where F1' and F2' are the orthogonally polarised versions of F1 and F2.

The use of such an arrangement in "all-digital" scenario would allow an increase in spectrum utilisation (compared with PAL networks) of around five times.

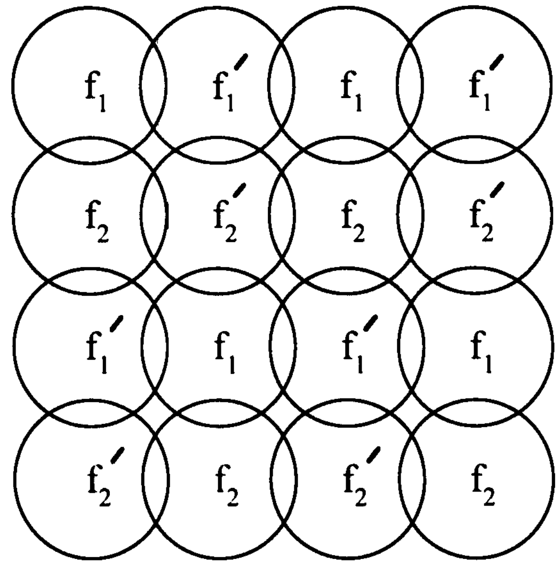


*Figure A1.10: Dual frequency network plan*

#### 1.5.3.3. OFDM protection ratios

A). QPSK modulation

Laboratory measurements on the OFDM system used for field trials in the UK in the SPECTRE project are reported in *[25]* and are summarised below in Table A1.8.

| Wanted Signal | Interfering Signal | Protection Ratio |
|---|---|---|
| PAL-I | QPSK-OFDM | 45dB |
| QPSK-OFDM | PAL-I | 3dB |
| QPSK-OFDM | QPSK-OFDM | 14dB |
| QPSK-OFDM | Noise | 14dB |

*Table A1.8: SPECTRE system CCI protection ratios*

The values given in Table A1.8 are rather conservative. They take account of implementation margins and effects due to actual picture dependency and assume a PAL grade 4.5 picture.

B). 16QAM

Preliminary Frequency Planning Parameters for the HD-DIVINE COFDM-TV system integrated into a G-PAL AM UHF network are given in Reference *[8]*. The main CCI protection ratios may be summarised as follows:

| Wanted Signal | Interfering Signal | Protection Ratio (dB) |
|---|---|---|
| PAL-G | COFDM (continuous) | 40 |
| PAL-G | COFDM (tropospheric) | 33 |
| COFDM | PAL-G (continuous) | 13 |
| COFDM | PAL-G (tropospheric) | 13 |
| COFDM | COFDM | 20 |
| COFDM | COFDM | 19 |

*Table A1.9: HD-DIVINE CCI protection ratios*

Here the requirement to maintain a PAL picture of grade 4.5 has been relaxed to allow a reduction of C/N of around 5dB in the continuous case and further relaxed for tropospheric interference.

It is interesting to note that if the SPECTRE QPSK results were extrapolated to 16QAM (allowing an additional 1dB in implementation margin), the protection ratio figures of Table A1.9 would become 45dB, 45dB, 11dB, 11dB, 21dB and 21dB from top-to-bottom respectively.

### 1.5.3.4. OFDM frequency planning studies and field trials

Extensive Frequency Planning Studies and Field Trials have been carried out within the SPECTRE project in the UK and a summary of results was presented in a recent discussion document from the ITC *[4]*.

The OFDM system used as the basis for these studies and trials as described in *[4]* has the following characteristics:-

- digitally compressed (MPEG) video coding: 12Mbit/s

  digitally compressed (MPEG) audio coding: 256kbit/s
- combined video and audio with Reed - Solomon (255,239) burst-error-correcting code: 13Mbit/s
- OFDM with 64μS Symbol Period and built-in rejection to the PAL vision, sound and colour carriers and sub-carriers.

The planning studies were based upon the protection ratios given in Table A1.8, a location factor of 50% and the assumed minimum wanted field strength values for PAL-I in Bands IV and V as 64dB and 70dB respectively. Only taboo channels were assumed since the planning considered only existing broadcast frequencies and took account of all existing (and planned) PAL transmissions. UHF channels 35 and 37 were not included in the study, but are the subject of a separate frequency planning exercise currently being undertaken on the ITC's behalf by NTL. The study considered digital transmissions from only the main transmitters in the UK. The results are summarised in Figure 2.1 of Section 2. It can be seen that in some parts of the UK more than 20 digital channels might be available, while in others the availability is three channels or less. This latter case arises from the existing high concentration of relay transmitters in these areas which makes it difficult to introduce even low power digital services alongside the PAL transmission. If a threshold is set at a minimum of four available channels per transmitter, then the calculations suggest that while PAL transmissions continue, around 80% of the UK population could be covered digitally from the existing main transmitter sites. (The figure for a minimum of five available channels, rather than four, is almost the same). This compares with a figure for PAL coverage from these sites of approximately 90%. The relay transmitters which provide the remaining 9% or so for PAL were not included in this study. It is certainly the case that the identification of digital frequencies for the very large number of small relays currently in operation will not be feasible, although it is possible that use of some of the larger relays (covering, in aggregate, around 3-4% of the UK population) might be able to enhance digital coverage. Further planning studies for these relays are now being commissioned.

Frequency planning studies in France provide a first assessment of population coverage for EDTV and HDTV using the main stations in each of the four national UHF networks *[7]*. The results to date suggest that population coverage for EDTV services would vary from 52% to 69% across the four networks. The equivalent figures for HDTV services are from 30% to 51%. It should be noted however that a very conservative location factor of 90% was used in this analysis and that there is therefore reason to believe that the considerably higher population coverages using a 50% location factor will prove to be possible.

While no details of the population estimates are available for Sweden, the feasibility of one or two HDTV channels is reported. It is believed that transmit powers ~24dB down on the PAL levels are under consideration.

### 1.5.3.5. OFDM with hierarchical source and channel coding

The OFDM system is flexible, in the sense that services of different qualities can conveniently be carried within the same multiplex. This convenience occurs because different carriers within the same OFDM signal can be modulated and error-protected differently. If the source coding of an HDTV service is arranged to yield a structure which contains a lower resolution signal embedded within it, this basic service could be sent on OFDM carriers, using say QPSK, and the additional HDTV bits sent on separate carriers, using say 16QAM. The 16QAM carriers would be able to convey twice the bit-rate of the QPSK carriers but the QPSK carriers would have around 7.5dB better noise and interference performance.

Numerous arrangements are possible, depending upon permitted transmitter power and the quality/coverage requirements, including the simultaneous transmission of 64QAM, 16QAM and QPSK to achieve the multi-resolution transmission characteristic illustrated in Figure A1.11.

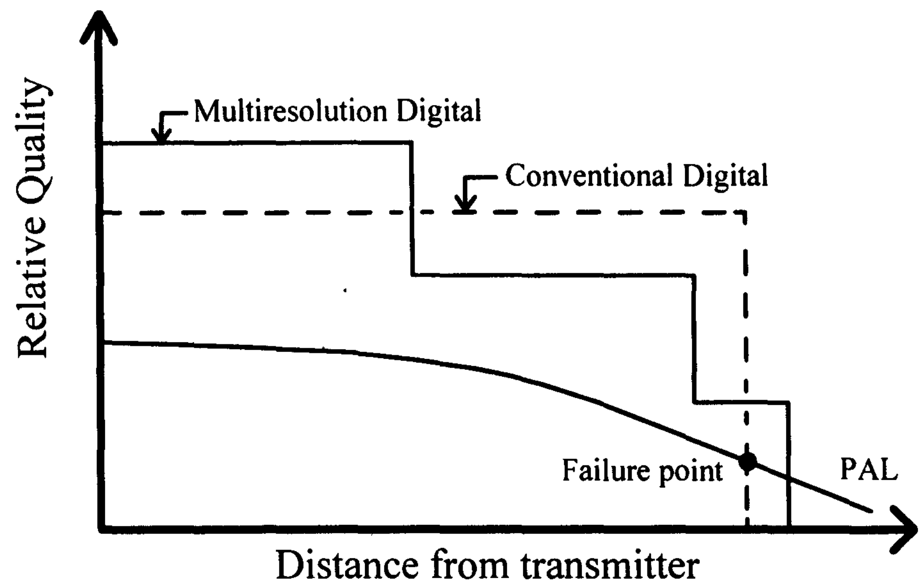


*Figure A1.11: Multi-resolution transmission characteristics*

Thus the transmission characteristic can be shaped to provide step-wise graceful failure plus higher definition in the inner coverage area where the field strengths are naturally at a higher level.

In hierarchical systems, configured to provide the characteristics of Figure A1.11, the video source coding and channel coding are configured to provide a transmission format where the different quality levels of the source video are "nested" together in transmission in such a way that they are acceptable to a hierarchically-configured receiver, or to receivers of different types. Such systems have the advantage that:-

a) they provide the means of "matching" the quality level available to the receiver to the signal strength available. (HDTV or EDTV with graceful degradation);

b) they can improve the quality and availability of portable reception when services intended primarily for fixed reception;

c) they can provide a means for up-grading services when increased transmit power is permitted (from EDTV to HDTV, say);

In such systems the source coding algorithms provides a data stream in which the different video quality levels are clearly separated. Different channel coding techniques are then applied to the segments of the source coded bit-stream that represent the different levels of picture resolution in the source coding hierarchy. The coding techniques can be based upon the methods illustrated in Figure 1.12, where the highest priority data is coded in QPSK and the lowest in 64QAM.

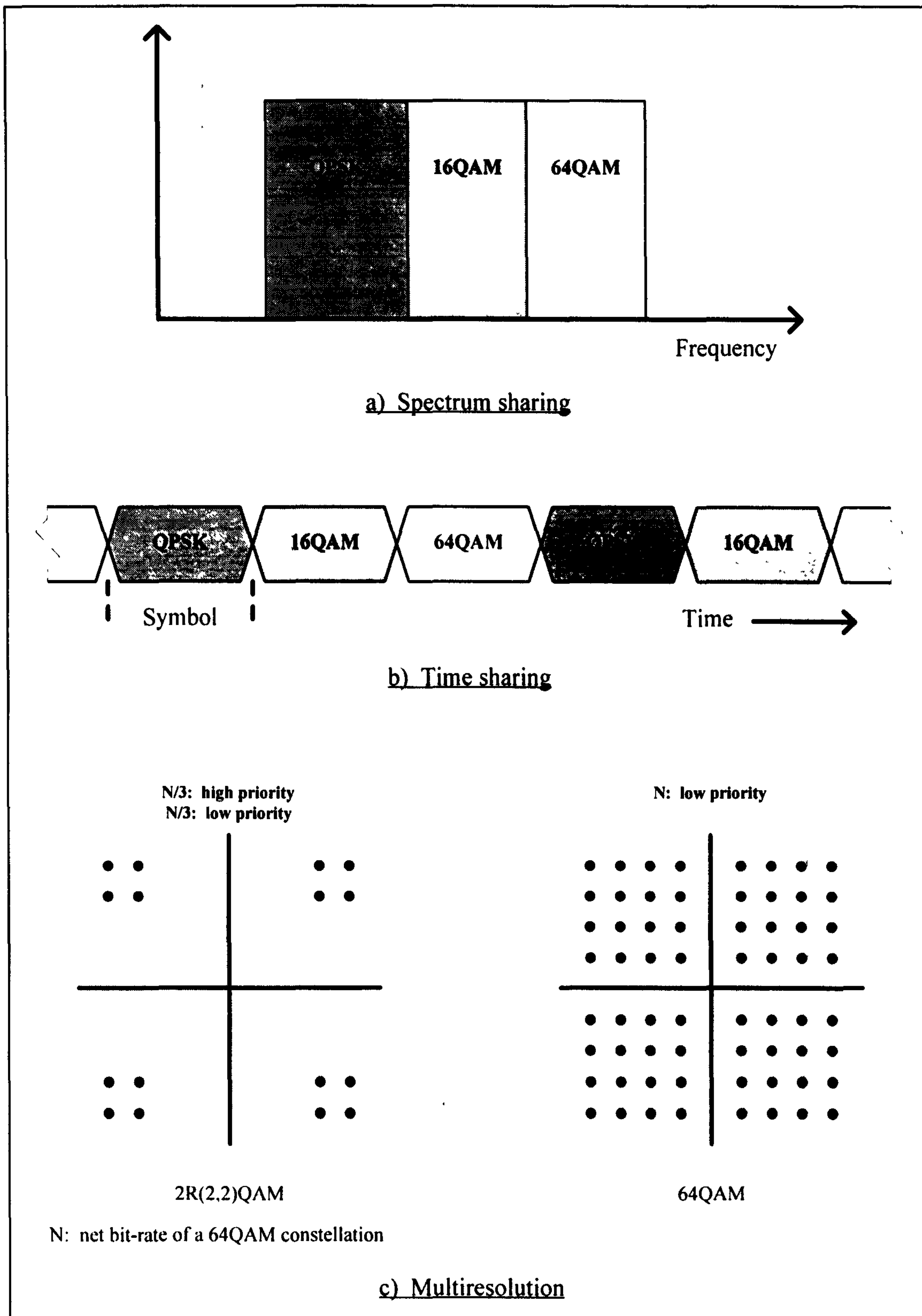


*Figure A1.12: Hierarchical modulation*

Additional protection can be given to the high priority data by transmitting its information at higher power, or, with some loss in transmission capacity, by the use of error protection. In an OFDM spectrum sharing system working in a PAL environment, for example, it should be possible to align the high priority QPSK signal with the H.F. components of its PAL co-channel signal and utilise a somewhat higher power level for the QPSK components of the multiplex.

Notional comparison with QPSK only transmission (purely for illustration)

QPSK only model

Assuming an 8MHz channel with a transmission bandwidth of 7.5MHz, the useful gross bit-rate with QPSK is around 15Mbit/s.

Allowing:-

~0.5Mbit/s for audio, SI, etc.

~1.5Mbit/s for spectral shaping

leaves ~13Mbit/s for video (i.e. sufficient for 1 EDTV service with 1 LDTV "simulcast" service for graceful degradation)

Hierarchical model

In a hierarchical system with 64QAM, 16QAM and QPSK modulation components, each sharing one third of the multiplex, the average gross useful bit-rates for each modulation level would be:-

| | | |
|---|---|---|
| for 64QAM: | 15Mbit/s | (≡ 45Mbit/s ÷ 3) |
| for 16QAM: | 10Mbit/s | (≡ 30Mbit/s ÷ 3) |
| for QPSK: | 5Mbit/s | (≡ 15Mbit/s ÷ 3) |
| Total available bit-rate: | 30Mbit/s | |

Allowing:

~ 0.5Mbit/s for audio, SI, etc. from the QPSK signal

~ 4.5Mbit/s for spectral shaping from the 64QAM signal

reduces the average bit-rates at the 64QAM and QPSK levels to 10.5Mbit/s and 4.5Mbit/s respectively. The resulting hierarchy of video bit-rates, together with their respective C/N requirements, is summarised in Table A1.10

| Modulation Level | Average Bit-rate (Mbit/s) | C/N Requirement |
|---|---|---|
| 64QAM | 10.5 | 29dB |
| 16QAM | 10.0 | 22dB |
| QPSK | 4.5 | 14dB |
| QPSK + 16QAM | 14.5 | ~ |
| QPSK + 16QAM + 64QAM | 25 | ~ |

*Table A1.10: Possible hierarchy of video bit-rates*

Adding 1 in 2 FEC to the 16QAM signal and increasing the power of the QPSK signal by 3dB could result in the transmission characteristic (notionally) described in Table A1.11.

| Bit-rate Level | Service Component | Coverage wrt EDTV/QPSK only | |
|---|---|---|---|
| 20Mbit/s | HDTV | limited | (-14dB) |
| 9.5Mbit/s | EDTV | reduced | (-1dB) |
| 4.5Mbit/s | SDTV | increased | (+3dB) |

*Table A1.11: Illustrative (notional) example of a hierarchical system*

It should be emphasised that the above example is based upon simplified theoretical considerations. Hierarchical systems are under study but as yet the development of such systems is in its infancy compared with the development of QPSK-only or 16QAM-only systems. It should be noted also that, in principle, the spectrum shaping "notches" could be created adaptively in the receiver. In this case a higher degree of error protection could be used to improve the coverage of one or all of the bit-rate levels.

Alternative coverage estimates for introduction phase hierarchical systems

More realistic quality/coverage estimates for the introduction phase are realised if the modulation levels are restricted to QPSK/16QAM for the case that transmitter power is limited to 30dB below PAL/SECAM levels, and to

16QAM for the cases where operation at 24dB below PAL/SECAM levels is possible. In the following examples a 64µS symbol period is assumed as well as Reed-Solomon (255/239) error protection.

**Example 1: $P_T$ = -30dB w.r.t. PAL/SECAM**

QPSK only Model

Assuming 7.5MHz useful bandwidth and a gross bit-rate of 15Mbit/s.

Allowing data transmission losses:

| | | |
|---|---|---|
| - | guard interval: | ~6.3% |
| - | Reed-Solomon (255/239) error coding: | ~6.3% |
| - | 0.5MHz "slot" for PAL vision suppression: | ~6.7% |
| | Total losses: | ~20% |

Hence useful bit-rate ~12Mbit/s

Further assuming:

| | | |
|---|---|---|
| - | a nested LDTV signal (1.25Mbit/s) with 1 in 2 FEC: | 2.5Mbit/s |
| - | a sound/data/synchronisation with 1 in 2 FEC: | 0.5Mbit/s |
| Hence bit-rate for Vision ~10.25Mbit/s: | | (12 - 3 + 1.25)Mbit/s |

- Therefore capacity for 1 EDTV Service of full coverage with graceful failure to LDTV.

**Example 2: $P_T$ = -30dB w.r.t. PAL/SECAM**

16QAM only Model

Assuming C/N reduced by ~7.5dB w.r.t. QPSK and a gross bit-rate ~30Mbit/s.

Allowing:

- ~20% Data transmission losses as per Example 1 ~6Mbit/s

Hence useful bit-rate ~24Mbit/s

Further assuming:

- 1 in 2 FEC gain ~7dB for full coverage

- Hence system service options as for Example 1.

**Example 3: $P_T$ = -24dB w.r.t. PAL/SECAM**

16QAM Fixed Format Model

Assuming C/N reduced by ~1.5dB w.r.t. QPSK at -30dB and a useful bit-rate ~24Mbit/s.

Further assuming:

- 1 in 6 FEC to regain loss w.r.t. QPSK at -30dB

Useful bit-rate now ~20Mbit/s

- Hence 1 HDTV Service of similar coverage to Examples 1 and 2 but no graceful failure.

**Example 4: $P_T$ = -24dB w.r.t. PAL/SECAM**

16QAM Hierarchical Format Model

Assuming C/N reduced by ~1.5dB w.r.t. QPSK at -30dB and a useful bit-rate ~24Mbit/s.

Further assuming:

- 3-level hierarchical coding with each component using 1/3 of the multiplex to give usefulbit rates of:

| | | | |
|---|---|---|---|
| HDTV component | (no FEC) | 8Mbit/s | (~ -1.5dB w.r.t. QPSK) |
| EDTV component | (1in 6 FEC) | 6.7Mbit/s | (~ 0dB w.r.t. QPSK) |
| SDTV component | (1 in 3 FEC) | 5.3Mbit/s | (~ +3dB w.r.t. QPSK) |

Hence:

- HDTV coverage to around 70% of service area radius.
- EDTV coverage to around 100% of service area radius.
- SDTV coverage exceeding well beyond service area radius.

### 1.6. Cable television

Cable Television Systems have grown piece-meal throughout Europe and today the percentage of the population to which cable television is available varies enormously from country to country. This pattern is reflected in the figures for the cable television subscriber penetration which are at their highest in the Benelux countries (88-96%), relatively high in Spain, Scandinavia and Germany (30-50%), but low (<5%) in a number of other countries including France, Italy and the UK.

Currently most systems are broadband/co-axial cable (able to support >30 TV channels) typically carrying 20 or more programmes to the public. The pressures for more programme capacity continues to increase however and there is a continuing process of up-grading the networks to increase capacity. While the use of optical fibre for the trunk grading the networks to increase capacity. While the use of optical fibre for the trunk network is now commonplace in many countries, means for realising the dream of providing fibre connections to the home remain elusive. At present few in the European cable industry have clear plans for introducing digital television distribution although they are following satellite developments in this area with a great deal of interest. In addition to the extra transmission capacity that may be needed, questions related to the control of conditional access (CA) are of serious concern. Here many cable operators fear that their control over the selling of "packages of programmes" (traditionally their "bread and butter" in some countries) will be threatened by digital satellite television broadcasters with a "built-in" subscription or pay-per-view facility via the satellite CA system. Technical concerns are focused more on ways and means of adjusting existing network arrangements to digital distribution rather than on problems of a fundamental nature. The problems of finding spare capacity to accommodate new satellite-delivered digital channels may be more difficult in the Benelux countries, where plant is relatively older and usage very high, than in Germany, France and the UK where most plant is relatively young and extendable for use throughout the UHF band.

### 1.6.1. Characteristics of cable systems

Topography

There are no common standards for cable network configurations in Europe, although there are similarities between the arrangements of many operators. The most common arrangement is "tree and branch", whereby all the channels in the long distance trunk network are delivered to the home and programme selection is made at the receiver. However there are also "star" sub-systems in use in the UK and in the Netherlands. In certain "star" arrangements, only selected programmes are delivered to the home from a local distribution point.

In "tree and branch" networks, subscriber drop cables are tapped across feeder cables running down streets. Feeders are fed from splitters and/or "line extender" amplifiers. Usually not more than two of these are in tandem. A feeder driving the first line extender will be fed from the high-level output of a "trunk bridger" amplifier, the trunk being to a high specification permitting many amplifiers in tandem. There is an increasing use of fibres in trunks, generally carrying the cable spectrum as a block..

In the UK, the subscriber cables usually radiate individually from the line extenders; this is known as "tree and bush". In the Netherlands some networks have "mini-hubs", where the frequency of certain input signals is up-converted, originally because some receivers could not cope with adjacent channels.

Capacity

While the highest frequency transmitted by most of the local distribution networks is around 850MHz, the upper edge of the overall bandwidth of many cable networks is limited by the capacity of the trunk networks (or, in the Netherlands, the mini-hubs) to around 300MHz. However most of the larger operators are up-grading to 450MHz and many to 600MHz. In general 7MHz channel spacing is used in the band up to about 230MHz and 8MHz channel spacing for the higher bands. The hyper-band, 300 to 420MHz, has generally been reserved for MAC and in some cases this band can be configured to 12MHz channel spacing for HD-MAC.

<u>Conditional access</u>

Premium cable services are generally scrambled, like many satellite services, except in a "star" configuration. New CA methods, allowing the subscription management to be controlled either by the satellite service provider or by the cable operator, may be required when digital signals are cable widely distributed by cable networks.

<u>Digital considerations</u>

In general, problems can be expected where there are echoes in the networks arising from misterminated drop cables and poor matching at the receiver interface. This may necessitate the use of directional couplers for all splitting and subscriber tap points as well as a general "tightening-up" of "matching" in the network.

Questions regarding digital/digital and digital PAL/SECAM operation on adjacent channels of the networks need to be addressed. At first sight such operations should be feasible, but extensive experiment and field trials will be required to prove satisfactory performance.

While there is good evidence that single-carrier QAM systems with delay equalisation will work satisfactorily in certain networks (see A1.6.2 below), little information has been reported regarding the use of COFDM for cable distribution although this method, with its resistance to short-term echoes, would seem to have advantages.

The situation has been assessed by the CCETT from a theoretical point of view, and using simulations of a typical network to assess "sharing" problems in a mixed digital/analogue scenario *[31]*. A number of approaches have been identified including:

- single carrier modulation of the QAM type;
- single carrier modulation of the VSB-AM type;
- multicarrier modulation of the COFDM type;

Their provisional conclusions may be summarised in the following salient points:

- While it would appear to be feasible to transmit bit-rates of 40Mbit/s in 8MHz channels in networks also carrying analogue transmissions, it is unlikely that higher bit-rates could be achieved with filters considered practical for consumer equipment;
- There are doubts regarding the implementation margins to be assumed for cable networks and user terminals;
- The choice of a particular modulation method is highly dependent on the likely cost of the demodulator in a "consumer electronics environment". "A great deal of research remains to be carried out, in conjunction with industry, in order to determine the best possible compromise between price and performance."

### 1.6.2. Digital cable television experiments

Considerable testing of the performance of 16QAM and 64QAM modulation in cable systems has been carried out in the USA.

The Jerrold company has reported the results of field trials using 16QAM on cable trunks containing 20 amplifiers, showing that it is satisfactorily rugged. Additionally, testing of a 64QAM system over several cable systems is reported *[32]*. The system configuration included trunks with 23-41 cascades amplifiers plus distribution systems consisting of 1 or 2 line extenders and up to 26 taps. The use of SAW filters and adaptive equalisers in these arrangements resulted in uncorrected recorded BER's of $10^{-5}$ to $10^{-6}$ in tests ranging from 2 to 51 hours in length. Based upon these results a digital satellite to cable distribution scheme which will carry 26.9Mbit/s of useful data plus 2.34Mbit/s of error protection in a 6MHz bandwidth: (i.e. scaling to an 8MHz bandwidth ~36Mbit/s of useful data) is proposed for further proving. The conclusion reached is that modulation levels up to 64QAM are almost certainly feasible on the cable itself, but that further investigation into the ramifications for ancillary equipment is desirable.

### 1.6.3. Proposals for transmission standards and harmonised bit-rates

Following the same general principles as those proposed by Jerrold, Thomson Consumer Electronics suggests the use of a transmodulator for head-end processing of the digital satellite off-air signals for cable distribution *[33]*. Following QPSK demodulation and error correction, the digital television signal is re-protected for cable distribution at an appropriate QAM level.

In considering the question "Common elements for Cable, Satellite and Terrestrial Networks?", Stenger *[34]* suggests a useful bit-rate of 34Mbit/s as the common rate for use by satellite and cable (as well as, possibly, for the "Final" terrestrial system). Standardising the useful bit-rate for all the media would of course underpin the Thomson/Jerrold proposal for "Transmodulation" – all useful satellite data being relayed on in standard format with no requirement for decoding and re-coding. With suitable error protection, the gross cable distribution rate would be ~40Mbit/s – possibly achievable with 32QAM in 8MHz cable channels but requiring 64QAM in the 7MHz channels. Capacity for such transmission in 8MHz channels in the German cable system is available in the "extended special supra-band" or "hyperband" stretching from 302 to 438MHz. Similar capacity exists in many other European cable networks systems (albeit sometimes in different frequency bands).

# *PART 3 - RESEARCH AND DEVELOPMENT*

Chapter 5

**Research and Development -**
**A European overview**

# Table of contents

# 1. INTRODUCTION

Unlike conventional analogue television broadcasting, which employs technologies that have been developed specifically for that purpose, digital television broadcasting is being developed largely by the integration of generic digital telecommunications technologies, such as multiplexing and channel coding, which have many other potential applications. While this provides valuable opportunities for facilitating interworking between digital television and other digital telecommunications services, the need to take these opportunities into account, along with the wish to ensure that full advantage is taken of the potential advantages of digital techniques, such as more efficient use of the spectrum and a substantial increase in the number of channels without loss of quality, has provided the impetus for extensive research in Europe and elsewhere.

European research into digital television broadcasting techniques is being undertaken in a number of projects. Taken together, these efforts stretch from one end of the transmission chain to the other: from source coding and multiplexing to channel coding and modulation. Furthermore, their scope ranges from the pre-competitive development of generic technologies to their practical implementations in feasibility studies within particular environments. Apart from the significant investments of the actors themselves, research is funded in three ways: through national projects; through transnational collaborative projects in the Eureka mechanism; and through the transnational collaborative projects of the Community's RACE programme.

The range of research issues and the three different funding mechanisms mean that the research environment is complicated, and it is difficult for those outside the research environment to assess precisely what is happening in European digital television research. One must consider not only the individual projects, but the relationships between them, since many of the researchers are involved in all three research mechanisms identified above, perhaps investigating slightly different combinations of topics in each. The relationships between apparently overlapping projects cannot simply be characterised as unnecessary duplication. Particularly in the area of channel coding, there are a wide range of different conditions in Europe and researchers have sought to address these as well as seeking commonality.

The aim of this chapter is threefold: to provide descriptions of the main projects; to assess their collective contribution to standardisation and implementation efforts; to consider future research requirements.

## 1.1. Acknowledgements

The Descriptions of the RACE dTTb project and the main national research projects are edited from articles that first appeared in the *EBU Technical Review.* They appear with the assent of Dr. George Walters, Technical Director of the European Broadcasting Union. Several contributions made by Prof. Dr.-Ing. Ulrich Reimers of Brunswick University, the Chairman of the Working Group on Digital Television Broadcasting (WGDTB), were also important inputs.

## 1.2. Scope

Note that the scope of this section is strictly limited to source coding, multiplexing, channel coding and modulation; it does not cover other research areas like studio systems and consumer electronics products.

## 2. COMMUNITY CONTRIBUTIONS

The RACE programme (Research into Advanced Communications, Europe) began in 1986 with a definition phase, proceeding through the RACE I phase 1988-92 under the 2nd Framework Programme; RACE II is currently active under the 3rd Framework Programme (1990-94), and additional research is envisaged. The primary focus of both RACE I and II is on developing "integrated broadband communications"; that is, taking advantage of recent developments in digital technology to enable the present incoherent set of networks dedicated individually to specific analogue telecommunications services (telephony, image transmission etc.) to be replaced by high-capacity digital networks able to carry all such services, up to HDTV. Both RACE I and RACE II thus include several projects addressing various tasks relevant to digital broadcasting. These are described individually below together with the effect they have had on standardisation activities, including MPEG. Further details of these projects are given in the RACE Annual Reports

### 2.1. RACE I coding projects

To reduce the cost of transmission and switching facilities, compression of digital information is required. However, the use of excessive compression may alter the information, resulting in a loss of quality; so an appropriate compromise has to be found on a case-by-case basis, between transmission cost, loss of service quality and interpretability. These topics were studied by FUNCODE, R1041.

This coding work for image communications had three main focal points: video-telephony (VT), sound and TV/HDTV.

- Video-telephone coding work concentrated on enhancing picture quality relative to CCITT rec. H.261 while maintaining compatibility with this earlier standard. Interoperation had to be guaranteed not only between 64Kbit/s and 2Mbit/s services, but also for interworking between STM and ATM networks. The architecture of a cost-effective solution using several ASICs has been designed and these components are being implemented. A high quality 2 M/bit/s codec was produced. The objective of the sound coding work was to select a suitable audio coding algorithm on the basis of an optimal trade-off between sound quality and hardware complexity.
- TV/HDTV coding work focused on three topics: adaptation of TV codecs for ATM networks; HDTV contribution links at 140 Mbit/s; HDTV compatible distribution.

HIVITS, R1018 - 19 partners led by Thomson - implemented the FUNCODE functional requirements in experimental hardware with the following results:

- A SDTV codec capable of decreasing the bit-rate required for transmission of pictures from 166 Mbit/s to 30 Mbit/s - contribution quality - and to 15 Mbit/s - distribution quality. This corresponds to a reduction of 16 bit/pel, for luminance and colour respectively, to 2.9 bit/pel and 1.45 bit/pel.
- An HDTV codec was built on the shoulders of the above work, using six parallel processors and giving a 140 Mbit/s bit rate contribution/primary distribution codec.
- Extension of video compression algorithms to HDTV rates met the requirements of contribution and primary distribution but was inadequate for secondary distribution of a mass medium with the perceived need to take into account the existing infrastructure, including the viewers' existing equipment. Studies of HDTV systems in Europe had been driven by the need for compatible evolution from SDTV to HDTV.

The HDTV secondary distribution system used sub-band coding allowing a Video-telephone (VT) terminal to receive a digital HDTV signal but with lower resolution. With this kind of approach, a digital HDTV signal could be received by VT, SDTV or HDTV terminals. Image quality is linked to terminal cost.

Video coding techniques in the context of digital video recording were also studied in RACE project R1001 (DVT), in which the Philips, Grundig and DTB companies developed solutions meeting the special requirements of a domestic TV/HDTV video cassette recorder. These solutions inevitably differ from those optimised for transmission networks, because with magnetic recording media the errors tend to occur in bursts, and the available information comes from different blocks of data depending on the play-back mode used (still-image to fast monitoring...).

### 2.2. RACE I transmission projects

In the optical fibre network context, R1010 applied coherent optical communication techniques to signals at up to 140 Mbit/s. Its results provided techniques for signal transport and switching of 140 Mbit/sec TV/HDTV signals. Its RACE II follow-up has achieved 2.5 Gbit/s HDTV signal throughput.

R1051 took a different approach, from the premise that high video-compression ratios would require complex algorithms and therefore expensive processing in the receiver. It developed methods for transmitting multiplexes of uncompressed digital HDTV signals through optical fibre networks at very high rates - up to 10Gbit/s - so that the decoding in the receiver would be simpler and cheaper.

R1026 studied an evolutionary approach for the replacement of the existing network of analogue point-to-point television and audio contribution links used for the exchange of television programmes among broadcasting companies in Europe, by more extensive digital transmission facilities. This "Eurovision" circuit network is largely routed via satellites, but terrestrial links are also used.

### 2.3. Some RACE II coding and modulation projects

RACE II is building upon the results from RACE I and has also expanded into new areas, including modulation and channel coding for digital terrestrial television transmission. Apart from the dTTb project R2082, (see 2.4 below), which builds upon work begun in the SPECTRE and STERNE projects discussed in sections 4.1 and 4.2, the following RACE II projects are also undertaking relevant studies:

- HD-SAT, R2075, is developing a bandwidth-efficient coding and modulation system for wide-RF-band HDTV (W-HDTV) satellite broadcasting in the 20 GHz band, taking advantage of the greater bit-rates available in this way to provide studio-quality HDTV images to domestic viewers. Accordingly, it involves detailed studies of the practical requirements for operation in 20 GHz band and for the interconnection of the space and terrestrial transmission networks.
- FLASH-TV, R2064, will provide broadcasters, news-agencies and telecoms operators with a flexible, cost-effective and advanced means of setting up and operating high-quality digital video contribution links by satellite in the range 34 to 70 M/bit/s. It is concentrating on developing a satellite modem and a video codec derived from that produced by R1018 HIVITS, but with improved quality and flexibility (through being

able to operate at four bit-rates), especially at the lower bit-rates. The work includes construction of prototype equipment.

- MAVT, R2072, addresses the problem of devising powerful video and audio coding algorithms for the transmission of moving and still image signals to mobile users, and the design of the corresponding terminal equipment. The results are expected to provide basic design guidelines relevant to the development of mobile personal communication facilities, including the display of low-resolution images.
- DISTIMA, R2045, is studying the capture, coding, transmission and presentation of digital stereoscopic systems for broadcast and non-broadcast applications. The work includes software development, human factors research, hardware design and interworking aspects. The basic concept is the extension of existing digital video data-compression techniques to the case of a pair of closely correlated signals, representing the output of cameras viewing the scene from the position of one individual's eyes.
- MORPHECO, R2053, intends to develop an advanced object-oriented approach to the coding of visual images, involving high compression rates, but retaining image quality. This project's studies are concentrated on the use of mathematical morphology, and involve the development of new image coding algorithms based on the analysis of the fundamental shape of the objects to be represented. The approach allows the image to be described in terms of information about its contours and its underlying skeleton; it is expected that this technique will ultimately enable higher compression ratios than those currently attainable to be achieved.

## 2.4. The dTTb project

This is RACE II's flagship activity in respect of terrestrial broadcasting. The following profile is an edited extract from a description published in the *EBU Technical Review*, No. 256 - Summer 1993.[1]

### 2.4.1. Context

The dTTb project is operating under the umbrella of the EEC RACE projects aiming at the definition of integrated broadband communications systems. This context reinforces the importance of considering digital television broadcasting as a branch of data communications, like HDTV-T (see 4.3), and explains the interest of several telecommunications operators in the project (Table 1).

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1 European activities on digital television broadcasting: from company to co-operative projects, B. Marti, N. Lodge, P. Bernard, R. Schäfer

| |
|---|
| *European industrial groups:* |
| Philips |
| Thomson |
| Seleco |
| *European broadcasters:* |
| BBC, ITC (United Kingdom) |
| TDF (France) |
| RAI (Italy) |
| IRT (Germany) |
| Retevisi6n (Spain) |
| *European Broadcasting Union PTT Administrations:* |
| DPB-T (Germany) |
| France Telecom |
| Telecom Denmark |

*Table 1: Members of the dTTb consortium*

The service model adopted is the one defined by the European Launching Group for Digital Television Broadcasting, and this serves as the basis for the technical co-ordination.

The title of the dTTb project contains two key words: digital and terrestrial. Nonetheless, the situation is rather different in Europe compared to the situation in the United States, where digital terrestrial television is also under development. The partners in the dTTb project are convinced that, because of the specific nature of the European context, it will be easier to reach some degree of standards unification if the studies begin with the terrestrial part of the problem. The reason is that *terrestrial* implies certain constraints, relating to frequency planning in particular, that other network environments do not have (or, at least, not to the same extent). The necessary long-term coexistence of the analogue systems of today and the future digital services (simulcasting), possibly extending for several decades into the next century, means that the situation is completely different in the terrestrial environment and that it would be certainly easier to move from a system designed initially for terrestrial use towards a compatible system on cable or satellite, than it would be to move in the opposite direction.

The second reason for this approach, at least in Europe, is that services have already started an evolution towards higher quality on cable and satellite, using the MAC systems, and it is very unlikely that this kind of system can be adapted for terrestrial broadcasting. It may be technically possible, in some cases, to adapt D2-MAC, but there is no interest, in terms of benefits for frequency planning, in adding a new analogue terrestrial broadcasting system. It is only through the introduction of a digital system that there is a chance of solving the problem of increasing programme quality in the terrestrial environment. It is in this domain that the broadcast network operators have much to gain. Only by using digital techniques can terrestrial broadcasting evolve. In terms of service quality, terrestrial networks offer an advantage that satellite probably - and cable certainly - cannot offer: portability, which means the possibility of having non-connected, movable receivers.

The technical objectives of the dTTb project are addressing to this problem, and the first step is the definition of techniques allowing the use of single frequency networks as a means of simplifying the frequency planning problems. The term "single frequency network" can be used a various levels. Europe is very regionalised, so SFN can be considered at national level and also at local or sub-local levels. The problem is how to operate transmitters and rebroadcast transmitters all working at the same frequency. The DAB experience has shown that this is possible using spread spectrum techniques, and OFDM in particular.

The dTTb project is currently exploring three types of reception conditions, corresponding to three levels in a hierarchy of sensitivities of the signal to errors. These are the *portable* level which is given a certain priority because it represents the specific feature of terrestrial broadcasting, traditional reception with *fixed* receivers and roof-top antennas, and, in continuity with the DAB project, the possibility of reception by *mobile* receivers, even if the commercial incentives for this last application may be thought less significant that for the first two. In the same way, four types of picture quality are being considered.

SPECTRE AND STERNE can be considered as elements of the contributions from ITC and the CCETT to dTTb, while close co-operation with HDTV-T has been agreed through the form of joint meetings of the different working parties.

### 2.4.2. Goals

In line with the objectives explained above, the dTTb project has two principal goals:

- provision of the technologies required for the broadcasting of digital signals at a bit-rate which will allow the broadcast distribution of the digitised and compressed television signals being studied in other projects,
- contribution to the establishment of the necessary standards for the modulation, channel coding and multiplexing systems.

The strategy for the introduction of operational digital television services within Europe requires both technical and marketing studies, although the latter fall outside the framework of the dTTb project. The objectives will have been attained if the required specifications and technologies are ready at the time when the political and economic opportunities for starting such a service occur. Such an opportunity for the introduction of digital television services will probably arise shortly.

The project is concentrating its contributions on channel modulation/coding and broadcast system aspects. The areas covered are:

1. A broad study of all possible data rates for existing VHF and UHF channels, in order to determine which digital video broadcasting services and products are feasible. This study is to ensure that account is taken of possible future extensions offering more television channels and/or higher picture quality.

2. As a first step towards defining a service/product, demonstrators will be constructed for broadcast television programmes and will be aimed towards portable (i.e. "plug-free"), or mobile receivers. These demonstrators will serve for the investigation of the network configuration aspects, extendibility towards future services, frequency allocation aspects and chip area analysis. The aim is a service with higher quality than the existing PAL/SECAM systems and featuring 16:9 capability.

### 2.4.3. Channel coding and modulation

A primary consideration of the project as a whole is to develop the most appropriate modulation and channel coding techniques for the efficient exploitation of the available channel capacity for the delivery of digital television. The identification of what constitutes optimal use of the channel can only be achieved through consideration of very complex trade-offs involving the performance of source coding methods for video and sound, the data capacity allocated to ancillary services, the service coverage area, the mode of reception ("plug-free", mobile or fixed), the basic service quality required, the target cost of the receiver and the frequency planning criteria that are adopted.

In the domain of channel coding and modulation, high priority is given to OFDM which allows single-frequency networks to be used. This capability is of prime importance in order to cope adequately with the frequency-planning situation in Europe.

All these techniques are under thorough examination and comparison, including the LSI complexity evaluation, within the dTTb project.

A first set of results is shown in Figure. 1 which compares several modulation and channel coding schemes that have already been tested. It can be seen that the channel efficiency typically needed for mobile reception is around 1 bit/s/Hz, around 2 bit/s/Hz for portable reception, and around 4 to 5 bit/s/Hz for fixed reception.

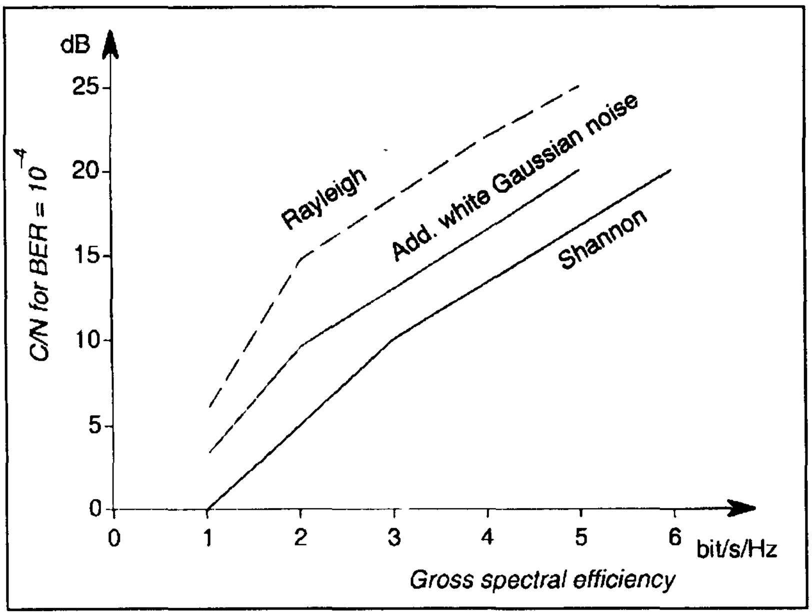


*Figure 1: Gross spectral efficiency and bit-rates*

### 2.4.4. Service and system characteristics

It is difficult to establish parallels between the qualities of digitally-coded pictures and analogue pictures because the impairments created by digital compression schemes are very different from the usual effects of noise, echoes or intermodulation common to analogue systems. It is nonetheless usually reckoned that, to obtain a digitally-coded picture having the quality of a normal composite PAL or SECAM picture, a bit-rate of the order of 4 to 5 Mbit/s is needed. This quality level has become known as "standard" quality (SDTV). In the same way, extended quality (EDTV) is defined as being "near-transparent" to the quality of the CCIR Recommendation 601 studio standard; this seems to require some 10 Mbit/s. A third level, known as limited quality (LDTV), is also defined; it is often compared with VHS and requires 1,5 Mbit/s. Finally there is HDTV, preliminary reckoned to require between 20 and 30 Mbit/s, and this is to be at least as good as HD-MAC. Table 2 shows the service possibilities offered by the combinations of these four quality options and the three receiving environments mentioned earlier.

| Designation | Equivalent quality | Bit-rate (Mbit/s) |
|---|---|---|
| LDTV | MPEG-1 - VHS | 1.5 |
| SDTV | Composite PAL/SECAM | 5 - 6 |
| EDTV | Studio (CCIR Rec. 601) | 9 - 11 |
| HDTV | ≥ HD-MAC | 20 - 30 |

| Reception Conditions | Efficiency (bit/s/Hz) |
|---|---|
| Fixed (roof-top antenna) | 4 - 5 |
| Portable ("rabbit-ears" antenna) | 1.3 - 2 |
| Mobile (Rayleigh channel) | 1 |

*Table 2: The four service quality options and the three reception environments studied by dTTb*

This range of possible configurations explains why, within the framework of the dTTb project, a very comprehensive multiplex structure is being studied. Another factor relates to two complementary statements:

- the picture quality of digitally-coded television is dependent on picture content;
- the bit-rate of digitally-coded television is dependent on picture content.

  Both those statements are true, depending on whether the emphasis is on the quality or on the bit-rate. The multiplex will have to cope also with the situation where a broadcaster imposes a form of "cultural revolution" and considers spectrum space as a statistical entity rather than as static frequency allocation, enabling better advantage to be taken of the natural flexibility of digital coding.

  It will probably be much more difficult to agree on the design of the multiplex, and on how to give due consideration to all the service parameters, including access control, than it will be to agree on the choice of an efficient modulation or coding scheme. It is now understood, in Europe, that the multiplex is the place where the "digital television war" can be won or lost.

### 2.5. Accompanying measures and preparatory actions

During 1993, it became apparent that there was a need for additional accompanying measures to collaborative international research and development in the field of digital image transmission, with particular emphasis on the social and economic implications of the transition to integrated digital transmission. Agreement was therefore reached in 1993 on the provision of additional Community funding of up to 12 MECU over a period of 18 months for such "accompanying measures and preparatory actions" intended to stimulate consensus on European standards and implementation strategies in this field. The following three tasks were identified:

- acceleration of European consensus development on technical specifications for digital image transmission and system implementation strategies;
- demonstration of digital image transmission systems;
- analysis of potential social and economic implications of the transition to digital image transmission.

The results expected to be obtained from the work to be undertaken on this basis should include:

- identification of the requirements of all actors (programme-makers, broadcasters, telecom operators, equipment manufacturers, users);
- identification of open-architecture terminal requirements;
- identification of the technical options and establishment of a consensus on the favoured technical solutions;
- estimates of the cost of equipment and implementation;
- contributions, including transmission system specifications, to international standardisation bodies;
- implementation, demonstration and testing of the retained coding and transmission systems, on their own and as part of an integrated digital transmission chain.

The corresponding Call for Proposals was published in August 1993, and it is expected that the work on the projects will start at the beginning of 1994.

## 2.6. Impact upon standardisation

The HIVITS project from RACE I, which was undertaken by a consortium consisting of many of the leading European organisations in this field, has had a widespread and pervasive impact upon image coding standards, including the following:

- CCIR Rec. 723, TV and HDTV contribution: project members built experimental codecs as described above.
- CCITT Rec. H.261: Most HIVITS partners were involved in this standardisation process. The joint simulation effort together with the building of two video telephone demonstrators contributed to the definition and optimisation of the H.261 architecture. The analysis of codec evolution and preparation of functional specifications made by FUNCODE was also influential.

The H.261 work was one important input into the MPEG standard. Many former and present RACE participants are involved in MPEG. The main contributions are listed below:

- hierarchical coding: the idea of multi-resolution, TV/HDTV compatible coding originated within HIVITS; it was taken up by MPEG and renamed `scalability'. Other organisations have also taken it up, notably CMTT/2-SRG, CCIR TG11-3, dTTb, HDTV-T.
- low delay for conversational services: this idea was initiated by HIVITS.
- MPEG-2 test models: HIVITS defined the original specification and oversaw their evolution.
- decoupling of source coding and channel coding: this enabled a generic source coding usable by both broadcasters and telecom operators.
- two layer coding for ATM: the concept of two layer coding for asynchronous networks was an extension of the H.261 standard developed by HIVITS and later applied to an MPEG coder.
- cell-loss model for ATM: this was introduced by HIVITS to ETSI/NA5/VCM and forwarded to CCITT SG XV AVC and ISO/MPEG.
- cell-loss concealment: HIVITS introduced a scheme which will be influential on MPEG coding when transmitted through asynchronous networks.
- audio coding: MUSICAM is one of the data reduction schemes proposed for MPEG audio. It originated in HIVITS and is now part of ISO 11172.

Current RACE projects such as MORPHECO, MONALISA and MAVT are pooling their coding efforts in order to contribute to object-based, second generation coding schemes, particularly MPEG-4 which is scheduled for late 1994.

Many partners who worked on image coding in RACE I have moved on to study channel coding and modulation problems in RACE II projects like dTTb, FLASH-TV, HD-SAT or DISTIMA. They are also active in other organisations like the European Launching Group and national research projects like HDTV-T.

# 3. EUREKA CONTRIBUTIONS

Eureka provides another mechanism for cross-border collaboration between companies and other institutions. Like the Community research programmes, it combines public money with private resources. National governments provide the public funding. The Ministerial Conferences which approve Eureka projects include Member States and other European administrations.

The following profiles are edited from ECHO database fiches on the projects, describing the state of the art when they began.

## 3.1. EU 625 -VADIS

### 3.1.1. Project description

The primary goal of the project is the development of a European enabling technology for digital television at bit rates up to about 10 Mbit/s, including the development of the related microelectronics. The project is classified in the area of Information Technology. The motivations for the project are the current activity carried out within the ISO/IEC JTC (MPEG) intended to define, by the end of 1992, a generic standard for coded representation of audiovisual information at bit rates up to about 10 Mbit/s.

There is common agreement that such a generic standard will open up new possibilities of products and services, such as interactive multi-media applications and will be used in several different application areas e.g. digital broadcasting, digital distribution over cable, storage and retrieval on digital storage media, television distribution and audiovisual communication over ATM networks (e.g. B-ISDN, LAN, MAN, etc.).

In the worldwide standardisation area of audiovisual coding matters there is a strong North American and Far Eastern presence, the former bringing the excellence of their VLSI technology and the latter the power of their product development and manufacturing capability. Unless a strong and unified European presence is secured by means of a co-ordinated action there is a risk that the definition of the standard and its subsequent exploitation will confine the European hardware and software industry to a marginal role.

To achieve the goal of enabling the European industry to timely exploit the standard, the project will have a duration of 30 months and will be organised in the following work packages (WPs):

### 3.1.2. Schedule

Requirement definition (months 1 - 6)

In this WP the constraints coming from the different application areas envisaged will be analysed and its impact on the compression algorithm identified. Virtually all partners in the project are involved in this WP in order to make sure that all possible application areas are duly taken into account.

Algorithm development (months 1 - 18)

This WP is clearly of the highest importance because so far there is no known solution to the problem of coding audiovisual information at about 10 Mbit/s with an overall quality better or at least equal to the one obtained today by traditional analogue techniques, although there is confidence both at ISO and at this project level that the goal is attainable in the given timescale. This explains the large amount of manpower assigned to this WP (about 40% of the total).

System aspects (months 4 - 18)

Once the audiovisual signal has been digitised there is by no means a straightforward way to exploit the flexibility used by the digital representation. The multiplexing of several audio and video streams in a single (or multiple) stream, the inclusion of protection mechanisms, access control, interactivity, etc. are some of the problems that need to be resolved.

Demonstrators/VLSI (months 10 - 30)

Once the coding algorithm has been defined by means of computer simulation, a check by hardware operating in real time is mandatory. Because of the short time scale of the project, it is quite likely that some of the VLSI chips developed for use in the subsequent industrial phase will already be used for the purpose of demonstrators. All VLSI needed for the standard will be designed in this WP.

Field trials (months 22 - 30)

The demonstrators developed in the WP above will be used to check the applicability of the standard to the intended application areas.

Standardisation (months 1 - 30)

The project will provide a co-ordination forum to enhance European presence in ISO, contributing to drafting the standard, etc.

### 3.1.3. Developments

There are indubitable advantages in the handling of audiovisual signals in digital form. The reason why the penetration of digital techniques was initially so slow is that the straightforward PCM representation of the audiovisual information yields exceedingly high bit rates. As an example, studio-quality digital television requires 216 Mbit/s and compact disc audio 1.41 Mbit/s. Redundancy-reduction techniques can be applied, but in order to retain sufficient quality a bit rate of about 34 Mbit/s was the minimum that could be achieved at the start of the project. Such a bit rate is, however, too high to be accommodated in the digital channels that are in widespread use. The project will therefore develop:

- a compression coding algorithm yielding high quality for the coded audiovisual signal.
- the microelectronic components to implement the algorithm and,
- an agreed international standard (ISO/IEC MPEG).

### 3.1.4. Applications

The application areas likely to exploit the results of the project are:

- Digital terrestrial broadcasting: about 10 Mbit/s can currently be carried by the 7 MHz VHF and UHF channels assigned to television broadcasting with an improved quality of service compared to the analogue systems of today.
- Digital satellite broadcasting: several 10 Mbit/s signals can be carried simultaneously by one of the 27 MHz satellite channels currently assigned to television broadcasting, compared to one analogue programme signal today, thus making improved exploitation of these channels possible.
- Digital distribution over cables of various types: copper, optical fibres, passive optical technology.
- Storage and retrieval using advanced optical, magnetic and magneto-optical recording technologies for discs and tapes.
- Digital television distribution and audiovisual communication through the future B-ISDN; although B-ISDN is intended to be able to deliver 155 Mbit/s to the customer, there will be merit in reducing the bit rate of the audiovisual signal in order to increase the number of simultaneous channels delivered to the subscriber and reduce the bit rate in the trunk network for interpersonal audiovisual communication.

### 3.1.5. Markets and standards

The enabling technology developed by the project can be used in products with wide application areas (recording, telecommunications, broadcasting, etc.) on a worldwide scale. The power offered by digital audiovisual technologies will improve existing services and products and open up completely new application areas, particularly in the area of interactive services and products. Here are some examples:

Storage and retrieval: digital tape recorders, next generation read-only optical discs, write-once or rewritable optical discs, Winchester discs. The combination of these recording technologies and the digital audiovisual technology developed by the project are bound to create a market which, in many cases, will be totally new.

Telecommunications: B-ISDN, LAN, MAN. The combination of these high-bit rate network technologies and the digital audiovisual technology developed by the project will enable the integration of conversational, distributive and storage/retrieval services on the same physical medium.

Broadcasting: satellite and terrestrial digital channels, distributive digital networks. The combination of the technologies for digitising existing or emerging broadcasting/distribution channels with the digital audiovisual technology developed by the project will improve the existing quality of service.

The current plan for the definition of the standard assumes that a technically-consistent draft will be available by the end of 1992. In view of the strong need for products based on the standard, it can be assumed that, by the end of 1993 European industry should have both hardware and software products ready.

## 3.2. EU 256 - HDTV contribution codec

### 3.2.1. Project description

The main goals are the definition of an algorithm and a codec structure for bit-rate reduction for HDTV transmission in contribution links, and the implementation of codec prototypes; consideration will be given also to a wide range of applications related to an HDTV contribution codec such as a distribution HDTV codec, and a contribution codec for conventional TV. The following main constraints will be taken into consideration during project execution:

- evolution and definition of a production standard for 50 Hz HDTV, as emerging in established European research activity on the HDTV subject.
- high bit-rate reduction being one of the project goals, the investigated algorithms will be based on the discrete cosine transform (DCT); results coming from any other research on the subject will be taken into account.
- project details will be adjusted so as to take into account the results of discussion and proposals coming out from standardisation bodies.

### 3.2.2. Developments

- Development of an algorithm for bit-rate reduction with highest efficiency.
- Development of large-scale semi-custom circuits to make the algorithm complexity manageable.

### 3.2.3. Markets

The products developed during the project are aiming at a worldwide market in both 50 Hz and 60 Hz field-rate countries.

## 3.3. Impact assessment

VADIS has co-ordinated European inputs into MPEG and provided financial assistance to participants.

EU 256 has completed development of an HDTV codec operating at 140, 70 and 45 Mbit/s.

Consumer electronics companies also report that their earlier EU 95 HDTV work on digitally-assisted television techniques for HD-MAC contributed to more recent work.

# 4. NATIONAL PROJECTS

The following paragraph, and the subsequent profiles of national research projects, are edited extracts from two articles published in the *EBU Technical Review*, No. 256 - Summer 1993.[1]

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1 Marti et al, ibid.; see also: HD-DIVINE, a Scandinavian terrestrial HDTV system, P. Appelquist

Stimulated by recent advances in image compression techniques (for example those achieved within MPEG), and by developments in the United States (the competitive evaluation of proposed standard digital terrestrial television systems), digital television broadcasting has become an important topic in Europe at both the political and technical levels. Early in 1991, several companies started activities dealing with terrestrial broadcasting, and a series of projects took shape: SPECTRE, from the IBA in the U.K. (now ITC and NTL), STERNE at the CCETT in France, DIAMONDS at Thomson-LER, also in France, HDTV-T in Germany and HD-DIVINE in the Scandinavian countries. All these projects have a lot in common - and each has its specific features. They all aim at terrestrial broadcasting of digitally-coded video signals using spread-spectrum techniques of the type known as OFDM or COFDM (coded orthogonal frequency-division multiplexing). However, the projects do not all have the same objectives: some aim at the short-term introduction of HDTV, others (SPECTRE, STERNE) aim at proving the technical feasibility of various scenarios for the introduction of digital television.

## 4.1. SPECTRE

### 4.1.1. The concept

The SPECTRE (*S*pecial Purpose *E*xtra *C*hannels for *T*errestrial *R*adio communication *E*nhancements) system is an application of the OFDM concept, with an implementation which is flexible enough to allow variations on the given scheme and various measurements to be performed. It also makes use of an advanced form of source coding (motion-compensated discrete cosine transform: MC-DCT) for the experiments, although most of the channel error measurements have been using pseudo-random binary sequences (PRBS).

The video code provides a data-stream of 13 Mbit/s including internal protection using a Reed-Solomon (255, 239) burst-error correction code. The video code is designed to operate over a range of bit-rates and is up-upgradeable to permit the processing of higher-definition signals having a bit-rate of the order of 24 Mbit/s.

### 4.1.2. OFDM modulator

The OFDM hardware is designed to be a flexible laboratory tool. It is structured in a rather unusual manner in that it uses a large number of digital signal processing (DSP) chips working in parallel to constitute a powerful computing engine. The modem uses software running in real time to execute the fast Fourier transform (FFT) function which modulates and assembles some 400 carriers into the orthogonal frequency division multiplex.

The modem can be programmed to perform a variety of digital modulation schemes. At present, QPSK, 8-PSK and 16-QAM have been implemented. The demodulator performs coherent carrier recovery on all 400 carriers and also determines the FFT block synchronization and automatic frequency control (AFC) from within the data. In fact, all the synchronisation information is found by correlative techniques within the normal transmitted information, so no special synchronization signals are necessary. The modulator mixes the signal to the required UHF channel and presents this signal to the high-power amplifier (HPA) consisting of two linear 200 W valve amplifiers operating in parallel to produce 400 W. This is backed-off by 6 dB to accommodate the large dynamic amplitude range of the OFDM signal. OFDM, like noise, has an approximately Gaussian amplitude distribution. This means that it has a high peak-to-mean power ratio, so the HPA needs to be operated at a much lower level than its normal rating for PAL signals. The chosen back-off level will depend on the HPA design, and a solid-state amplifier is likely to behave differently from a valve amplifier. The reason for using a valve amplifier was simply that this equipment had recently been removed from a relay station site and was available to the project at the right time. The specific requirement was that the HPA be linear, to give the best performance, and that sufficient power be available at the site should an increase in power be necessary during the trials.

### 4.1.3. Field trials

The first trials were performed using two existing transmitting sites: Stockland Hill and Beacon Hill. The first is configured as the wanted signal, the second as the interferer. The wanted signal is a transmission of the OFDM modulated compressed video signal, whilst the interfering signal is an OFDM signal fed from a PRBS generator. It is not necessary to provide a video signal source at the interfering site because the properties of coded video are noise-like and can be simulated using the random bit-stream from a PRBS generator. The effective radiated power (ERP) of the SPECTRE transmission is 250 W. This is 30 dB below the power of the existing PAL transmissions (250 kW at Stockland Hill). The beacon Hill site is similar to Stockland Hill, except that the antenna height is lower (50 m) and the ERP of the SPECTRE transmission is only 100 W.

Stockland Hill is a good site for these experiments, for a number of reasons:

- it includes a sea path which enables tropospheric co-channel interference from the French SECAM system to be studied;
- co-channel interference from the PAL transmitter at Rowridge (United Kingdom) will also be present;
- the site offers a range of "taboo" channels, so various experiments can be carried out;
- the number of rebroadcast transmitters in the area provides an excellent environment in which to study interference from OFDM.

The receiving equipment is contained in a Renault Espace, modified as a survey vehicle. The vehicle is fitted with a compressor to raise its telescopic mast to 10 m, and a generator is supported on a platform at the rear to power the receiving equipment which is mounted in special racks inside. The OFDM demodulator and video decoder are constructed in identical form to the transmission equipment. The survey vehicle measures quality around the field trial service area using a log-periodic antenna and results are logged by a lap-top computer for analysis at the laboratory.

A relay site which receives Stockland Hill transmissions off-air could be used for longer-term monitoring studies. Indeed, at a later time it might also be possible to use this site for the installation of a SPECTRE relay station. Since the SPECTRE transmission from the main station could use up all the theoretical redundancy in the frequency plan in some localities, there may be no additional channels available for the planning of SPECTRE relays. However, since OFDM has good anti-multipath properties, the relays could be on the same frequency as the main transmissions, using active deflection techniques. The relay would then consist only of a UHF amplifier, without any frequency transposition. Careful site shielding would be needed to prevent oscillation problems, but interference between the main station and the relay transmission should not cause a problem because one signal looks like an active multipath reflection of the other.

### 4.1.4. Laboratory measurements

Before it had become possible to make measurements from off-air transmissions, several laboratory tests had been carried out to determine the noise and interference properties of the QPSK-OFDM version of the system. The spectrum has "holes" at the PAL vision and sound carrier positions, and also a small "hole" at the PAL subcarrier. This is found to help when the PAL pictures contain highly-saturated colours. Each OFDM carrier is spaced 15625 Hz from its neighbour. The measurements included the degree of co-channel interference from PAL signals, and adjacent-channel interference. OFDM signals would be transmitted from the same geographical locations as existing PAL-I services, but in "taboo" channels at -30 dB power level relative to them. It is therefore essential that the adjacent-channel and image-channel protection ratios are better than -30 dB.

Field trials have recently been carried out in the area of Exeter, which lies in a valley with a shadowed area. These measurements have shown that, in this configuration, the coverage of an OFDM signal is comparable with that of a co-sited PAL-I transmission.

## 4.2. STERNE

### 4.2.1. Introduction

The CCETT has played an important role in recent years in the definition of standards for television contribution coding, as well as in the EUREKA 147 DAB project where the principles of COFDM were successfully defined and defended. On the basis of this long experience, the CCETT decided, in 1991, to develop a demonstrator of a digital television broadcasting system using COFDM techniques. The project is known as "STERNE" (*S*ystème de *T*élévision en *R*adiodiffusion *N*umérique) - which is also the French word for the tern (sea-swallow).

The objectives of the STERNE project are:

- to establish itself in the "gap" in terrestrial broadcasting, taking maximum advantage from the advantages of digital techniques in this vital market for broadcasters;
- to establish as the principal, imperative objective of the project, from the very outset, the broadcasting of conventional television signals to portable receivers, with service options linked to access control and to the transmission of assistance data, as well as the multiplexing of several programmes in a single channel;
- to extend the scope of applications to the broadcasting of HDTV to fixed receivers.

The principal objective is based on the following technical characteristics:

- channel bandwidth: 8 MHz
- quality approaching that of D2-MAC/packets, both for the 625-line image and for the sound component.

### 4.2.2. Channel coding

The STERNE project has selected an adaptation of the COFDM system, modified for the requirements of television broadcasting.

The masking conditions applying in reception on a roof-top, in an apartment or two metres above ground are quite different, and there is a substantial difference in terms of propagation losses between fixed and portable services.

The implementation of ambitious coding and modulation techniques for terrestrial broadcasting in a Rayleigh channel remains a challenge to engineers and designers. For instance, to broadcast to portable receivers, the efficiency achieved in the DAB radio system would have to be doubled, for example by the use of 2*4AM modulation with a 1/2-rate code.

### 4.2.3. System characteristics

The multiplexer will perform the synchronization of the various programme components (including access entitlements) as well as their integration within the transmitted data stream. The cost of the multiplex will be reduced as far as possible, whilst ensuring that services can evolve, on the basis of a mixed circuit packet approach, as this is an optimal solution for the consumer electronics industry. This approach also leaves the door open to data broadcasting services, whether they be broadcast telematic services or interworking with future ATM networks, which is also made easier with a packet-based approach.

The proposed system asserts the modular and programmable nature of the different configurations of a broadcast service, while giving high priority to the control of the different channel configurations attributed to broadcasting and to the analysis of the logical and physical architecture of the networks.

The EUROCRYPT access control system, currently being standardised at European level for MAC/packet services, is also being adopted for the STERNE project.

### 4.2.4. Calendar

Three stages of development are foreseen in the STERNE project:

- Development of a prototype demonstrator by the end of 1992. This is a derivative of the DAB project and permits the broadcasting of one programme of SECAM-like quality to portable (and mobile) receivers.
- Second demonstrator, at the end of the second quarter of 1993. This will be able to broadcast one programme of 4:2:2-like quality, two channels of SECAM-like quality, or three programmes of MPEG-1 quality, to portable receivers.

- The third stage, scheduled for the end of 1993, will demonstrate the possibility of a high bit-rate broadcasting service to fixed receivers, carrying either conventional or high-definition television signals.

The prototypes will serve essentially to illustrate the potential of an overall approach to the broadcasting and distribution of digital television, and will contribute to the attempts to standardise in this area.

## 4.3. HDTV-T

SPECTRE and STERNE are each the fruit of work in an individual organisation. They are concerned mainly with channel coding, the source coding being considered not as innovative but simply as part of an assessment of overall feasibility. HDTV-T, in contrast, is a German national cooperative project with ten partners (Table 3), and it has a budget of about 200 MDM up to the end of 1995

| |
|---|
| Bosch |
| DLR-Institut für Nachrichtentechnik |
| Grundig |
| Institut für Rundfunktechnik (IRT) |
| DAB-Platform |
| Heinrich-Hertz-Institut (HHI) |
| ITT-Intermetall |
| Siemens |
| Telekom-FI/DBP |
| Thomson Consumer Electronics Corporate Research |

*Table 3: Partners in the German HDTV-T project*

The project is led by a steering committee under the chairmanship of the Heinrich-Hertz-Institut; each participant is represented by one delegate. The work is organised in working groups on system aspects, image compression, modulation and hardware, and in several ad-hoc groups on specific topics. The objective of the project is to investigate the possibilities for digital television and HDTV broadcasting over terrestrial channels, satellite channels and broadband cable networks, by means of theoretical analysis, computer simulations, hardware developments and field trials. On the basis of this work it will then be possible to develop scenarios for the introduction of digital television and HDTV broadcast services offering practical alternatives to the systems of today.

Scalability, graceful degradation and compatibility between future television and HDTV services, as well as flexible utilisation of the multiplex, are the basic requirements of future broadcast systems. These systems will be data broadcasting systems, rather than television-only systems, so the emphasis is also placed on inter-connectibility with broadband cable networks, satellite channels and ATM-based B-ISDN.

In contrast to SPECTRE and STERNE, a large part of the HDTV-T project is devoted to source coding. In this respect, and also for the multiplexing aspects, the project relies on the proposals made by MPEG; the project is indeed contributing actively to MPEG. OFDM is currently the preferred solution for modulation because of its potential use in single frequency networks.

The HDTV-T project is organised in three overlapping phases:

- system definition
- hardware realisation;
- systems trials and optimisation.

Although the definition phase is still under way, the first hardware developments have already started. It is intended that hardware will be available by mid-1995 so that field trials can begin before the end of that year.

## 4.4. HD-DIVINE

In the summer of 1992, a digital terrestrial HDTV broadcasting system was demonstrated as a result of a collaborative Scandinavian study. The system included a motion compensated hybrid DCT video codec, a 512-carrier OFDM 16 QAM modem and four ISO/IEC Layer II sound codecs. The complete system was implemented in hardware. Since then further refinements of the vision codec and final assembly of the modem have resulted in a fully operational demonstration system and field trials started in January. The current implementation uses the 1250/50/2:1 studio standard and fills an 8 MHz UHF channel. A second implementation, ready in March 1993, will allow reconfiguration from 1 HDTV input signal to 4 input signals conforming to CCIR Recommendation 601.

HD-DIVINE is a project that developed a demonstration system to prove the feasibility of HDTV in terrestrial channels. The system, as it stands today, will not be proposed as a standard, either in Europe or anywhere else. It has, as its main building blocks, a video codec and an OFDM modem.

Since the first demonstration at the International Broadcasting Convention (IBC) in July 1992, a demonstration system of a modern teletext service has also been developed. It was felt that this area of television has not been fully covered in the discussion of the next generation of television broadcasting standards. This Broadcast Multimedia system was presented at the NAB '93 Multimedia World Conference. A second codec was also developed with minor changes compared to the first. The most important additional feature is the possibility of splitting the service into 4 parallel television services each with 4:2:2 subjective video quality and stereo sound. The future plans include a further updated MPEG-2-based video-coding algorithm, a more elaborate service multiplex, a broadcast multimedia function and further studies of the basic modulation parameters for the terrestrial modem in connection with field trials.

### 4.4.1. The video coding algorithm

The net data rate for the video in the HD-DIVINE system is roughly 24 Mbit/s. With a sampling rate of 54 MHz for the luminance and 13.5 MHz for each of the chrominance components and 8 bits/pixel the overall source data rate is around 650 Mbits/s. Thus the compression ratio is of the order of 30:1.

The vision codec is implemented as four parallel standard-resolution codecs with the HDTV screen divided in four vertical stripes for each of the codecs. Using a statistical multiplex the outputs of the four codecs are assembled into a single data stream. The interface of each of the codecs conforms to CCIR Recommendation 601, allowing for the reconfiguration of the system for the distribution of four standard-definition programmes.

Each of the four codecs uses a discrete-cosine transform (DCT) with a block size of 8x8 combined with adaptive quantization. DPCM with motion compensation is used between frames. Huffman coding is used as a variable-length code. There is a movement vector for every 4x2 pixel block. The maximum magnitude of the movement vectors is ± 32 pixels (horizontal and ± 16 pixels (vertical) with half-pixel accuracy. Estimation and compensation is done on a field basis and is done only between fields of the same parity. The field of motion vectors is processed in the same way as the image data i.e. in a hybrid-DCT loop but without motion compensation. This reduces the bit-rate required for the motion vector field.

It is not always possible to reduce the amount of data in a video sequence as much as is done in HD-DIVINE without introducing visible distortion. To smooth this effect, adaptive spatial prefiltering is implemented. This has another very significant effect on image quality. In effect, the filters used are very efficient in removing noise. Noise is not only unpleasant to look at, it also costs bits to code. The postfilter further reduces quantizing noise introduced in the coding loop.

The physical dimensions of the hardware implementation are quite small, bearing in mind that it is a prototype system and therefore full integration is not utilised; the vision decoder has just 16 boards, each measuring 9x11 inches.

### 4.4.2. The OFDM modem

The technique used in the HD-DIVINE modem is not new. Several laboratories are studying COFDM, including National Transcommunications (NTL) in the United Kingdom and the CCETT in France, and it is currently used in digital audio broadcasting (DAB).

OFDM is a multi-carrier system. The spectrum is shaped to provide robust co-channel interference properties for co-existing PAL transmission by cutting out the carriers which coincide with the PAL vision and sound carriers. Further carriers are left out at the edges of the spectrum to improve an adjacent-channel interference properties. In HD-DIVINE, 448 out of a possible total of 512 carriers are used.

The spectrum of each adjacent carrier overlaps those of its neighbours. This results in a rectangular overall spectrum and an efficient use of the available bandwidth. Demodulation of the carriers is possible if they are mutually orthogonal during the symbol time. This is valid if the frequency spacing between successive carriers is Δ

$f=1/T_S$. The symbol time $T_S$ in HD-DIVINE is 64 μs. To improve the immunity against inter-symbol interference, the symbol time is extended by a period of $\Delta$ T. This guard interval is set to 2 μs. in HD-DIVINE. Dynamic equalisation is used for each carrier to compensate for the channel. The method used requires that the receiver is stationary.

Each single carrier is modulated using 16 QAM with a resulting total bit rate of $4x448x1/(Ts + \Delta T) \pm 27$ Mbits/s. The error correction code is a Reed-Solomon code (208,224) which leaves a net bit-rate of $27x208/224 \pm 25$ Mbits/s.

The hardware implementation of the modem is also physically size; the modulator has only one board, measuring 28 x 63.5 cm plus two smaller ones.

### 4.4.3. Network planning

So far, only preliminary network planning studies have been carried out within the project. The basic assumption is that directional receiving antennas are used. The calculated requirement for a C/N ratio of 20 dB or less will be checked during field trials starting in January 1993. With techniques for integrated coding and modulation (e.g. trellis-coded modulation) the required C/N ratio will be even lower. The single-frequency network (SFN) is an interesting concept. In particular, the idea of regional sub-division by using a number of low-power transmitters in regional SFNs is attractive for a public service broadcaster with the need to reconfigure its nation-wide network for part of the time.

Laboratory tests are being carried out on the modem, and the protection ratios for co- and adjacent-channel interference are being measured. Full-scale field experiments are planned for February 1993.

### 4.4.4. Conclusions

The feasibility of digital terrestrial HDTV broadcasting has been demonstrated in real-time over-the-air transmissions using the HD-DIVINE hardware. The working assumption for HD-DIVINE is that the same primary parameters should be used for a new television broadcasting system, regardless of the specific broadcasting media. This means that the most difficult problem of HDTV is in terrestrial channels must be solved. The key word associated with the digital approach is *flexibility*. The future studies will concentrate on how to implement flexibility in the HD-DIVINE system, and to what degree.

## 5. TOWARDS IMPLEMENTATION

### 5.1. Relationships between projects

Implicit in much of the above-mentioned research and development activity is the desire to contribute to the establishment of a framework of new digital television standards intended to provide a solid and enduring basis for the introduction of new broadcast services and the corresponding receiving equipment throughout Europe in an orderly and rational manner. Although the existing technical barriers due to incompatible analogue television broadcasting systems have not completely prevented the introduction of pan-European services, their relative lack of success has clearly shown the need to make sure that digital television is not similarly fragmented among incompatible regional or linguistic variants that all perform the same basic functions.

To a large extent, the research is being undertaken by organisations in public-service broadcasting, telecommunications and consumer electronics, that are well aware that it is in their own interest to prevent such fragmentation and are therefore maintaining close liaison with each other. But it is already becoming evident that unless they can persuade the regulatory bodies to recognise this fundamental truth, this unique opportunity will be lost through the desire to give the economy in certain member-States a brief stimulus by allowing the introduction of incompatible proprietary systems by commercial operators concerned more by the prospect of short-term gains than that of the orderly development of an open market. In order to obtain competitive advantage, few if any of these operators are participating in the collaborative research projects.

As has been explained elsewhere, the European Launching Group has been set up to co-ordinate the preparations for the introduction of digital television broadcasting, and its Working Group on Digital Television Broadcasting (WGDTB) has already examined the relationships between the various research projects and identified a number of potential overlaps and gaps in their coverage, from the viewpoint of the development of a single European digital television broadcasting system with a range of technical quality levels.

Figure 2 summarises the relationship between all the research projects and includes two EBU Working Parties with related tasks.

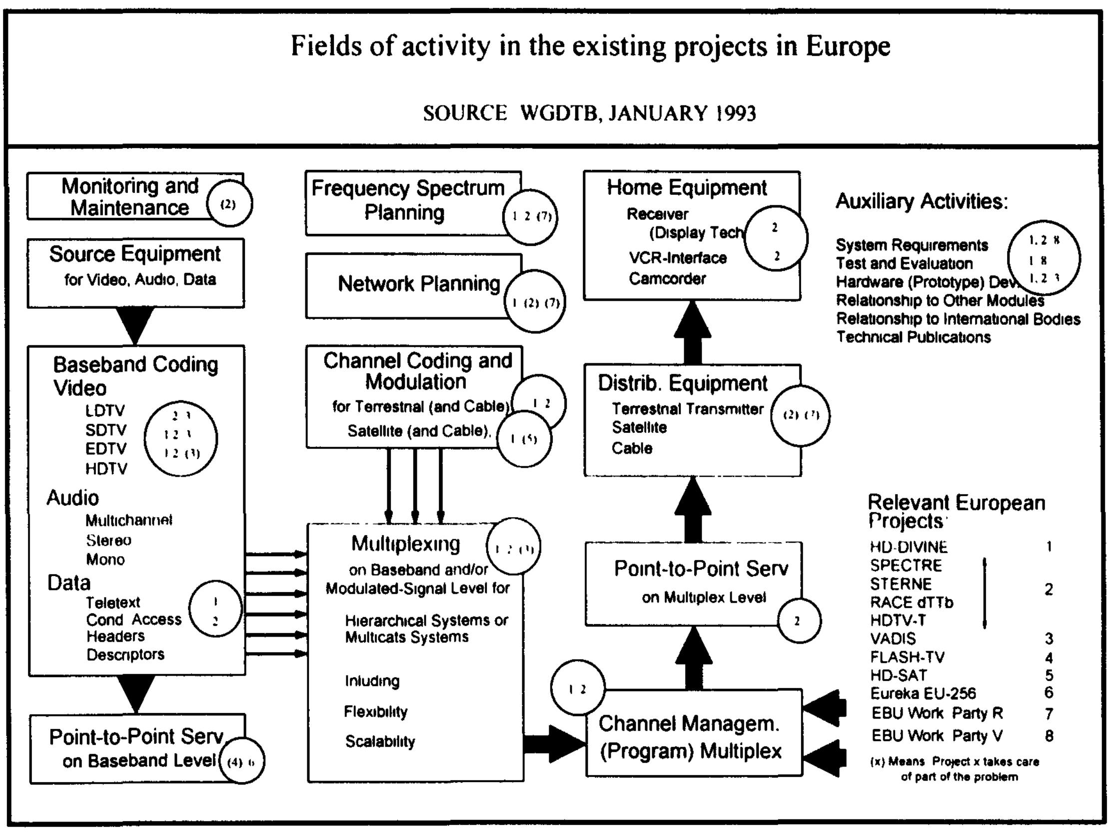


*Figure 2: Fields of activity in existing projects in Europe.*

## 5.2. Further research requirements

At this stage, most of the basic research work necessary to enable the framework for the introduction of operational services has already been completed, but much remains to be done in determining the optimum parameter values, which represent crucial choices with regard to the prospects of interworking, equipment design, quality of service and future extensibility. An iterative process involving the preparation of draft solutions, widespread consultation and consideration of the likely impact of potential developments outside the broadcasting industry's control is envisaged. Ultimately, this work is intended to result in the preparation of the input required by the European Telecommunications Standards Institute for formal approval as a set of European Telecommunications Standards.

The following new research projects have been identified in this context:

### 5.2.1. Project Manager and Project Office for the co-ordination of DTVB-related activities

The complexity of the DTVB scenario as well as the large number of on-going projects requires co-ordination. It is therefore proposed that a project manager; who should be independent from any major player in the field, together with a project office are urgently required. Funding of the order of 260 KECUs per year over the period 1993-98 are estimated to be necessary to support this co-ordination activity.

### 5.2.2. DTVB on satellite and cable

The existing projects in Europe are mainly concerned with the implementation of DTVB on terrestrial networks and in cable networks. What is missing is a significant part of the R&D necessary for the implementation of the services to be broadcast by satellite. The conversion of the signals broadcast by satellite into a physical form suitable for subsequent distribution by cable and possibly in B-ISDN networks would also have to be part of such a new project, because the modulation to be used on the satellite will not be appropriate for use on cable and B-ISDN. The new project could be seen as an activity in parallel to HD-SAT, but is outside the scope of the currently-defined tasks of HD-SAT. On the assumption that MPEG source coding and multiplexing can be adopted, the new proposal should be concerned mainly with channel coding and modulation, the overall system design, implementation in hardware, testing and drafting standards. It is envisaged that about 7 man-years over a 2 or 3 year period would be needed.

### 5.2.3. Investigation of the impact of cascading data compression systems

In future, a television signal will have to pass through a long and variable chain of data compression processes, of several different types. In the production centre, digital video-recording equipment using its own data compression will soon be in use; in post-production today, it is not unusual for the signal to pass ten times through such equipment. The signals may then pass through contribution links with their own data-compression processing, and on to the terrestrial transmitters via a satellite distribution network, again with its own data-compression. Further stages of data-compression, including those in domestic video-recorders, can be expected to occur before the signals finally reach the viewer's television set. As most of these links in the transmission chain are being developed independently, it would obviously be advisable to investigate what their cumulative effect on the signal is likely to be, and possibly to specify the precautions necessary to prevent significant impairment. A small project of about 4 man-years is envisaged.

### 5.2.4. Task force for the testing and demonstration of DTVB services

The complexity of the DTVB scenario and the past experience with MAC demonstrations, some of which were inadequately co-ordinated, has led to the conclusion that a professional organisation should be formed, possibly in mid-1994, to undertake the testing of the overall performance and functionality of DTVB systems, and to take on the responsibility of organising effective demonstrations of them to specialists and to the general public. A group of some 5 to 10 people would be needed during the period 1994-98.

### 5.2.5. Development of an overall scenario for the introduction of DTVB services in Europe

In the past, the telecommunications administrations, the broadcasting companies, the domestic electronic equipment makers, the computer industry and viewers have too often been unaware of each other's requirements, constraints and objectives in this context, which has greatly hindered the development of a coherent scenario for the introduction of digital television broadcasting. In order to overcome this problem, it is proposed that a Project consisting of a few people, each with an in-depth knowledge of one or more of the above-mentioned sectors who - after thorough discussions with the whole spectrum of potential players in the broad field of image communications - should propose an overall strategy for the introduction of DTVB in Europe, by about the end of 1994. This is estimated to require a total of about 4 man-years over two or three years.

### 5.2.6. Selection and standardisation of scrambling and conditional-access systems for DTVB

In many cases, providers of DTVB services will require to be able to control access to them, notably in order to obtain revenue and to restrict viewing to territories for which they hold copyright. While each service-provider may wish to have complete control of its conditional access system, from the viewpoint of the potential user, however, it would be impractical to have more than one or two descrambling boxes specific to the various programme services (s)he wishes to receive. It would therefore be very desirable to devise a conditional-access system that provided all the required functionality specific to the individual services by means of plug-in units, e.g. "smart cards". The total effort required to develop such a system is estimated to be some 12 man-years, during the period 1993-95.

## 5.3. Community R&D in the Fourth Framework Programme

The R&D work in relation to multimedia systems and services planned for the Fourth Framework Programme has still to be refined with the help of experts and the member states. It can be expected, however, that the work will broadly cover the following areas:

### 5.3.1. Second generation image coding

Pixel-based compression algorithms have now reached their limit in terms of compression ratios. They have proven to be quite powerful and have opened the way to the new opportunities which promise to revolutionise the audio-visual landscape in the decade to come. However, it seems that they will not be appropriate to meet new challenges such as very low bitrate coding (8 kbit/s video stream) for applications such as mobile video communication or very high definition television (source signal at 5 Gbit/s for a 2000 x 2000 pixel image at video rate). Research is needed in image segmentation and interpretation, region and model-based coding. It is vital for Europe to actively promote and support collaborative R&D work in this area.

### 5.3.2. Beyond High Definition Television

In areas such as medical, imaging, film making, printing or art diffusion, there is a need for very high definition imaging systems. As a consequence, research is needed aimed at developing very high definition (2000 x 2000 and beyond) image acquisition, coding storage, transmission and display systems. This will necessitate the development of faster and higher density image sensors with extended color reproduction range. To cope with the dramatic increase in the amount of information to be transmitted or stored, new, more efficient coding methods, will have to be developed. Higher density recording systems and higher speed transmission media will be necessary. Finally, very high resolution display units will have to be developed. High-end professional applications will then pave the way for the timely and market-driven evolution of consumer products to higher resolution levels.

### 5.3.3. Advanced imaging for Telepresence

The use of video conference facilities is not so high as one would wish for e.g.the reduction of travelling. One of the reasons could be that the present video conferencing systems do not yet fulfil the quality requirements of the users. Perhaps the telepresence can be increased by higher image resolution and larger displays. The degree of telepresence that can be achieved is a function of the accuracy of the sensory information conveyed back to the operator. This in turn depends on the developments taking place in sensor design. Sensors likely to contribute to bandwidth utilisation are: "magic glasses" (high resolution colour displays one for each eye), microphones, earphones, navigational aids (tracking the position and orientation of the glasses and consequently the head of the wearer), "magic gloves", sen-sorised/motorised "coat" and a high mobility navigation platform.

### 5.3.4. Advanced image generation

At a time when it is possible to dramatically increase the number of TV channels and when a profusion of new multimedia interactive services are about to be introduced, there is more than ever a need for new, audio visual material to be generated. Appropriate steps should be taken to strengthen the technologies of image analysis, image understanding and interpretation, image modelling and image rendering. These technologies will have a major impact on other areas such as second-generation image coding for applications ranging from low bitrate to HDTV, telepresence and advanced man-machine interaction for intelligent terminals.

### 5.3.5. Channel coding and modulation

UHF and VHF bands are used for the distribution of analogue video programmes and could be made much more efficient by the use of digital modulation and coding techniques. Convolutional coding and various multi-level modulation schemes have been applied in radio channels in general. Careful consideration is needed of the application of these modulation and channel coding techniques to the distribution of HDTV and TV programmes, with particular reference to channel bandwidth, signal-to-noise ratio, echoes, etc. The goal should be to optimise the exploitation of channel capacity and to minimise the cost of a dedicated receiver.

a) definition of modulation, channel coding and error protection strategies for bandwidth limited media taking account of: Spectrum-efficient systems on satellite, cable and UHF/VHF channels, taking into account, in this later case, the present occupation of the spectrum by analogue services and the requirement of suitable transition scenarios. Processing of distortions resulting from echoes by easy-to-implement means such as simple equalisation schemes, guard intervals, channel coding. Balance between the amount of processing required at the interface and that required for the receiving equipment. The multiple reception configurations of UHF/VHF signals (portable; indoor or outdoor; mobile; fixed with adapted antennae).

b) definition of system and service parameters for the distribution of video programmes, including:

   the consideration of various quality levels enabled by digital coding (compromise between quality and bit rate): The possibility of a graceful degradation of received signal quality on adverse reception conditions (rains for satellite, distance, echoes, interferences for UHF/VHF). The definition of a flexible multiplex allowing various programme configurations for TV/HDTV, sound or data programmes. Access control. Network architectures of UHF/VHF systems (central transmitters/cellular or dense networks). Compatibility and inter working problems between the three distribution media (cable, satellite, UHF/VHF) and between those and contributive transmission networks (satellite, ATM, others). Choice of suitable image coding systems and definition of their interface to the distribution networks.

### 5.3.6. Network management and audio-visual services traffic

The introduction of video services carried by IBC to subscribers served by cable distribution (CATV) or MVDS depends on the development of the appropriate interfaces that will enable IBC to built upon this large customer base. Communications satellites are also a flexible medium for digital HDTV news gathering and studio-to-studio connections, and these sources have also to be connected to IBC.

Video traffic, together with traffic from other sources, such as voice terminals and high-speed computers, will be multiplexed over physical links of fixed total capacity, with each traffic component having different performance requirements for end-to-end delay and data loss. In the past years, the introduction of asynchronous networks has led to expectations that such transmission media would lead to services providing variable bitrate and constant quality. Active research in this area has been and still being carried out, producing valuable insights concerning the statistical behaviour of asynchronous networks and laying the foundations to coding architectures well adapted to the specific requirements of these networks.

### 5.3.7. Mass storage

Digital audio-visual services depend to a large extent on the economic availability of appropriate storage facilities in both the systems of the service providers as well as the end users. Appropriate coding algorithms can increase the amount of storage capacity, but also considerations for rapid (non sequential) access should play a role in the development of these systems.

It is expected that one of the services to appear first may well be video on demand or pay per view, as it seems to lend itself to immediate exploitation.

The most central issue in the design of multimedia information systems is the mechanism through which information is fetched and supplied to the user. If multimedia systems are to be accepted they would have to be fast and easy to use. Ease of use, for untrained users, pertains to the utilisation of natural means of communication such query by content or query by example. Both methods are expected to have a significant impact on network performance. This can be explained by the number of on-line users and the amount of information communicated at any moment in time. Research on query mechanisms can lead to a massive reduction in the size of query sets thus reducing excessive demands on system bandwidth

### 5.3.8. Advanced multimedia terminals

It is likely that the introduction of multimedia services will make it more difficult for average users to make full use of the information at hand. Against this background, efforts are necessary to promote research in intelligent terminals able to assist the user in understanding the extent of the information and services available, finding relevant material, retrieving data, accessing services and contacting other parties. For this, the introduction of advanced searching techniques based on artificial intelligence is essential. In addition, high level man-machine interaction mechanisms should be developed making wide use of natural human-to-human communication means such as speech recognition, lip and sign language reading, facial expression and eye position reading etc.

### 5.3.9. Multimedia networks and services, interactive TV

Since connections have to be established prior to any transmission, mechanisms that verify the requirements of the connection, while taking into account the current state of the network are needed. Call admission mechanisms will be based on powerful decision-making algorithms which will accept or reject a new connection by analysing the transmission links along its path and by querying network databases for long term as well as instantaneous link related information in real-time. Call admission mechanisms may have to be followed by congestion control procedures in cases where the multimedia source behaviour deviates. In these cases, preventive or reactive source monitoring and bandwidth enforcement procedures may be activated in order to prevent the network from reaching unacceptably high levels of congestion while maintaining the offered quality of service.

Developing efficient quality control and service management mechanisms for multimedia transmission over broadband networks requires accurate modelling of individual or groups of multimedia sources in terms of their bandwidth requirements and their interaction with other services multiplexed along the same path, as well as flexibility in incorporating the underlying coding and decision-making algorithms into the appropriate layers of the network operating system. The results of research in this area are expected to contribute to the introduction of new codec architectures which can optimised for interactive multimedia applications and services.

### 5.3.10. Personalised access and protection of copyright in distributed systems

As the number of television programmes increases the need for effective access control and copyright protection systems increases. In order to avoid multiple access systems in the customers premises an effort should be made to develop a universal access sytems that can serve the needs of many different service providers.

## 5.4. Conclusions

From the foregoing description, it is clear that the European Research effort has already made a major contribution to the development of digital television broadcasting technology. This research has clearly demonstrated that the availability of powerful data compression algorithms and new modulation methods can enable substantial improvements to be made, notably in the efficiency of spectrum usage, in the number of programmes that can be broadcast simultaneously without loss of technical quality, and in the flexibility of the transmission multiplex. Indeed, this progress has been so dramatic that it is liable to transform broadcasting out of all recognition, from a situation of technical constraints into to one where they are mainly economic. And that is -- at least partly -- why implementation of the transition from today's analogue technology to tomorrow's digital world is likely to be so difficult.

Sufficient progress has thus been achieved to confirm that digital television broadcasting is likely to be technically and commercially worthwhile without further fundamental appraisal. But that does not mean there is no place for further research. On the contrary, provided digital television broadcasting in Europe corresponds from its beginning to the coherent framework now being developed, the scope for compatible future enhancements is virtually unlimited. And it is important to appreciate that the purpose of such research is not simply to devise technical solutions to the practical problems of digital television broadcasting, but also to ensure that the European broadcasting industry is at the leading edge of such advanced technologies and is thus able to evaluate the merits of alternative approaches in the international standardisation bodies and to acquire valuable intellectual property rights in the field.

# *PART 4 - THE STANDARDISATION PROCESS*

Chapter 6

**Prospects for common worldwide digital television systems**

David Wood, European Broadcasting Union.

# Table of contents



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## 1. INTRODUCTION

The purpose of this paper is to provide a brief summary of the current prospects for worldwide common standards for digital television broadcasting.

- Section 2 of the paper is a review of the key elements which make up a digital television broadcasting system.
- Section 3 considers the value of standardisation.
- Section 4 outlines potential standardisation bodies, and the areas they cover.
- Section 5 gives the current situation in the three key world economic Regions; Europe, Japan, and North America.
- Section 6 is a short analysis of the prospects for standardisation, building on the earlier Sections.
- Section 7 draws some conclusions.

The paper is complemented by three Appendixes. Two provide background information on commonality, quality, and other issues. The third, a recent document from ISO/IEC JTC1 MPEG, gives currently agreed parameters for certain system elements.

## 2. THE OVERALL STRUCTURE OF THE DIGITAL TELEVISION BROADCASTING SIGNAL.

Understanding fully the elements of a digital television broadcasting system takes considerable time and energy, but is is possible to grasp certain concepts without being a specialist. It is useful to do so, in order to see why different bodies are, or may be in future, standardising different parts of the system, and how they will eventually fit together.

One way of dissecting a digital television broadcasting system is to use a tool known as the OSI (Open Systems Interconnection) Layer Model. This is a generic method for subdividing an overall system into the elements which are always found in any system for carrying information from one place to another.

The Layer model is a kind of onion. The total system is achieved by starting with a core (Layer 7) and building up successive layers around it (Layers 6 - 1). The information that you ultimately want is in the central core; and, to get it, the receiver has to successively unwrap the layers around it. The layers thus represent the packaging. The packaging is needed to collect together all the items that need to be transported at one time, and to adapt the information to the particular method of transportation being used.

What the terms mean, that are used to describe each Layer, is subject to some variation in interpretation. One relatively common interpretation, however, is given below.

## THE OSI LAYER MODEL APPLIED TO DIGITAL TELEVISION BROADCASTING

- LAYER 7: THE APPLICATION LAYER

  In this Layer are the television and sound source signal formats, together with the base formats of any other services to be provided, like teletext.

- LAYER 6: THE PRESENTATION LAYER

  In this Layer is the digital compression system or digital coding system which is applied to the vision, sound, or data services.

- LAYER 5: THE SESSION LAYER

  In this Layer is any conditional access system or scrambling system which is to be applied to the vision, sound, or data.

- LAYER 4: THE TRANSPORT LAYER

  In this Layer the data rate conversion of all the different elements of the total service is done, so that all can fit into the final available bit-rate. The information is made ready for the multiplex.

- LAYER 3: THE NETWORK LAYER

  In this Layer the baseband multiplex of the different service elements is created, together with a service identification system.

- LAYER 2: THE DATA LINK LAYER

  In this Layer appropriate error correction coding is added, to make the overall system work in the particular environment used for transporting the signal. There may also be other RF-channel-related service identification elements to be added.

- LAYER 1: THE PHYSICAL LAYER

- In this Layer the modulation and channel coding, which is needed for the particular environment used for transporting the signal, is applied.

Of course, broadcasting standards have many dimensions, other than technical ones, to take into account. For example, Charles Poynton, a strong advocate in the United States of 'interoperability', points out that in North America there is actually an additional layer, which also has to be catered for. This is Layer 8: The Political Layer. We may also have some additional Layers in Europe.

In the global world of vision and sound services and communications, there are many ways to carry programmes or pictures to the viewer. There are also many different levels of picture quality that need to be provided. This means that we cannot use an identical system in all cases.

In the parlance of digital television, elements which can be always used, or at least used in many circumstances, are called *generic*. The elements that have to be changed according to circumstances are termed *application-specific*. The digital television system will be a mixture of Generic and Application-specific elements.

The objective of striving for *interoperability* is to make, starting from Layer 7, as many of the Layers as possible of the OSI model 'generic', and as few as possible 'application specific'. The objective is to move as far outwards from the core as far as possible, while still maintaining the possibility of using the system for different applications. The different applications include not just the different broadcasting applications, but also computer workstations and video telephones.

Since we must always start from Layer 7, the prospects for interoperability lie largely in the higher-numbered Layers of the model. These Layers, as explained later in this report, are being standardised in the ISO/IEC Joint Technical Committee MPEG. It is therefore largely in the MPEG work that the key to interoperability lies.

Having grasped the concepts of 'generic' and 'application-specific', you may also realize that there are two dimensions across which the 'generic-ness' can apply. These are the *application dimension* (the use of the same system across different types of transport and function) and the *international dimension* (the use of the same system across different countries or Regions).

Both of these dimensions provide important opportunities for reducing consumer costs, and thus for increasing the total market for equipment and services, but if a choice has to be made, the general public's interest will probably be served more by commonality between applications in a given region, than by commonality between Regions for the same transport or type of service. Nevertheless commonality in both dimensions would be well worth striving for.

Appendix A to this report provides additional background on the dimensions of commonality. This is a technical paper prepared by the author in 1992.

## 3. THE VALUE OF STANDARDISATION

In broadcasting, the primary aim of standardisation is to allow the interchange of signals and programmes. An ideal world would be one in which common standards were used everywhere, where programmes produced in Japan, the United States or Europe could be readily transmitted in any other part of the world, without the need for conversion, and where radio and television signals and receivers were common and interchangeable.

Alas, this is not the present situation. There are four colour television systems (PAL, SECAM, NTSC and MAC), with many variations in different parts of the world. Bands in the electromagnetic spectrum are also unevenly allocated, with different bands in use in different parts of the world. Great success has not always been achieved in broadcasting standardisation, especially where this would reduce the utility of major investments.

It is self-evident that some facets of broadcasting should be standardised before such major investments are made, if possible. Future generations would pay heavily for their lack. Common signal formats make it possible to achieve the economies-of-scale in equipment production, and thus bring unit costs down. But however true this may be, there are many opposing forces. It can be argued that standardisation does not allow the use of ever-improving technology, because it freezes the technical form of the system at a particular instant in time. Equally, there may be occasions when having your own personal standard can provide a commercial tool in a competitive situation.

But even where it is a good thing on balance, there may be even greater hurdles to overcome in achieving standardisation. Companies from market-economies need much coaxing to come together and bury their different interests for long enough to secure a standard. Even if they can manage this, they still have to persuade the standards bodies themselves to adopt it.

To complicate matters, evolution in broadcasting technology has shown something of a cyclical pattern in the last ten years, and Appendix A provides some reflections on this.

The world standardisation landscape seems like a jungle of committees and procedures. This is in some ways an accurate impression. There is duplication of effort and some bureaucracy. But there is no other choice but to try to understand how the various bodies work.

In any field of human endeavour, whenever collective decisions are to be made, or agreement is required, there is always a human dimension. Individuals are biased in their judgement by human failings, such as the need for recognition, or the need to exert authority, or the need to reinforce one's own ideas and reject those of others. Standardisation is no exception. Success or failure depends, as much as always, on the individuals involved, and particularly on available leadership capacity. The infrastructure plays a role, but experience shows that it is not at all the only determining factor in success or failure.

The standardisation process can be an uncomfortable one. Reconciling different perspectives, deciding on what will be a satisfactory cost/performance/trade-off, and knowing when is the right time to freeze technology, all require their own special kind of genius. The only way forward is to hope that we succeed more often than we fail. Only in standardisation can we create the real framework for true competition in broadcasting: competition in the creative content of the programmes themselves, and among equipment manufacturers serving the same market.

## 3.1. Standardisation organisations and groups

### 3.1.1. The ITU/RS

The most important world standardisation body for total broadcasting systems is the International Telecommunications Union (ITU). The ITU has a hierarchical structure of committees which progressively enter into more and more detail. Periodically, the ITU publishes details of its deliberations.

The ITU subdivides into two sectors: the Radiocommunications Sector (RS), which is broadly responsible for broadcasting matters; and the Standardisation Sector (SS), which is responsible for standardisation of other telecommunications matters. The ITU/RS incorporates the former CCIR.

The ITU/RS publishes its results in a formally agreed way every four years. During the intervening periods, results of its immediate studies are available in temporary forms, and can be approved by correspondence. The ITU/RS thus operates on a four year cycle, which is termed a 'Study Period', but Recommendations can be agreed at any time provided there is consensus.

Within the ITU/RS there are currently eleven different study groups. The two largest groups are those which directly concern broadcasting. These are Study Group 10 for sound broadcasting and Study Group 11 for television broadcasting. Within these study groups there are Working Parties and Task Groups. The unspoken reason for this substantial inverted-tree structure is that useful discussion cannot be held if the committee size is too large.

The participants in ITU/RS meetings are the representatives of the governments of the Member States, because the 'airwaves' are considered to be a national resource for each state. In practice, however, the representatives are often from national industry, or national broadcasters, or the national carriers. Recognised Private Operating Agencies or International Organisations can also take part as such in the work, but do not have the right to vote.

The principal objective of the ITU/RS is to produce 'Recommendations'. There are at least morally binding on the member states, but not legally enforceable as such. However, they may be taken as the basis of mandatory national regulations. The ITU/RS also draws up Reports, Questions, Study programmes, Decisions and Opinions. These are the collected opinions and studies of the members and suggestions for future work. They can be useful, but are very much less valuable than Recommendations.

Much as one might wish it to be otherwise, the ITU/RS meetings are usually forums for reporting studies and passing information, rather than bargaining occasions. If there is to be agreement, it usually has to be pre-arranged between the key interest groups before the meetings. Delegates attend meetings with national briefs to put forward a particular position, but rarely with the freedom to negotiate.

There certainly have been some successful Recommendations, but even the most ardent ITU/RS supporter would admit that it is extremely difficult to reach Recommendations. Even when they are agreed, there is no guarantee that member states will adhere to them.

In the 1960s, the ITU was not able to reconcile the different views on the PAL/SECAM/NTSC colour television systems. In the end it could only recommend that the ITU member states should choose their colour television broadcasting system from among about 20 variants.

In the 1980s, following prior agreements between the key parties, the ITU was able to agree on a Recommendation for a new digital television studio standard. It is termed Recommendation 601. This was followed by a Recommendation for a tape recording format to be used to record Recommendation 601 signals, and a signal interface to connect equipment together. The stage was set for a new age of universally interchangeable equipment for studios and television production throughout the world. This has not happened. In the United States and Japan, manufacturers agreed on their own different digital standard and tape formats. There were no legislative powers available to prevent it.

The ITU/RS has also been struggling with the question of High Definition Television since 1983. Everyone agrees on the value of unique worldwide HDTV standards, but support for particular standards depends up on geographical points of view.

The two key groups for digital television broadcast standards are probably Working Party 10-11/S, which is concerned with satellite systems, and Task Group 11/3 which is concerned with digital terrestrial systems. So far, it has not been possible to agree substantive draft Recommendations for digital television broadcasting systems in either of these groups, but many believe that this is largely because the subjects are not yet sufficiently mature. Both groups will meet next in Autumn 1993.

Task Group 11/3 has appointed a Special Rapporteur to take a global look at the prospects for a common digital terrestrial system, or for common elements.

In the scheme of things, the above two ITU/RS groups are likely to be the focus for any worldwide standardisation of the modulation and channel coding systems to be used for digital broadcasting. These are Layers 1 (Physical) and 2 (Data Link) of the OSI Layer model.

### 3.1.2. The ISO/IEC

The International Standards Organisation (ISO) and the International Electrotechnical Commission (IEC) are agencies concerned with the standardisation of equipment, consumer and industrial products. The work of the IEC does have a relationship to broadcasting, because it is empowered to define standards for electrical equipment used in the studio or the home. The IEC has been concerned, for example, with the standardisation of a digital audio interface, developed by the EBU in conjunction with the Audio Engineering Society in the United States. It has also been concerned with the standardisation of various television and audio recording formats. Participants at the IEC are generally from large manufacturing companies. Usually their interest is to gain recognition for a standard they themselves have developed (and for which they hold intellectual property rights), rather than be involved in negotiation to arrive at a compromise.

The ISO is a worldwide general equipment standardisation body, and the IEC is a worldwide electrical equipment standardisation body. The area of Information Technology overlaps the areas of competence of the two standards bodies; and so several years ago, a Joint Technical Committee, JTC1, was formed to standardize 'IT'-related equipment.

One of the JTC1 subgroups, the Motion Picture Experts Group MPEG, has already standardized a baseband video compression and multiplexing system for VHS-quality video (MPEG-1) and CD-quality audio. The same group has now moved on to systems with higher video qualities, and this work is termed 'MPEG-2'. This activity is critically important for digital broadcasting.

The MPEG-2 group is now renowned for its speed of working and its considerable success. Participation is extremely large, but in spite of this it is making considerable progress. One of the keys to this is in its fine leadership.

In the current scheme of things, the ISO/IEC MPEG group is likely to be responsible for standardising the video coding, baseband multiplex, and baseband identification system for digital television broadcasting. These functions correspond to Layers 4 (Transport), 3 (Network), 6 (Presentation), and 7 (Presentation) of the OSI model.

### 3.1.3. The EBU

The European Broadcasting Union (EBU) is an organisation of broadcasters in what is termed by the ITU the 'European broadcasting area'. This means west and east Europe, and the countries on the southern side of the Mediterranean basin. Broadcasters from other parts of the world can join the EBU as Associate Members. At present there are about fifty members in the EBU, and over sixty associate members from other parts of the world.

The EBU Technical Committee is divided into five Working Parties concerned respectively with new systems and services, broadcasting technology, production, transmission via Eurovision, and training. The largest single EBU Working Party concerned with standardisation is Working Party V, dealing with new systems and services.

The EBU establishes and publishes Recommendations and Standards. These are also often submitted to the ITU and/or the IEC for consideration as World Standards. These EBU Standards represent nothing more than a voluntary commitment by the broadcasters of Europe to a particular system. There is no regulatory force behind them. As with the ITU, circumstances have arisen where Members, for one reason or another, are not able to abide by an agreement.

The list of Standards agreed by the EBU Technical Committee is considerable. The most important include: the digital studio production standard and tape recording format, the digital video and audio interfaces for equipment, the Radio Data System, the MAC-packet family of standards, and the recent Programme Delivery Control system. The EBU has regular liaison with the European consumer manufacturers.

One of the fundamental problems in developing new systems for broadcasting is that such systems have to be ever more technically complex and sophisticated. This translates into a requirement for ever larger resources in terms of money and manpower, to develop the systems. In the EBU there are certainly a number of excellent laboratory facilities. But, as time goes by, these prove not to be large enough for complete development programmes. There is an increasing need for joint projects with manufacturers. In the early 1980s the EBU members had, essentially, the resources among themselves to develop the MAC-packet system. However, for a system several times more sophisticated, like HD-MAC, a pan-European industrial consortium was essential. This is also the case for digital television broadcasting systems.

The EBU welcomed the establishment of the European Launching Group (the ELG) and its Technical Sub-committee, the WGDTB, which now globally coordinates European Research and Development in digital television broadcasting, and the EBU contributes substantively to its work. It makes a major contribution to system requirements, system evaluations, and frequency planning.

### 3.1.4. Collaborative projects

There are currently many pan-European research and development programmes, which include broadcasters, universities and industrial interests. The funding of such projects can come partially from the European Community or National Governments. The projects relevant to broadcasting include Eureka 95 on HDTV, Eureka 256 on digital video coding, Eureka 147 on digital audio broadcasting, RACE HDSAT, RACE dTTb, etc.

The WGDTB endeavours to coordinate the work of these collaborative project and to lead Europe in a common technical direction.

### 3.1.5. EBU/ETSI JTC

The European Telecommunications Standards Institute, ETSI, was established about five years ago at the proposal of the European Committee for Posts and Telecommunications (CEPT), with the aim, among others, of establishing standards that could be made mandatory by legislation in the participating countries (eg an EC Directive within the European Community).

ETSI has a small permanent staff, and admits participating members from all sectors, including regulators, manufacturers and broadcasters. The organisation of ETSI is structured in a series of hierarchical committees, which devolve discussion of ever more specialized topics to lower groups. The apex of the pyramid is the General Assembly. Below this is the Technical Assembly. The Technical Committees (TCs), which report to the Technical Assembly, are empowered to establish Sub-groups or Project Teams to work on particular standards. The Project Teams are groups of individuals dedicated to a particular task and may be stationed for periods of time at the ETSI headquarters, or elsewhere, to do the work.

ETSI has provision for an initial indicative vote. This shows how people would be likely to vote, if a vote were actually called. Following this the minority may agree to joint the majority. In this way, consensus can be achieved without formal voting. Before standards are finally approved however, they are subject to a Public Enquiry, which is effectively a formal voting procedure.

A feature of ETSI is that when members joint ETSI they agree to abide by its decisions.

The European Broadcasting Union and the European Telecommunications Standards Institute (ETSI) have formed a Joint Technical Committee, the JTC. This body is responsible for European standards for broadcast signals, and point-to-point transmission of broadcast signals. This reports to both the ETSI Technical Assembly and the EBU Technical Committee.

Formal approval and Public Enquiry for any European digital television broadcasting standards will probably need to be via the EBU/ETSI JTC, following ETSI procedure.

### 3.1.6. CENELEC

The European Committee for Electrical Standardisation (CENELEC) concerns itself with standardisation of consumer and industrial equipment, rather than the broadcast signal format. Its standards are subject to national inquires among Member States. Its work related to broadcasting could concern receiver standardisation. It has Technical Committees concerned with television and radio receivers, conditional access, and cable distribution systems.

CENELEC has standardised a Conditional Access system for MAC/packet services, Eurocrypt M. It may thus be the appropriate body to standardise the corresponding system for digital television broadcasting, associated with OSI Layer 3 (the Session Layer).

### 3.1.7. SMPTE/ATSC/ACATS

The Society of Motion Picture and Television Engineers (SMPTE) is a common interest group of engineers in the broadcasting and film industries. Most of the members are in the United States and its work in the standardisation area is usually centred on the needs of North America. The SMPTE has a large range of committees covering aspects of programme production equipment and agrees on standards which are voluntary.

After the difficulties in convincing the CCIR to accept the 1125/60 HDTV production standard in 1985, the SMPTE drew up its own internal standard for an 1125/60 interlaced production standard, which is known as SMPTE 240M. This has not gained acceptance by the US national standards body, the ANSI, but it is widely used.

In the early 1980s it became clear that there were a number of societies and bodies in the United States with a stake in future television systems. There was, in particular, a pressing need to establish a United States view of HDTV and enhanced television. All the interested groups agreed to the establishment of one new body to address these issues, which is the Advanced Television Systems Committee (ATSC).

In the 1950s there was a National Television Systems Committee (NTSC) which successfully steered the United States to agreement on a unique colour television system, itself called the NTSC. In structuring and naming the new committee in a similar way to the earlier one, there was an implied hope that the new committee would similarly steer the United States towards a unique HDTV system. The ATSC is the primary definer of the United States position on HDTV at the ITU. The ATSC achieved recent success in standardising a ghost-cancelling system.

The Radio Regulatory body in the US is the Federal Communications Commission (FCC). Several years ago the FCC set up the Advisory Committee for Advanced Television (ACATS). This group created a large voluntary committee structure, which analysed many aspects of the implementation, introduction and used of advanced television systems.

The FCC launched a call for candidate systems for a terrestrial advanced television several years ago, and the proponent systems were tested in a collaboratively funded laboratory, the Advanced Television Test Centre, ATTC, and in the Canadian Advanced Television Evaluation Laboratory, ATEL, and in the cable industries joint laboratory, Cablelabs.

The US has thus set up its own national standardisation procedure for digital terrestrial television. The situation for digital satellite systems is currently that no formal standardisation procedure has been adopted.

## 4. CURRENT STATUS IN DIFFERENT REGIONS

### 4.1. The Japanese situation

The Japanese Ministry of Posts and Telecommunications recently announced a timetable for using the analogue MUSE HDTV system for satellite broadcasting, continuing into the next century. However, large scale studies of digital systems are continuing, and the LOT approach (a Lapped Overlapping Transform system for source coding) seems particularly promising, and may be more robust than conventional DCT.

The COFDM approach is also being seriously evaluated for terrestrial broadcasting, and the MPT has licensed an experimental station for propagation/planning evaluation.

The concept of a Broadband Integrated Services Digital Broadcasting system is being vigorously studied. The basic principle is that a common fixed length packet system (2Kbit) is used for transporting all types of digital service, including audio, video, teletext, etc... The same packet system would be used for all types of delivery medium including satellite, terrestrial, cable, etc... In effect, the broadcast channel would become a broadband integrated services digital broadcasting channel, in a way parallel to the BISDN concept for fibre optic networks. The concept combines the advantages of flexibility of use with commonality of receiving equipment, and is thus a very powerful idea, which deserves world-wide consideration for all digital delivery system studies. OSI Layers 3 and 2 would need to be arranged so as to accomodate the ISDB packet.

A satellite ISDB is being developed for the 21 GHz band. This will be a multi-beam variable-EIRP system, with a capacity of 150 Mbit/s.

In general the Japan MPT's policy is thus to continue analogue HDTV and EDTV broadcasting, for terrestrial and 11/12 GHz bands, but studies of digital broadcasting possibilities are considered very important, and a flexible digital broadcasting system for 21 GHz is being developed.

Terrestrial transmitter frequency congestion in Japan is among the most severe in the world, and thus digital terrestrial television broadcasting is correspondingly more difficult to implement.

## 4.2. The North American situation

One of the results of the FCC ACATS evaluation of the terrestrial ATV candidate systems in early 1993 was that a decision was taken to pursue uniquely a digital system for terrestrial television. No single digital candidate system emerged a winner on all counts. The four digital candidate systems were encouraged to come together of their own volition, to create a combined system which makes use of the best of each of the candidate systems.

In May 1993, the four proponents announced that agreement has been reached on a combined system. This is termed the "Grand Alliance". The initial formulation of the system is that it is a multi-scanning format system which uses some (but not all) of the coding tools used in the MPEG-2 system. Some new tools not in the MPEG-2 kit are also included.

The system will be formally evaluated in about six months, when hardware is available and a decision on acceptance will be taken early in 1994.

In practice this means a longer time window for international agreement has been opened.

Satellite digital television broadcasting is planned to start in 1994, and the first such service is understood to be likely to be 'DirectTV', using its own system. This system will use DBS channels in the 11/12 GHz band to provide multiple channels of conventional quality (in a given transponder).

The system is developed as a 'closed' system, such that only the developers will be allowed to produce receiving equipment for the first two years of service. The system will use conditional access based on a smart card.

Informally, it is understood that the video coding system used for DirectTV is likely to be the so-called MP/ML option of MPEG-2 (see Section 6) . However, the system is likely to use its own programme identification system. Only partially successful efforts were made at the MPEG-2 meeting in July 1992 to make the MPEG-2 identification system compatible with the DirectTV system, as it was said that it was too late to change the latter.

### 4.3. The European situation

The studies of a digital HDTV-based terrestrial broadcast system continue in Europe, either on the basis of a multi-layer system, capable of meeting several different market needs simultaneously, and/or with mechanisms for graceful degradation, or as a single layer system. In both cases, provision is being sought to make the system reconfigurable, to allow several conventional quality channels, either with or without increased robustness.

The study of a digital satellite broadcast system, however, is now seen as of somewhat greater urgency in Europe, because of specific plans to begin operation of such services in 1995. Considerable effort is being expended to develop the main elements of a common digital satellite broadcasting system before the end of 1993. The core activity for the system will be to provide multiple conventional quality channels from a single satellite transponder, but provision is being made to also allow widescreen and HDTV, if there is a market demand for such services at some stage.

## 5. PROSPECTS FOR COMMONALITY

The single most important element in the issue of common digital television standards now appears to be the work of the ISO/IEC/JTC1 MPEG group.

Consideration of the views of the major actors in digital broadcasting in Europe, Japan, North America, and Australasia during April and May 1993 lead to the belief that the best hope for a common standard, or at least common elements of a standard, will be the universal acceptance of the MPEG-2 baseband coding and multiplexing system for digital broadcasting. This appears to be a willingness to do so, provided that the MPEG-2 system parameters are chosen not to conflict with broadcasters' requirements. However, there are problems and uncertainties, and though the situation is promising, success is not assured.

Furthermore, it is important to note that the MPEG-2 system will provide only the basis for commonality of Layers 7 (Application), Layers 6 (Presentation), 4 (Transport), and 3 (Network). The basis for commonality of Layers 5 (Session), 2 (Data Link), and 1 (Physical) will need to be found elsewhere.

The MPEG-2 group defined, in Spring 1992, the elements which will be used for a conventional quality MC hybrid-DCT codec, operating at between 3 and 15 Mbit/s. This codec is seen to form the focal point of a "matrix" of codecs which will provide a range of qualities and have a range of complexities. The MPEG matrix extends to include HDTV-quality systems, operating at up to 60 Mbit/s.

The matrix itself, and other requirements of the MPEG-2 system, are given in Appendix B to this report, which is from the MPEG-2 meeting in July 1993.

Appendix C to this report, which is a paper written by the author earlier in 1993, provides some further background on the issues of picture quality and bit-rate.

The MPEG group has also developed a packet multiplex system, and an identification and signalling system, which can be used with the codecs, and for a combined video and related sound channel.

The MPEG-2 matrix envisages three "profiles". These are called currently *simple, main and next.* Each profile takes a selection of bit-rate reduction techniques from a pool of possibilities. The tools used for the *main* level are all those used for the *simple* profile plus additional elements. Similarly the tools for the *next* profile are an extension of the main profile. The different profiles are said to have different 'functionalities'.

The major difference between the *main* and *next* profile is that the *next* profile allows the option of 'scalability'. That is to say, it allows the option of embedding lower quality bitstreams in a higher quality bit stream, in such a way that either the higher or lower quality signal can be retrieved and used at the same time.

Each of the three profiles has in principle four levels which each relate to scanning formats. These are currently termed '*low*' '*main*' 'HIGH/1440' and 'HIGH'. Each Level will be capable of operating over a range of bit-rates, and thus provide a range of picture qualities. Each 'level' starts from a different quality point as input signal, but the output quality, while having the same scanning format as the input, will depend on the particular bit-rate used.

Thus the MPEG-2 system is a family of related standards with an arranged degree of commonality and compatibility. Even if the all regions of the world accept to use the family, the degree to which receivers can be commonly used will depend on which parts of the matrix have been chosen. Different regions of the world, and different transports, will not necessarily use the same part of the matrix.

In Europe, for digital terrestrial television, interest in the WGDTB is currently focused on the HDTV-1440/NEXT system. This will allow, in principal, two levels of quality to be simultaneously broadcast and received, HDTV-1440 and MAIN.

In Europe, for satellite broadcasting, the multichannel conventional-quality system will probably use the MPEG-2 ML/MP video coding system. For digital HDTV broadcasting by satellite the situation is less clear. In principle there are six options: three degrees of complexity and two degrees of picture quality. These options will be evaluated in the coming months.

In Japan, the most likely outcome will be for the *high* level of the *next* profile to be used for wider band satellite broadcasting services. This is the highest available quality system, and will call for very large RAM size in the receiver, which will probably not be available for some years as a consumer product.

In the United States, the most likely choice for terrestrial ATV will be the *high* level of the *simple* profile. This will call for a much more modest amount of RAM in the receiver, and thus could be made available relatively quickly. *High level/simple* profile receivers will also receive and display *main level/main* profile pictures, which are likely to available from many other sources.

The Grand Alliance has certainly not yet made a commitment yet to use such an approach, but will consider the options over the coming months. Compliance will require them to modify, to what seems to the author to be a relative modest extent, the current form of the Grand Alliance system.

There are few prospects that either the 60Hz or 50Hz world would agree to adopt the other's field rate, but nevertheless it is possible that common receiver ICs could be made available. The MPEG-2 profile calls for systems which can work within upper bounds, which encompass the highest field rates, lines/picture, and samples/line.

In principle the scrambling and key management elements of a conditional access system could be the same for all Regions, and even for many different applications. However, this part of the system is not being considered in the MPEG-2 group, and it it is difficult to see what may be the vehicle for standardisation. Last December, the ITU/RS Task Group 11/3 invited Members to make proposals for a common worldwide conditional access system ("Worldcrypt") for digital terrestrial television. No submissions have yet been received.

In North America, the Conditional Access system to be used for DirectTV is not being proposed for standardisation. No system is being developed for digital terrestrial television currently in North America. No studies are known also in Japan. In Europe there is currently an *ad hoc* Group of the European Launching Group analysing prospects for European agreement on a common system, but at the time of writing this paper no active technical steps are being taken to develop a common system.

This may be something that we will live to regret. It seems certain that in Europe at least, Pay TV will be the only means available to finance further television services, given that the advertising and public licence cakes have a finite size and limit. If there is a common 'clear' system, but with a large number of incompatible application-specific conditional access systems, many of the economic benefits of commonality may not be realised.

The Data-link and Physical layers are beyond the scope of the MPEG-2 standardisation. Worldwide international standardisation will need to be undertaken in the ITU/RS TG 11/3 and WP 10-11/S.

Initial proposals have been submitted by European bodies for the use of a multi-carrier system, COFDM, for digital terrestrial television broadcasting. This concept is receiving serious attention in Japan. In North America, none of the proponent systems call for COFDM, but instead propose simpler types of modulation. There is universal recognition that a COFDM system would provide greater immunity from multipath distortion, and allow the operation of networks of transmitters all using the same frequency. However, in North America, there is no need for such networks because of the broadcasting structure. Television stations are based essentially on the town or area in which they are located, and are not allowed to provide service throughout large areas. Furthermore there is some skepticism as to whether the benefits of the multipath immunity will outweight the additional receiver costs of using COFDM. Nevertheless, the question of COFDM is being kept under review in North America.

Essentially the digital television broadcasting system will be a digital container, able to accept vision plus sound for television, or separate radio stations, or indeed other data services. In this circumstance, we are obliged to look also at the Digital Audio Broadcasting (DAB) system. DAB is a digital broadcasting system intended to work terrestrially, or by satellite, and capable of carrying radio stations or data in a flexible way. DAB uses COFDM for its Layer 7. This permits reception in cars under multipath conditions.

Twelve years ago, the MAC/packet and Digital Satellite Radio (DSR) system were developed in parallel, and although their data transport systems did very much the same things, there were small differences that made them incompatible. For example, DSR operates with a only marginally different clock rate to the MAC/packet system. It would be difficult to argue that either the MAC/packet system or the DSR system benefited from the small differences. Having a common transport system might actually have helped both, in a business sense.

We are approaching a similar situation again. Two systems are being developed separately. One is a flexible digital broadcasting system mainly intended for radio station broadcasting, and the other for television station broadcasting. However, there could well be a mutual benefit if both used the same transport system, because this would simplify the design of receivers intended to receive both services.

Currently, there is no enthusiasm on the side of the DAB developers, or on the part of those developing digital television systems to pursue the question of potential areas of commonality. This is a probably something that we will live to regret in the years ahead.

## 6. CONCLUSIONS

The best prospects for worldwide commonality lie in OSI Layers 7, 6, 4, and 3 for digital television broadcasting. There is a reasonable chance that there will be agreement to use an interconnected family of standards for these Layers throughout the world.

The prospects for a worldwide common OSI Layer 5 (Session/Conditional Access) are not striking, and positive action is needed to make anything happen in this area.

The prospects for worldwide common OSI Layers 2 and 1 for terrestrial broadcasting are modest, largely because the different broadcasting structures used in the USA and in other parts of the world place different demands on the technical solution needed. Nevertheless, the possibility should not be completely ruled out, and we should be able to understand the prospects more clearly at the Autumn 1993 meeting of the ITU/RS, both for terrestrial and satellite systems.

In general, the outlook is reasonable for some worldwide standardisation and a degree of interoperability. The prospects for standardisation certainly look at least as good as they have done for forty years.

# APPENDIX A

## A COMMON WORLDWIDE STANDARD FOR DIGITAL TERRESTRIAL TELEVISION - DREAM OR REALITY ? -


## ABSTRACT

*A Global Village needs a Global Television system. But agreeing durable common world standards for broadcasting systems has proved frustrating and difficult. The paper explores some of the issues associated with a possible common digital terrestrial television system. It looks at the lessons learned from the past and the widening dimensions of potential commonality. Examining the various elements that will make up the system, it suggests that a common international multiplex system may be a practical starting point for discussion. The difficulties of achieving common systems include different timescales and quality objectives.*


## 1. INTRODUCTION

Why bother with worldwide standardisation for Advanced Television ? Agreement within North America alone will be difficult enough, you might say. You have a large enough internal market, you can achieve your own economies of scale, without needing a larger market anyway. On top of that, worldwide standardisation in broadcasting in the last few years, could hardly be called a runaway success story.

However this may be true, there is no question that single worldwide standard for Advanced Television would be a golden apple. We surely owe it to future generations of broadcasters and viewers, at least, to examine whether we can try to grasp it for them.

The traditionally cited benefits of worldwide standardisation are lower costs and more choice for the public, and many other non-financial, cultural and morale-boosting benefits. But the other side of the same coin - what happens if you do NOT have a unique standard is also worth looking at.

If we look at the evolution of new broadcasting systems over the past ten years, a rather surprising (except with hindsight) pattern emerges.

Key milestones might be seen to be: first MAC in 1982, then MUSE in 1985, then HD-MAC in 1989, and now the soon-to-be-selected North American ATV system (1993). Let us assume hypothetically that the US system-selected will be a hybrid-DCT system.

Each new system has arisen in a different part of the world, after a three-to-four year period following the last system.

MAC's novel feature was the use of time-compressed component coding. MUSE had all this plus a new idea, a four-field sequence for high-definition stationary (or panned) pictures. HD-MAC had the component coding, the four-field sequence, plus a new idea; block-based adaptation using a-priori signaling, and motion compensation. The new digital ATV system may have all the above (if it is hybrid DCT) plus a new idea: direct digital transmission of the samples, which allows transform coding and subsequent variable-length coding.

In each case, the new system is, to some extent, a more sophisticated version of the previous system.

As an engineer working during this entire period, the author was convinced, at each juncture, that each system in turn (MAC, MUSE, and HD-MAC) represented the last word in new broadcast technology. In each case, he was wrong. Unfortunately, Providence has not given us the possibility of second sight.

However, it does seem reasonable not to make the same error of judgment four times in a row.

It does seem likely that if a system is developed today in North America, then later on in the nineties, in another Region (which may be Japan or Europe or jointly) it will be possible to develop an improved variant.

Even if this proves the case, it may not have to lead to completely independent systems. It does make sense to examine what may be done in terms of common standards. At worst, the feedback may well be beneficial for the internal North America discussions.

Just because a new system uses a greater amount of digital processing than a previous system does not, unfortunately, mean that it is the last word or final technical solution. It may be just one more stage in technical evolution. It is probably a mistake to assume there is nothing over the horizon, just because we cannot see beyond it.

The intention of this paper is not to provide the complete formula for worldwide commonality, but more modestly to make a first excursion into the subject. Hopefully it will encourage reflection of the subject.

## 2. WHAT LESSONS CAN BE LEARNED FROM THE PAST?

The NTSC system was developed in the early 1950s. The European economic climate made the transition to colour conceivable about ten years later. By then it was possible to devise a more refined colour system, (ironically) using the experience gained with NTSC. Since the basic scanning standards in use in Europe and in the 60Hz world were different, persuading Europe to adopt the NTSC colour system was a hard sell, and a 50Hz/60Hz world split seemed inevitable. More difficult to justify, a single system within Europe, was also not achieved. Alas, in different parts of Europe, different selection criteria were used: degree of backwards compatibility, studio-signal processing capacity, national prestige, etc.

On balance, useful elements demonstrated by this story may include;

- unless everyone chooses a standard at the same time, it is quite likely that latecomers will be tempted by technological advances and refinements, which experience with the first system has shown to be valuable,
- the system selection criteria used by all those choosing a system must overlap to some degree, if common ground is to be found, and a single standard is to be agreed.

In the late 70s and early 80s, a digital studio standard which had only two variants was agreed; CCIR Rec. 601. A totally unique system was not possible, but the standard was still seen as a major breakthrough. In principle, it allowed common 50/60 Hz switchable equipment to be made simply. The agreement was the result of a concerted effort by the SMPTE and EBU in pre-CCIR discussions. After a few years, further agreement was reached on a common tape format to carry the system. The standard however has not been universally used, and the major reason seems to be that it is too expensive.

On balance, useful elements demonstrated by these events may include:

- basic agreement needs to be reached by all the key actors before a common standard is taken to the CCIR;
- whatever else is part of the selection process, the cost factor has to be brought into the equation. In the end, goodwill is not enough. The product has to be saleable.

In the mid 1980s all the world's Broadcasting Unions agreed that efforts should be made to agree common HDTV production, point-to-point, and broadcasting standards.

Conceiving the first users of HDTV to be the motion picture industry, attention was focused at first, on a common HDTV production standard.

Initially there was talk of a common 80Hz interlace/40Hz progressive system, but the 60Hz world thought there would be little to be gained from this, and put forward the 1125/60/2:1 system as a unique worldwide standard.

The 50Hz world was unable to accept this. They argued that 60Hz-50Hz is nearly the most difficult standards conversion there can be, and there also seemed no major psycho-physical benefit to 60Hz, since in both cases, up-conversion will be needed in the home receiver in the years ahead. European Administrations proposed a 50Hz progressive standard as the single world standard. No agreement was reached, and this situation remains today. The discussion never really progressed to common point-to-point or broadcasting systems.

On balance, the elements shown by these developments may include:

- common systems must offer more advantages that disadvantages to each side if they are to be agreed, and equal benefit/equal misery solutions are needed;
- a dominant consideration may be the influence of the existing infrastructure.

## 3. THE DIMENSIONS OF COMMONALITY

It is not possible in this paper to cover all aspects of the common standards question. However, it may be helpful here to have a brief map of the territory, so that at least we know which part of the (standards) discussion we are in.

There are essentially three dimensions for potential commonality among image systems.

a. <u>The international dimension</u>. If all parts of the world (or large parts) used the same systems there could be well-known benefits to the user (lower costs, because of competition and the economies of scale, and/or greater choice among available services and to the receiver maker (large market size). The <u>international dimension</u> is thus that concerned with maximizing the use of a unique system in different countries.

b. <u>The application dimension</u>. Given that the number of products and systems which could use image coding is growing (broadcast, home disc and tape, still photography, facsimile, B-ISDN, etc.) there could be benefits if the same (or a related) system was used in each case.

The application dimension is thus that concerned with maximizing the adoption of a unique system in different applications.

c. The quality-level dimension. Over the years the possibilities for achievable picture quality have continued to rise; furthermore, picture quality requirements are linked with screen size and viewing distance. There is, and will continue to be, a need for systems operating at different quality levels. There could be benefits if the systems used, at each quality level, were related; for example, if images were always receivable in some form, or simply transcodable between levels. The quality-level dimension is thus that concerned with optimizing the relationship between the systems operating at different quality levels.

Overlaying all these dimensions is a new conception of the way that image systems should be developed in future, and which is being discussed in many international forums. This is as follows.

It is supposed that there will be a large number of uses for image systems. Bearing this in mind, the objective (according to some) should be to develop TOTAL systems as the combination of TWO elements:

- Generic elements. These are elements which can be common to all applications, such as (possibly) basic image coding, basic image format;
- Application-Specific elements. These are elements which are tailored to the particular environment or use, such as forward error correction.

It is important to be aware of the map above, if only so as to be able to follow the discussions. However, for the purposes of this early review of the situation, we will concentrate here on the international standardisation of a broadcast signal format.

## 4. THE OPTIONS FOR COMMONALITY IN BROADCAST ATV

In broad terms, the ATV system (if digital) can be considered as consisting of the elements in Table 1.

In the best circumstances, all of the elements would be common throughout the world. However, different planning and infrastructure environments exist in different parts of the world. Some of these are as follows.

In Europe the current terrestrial broadcasting plan operates with 7 or 8MHz channels, and a similar situation applies with cable networks. In North America, the current terrestrial plan operates with 6MHz channels.

The maximum bit-rate which can be transmitted in a given terrestrial channel depends on the type of modulation used, and the planning constraints (which define the potentially available received eye-height, etc.)

| | | | | Ease of agreement | Priority |
|---|---|---|---|---|---|
| 1. Modulation system | | | | | |
| 2. Multiplexing system | | | | | |
| 3. Baseband system | a vision | i | error correction | | |
| | | ii | image coding alg. | | |
| | b sound | i | error correction | | |
| | | ii | sound system | | |
| | | iii | sound coding alg. | | |
| | c data | i | error correction | | |
| | | ii | teletext system | | |
| | | iii | PDC system | | |
| | | iv | conditional access | | |
| | | v | other | | |

*Table 1: Elements of commonality for ATV modulation used, and the planning constraints (which define the potentially available received eye-height, etc).*

The planning circumstances which exist in North America are different to those in Europe. There is also a rather different broadcasting infrastructure. In the USA there are essentially a very large number of local stations, each serving a given community. In some senses they are usually independent entities. In Europe, the pattern is more usually of national broadcasters, each using a jigsaw puzzle of transmitters to achieve nationwide coverage. Furthermore, the protection ratios which apply in Europe are generally more stringent than those used in North America.

For reasons such as the above, it may be necessary to move our horizon to maximum commonality, rather than total uniqueness.

It may this be useful to study the above table and establish which elements will have the most implications for receiver costs and quality. Furthermore, it may be possible to establish the degrees of potential difficulty in achieving commonality in the various elements. In the above table, these columns are not complete. Hopefully the reader will be stimulated to include his/her own ideas. The author consoles himself for his inadequacy with Plato's maxim "The beginning is the most important part of the work".

To make a positive suggestion, one option is to begin with discussion of a common multiplexing system. It should be possible to design a unique system which can cope flexibly with present and future requirements in all parts of the world. From this, progress may be easier with other elements.

## 5. OBSTACLES TO STANDARDISATION

There are certainly many obstacles to standardisation in the other boxes of the above table. Different countries and different regions will have different selection criteria for choosing a system. If we are to complete the table, we need to understand what these are. Two of the most critical factors may be timescales and quality/ruggedness criteria. These are briefly considered below, although by no means exhaustively.

### 5.1. Timescales

The current timescale for the internal standardisation process and commencement of ATV services in North America is (understood to be) that a decision will be taken on a system in 1993, and that arrangements are also being made to strongly encourage services to start within the immediately following years.

In Europe, there is no unique or formal position on standardisation on introduction timescales for a digital terrestrial system. An enhanced PAL system has been developed, which could provide compatible widescreen services terrestrially from about 1995. The studies of digital terrestrial systems are only just beginning, and effectively have begun several years after North America. This probably means that Europe will be some years behind North America in starting services.

Of course it would be for the Japanese to explain their own timetable if they wished to do so. However, as understood, it seems that their introduction scenarios are not too dissimilar to those in Europe.

This timescale difference between Europe and North America may well present a problem as far as appropriate technology is concerned. However, the best way forward now could be not to prejudge the matter before discussions have begun.

### 5.2. Quality and ruggedness objectives

#### 5.2.1. The modulation system

In Europe, the Digital Audio Broadcasting system ('DAB') has been developed over the last three years. One of its key novel features is the use of modulation system termed COFDM (Coded Orthogonal Frequency Division Multiplex). The signal to be broadcast is shared in a defined way between a large number of simultaneously-transmitted closely-spaced carriers. When there is multi-path propagation, the effect in the frequency domain is that some of these carriers are reduced, and others are increased. The receiver processes the totality of the carriers, and the result is that the received signal quality is much less affected by the multi-path. This means that the requirements for receiver antennas are less stringent. This is a very attractive feature for in-car reception in DAB. It will also be attractive for a future European high-quality digital television service. We may need to ensure that noise or other impairments do not mask or remove the advantages of the extra definition available. There is clear European interest in using COFDM-type techniques for digital terrestrial television. This would need to be reconciled with North America criteria.

### 5.2.2. The baseband system

The picture-quality versus bit-rate curve is one which is notoriously difficult to define. It seems, to some extent, to be a moving target as technology evolves.

At the current time however, there is no evidence that it will be possible to develop a system with a better bit-rate to picture quality ratio than hybrid DCT with motion compensation and VLC (although systems which are more flexible or simpler may be possible).

Within this overall framework there are still 'adjustables' which will affect picture quality, for example; motion vector accuracy, type of VLC, etc. We cannot draw absolute conclusions at this stage, nevertheless, the following seems to be the general situation.

The laws of nature tell us that the lower the bit-rate, the poorer the picture quality. As the bit-rate is decreased, however, with this type of system, the predominant effect is to reduce the proportion of moving pictures which can be conveyed impairment-free, rather than to systematically impair all pictures. When a given information content is exceeded, mechanisms kick-in (albeit smoothly) which effectively add noise or lower the resolution. Lowering the bit-rate lowers the 'kick-in' point and increases the extent of the impairment. If, however, you start from a lower quality source (i.e. a source with fewer samples) you arrive at the equivalent kick-in point later.

The design game is therefore to choose the best combination of systematic (i.e. source system) reduction and coding system algorithm parameters for the best balance of picture quality and receiver costs.

It is always necessary to have an open mind. The results of the FCC Advisory Committee evaluations may indeed confirm the optimism of the many respected engineers who anticipate that HDTV picture quality will be achieved with bit-rates as low as 14Mbits/s. Currently, however, other evidence leaves many Europeans rather uneasy about this.

For example, the ISO/IEC JTC MPEG Phase 2 system evaluations in November 1991 examined a large number of digital codecs operating at about 4Mbits/s and 9Mbits/s. These simulations of codecs used a series of moving picture sequences from a 4:2:2 source. The test material was well chosen to explore the quality limits of the codecs.

What comes from the results is that 9Mbits/s seems enough to give 4:2:2 picture quality with material which is 'critical but not unduly so', with a well-designed hybrid DCT system. However, at 4Mbits/s there is a perceptible loss in quality (with this kind of material). This would mean that normal programmes would be occasionally impaired.

A useful shorthand way to describe the way a codec operates is the "bits/pel" ratio. This is the ratio of the codec's operating bit-rate to the source active pixel-rate. The active sample-rate of the 50Hz 4:2:2 system bits/s are thus about 0.39 and 0.87 bits/pel respectively. If the results of the MPEG tests are taken as some sort of guideline, we may conclude, in a source-independent way, that 0.8bits/pel can be effectively transparent to the source but 0.4bits/pel cannot. This might be called an occasionally impaired quality. The same kind of conclusions can incidentally be drawn from independent work made by the RAI and RETEVISION.

This is only a guide, because there will be some relationship between the degree of annoyance caused by an impairment and the total amount of information presented to the viewer.

We could use these data, however, for guidance to extrapolate the situation with digital terrestrial widescreen service.

For our digital terrestrial television system we could begin by examining the bit-rate needed for virtually transparent widescreen 4:2:2 quality. This would be 1.3x9Mbit/s = 12Mbit/s.

Virtually transparent reproduction of a 1152/1920/50/2:1 studio source (with 110592 M samples/s) would need 44Mbit/s. A 1152/1440/50/2:1 source (82944 M samples/s) would need 36Mbit/s. If we drop to 0.2 bit/pel, and our 'occasionally impaired system', then the above bit-rates would be halved.

It seems likely therefore that, in the range of available bit-rate for digital terrestrial television, which is likely to be up to about 24Mbit/s, there will be a difficult choice. Which is likely to be more attractive to the viewer? A system which is always impairment free, but has lower resolution, or a system which has occasional impairments, but a higher resolution, provided the picture does not contain more than a given amount of information.

This is a far from easy question. To complicate the argument, it is possible to up-convert lower resolution signals in the receiver, and in certain types of programme material such as feature films, at anything more than 3H, the difference will probably be difficult for the normal viewer to see.

The FCC evaluations themselves will certainly add important evidence and information to this issue, of value not just to North America. However, this paper was written before that evidence was available, and so for the time being we are not able to draw on it.

In the coming months it should be possible to put together a clearer picture of the relationship between picture quality and bit-rate.

## 6. CONCLUSIONS

Efforts to agree unique world standards in the past have met with varying amounts of success. It proves to be a very difficult business, complicated by different needs and requirements in different parts of the world. To complicate matters further, technology continues to increase its boundaries, making ever more sophisticated systems possible, at apparently regular intervals.

The histories of past standardisation sagas may give us some clues about what to do, and what not to do, for the maximum likelihood of success.

The commonality discussions are wider than just the international dimension, and include commonality across different applications and different quality levels. Initially, it may be more manageable to look at international standardisation.

One way forward is to examine the various elements which will make up a digital television system, and to establish priorities for commonality etc. <u>Opening a discussion on a common multiplex system may be a useful beginning</u>.

The obstacles to be overcome include potentially different conceptions about the type of modulation method to be used, and the degree of ruggedness needed. Furthermore, the optimum quality balance is not clear.

A major difficulty is also the different time scales which are sought for such systems in different parts of the world.

For all the difficulties and potential obstacles, standardisation is still worth the attempt. Thomas Jefferson said "I like the dreams of the future better than the history of the past". He would have tried.

## ACKNOWLEDGMENTS

The views expressed in this paper are those of the author. However, the studies reported were made by many groups, in particular EBU groups V1/RDB and V1/HDTV. Particular help was provided by Dr Marzio Barbero of the RAI.

# APPENDIX B

## INTERNATIONAL ORGANIZATION FOR STANDARDIZATION
## ORGANISATION INTERNATIONALE DE NORMALISATION

## ISO/IEC JTC1/SC29/WG11
## CODING OF MOVING PICTURES AND ASSOCIATED AUDIO

Source: Requirements sub-group
Title: Agreements on Profile/Level
Version: 10 AM, 16 July 1993
Status: Agreed

ISO/IEC JTC1/SC29/WG11 NO489
(Revised)

MPEG93/
16 July 3

### 1. Level parameters

Upper bounds are as follows:

| Level[1] | | Profile | | |
|---|---|---|---|---|
| | | Simple 4:2:0 | Main 4:2:0 | Next 4:2:2 |
| | Layer: | single | single | scalable |
| High (up to 60 Mbit/s)[4] | Pels/line | 1920 | 1920 | 1920 |
| | Lines/frame | 1152 | 1152 | 1152 |
| | Frames/sec | 60 | 60 | 60 |
| | Pels/µsec | 62,7 | 62,7 | 62,7 |
| High-1440 (up to 60 Mbit/s)[4] | Pels/line | 1440 | 1440 | 1440 |
| | Lines/frame | 1152 | 1152 | 1152 |
| | Frames/sec | 60 | 60 | 60 |
| | Pels/µsec | 47 | 47 | 47 |
| Main (up to 15 Mbit/s)[4] | Pels/line | 720 | 720 | 720 |
| | Lines/frame | 576 | 576 | 576 |
| | Frames/sec | 30 | 30 | 30 |
| | Pels/µsec | 10,4 | 10,4 | 11,06[2] |
| Low (up to 4 Mbit/s?) | Pels/line | 352 | 352 | |
| | Lines/frame | 288 | 288 | Not decided |
| | Frames/sec | 30 | 30 | " |
| | Pels/µsec | 2,53 | 2,53 | " |

Notes:
1. Level for the Next profile indicates the upper layer of the two resolution scales.
2. 720 x 512 x 30 has been considered to accommodate 483 active lines of 525/60 TV. The extension of the upper bound of the PEL rate to cover half SCIF is an open issue.
3. Multiples of 16 up to the upper bound are supported in number of PELs per line and number of lines per frame.
4. Data rates for other than MP @ ML are to be determined.

## 2. SPECIFICATIONS FOR NEXT PROFILE

The NEXT profile will support the following functionalities in addition to those of the Main profile:

| | Functionality | NEXT | Main |
|---|---|---|---|
| 1. | Chroma format | 4:2:2 and equivalent formats in other resolutions | 4:2:0 |
| 2. | Flexibility in bit-rates (range of bit-rates, CBR/VBR) | Independent control of bit-rate on each layer is required. | Yes |
| 3. | Random access/ channel-hopping | The same as main -the video decoder contributes only a part of the overall access time as seen by the viewer; thus no restrictions are set on the codec delay itself.<br><br>A desirable figure for video and audio delay[1] is less than 0.5 seconds | Yes, but not neccessarily at each frame |
| 4. | Editability | Yes as Main Profile | Yes, but not neccessarily at each frame |
| 5. | Resilence | "The same as Main;" in addition increased error resilence and graceful degredation[2] may be provided by the use of scalable coding. See §9. | Yes |
| 6. | Video windowing (eg 4:3 from 16:9 pictures) | 16:9 needs to be provided for both HDTV and EDTV transmissions.<br><br>For display on 4:3 (TV) receivers, signalling of the proportion of a 16:9 picture to be displayed is required.<br><br>Video windowing should be possible in each layer. | Yes |
| 7. | Low delay | The same as Main | Yes |

| | | | |
|---|---|---|---|
| 8. | Trick modes | Simple fast forward and reverse are required.<br><br>It should be possible that data rate in FF/FR mode be kept at less than normal mode rate | Yes, those automatically supported by the basic syntax. |
| 9. | Scalability (Hierarchical) | The following should be supported at High -1440 or High levels:<br><br>• up to 2 resolution scales<br>• up to 4 scales in total as combination of resolution scales and SNR scales<br>• up to 2 SNR scales at the upper resolution scale<br><br>At Main level:<br><br>• up to 2 resol. scales<br>• up to 3 scales in total as combination of resolution and SNR scales.<br><br>It shall be possible to decode single resol- ution scale as degenerate case.<br><br>Flexibility in formats at each layer (eg resolution ratio other than 2:1) is required for format compatibility and for letter-box.<br><br>Scalable technique with better efficiency than simulcast is required. | No |
| 10. | Compatibility | Forward compatibility with main profile is required. Backward compatibility with MPEG-2 (MP), MPEG-1 and H.261 is required to be possible when MPEG-2 (MP), MPEG-1 or H.261 is used for the base layer of multiple layers[3].<br><br>Downward/upward compatibility is inherent in scalable requirement. | Yes, but only MPEG-1 forward compatibility. |
| 11. | Quality | It should be possible to trade subjective picture quality for bit rate and coding complexity. See §2. | |

| | | | |
|---|---|---|---|
| 12. | Flexibility in Implementation | Encoder flexibility in the set of modes provided. M-value, in mode selection criteria, motion estimation, preanalysis, quantisation strategy, and other issues not being part of the standard.<br><br>Target of software decoding of the lower level is foreseen for IT applications | Yes |
| 13. | Copyright and Copy Management (new requirement) | The same as Main | To be contained in the Systems layer. |

Notes :

1. Time from reception of the first coded bit to the representation of a visually recognisable picture

2. Graceful degredation provides the ability for the progressive reduction of picture quality as the error rate in the bit-stream increases; The reduction in picture quality may be continuous or in discrete stages, and may be achieved by mechanisms such as reduction in picture resolution or in picture PSNR.

3. Backward compatibility may not be an important consideration in some applications, particularly if it affects any of the other requirements.
Forward compatibility with MPEG-2 MP@ML is required for conformance atthe corresponding level. Other levels may not require this functionality.
Forward compatibility with MPEG-1 constrained parameter is required forconformance at this level. Other levels may not require this functionality.

## 3. Specifications for the Simple Profile

Main Profile minus "B-pictures". Dual Prime is retained in the Simple Prime

## 4. Forward compatibility between different Profiles@Levels

"X" indicates the decoder must be able to decode the bitstream:

| Decoder \ Bitstream | NP @ HL | NP @ H14 | NP @ ML | MP @ HL | MP @ H14 | MP @ ML | MP @ LL | SP @ HL | SP @ H14 | SP @ ML | SP @ LL |
|---|---|---|---|---|---|---|---|---|---|---|---|
| NP@HL | X | | | | | | | | | | |
| NP@H14 | X | X | | | | | | | | | |
| NP@ML | X | X | X | | | | | | | | |
| MP@HL | X | | | X | | | | | | | |
| MP@H14 | X | X | | X | X | | | | | | |
| MP@ML | X | X | X | X | X | X | | X[1] | X[1] | | |
| MP@LL | X | X | X | X | X | X | X | X[1] | X[1] | X[1] | |
| SP@HL | X | | | X | | | | X | | | |
| SP@H14 | X | X | | X | X | | | X | X | | |
| SP@ML | X | X | X | X | X | X | | X | X | X | |
| SP@LL | X | X | X | X | X | X | X | X | X | X | X |

1. Note that SP@HL and SP@H-14 decoders are required to decode MP@ML and MP@LL bitstreams, and similarly SP@ML decoders are required to decode MP@LL bitstreams.

## 5. Coding tools not included in defined Profiles

There are some coding tools (4:4:4, 10/8 bit scalability etc.) which are defined in the syntax but not included in any Profile. These tools may be included in one or other Profiles which can be added in the future if found necessary, according to the existing procedures (eg in the form of Amendment). Another possibility is that their use may be defined in application standards.

The November 1993 CD includes only Simple, Main and Next Profiles.

## 6. Pixel aspect ratio of existing or planned formats

The following values should be included in the WD:

- 0.7500 1440x1080, 16:9
- 0.7826 1440x1035, 16:9
- 1.0435 1920x1035, 16:9

## 7. Open issues

1. Appropriate naming for Profiles and levels other than Main
2. Upper bound for the base layer of the Next profile in terms of resolution and bit-rates
3. Create a table which relates syntactic elements and parameter values with Profiles and Levels
4. Compatibility with SCIF video format
5. Bit-rate bounds for Profiles and Levels other than MP@ML

END

# APPENDIX C

## EUROPEAN PERSPECTIVES ON DIGITAL TELEVISION BROADCASTING-QUALITY OBJECTIVES AND PROSPECTS FOR COMMONALITY

This is an edited version of a paper first delivered at NAB HDTV World, Las Vegas 1993. It has also appeared in *EBU Technical Review*, Summer 1993.


## ABSTRACT

*In Europe, there are a series of collaborative projects developing elements of digital terrestrial and satellite broadcasting. Efforts are being made to encourage these projects to work towards a common standard for Europe. The paper outlines some of the proposals already made for an initial target system. Particular explanations are given of the current quality goals, and how they were arrived at. Other issues considered are coverage problems, commonality of terrestrial with satellite, common multiplexing, and conditional access.*


## 1. INTRODUCTION

North American was some years ahead of the rest of the world, particularly in its faith in what could be done with very high image-compression systems. The open decision-making process which was taken place in the United States for ATV is a considerable achievement, and a great credit to the many individuals involved.

In Europe, there has always been considerable expertise in image compression and digital modulation, but many of different factors and circumstances have influenced the profile given to digital terrestrial television broadcasting studies.

These have included pessimism that the planning environment in Europe would allow the development of digital high definition terrestrial television with reasonable coverage, and pessimism that sufficiently attractive picture quality could be achieved with the bit-rates possible terrestrially.

Today there is a clear recognition that we must press rapidly on, undaunted by the problems, to explore the potential solutions, because the prize will be considerable.

An international committee, the European Launching Group (ELG), has been established to try to coordinate those projects which are developing digital terrestrial television broadcasting, or indeed related systems, in Europe. This committee has a technical sub-committee, the Working Group on Digital Television Broadcasting (WG-DTVB).

Current projects in Europe include the following:

| | |
|---|---|
| RACE dTTb: | a major multi-national project, centred on the development of modulation systems appropriate for digital terrestrial television. |
| RACE HD-SAT: | a multi-national project, centred on the development of modulation systems for 20 GHz digital satellite television |
| RACE FLASH TV: | a multi-national project, centred on the development of digital HDTV satellite point-to-point systems |
| HD-DIVINE: | a Scandinavian project, developing all aspects of digital terrestrial television |
| HDTV-T: | a German project, developing all aspects of digital terrestrial television |
| EUREKA-VADIS: | a multi-national project, developing baseband coding systems |
| EBU-Working Parties V&R: | continuing pan-European studies of terrestrial and satellite planning, requirements, and testing |

The WG-DTVB has examined the aims of the current known collaborative projects, and their timescales, and looked at the potential uses of digital television broadcasting. It arrived, at the end of last year, at a work plan, intended to make it possible for Europe to achieve common standards for digital television broadcasting, in the next few years. The purpose of this paper is to outline some features of this plan, and to give background to the conclusions reached.

To develop a strategy for future broadcasting, you need to make a series of reasoned assumptions, and then plot the logical consequences of them. If, in the end, you find that what you can achieve is not sufficiently valuable, or practical, then you return to re-examine your original assumptions. What the WG-DTVB has done is to develop a first scenario which needs now to be taken up by experimental work.

A fundamental limitation on the quality and ruggedness of terrestrial television services will be the terrestrial channel capacity. In Europe, the VHF/UHF broadcast television bands use either 7 or 9MHz channels. The working assumption has been that we need a system bandwidth of about 7.5MHz. The prospects of using more than one channel in a contiguous way for a single broadcast service seem remote, and the prospects of obtaining new frequency allocations with a wider channel spacing, even more so. Given a 7.5 MHz channel, it seems that the upper bound on gross bit rate is likely to be about 30Mbit/s.

The first task the WG-DTVB undertook was to evaluate the options which seemed most likely to be attractive and saleable to the European Consumer in the next century, in the light of what we could see, or predict, as trends in society.

There is no doubt that the quality expectations of viewers are rising, and that the long term future of television lies with HDTV. Nevertheless, the group was also conscious that viewing habits are changing as society evolves.

Therefore, in setting system goals there are dimensions other than quality which need to be taken into account. It is not sufficient to ask what the public may want, we also need to ask when and where they will want it. Furthermore, that Holy Grail of a large flat-screen HDTV display seems nearly as far away as ever.

One underlying trend in society is toward individual activity, rather than group activity. A second element to consider is mobility. Essentially sound-radio has migrated from a group experience in the home, to a near-individual activity in the car. We could reasonably ask if some of the same evolution will apply to television to any degree, or at least whether television will also have to cope with a mobile environment.

There seemed to be four options, essentially linked to different viewing environments, which were worthy of most attention. These were as follows:

HDTV: (High Definition Television) services to viewers with very large screen receivers, using fixed rooftop aerials.

EDTV: (Extended Definition Television) services to viewers with medium to large screen receivers using fixed rooftop aerials.

SDTV: (Standard Definition Television) services to viewers with portable televisions using set-top aerials.

LDTV: (Limited Definition Television) services to viewers with small screen receivers using whip/stub aerials in a mobile situation (e.g. in a car)

In order to translate these concepts into practical systems, it is necessary to decide what is meant precisely by the quality in each case, and what is meant precisely by each of the receiving environments.

Picture quality is difficult to quantify in absolute terms, because it is the net effect of a series of factors such as resolution, sharpness, noise, artifacts, etc. It is by no means only related to the scanning standard. Picture quality achieved will be related to the source quality, the sophistication of the compression algorithm, and the bit-rate used.

The receiving environment can be defined somewhat more easily. It is related to the bit-error distribution in which the system is required to work. In other words, it is associated with the ruggedness necessary to achieve impairment free pictures of the intended quality. As a first assumption in the WG-DTVB, the rooftop environment is considered to be associated with a spectral efficiency of 4 bit/s/Hz. The portable environment is considered to need 1-2 bit/s/Hz, and the mobile environment is considered to need 1 bit/s/Hz.

## 2. THE DIMENSIONS OF PICTURE QUALITY

### 2.1. HDTV

High Definition Television is defined rather loosely by the ITU as a system which has about twice the horizontal and vertical definition of conventional television. This still leaves open the amount of noise or artifacts that are permitted, and which affect the picture quality just as much as definition. Furthermore, there is a relatively wide range of definitions available within the term "conventional television". In addition, systems with interlace have a triangular vertical-temporal response, so knowing where the "twice resolution" applies is difficult.

To pin down HDTV, we have to look at the combined effect of all the quality factors on the picture; and to some extent make up new rules.

When deciding on a required picture quality we have to bear in mind the target viewing distance, and the need to ask responsibly for no more than is necessary.

Digital compression systems all work in a similar way. The information content of the source picture varies from scene to scene. The system reproduces the content of the input picture essentially intact, until the point is reached where the transmission bit rate will be exceeded if nothing is done. At this point, a series of approximations are made to parts of the scene. The output scene can thus have (apparently) added noise or loss of resolution, to an extent depending on the original scene content.

For any practical system there will always be scenes which are reproduced perfectly, and others which are impaired. The design intention is to make the impairments occur as little as possible, and be as unobtrusive as possible.

## 2.2. Scene-content dependent failure characteristic

The first approach examined by the WG-DTVB to specify the quality needed is termed, the "scene-content failure characteristic". This is a logical and scientific method, but it is also relatively expensive to use.

The basic element to specify is the proportion of total programme time which should be free of artifacts. "Freedom from artifacts" is considered to be associated with a minimum mean score (12%) on a DSCQS subjective evaluation. This is somewhat arbitrary figure, but much experience shows it to be a good rule of thumb for virtual transparency.

The technical challenges are then, firstly, to decide what constitutes a sensible proportion of time for which impairment free pictures should be demanded. The second challenge is to assemble statistical evidence about the relative occurrence of different kinds of scene content, so that we are able to check if the requirements are met.

In choosing the proportion of time for which impairment-free pictures could be expected, we can look to the other "statistical" domain of picture quality, which is the propagation failure characteristic, used as a planning criterion. For example, in BSS satellite systems, quality is required to be maintained for a percentage of the worst month of the year. If this kind of guide-line is acceptable for satellite systems, would it also be acceptable for our terrestrial television broadcasting?

Unfortunately, the answer is "quite". In satellite broadcasting the "outage time" is used up in rain-fades, which occur over a period of, say, half-an-hour. The quantisation-noise artifacts introduced by digital coding will probably be more spaced out than this, and thus their effects will be less severe on viewers' overall conception of quality (This is sometimes called the "forgiveness effect"). However, it may be appropriate to start our scene-content, failure characteristic requirement somewhere near this value.

We have tentatively begun by using 99.7% transparency, as the requirement for the digital terrestrial HDTV service. Coupled with this, we make the assumption that the reference quality is a 1250/50/2:1 HDTV studio signal, with 1440 samples/line.

We do not yet have a catalogue of HDTV picture sequences and their places on a codec "criticality table", but we do have some experience from former 4:2:2 codec studies. These suggest that to achieve the transparency, the codec would need to pass unimpaired, almost all the test pictures so far devised, including the second most stringent CCIR sequence "mobile and calendar" (critical, but even so only in the area of 80%-90% criticality).

The quality target is very high, and may not be achievable at the available bit-rate. But it certainly is worth aiming high at the start. We know from past experience that HDTV source and display equipment quality will improve, and a system which will last well into the next century would be valuable.

The $64,000 question is what quality we can achieve with 20-30Mbit/s. Initial tests may be possible this autumn with the HD-DIVINE system and will provide the first clues.

## 2.3. EDTV

The second quality level targeted is termed EDTV, Extended Definition Television. This is not a particularly appropriate name, because the scanning standard for the system would be the normal 625-line system.

The level is included because large-screen HDTV receivers, which have an HDTV dot-pitch, will be very unwieldy and very expensive for many years to come. An EDTV level would fulfil a need for a lower cost and lighter receiver. Probably at screen sizes less than about 30", it would not be dramatically inferior to an HDTV display in perceived quality. There may also be living rooms which are not large enough to take a true HDTV receiver.

The source format for EDTV is assumed to be a CCIR Rec. 601 signal, with 720 samples per line and a 16:9 aspect ratio. The domestic display would probably be subject to upconversion.

The codec transparency required, in terms of the percentage of programme time unimpaired, would be roughly the same as for the HDTV level (although in this case with respect to the 4:2:2 source).

The best information we have to date is that in order to achieve this level of transparency, probably about 9-11 Mbit/s is needed for an MC hybrid DCT system.

## 2.4. SDTV

The third quality level considered is SDTV, Standard Definition Television. This is specifically intended to be matched to the quality needs of portables.

On small-to-medium screen sizes, even today's PAL/SECAM quality is very good. Thus, for our SDTV level, we need a system which has a 625-line scanning format, but we can accept some artifacts, as is the case for PAL and SECAM.

The kinds of artifacts associated with PAL/SECAM and with a digital MC hybrid DCT system will be different, but we believe that to achieve, globally, about the same overall quality, about 5-6 Mbit/s data rate is needed.

## 2.5. LDTV

The fourth quality level is LDTV, Limited Definition Television. This is intended to match the needs of very small screen receivers, which might be used in cars, or as for personal, portable reception.

It has to be said that broadcaster members of the group are not yet convinced that there would be a need for such a level, but it has certainly been included in the discussion. The quality requirements of this level would be about the same as the MPEG-1 codec or about VHS level. This means about a quarter of the resolution potential of CCIR Rec. 601.

Specifying the quality requirements, and evaluating the systems in terms of scene-content failure characteristics will be a major technical challenge, principally because of the need to establish how often scenes of a particular type of content are likely to occur. There may be alternative simpler approaches which will also help to understand and quantify the systems' behaviour. One such approach is outlined in the next section.

## 2.6. The concept of quality space

Another potential quality evaluation criterion, which the WG-DTVB group has been asked to consider, is associated with the concept of "quality space";

Our perception of the picture quality of a given system is directly influenced by our viewing distance. The further from the screen, or the narrower the viewing angle, the less discriminating we are in terms of resolution or artifacts.

One way, therefore, to see the various quality levels, is by imagining that there is a "quality space", which is a graphical representation of picture quality-versus-viewing distance. For the picture quality axis, we use the same axis as for DSCQS evaluations. This is five contiguous equal intervals characterized by the quality descriptors: excellent, good, fair, poor, bad. For the viewing distance axis, we used multiples of picture height as markers 3H, 4H, 5H, 6H, 7H, and 8H.

In this quality space we assume an unimpaired reference in each case, and define an EDTV system as one for which the mean results of all assessments must fall within the "excellent" band at 3H. Similarly, we specify an HDTV system as one for which the results of assessments must fall in the excellent band at 4H. SDTV systems are those for which the results must fall in the excellent band at 6H, and an LDTV system as one for which the results must fall in the excellent band at 8H.

This seems a relatively clear means of defining and distinguishing between the quality levels, but experimental work remains to be done to establish its viability in practice.

## 2.7. The impact of source scanning parameters on overall quality

Another interesting dimension to this question of picture quality concerns the real impact of source quality on final picture quality.

Compression systems may show a characteristic whereby there could be considered (in a simplified way) to be two regions within their characteristics of quality-versus-bit rate. In the first region, which extends down to a given bit-rate, the codec is essentially transparent.

In the second region, beyond a kind of knee in the quality/bit rate curve, there is a progressive increase in artifacts as the bit rate is reduced.

The point at which the knee begins is related to the "overload" level of picture detail. If there were two systems, one 1250-line and the other 625-line, the quality associated with the flat region of the characteristic for the 1250-line system would be a grade or so above the flat region of the 650-line system. However, the knee of the 625-line system would arrive later, at a lower bit rate, because there is less picture detail to handle. Since the roll-off of the 1250-line system curve is relatively steep, it is possible that the two will cross, making the quality achievable with the 625-line system higher than with the 1250-line system, over a given range of (low) bit-rates.

With very-high-compression systems, which are actually operating on their quality roll-off, there may thus, surprisingly, be better results with lower resolution sources that with higher resolution sources. The loss in resolution may be less annoying than the artifacts introduced. We need to resolve this issue when experimental systems are available.

## 3. MULTI-LAYER SYSTEMS

### 3.1. The concept

Having established the desired quality playing field, the WG-DTVB moved forward to develop the concept of a multi-layer system.

Bearing in mind the potential of modern compression methods, is it possible to devise a system that will allow the reception of several quality layers simultaneously ? If this were possible, there would be considerable benefits for the size of the service's market.

There are, in theory, two approaches to multi-layer systems. The first, and simplest, is to broadcast the services separately side-by-side. This is termed "multi-cast", in the group's vocabulary. The second is to embed one layer within a second, and this is termed "hierarchical coding" in the group's vocabulary.

Each of the layers has to be arranged to have its own baseband coding elements and its own modulation elements, which give the required ruggedness or spectral efficiency.

Preliminary calculations suggest that taken overall, the multi-cast and hierarchical approaches will be about equally efficient for a two-layer system, but for a three-layer system, the hierarchical approach will be more efficient. This remains to be verified experimentally.

It would be foolish to deny that as it is going to be difficult to achieve the desired high quality in a stand-alone system, it will be even more difficult in a multi-layer environment. When all the evidence is in, we will have to weigh up all the trade-offs involved.

### 3.2. The WG-DTVB terrestrial "strawmen"

The WG-DTVB has drawn up a set of "strawman" multi-layer systems which it proposes should form the basis of studies for the projects associated with digital terrestrial broadcasting. These are explained more fully in the accompanying paper by Prof. Ulrich Reimers.

The first proposal is a three level multi-layer system which would simultaneously allow reception at HDTV, EDTV and SDTV, and be reconfigurable to two EDTV services. If the three-level multi-layer system proves impractical, a two-level system should be investigated. The second proposal is a three-level multi-layer system which would simultaneously allow reception of EDTV, SDTV, and LDTV.

## 4. Coverage

It would be a mistake to minimise the problems and uncertainties associated with terrestrial planning in Europe. Transmitter density is massive, and traditionally, broadcasters have achieved national coverage for their programmes. It appears from a first analysis that only in countries on the periphery of Europe (geographically speaking only) will it be possible to repeat the kind of coverage achieved today with PAL and SECAM with a digital HDTV service.

Critical studies still need to be made, based on hard data for protection ratios, and it seems possible that we may be faced with a difficult choice between providing HDTV coverage to a part of the population, and offering perhaps lower quality services to a greater part of the population, or some compromise.

The difficult planning situation is one of the main reasons why the use of COFDM as a modulation scheme for terrestrial services seems very important for Europe. It will make reception more immune from multipath, and it will allow the development of single-frequency networks.

## 5. CONSIDERATIONS OF COMMONALITY

### 5.1. Commonality with satellite systems

The CCIR agreed in 1992 that it was even more important to achieve common standards between satellites and terrestrial services in a given country, than to achieve international standardisation of terrestrial or satellite systems in isolation.

In the end, what matters most is how many pieces of equipment the consumer needs to allow him to receive the services available. International viewing is important, as are the economies of scale which an international system would bring. But more important, at the level of Mr and Mrs Joe Public, is having only one receiver for all his national services.

This is very much recognised by the members of the ELG and WG-DTVB, although everybody realises that there may be difficult commercial hurdles to overcome to achieve a common system for satellites and for terrestrial systems. Nevertheless, we will give the concept our best efforts.

At present there is serious interest in Europe in bringing into service digital multi-channel satellites around 12GHz. A number of programme providers are studying systems somewhat like the DirecTV concept. Our ideal would be to bring such systems into a family which encompasses terrestrial digital systems, and future HDTV services by satellite. This would be in the best interests of the public.

The WG-DTVB has prepared a further "straw man" which would provide a compatible bridge between digital satellite and digital terrestrial services. This is a satellite system which would be developed from the same tool-kit as the terrestrial system.

In the satellite case, there would be two key differences. Firstly, the usable system bit-rate would be about 45Mbit/s, and this could be used with any of the currently foreseen transponder bandwidths. Secondly, no attempt would be made to use systems with very low spectral efficiency, to help portables or mobiles, because the propagation conditions from satellites in the 12GHz region or higher would probably rule them out.

## 5.2. Multiplexing

A major step forward to achieving common systems, both between different types of transport and internationally, would be to develop a common multiplex system. This prospect is being seriously investigated in the WG-DTVB group. An analysis of the options available for a common multiplex is currently being made.

There seem to be a number of potential candidates for a common multiplex, and particular attention is focused on the MPEG proposals and the DAB system. A unique system for DAB and DTVB seems particularly attractive.

## 5.3. Conditional access

It is clear that the only way to finance many of the new digital services will be by Pay-TV using Conditional Access. Standardisation of Conditional Access systems has proved very difficult in the past. Some operators see the use of a proprietary Conditional Access system as a means to secure brand loyalty. Whilst this is true, the greater public good can be argued. It is unfair and unreasonable to ask the public to use a multiplicity of different decoders, all of which do very much the same thing in the same way. A more reasonable course may be to call for a common decoder system with a smart card interface. To use different services the viewer would just change smart cards, which would be obtainable from the programme provider and could include proprietary software, etc. This may be a way to balance the interest of the public, and the needs of the programme provider. Competition between programme providers on the basis of their programme content is the way to raise standards for everybody. Using technology differences to prevent people seeing the competition will not raise programme standards and certainly will raise costs for the viewer.

## 6. Conclusions

The European ELG and WG-DTVB have tried to commence their task, the coordination of digital television standards in Europe, with imagination.

The "strawmen" proposed are extremely ambitious, but it is right to begin the development of a television system for the next millenium in such a way. We have learned only too well in Europe that whilst we cannot give you the formula for success, we can give the formula for failure: try to please everybody all the time. If however the work continues as it began, with good will, cooperation, and pragmatism, it will succeed.

## ACKNOWLEDGEMENTS

The developments explained in this article were undertaken by the members of the Working Group on Digital Television Broadcasting. Their work, under the Chairmanship of Prof. Ulrich Reimers, is acknowledged.

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## *PART 5 - THE EUROPEAN GROUP FOR DIGITAL VIDEO BROADCASTING*

Chapter 7

# Report to the European Launching Group on the prospects for digital terrestrial television

Prof. Dr-Ing. Ulrich Reimers and others,
The Working Group on Digital Television Broadcasting

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# 1. INTRODUCTION

The 'Working Group on Digital Television Broadcasting' (WGDTB) was formed by the European Launching Group (ELG) for Digital Video Broadcasting in April 1992. The Terms of Reference of the WGDTB (Appendix 1) essentially require the group to propose directions for digital terrestrial television studies for Europe and to identify the research, development and preparatory work needed, to realize future systems.

Twenty-two specialists from seven countries cooperated in the activities of the WGDTB (Appendix 2) under the Chairmanship of Prof. Dr U. Reimers (NDR/Germany) and Vice-chairmen Dr I. Childs (BBC/United Kingdom) and Mr D. Sauvet-Goichon (TDF/France). Mr D. Wood (EBU) was Secretary of the group.

Their report, submitted to the ELG, is the result of many hours of work by all the members of the WGDTB. The group held three meetings, 25-26 June 1992 in Hamburg, 27-28 August 1992 in Kingswood Warren, and 3-4 November 1992 in Stockholm.

The Chairman of the group expresses his warmest thanks to all members, not only for their active work, but also for the positive atmosphere of pan-European cooperation that was experienced in the group[1].

# 2. ANALYSIS OF THE SITUATION TODAY

## 2.1. Digital video in the non-broadcast world

Image coding has made remarkable progress during recent years, and this opens up a large field of potential applications, with picture formats ranging from video phone to HDTV. This progress has become possible by the rapid development of VLSI technology, which allows the implementation of complex coding algorithms. The applications of image coding systems may also include printing, still photography, computer systems and many other areas.

Coding algorithm development can roughly be divided into two main fields: still picture coding and moving picture coding.

In still picture coding, mainly DCT based algorithms such as JPEG (see below) are used, although for some applications such as medical archiving, lossless schemes based on predictive or interpolative coding in connection with entropy coding are utilized as well.

Among the different techniques developed during the last few years for moving pictures hybrid-DCT, or hybrid SBC (sub-band coding), with motion compensation can be regarded as the most powerful techniques, yielding the highest compression rates. However, as far as practical implementations and standardization are concerned, mainly hybrid-DCT based systems predominate.

Image coding is already used in many areas, and new application fields are being added with progress in implementation, recording and transmission technology.

1 This report was first published in November 1992 with the reference number WGTDB 1063

Three main application fields outside digital broadcasting can be identified, although a clear distinction between them is not always possible:

- image communication,
- multimedia,
- television-studio.

### 2.1.1. Image communication

Classical applications in image communications are inter-personal communication via video conference and videophone. The CCITT standard, H.261, for video transmission with n x 64kbit (n equal or less than 32), which fits into the first level of the CCITT hierarchy (2Mbit/s), is one of the most developed coding standards for these applications. It uses the so-called common intermediate format (CIF) as video input format, and which provides about a quarter CCIR Rec. 601 spatial TV resolution, but reduced temporal resolution. The H.261 algorithm is a classical hybrid coding scheme, and several hardware codecs using this standards are now available on the market. The benefit of this standard is that the customer can select the data rate; and, with it, the image quality and the transmission costs, according to his requirements *[1]*.

The ISO/IEC/1172 standard 'MPEG-1' can be regarded as advance over the H.261 standard. Although originally designed for interactive CD-ROM based storage media with a data rate of about 1Mbit/s, MPEG-1 codecs can also be used for image communication. The main benefit of MPEG-1 is the fact that it represents a so-called generic coding scheme, where just the transmission protocol and the decoder architecture are standardized. So it is up to the user whether he wants to employ the MPEG-1 scheme in a simpler and cheaper form or in a more complex way. Furthermore he can select the coding parameters according to the requirements of his specific application (e.g. storage media or transmission) *[2]*.

MPEG-1 uses also a motion compensated hybrid-DCT and the advance in performance over H.261 has been obtained by using motion compensated interpolation in addition to motion compensated prediction. However, this increased performance has to be paid for by an increased number of required frame stores.

MPEG-2 is an advancement over MPEG-1 by using higher data rates (up to 10Mbit/s for a 4:2:2 input). The current main target (at the time of writing) is to provide an image quality subjectively transparent to the CCIR 601 studio standard, and the envisaged applications range from communications over interactive storage media, networked data base systems, serial storage media (VCR), to broadcast applications over cable networks, terrestrial broadcasting and satellite broadcasting *[3]*.

During the Rio meeting of MPEG in July 1992 the 10Mbit/s limit was removed and video formats higher than CCIR 601 are now under consideration. This means that MPEG-2 will, in principle, be usable for HDTV as well.

The standard is not yet finalized, but it is most probable that it will be an extension of MPEG-1, optimized for interlaced input formats, which means that the algorithm will again use motion compensated hybrid-DCT.

Television contribution networks can also be regarded as another form of image communication. The CCIR standard 723 or the ETSI (European Telecommunications Standards Institute) standard ETS 300 174 for contribution circuits 34Mbit/s is now the established format for studio to studio transmission of TV-signals in component format (4:2:2) and equipment is being installed in this format. The ETSI codec uses a motion-compensated hybrid-DCT, which is optimized for this data rate. Suitability of the algorithms for providing contribution quality has been demonstrated by a variety of tests (carried out by the EBU) within CMTT *[4]*.

HDTV contribution at 140Mbit/s according to the fourth level of the PCM-hierarchy is a rather new topic, for which no standard is yet available. However, there are already two different HDTV codecs on the market which are based on the ETSI codecs. One of the codecs has been developed in the RACE project HIVITS, and uses 6 ETSI (TV) codecs, working in parallel on six horizontal stripes of the picture *[4]*. The other proposal comes from the EUREKA 256 project which uses 4 ETSI (TV) codecs working in parallel on four vertical stripes of the picture *[5]*. The performance of both approaches is almost the same, and their suitability for contribution purposes has been demonstrated within the EUREKA 95 project *[6]*.

### 2.1.2. Multimedia

Multimedia applications with workstations or PCs have gained importance during recent years. This is the result of the availability of new transport media such as ISDN, and the progress in storage media such as Winchester disc drives and CD-ROMs.

An important coding algorithm for storage and transmission of still pictures, the ISO/IEC standard 10918 known as JPEG, has been standardized by the ISO *[7]*. This algorithm, which is DCT-based allows the coding of images with a compression rate which is content-dependent. Progressive build-up of the picture, with increasing resolution while decoding, is provided in order to allow fast visualisation of the stored or transmitted pictures. Meanwhile, JPEG chips and interfaces with JPEG codecs for PCs and workstations are available on the market *[8]*.

However, multimedia applications do not only combine computers with still image compression; moving image representation for videophone, multi-point teleconferencing and interactive video via CD-ROMs are also of importance. This means that JPEG, H.261 and MPEG coders/decoders have to be provided in one terminal.

As all these algorithms are DCT based, new chip generations will support all three standards - JPEG, MPEG-1 and H.261. First samples of these chips are already available on the market.

### 2.1.3. Television studio equipment

While video coding has been accepted widely for the above mentioned application areas, this technique is rather new in the broadcast field, including in the studio.

In the studio, image coding is mainly applied to contribution circuits and recording. While contribution systems with data compression are already used, coding for recording is a very new technology. As examples, Sony (1/2", Digital Betacam) *[9]* have announced and Ampex (3/4", Digital Component Technology - DCT) *[10]* have launched, data reduced digital VCRs with a compression factor of 2. Although few technical details are available, both recorders will use intrafield DCT.

While the introduction of image compression in television-recorders is mainly for economic reasons, the use of this technique for cassette-based digital HDTV-recorders is a necessity, because only by applying this technique can satisfactory recording times and robust operation (e.g. insert editing) be obtained for the largest cassettes available today.

Reduction factors between 2 to 4 can be obtained without visible artefacts using intraframe DCT schemes, depending on the application (2 for full post-production capability, 4 for limited post-production and archiving). In this case frame-by-frame editing and visible search can be achieved *[11]*.

If the two latter requirements are not necessary, for example if exclusively archiving with limited editing capability (e.g. cut is possible only every 10th frame) is needed, then MPEG-2-type algorithms can also be used, leading to very inexpensive recorders.

The use of computer workstations for post-production, using the JPEG algorithm, is also becoming more common.

### 2.1.4. DAB-Digital Audio Broadcasting

#### 2.1.4.1. Project organization

The DAB system is being developed by the Eureka 147 project, in collaboration with the EBU. Phase I of the project from 1989 to 1991 involved 360 man-years, and about 80MDM capital expenditure. The phase II, from 1992-1994 will involve about 170 man-years and 45 MDM.

#### 2.1.4.2. Background

The DAB system is designed to provide a high quality, multi-service digital radio broadcasting system for reception by fixed, portable and mobile receivers. Operation is required in the VHF and UHF range for terrestrial, satellite, and hybrid terrestrial/satellite delivery. Nevertheless, DAB is designed as a flexible integrated services digital broadcasting (ISDB) system supporting a wide range of sound coding options, sound program associated data, and independent data services.

#### 2.1.4.3. Technology

In both channel coding/modulation and audio source coding, sophisticated solutions have been developed. The audio coding part of the DAB system consists of the MUSICAM standard, recently approved as ISO DIS 11172-3 Layer-II. Compact disc quality is maintained at a bit-rate of 256kbit/s (a CD needs 14Mbit/s) and high flexibility is offered by different possible bit-rates between 32 and 384kbit/s.

The channel coding part consists of COFDM, which stands for Coded Orthogonal Frequency Division Multiplex. COFDM employs a wideband multicarrier transmission scheme, with convolutional error protection coding associated to soft decision Viterbi, decoding, in conjunction with frequency and time interleaving. The system concept uses a guard interval to overcome multipath propagation distortions. This will allow the use of single frequency networks, where all stations broadcast the identical ensemble of programmes on exactly the same frequency channel. DAB, as specified for its introduction, accommodates six high quality stereo programmes in a system bandwidth of 1.5MHz.

#### 2.1.4.4. Standardization specification

The technical development of the DAB system is taking place within the project EUREKA-147. The specification of the system, in order to prepare standardization and introduction, is a collaboration between EUREKA-147 and the EBU. The process of standardization started in ETSI (European Telecommunications Standards Institute) at the beginning of 1992, and it will be finished about Spring 1993. The system specification is designed in conformity with the flexible and broad ranging system requirements developed by the EBU and incorporated in the CCIR Draft Recommendations.

#### 2.1.4.5. Introduction of first service

Within the JESSI-DAB project (Joint European Submicron Silicon Initiative) the DAB specification is being implemented in ICs. The first chip-set is expected for early 1993.

The current proposal is that there should be an introduction period of about 10 - 15 years when DAB is broadcast in parallel with VHF/FM. A so called "parking position" is required for DAB, and the frequency channels most likely to become available for DAB in central Europe are TV channel 12.

Germany in particular is concentrating on channel 12, and by mid 1995 it is planned to remove all low-power television retransmitters assigned to this channel.

One TV channel covers 7MHz, and thus 4 blocks of 1:5MHz each (with six high quality programs per block).

After the phase of introduction, in approximately 2010 VHF operations will decrease continuously, and the additional broadcasting capacity will become available for use for digital audio broadcasting and other communication services.

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## 2.2. Situation outside Europe

### 2.2.1. Terrestrial HDTV transmission in the USA

In 1990, the Federal Communications Commission (FCC) ruled that only non-compatible systems would be considered for terrestrial transmission of HDTV. Transmission was to take place in taboo channels, simulcast with existing NTSC services. Four different digital systems were developed, all of them based on digital video and audio compression, channel coding, and modulation techniques:

| | |
|---|---|
| **Advanced Digital Television (ADTV)** | Advanced Television Research Consortium (ATRC = Philips NA, Thomson, D. Sarnoff Research Center, NBC, Compression Labs) |
| **Digital Spectrum Compatible HDTV (DSC)** | Zenith/AT&T |

| | |
|---|---|
| **Digicipher** | General Instrument Corporation |
| **Channel Compatible Digicipher (CCDC)** | Massachusetts Institute of Technology (MIT) / General Instrument Corporation |

A test facility (the Advanced Television Test Center - ATTC) was created to assist the FCC Advisory Committee in its choice of system. Testing procedures were defined and prepared by Sub-committees of the Advisory Committee for Advanced Television Systems (ACATS). Hardware prototypes of the four digital systems were tested one after the other, the last of the four systems is under test now. The winning system, in principle, will be chosen in January 1993.

The following analysis is an outline of the characteristics of the four proposed digital HDTV systems for the US, based on the information made available by the proponents for the ATTC test procedure. Starting with an outline of the principle common points, an explanation of some specific differences will be given.

### 2.2.1.1. Systems overview

All transmission systems consist of the same basic functions. These are source coding, formatting and error correction coding, channel coding and modulation, as shown in the block diagram of Figure. 2.1.

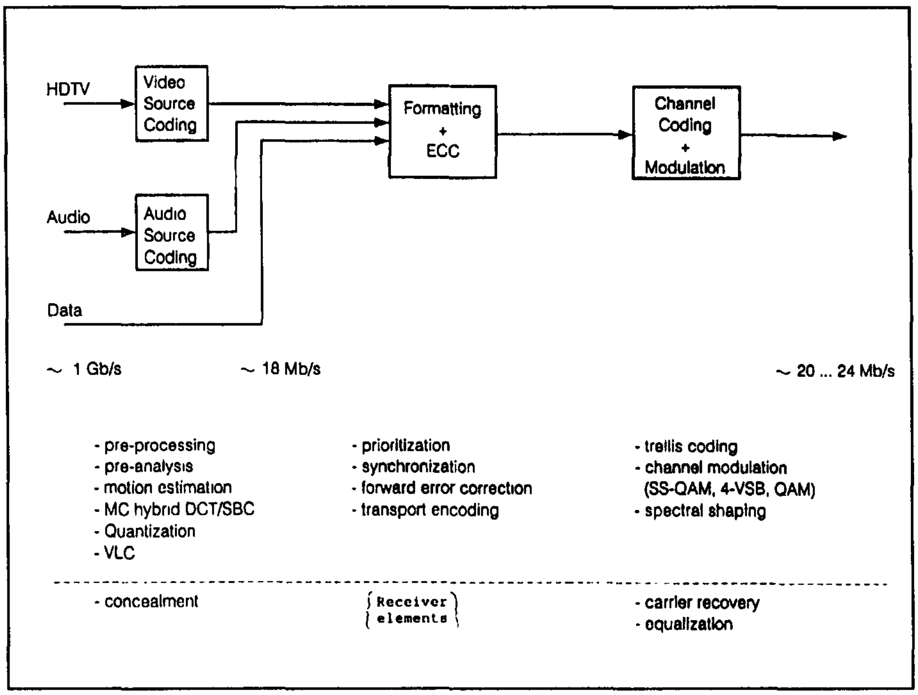



*Figure 2.1: Basic functions of the HDTV transmitter for the US systems*

The source coding has to reduce the data rate of the digitized HDTV signal from about 1Gbit/s to less than 18Mbit/s. All of the proposed systems use motion compensated (MC) hybrid DCT coding for this task, differences are given in the parameters of the various approaches, as well as in the implementation of the coding methods.

Formatting is the task of organizing the compressed video and audio data in a transport scheme, which provides for synchronization, data prioritization and forward error correction.

In the channel coding and modulation part of the systems different digital modulation techniques are used, in order to achieve an optimum transmission capacity, and to accommodate the data to the properties of the taboo channels. Modulation is done in conjunction with Trellis coding techniques, which allow for improved data recovery techniques in the receiver. After transport encoding and channel modulation, the data rate of the proposed HDTV systems is in the range of 20-24Mbit/s.

At the decoder, inverse operations are carried out: demodulation in conjunction with channel equalization, carrier recovery and Viterbi decoding, demultiplexing of the data and error correction, video and audio decoding, including error concealment techniques.

More details of the systems are given in Appendix 3 to this report.

#### 2.2.1.2. Conclusion

The four HDTV systems proposed for digital terrestrial transmission in the US reveal commonalities as well as significant differences. However, calculated, as well as some already demonstrated, performance figures indicate some convergence. In a 6-MHz channel, disturbed by NTSC interference and subject to multipath and other noise sources, not more than about 24Mbit/s total data rate can probably be transmitted, which leaves less than 19Mbit/s for HDTV signals, after audio, data and FEC data have been allocated.

All four systems have been demonstrated and tested in first hardware prototypes, with CCDC being under test in the ATTC now (Autumn 1992). In the ATTC, the systems are subjected to various moderately critical and normal source video sequences. Encoding and decoding is performed through transmission test bed conditions, which allow for a wide range of parameter settings, as well as noise and interference settings. Performance levels are measured and verified. In addition, subjective quality ratings are derived for a variety of transmission issues at the ATEL in Canada.

Field trials will be made next Spring and Summer of the selected system. Digicipher and DSC have been broadcast, organized by GI and Zenith, respectively. For ADTV, the ATRC has organized field tests with NBC. These are currently conducted in the Washington area.

Only the Digicipher results have been published so far by the ATTC, but it can be expected that further improvements of all systems will be developed. These will help to reduce particular weaknesses. They are needed to consolidate the performance levels for adverse conditions of channels and source signals, and will probably lead to slight improvements.

As the four digital system proposals have been originated 1-2 years ago, it is no surprise that alternative approaches are already being considered. Currently, Spread Spectrum modulation techniques are being discussed as an alternative to the QAM systems. It seems possible that one or other system could even adopt an OFDM modulation for reasons of improved spectrum efficiency, as well as better robustness against multipath and interference.

Another area of change could be source coding. The advent of the MPEG-2 standard, covering such issues as hierarchical HDTV/standard resolution coding, as well as graceful degradation, could lead to convergence on source coding with digital cable and satellite TV systems.

### 2.2.2. Current situation in Japan

Japan is the birthplace of High Definition Television. The MUSE HDTVsatellite broadcasting standard was designed in the early 1980s, as a means of broadcasting the 1125/60/2:1 scanning standard developed for programme.production. The basic MUSE system has a base-bandwidth of between eight and nine megahertz, and provides approximately 4 to 1 bandwidth compression by using a four-field sequence for stationary or panned scenes. These same compression techniques are used, in a somewhat more sophisticated way, in the HDMAC system.

The basic MUSE system was intended for use in 27MHz DBS channels. Fromthe late 1980s until 1991 there were many experimental broadcasts via the BS2 satellite. In 1991 an service of eight hours per day was launched.

MUSE receivers have been available for some years to the public but at very high costs, and with relatively low screen brightness. The expensive display tubes probably constitute the largest part of the cost, however they do provide a dot-pitch which can do justice to the MUSE potential static resolution. Recently, tubes have been made with gradually coarser dot pitch moving towards the periphery of the screen. This gives higher effective brightness to the tubes. This, combined with falling costs, may provide accelerated market growth for MUSE receivers.

For terrestrial broadcasting, the Clearvision system was introduced about three years ago. This provides sharper 4:3 pictures for viewers with the special Clearvision receivers. This is largely provided by improved receiver processing, and the use of enhanced signal processing in the studio. Programmes are broadcast regularly by all broadcasters in Clearvision, but very high receiver costs have limited sales of receivers.

The next phase of Japan's plan for improvements to the terrestrial television services is to develop a widescreen letterbox enhanced NTSC system, DTV. This should be finalised in 1993/1994. NHK developed an ingenious narrow band version of the MUSE system prompted by requests from the USA for a terrestrial HDTV broadcasting system. This system has continued to be a candidate in the FCC evaluations for a future US system. However most observers think it has little chance of winning. The digital systems may give somewhat better picture quality (thought not necessarily much better), and do have the advantage of being US-developed. NHK, NTT, and others have successfully developed hardware for wideband digital HDTV systems which may be used for future satellite broadcasting services.

Although to some extent conservative in its approach to innovation, Japan has enormous research and development capacity. As an example, NHK alone spends almost as much on research and development as virtually all Europe's broadcasters put together. Coupled to this are the strengths of some of the world's largest consumer electronics companies. At the time the CCIR digital studio standard was agreed, despite many years of European lead, Japan was able to make, evaluate, and agree the EBU proposal within a few months.

In general, Japan is able to respond very quickly and strongly to the need for technical studies.Traditionally, Japanese industry and broadcasters have been cautious about whether digital terrestrial HDTV can be achieved in practice, at least in Japan. There is a much higher density of terrestrial transmitters in Japan than in the USA, and in many parts of Europe (130 million people in a county the size of Sweden). They have therefore argued that the planning difficulties of a digital HDTV simulcast system would be very large. Japan has also argued that using the kind of levels of compression needed to squeeze the HDTV signal into a 6 MHz terrestrial channel would bring a level of artifacts which would negate the value of the HDTV quality for certain types of programme material. In short, there were not sure it was worth it, bearing in mind the high quality expectations of the Japanese public.

In spite of this, there is a growing tide of opinion that Japan must take digital terrestrial television more seriously, and it would be uncharacteristic if some planning studies and system studies had not already been carried out. Very high data compression techniques are proving more effective than anyone thought a few years ago.

There is considerable experience in digital compression systems in many Japanese laboratories, and also evaluations have been made of techniques such as vector quantisation, which was thought at one point to possibly provide the same quality as MC hybrid-DCT with simpler receivers.

No announcements have been made of Japanese projects to develop digital HDTV terrestrial systems, but this may be because of heartsearching decisions about the money already spent on analogue systems.

One firm point is that NHK have declared themselves ready to discuss a worldwide common multiplex system for digital broadcasting and transmission, which they term ISDB (Integrated Services Data Broadcasting). It will be important for Europe to take up the dialogue with Japan on this, if we are to achieve commonality in digital terrestrial television.

## 2.3. Current situation in Europe

The European technical and administrative environment for digital terrestrial television studies and standardisation is complex and multidimensional. Indeed, it is because of this complexity that coordination initiatives were first taken by the Launching Group. The following is a brief summary of the situation. It will necessarily be short, and it is inevitable that some elements will be omitted.

### 2.3.1. First steps in digital terrestrial television systems

It has been evident that digital HDTV would be the long term future of broadcasting for many years. Probably the first major discussions in Europe about digital terrestrial HDTV began in Autumn 1990, when the Scandinavian broadcasters asked the EBU to establish a Task Force to develop an EBU digital HDTV terrestrial television system. The EBU did not immediately set this up, but rather launched a review of the problem.

In 1991, a Scandinavian consortium was formed (HD-DIVINE) to implement a digital high-definition terrestrial television system, to prove the feasibility of such systems. Their objective was to develop a complete system in hardware by mid-1992. In 1990 the IBA (now ITC) began the development of a digital terrestrial television system (SPECTRE) initially based on 4:2:2 vision signals, and in 1991 the CCETT also commenced the development of a conventional quality digital terrestrial television system (STERNE).

In late 1991, a consortium (led by the CCETT and the ITC) of broadcasters and manufacturers put together a relatively large scale project plan for consideration under the RACE phase 2, due to begin in 1992. This was to become the RACE dTTb project. The project started with reduced funding from the EC. After the first year, the EC's understanding of the importance of the project increased, and at the time this report is being written, there are indications that major RACE funding will be available from 1993. The dTTb project is conscious of the need for systems which can be carried easily on cable networks (interworking). The RACE dTTb project will include the SPECTRA project, and parts of the STERNE and other projects.

In 1991, a further collaborative project to deliver digital HDTV terrestrially was formed, led by the Heinrich Hertz institute in in Berlin, now termed the HDTV-T project. The project anticipated German federal support, and therefore was only open to German organisations. Funding for the project has now been agreed.

In 1991 a further collaborative group was formed to develop source coding systems, the relatively large European VADIS project. It provides European input to the MPEG project (see 2.1.1).

In 1991, the German Administration began the process that led to the current Launching Group/WGDTB structure.

### 2.3.2. Satellite projects

There are two RACE projects concerned with digital HDTV satellite systems. FLASH-TV is developing a point-to-point digital HDTV transmission system operating at bit-rates between 34 and 70Mbit/s. HD-SAT is developing a digital HDTV broadcast system for use in the 20GHz band, and possibly the 12GHz band. Currently this is expected to operate at either 45Mbit/s, 70Mbit/s or 140Mbit/s.

The Astra group and News International/Canalplus are understood to be considering digital satellite systems of their own, but which may follow the DirectTV/Skypix pattern of multiple time-shifted broadcasts of the same programme with standard definition. Furthermore, the US-owned MTV (Music Television) is studying the use of digital standard definition broadcasting by satellite in Europe.

### 2.3.3 Transmission and studio system projects

There is also a COST project (COST 206), which will end in 1994 on digital HDTV contribution systems, and a RACE project HIVITS has just completed the development of a 140Mbit/s digital HDTV transmission system.

The Eureka EU-256 project has developed a flexible digital codec operating at 140, 70 and 45Mbit/s. Hardware implementation will be concluded in 1992. Tests and field trials on several transmission media are planned for 1993.

### 2.3.4. The Eureka VADIS project

The activities of MPEG group are explained in 2.1.1. This is in principle only a standardization body, but in practice the MPEG group works like a very large collaborative project. There is a European grouping, the EUREKA VADIS project, which essentially provides European input to the MPEG group, and provides for financial assistance for participants.

### 2.3.5. The EBU

The EBU has about 50 technical groups studying aspects of broadcasting, grouped into four areas: research and development, operational practice, training, frequency planning, Eurovision and Euroradio. The EBU has created a Specialist Group R2/DTV to analyse the pan-European situation regarding planning for digital terrestrial television. The role the EBU might usefully take in system studies is still to be decided. Currently groups V4 and V1 are analysing in somewhat more detail the practicability of the proposals suggested by the WGDTB, and hope to report their findings in Spring 1993. Furthermore, the Convenor of the VADIS/MPEG project has suggested that the EBU should be responsible, as an independent body, for evaluating the MPEG-2 algorithm.

The Eureka 95 project has developed the 1250/50/2:1 HDTV production system, which is about to be issued by the the EBU as a Technical Recommendation for EBU Members' HDTV production. All European HDTV distribution projects assume this will be the HDTV source standard.

### 2.3.6. World standardization studies

The studies and plans of the ISO/IEC JTC1 have already been mentioned. In addition to this, both CCIR Study Group 11 and the CMTT have groups studying digital coding systems. CCIR Study Group 11 has established Working Group 11B on digital source coding, and Task Group 11/3 on digital terrestrial television. Furthermore Task Group 11/4 studies HDTV (digital) harmonisation, and Working Groups 11A, 11C and 11D are also in principle concerned with various aspects of digital systems. In the CMTT, Task Group CMTT/2 and its Special Rapporteur Group SRG is endeavouring to standardise a hierarchical distribution codec for HDTV. The Working Group 10-11S is studying digital television systems for satellite broadcasting. There is thus a degree of overlap in all these Groups. It is not unknown for the same members to be involved with a different Chairman.

### 2.3.7. Timescales and overlaps

Fig. 2.2 attached (European digital television activities) is a summary map of the projects and their areas of interest. No information is available for the Astra/News International/Canalplus/MTV studies.

It is important to separate two types of work/study overlap, overlap in TECHNIQUES, and overlap in APPLICATIONS. Fig 2.2 shows the prima-facia overlap of applications only. Fig. 2.3 shows the prima-facia areas of overlap of techniques.

Fig. 2.4 shows the current broad timescales envisaged for the projects.

When considering the seriousness of the overlaps, it should be born in mind that there are benefits and drawbacks of competition and overlap. There are, furthermore, different kinds of competition which can be advantageous or disadvantageous to different degrees.

For the European public, the ideal POST RESEARCH AND DEVELOPMENT PHASE competitive environment would probably be where different manufacturers compete for the same market. This is functional competition and tends to lower prices and provide more user choice. This environment is achieved by standardisation of the basic system. A competitive environment in which there is not basic standardisation (as exists for example in the recording field) tends to increase prices, because captive markets are created and market sizes are smaller, and this may be seen as disfunctional competition.

For the European public, the ideal RESEARCH AND DEVELOPMENT PHASE competitive environment may be where different design teams compete for the same design goals or system targets, in order to bring out the best in them. This issue is discussed in the Section 5.4 of this report. On the other hand, the benefits of collaborative efforts can include cross-fertilization of ideas. It is in this light that any overlap of techniques or applications mentioned in this section should be judged.

### 2.3.8. Some consequences of the evidence

From the three figures 2.2, 2.3, 2.4 the following points may be made.

- The HD-DIVINE project HDTV hardware is well in advance of the other projects, perhaps by as much as two years. It will have completed hardware development before HDTV hardware development has begun in dTTb and HDTV-T. However, the SPECTRE 4:2:2-quality system hardware, and possibly other hardware is complete. Furthermore, there are more aspects to a system study than hardware development.
- Apart from HD-DIVINE, the earliest appropriate time window for the comparative hardware evaluations and trials which will lead to a final European system seems to be 1995. This leaves two years from the beginning of 1993 for agreement on targets, and for the development in both simulation and hardware of complete systems.
- If there were a unique system for satellites and terrestrial broadcasting of TV and HDTV there would be clear advantages for the general public. The satellite/cable

projects HDSAT and FLASH-TV, and studies in the EBU on the subject, need to take account of the terrestrial study time-scale. Terrestrial broadcasting is the more difficult technical challenge, and its constraints needs to be studied before a decision on satellite/cable is taken. Satellite and terrestrial studies should be encouraged not to go their separate ways before every effort has been made to reach a common system, or maximum commonality.

- Some of the existing projects include structures with up to four layers (Project Board - Project Management Group - Working Group - Sub-group). Even this many layers traditionally introduces long approvals procedures and communications difficulties. The EBU has a four layer structure (Technical Committee - Working Parties - Sub-groups - Specialist group), and this has been considered by some as unwieldy. If the Launching group adds additional layers of Pan-European management to the existing structures every effort must be made to reduce the number of management layers.

- The world standardisation situation is complicated by the fact that there are several bodies involved: ISO/IEC, CCIR Study Group 11, and the CMTT. At the European level, ETSI may be the only body involved, although, if the system is based on decoder/receiver algorithm standardisation, as in the MPEG group, it may be that CENELEC will be involved. Overall, it may be more realistic to concentrate on standardisation within ETSI/CENELEC at least initially, unless there are real prospects for agreement with the 60Hz world.

| STUDIO | TRANSMISSION | DISTRIBUTION | |
|---|---|---|---|
| | | Terrestrial emission | |
| | | dTTb | V4 |
| | | HD-DIVINE | V1 |
| | | HDTV-T | R2 |
| | | (VADIS) | |
| Eu 95<br>V1 | FLASH TV (HDTV)<br>(VADIS)<br>COST 203<br>HIVITS<br>Eureka 256 | Cable systems<br>HD-SAT<br>(VADIS) | |
| | | (VADIS) | |
| | | 12-GHz | |
| | | HD-SAT | V4 |
| | | (VADIS) | V1 |
| | | (Eureka 256) | R3 |
| | | (RACE HIVITS) | |
| | | 20-GHz | |
| | | HD-SAT | V4 |
| | | (VADIS) | V1 |
| | | (Eureka 256) | R3 |
| | | (RACE HIVITS) | |
| | | VCRs<br>(VADIS) | |

Notes: EBU Sub-group V4 studies modulation and multiplexing methods, V1 studies image systems. R2 studies planning for terrestrial television, and R3 studies planning for satellite broadcasting.

*Figure 2.2: European digital television activities*

------------------ <————FLASH-TV————> ------------------

------------------ <————HDSAT————————————>

<——V1——> <————V4————————> <——R2/R3——>

<————————————HD-DIVINE————————————>

<————————————HDTV-T————————————>

------------------ <————dTTb————————————>
(Inc. STERNE, SPECTRE)

<————VADIS————————>

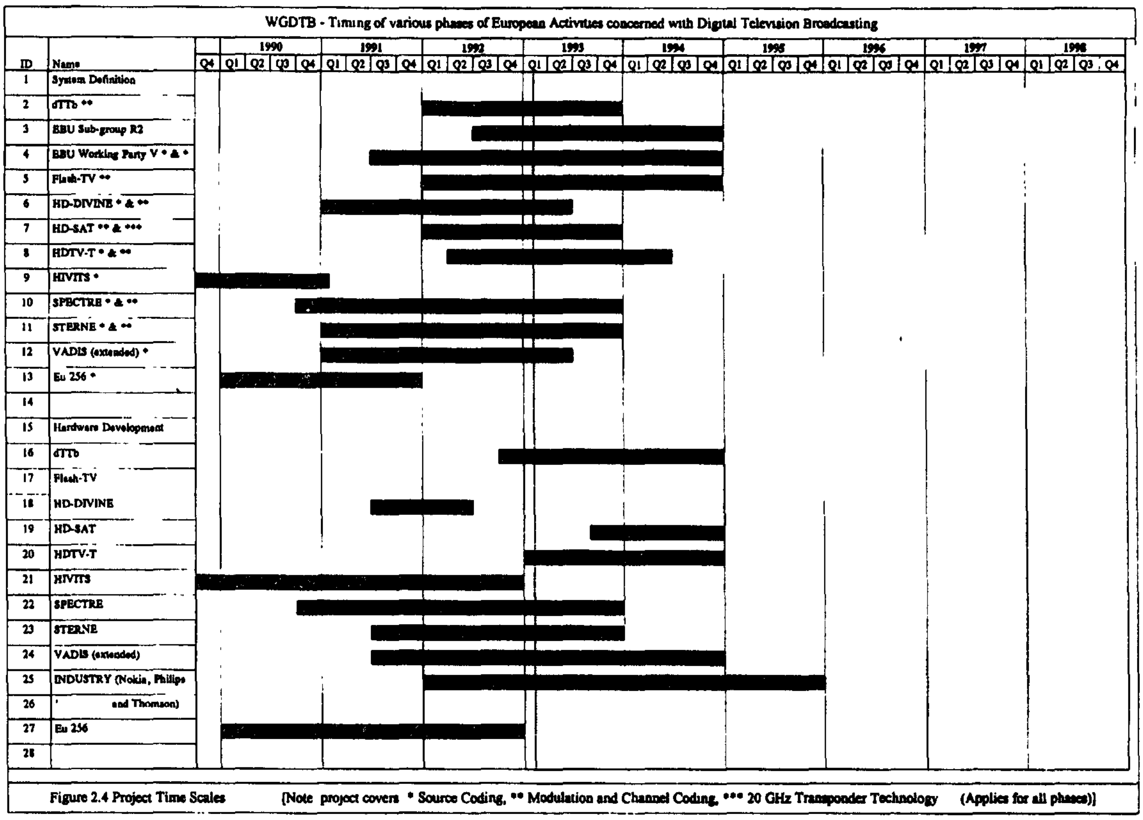
<————HIVITS————————>

<————Eureka 256————————>

| | | | |
|---|---|---|---|
| HDTV/4:2:2 BASEBAND CODING | MULTIPLEXING | MODULATION | PLANNING |

Solid line = primary effort of project
Doted line = secondary effort of project

*Figure 2.3: Overlapping technologies in European activities*



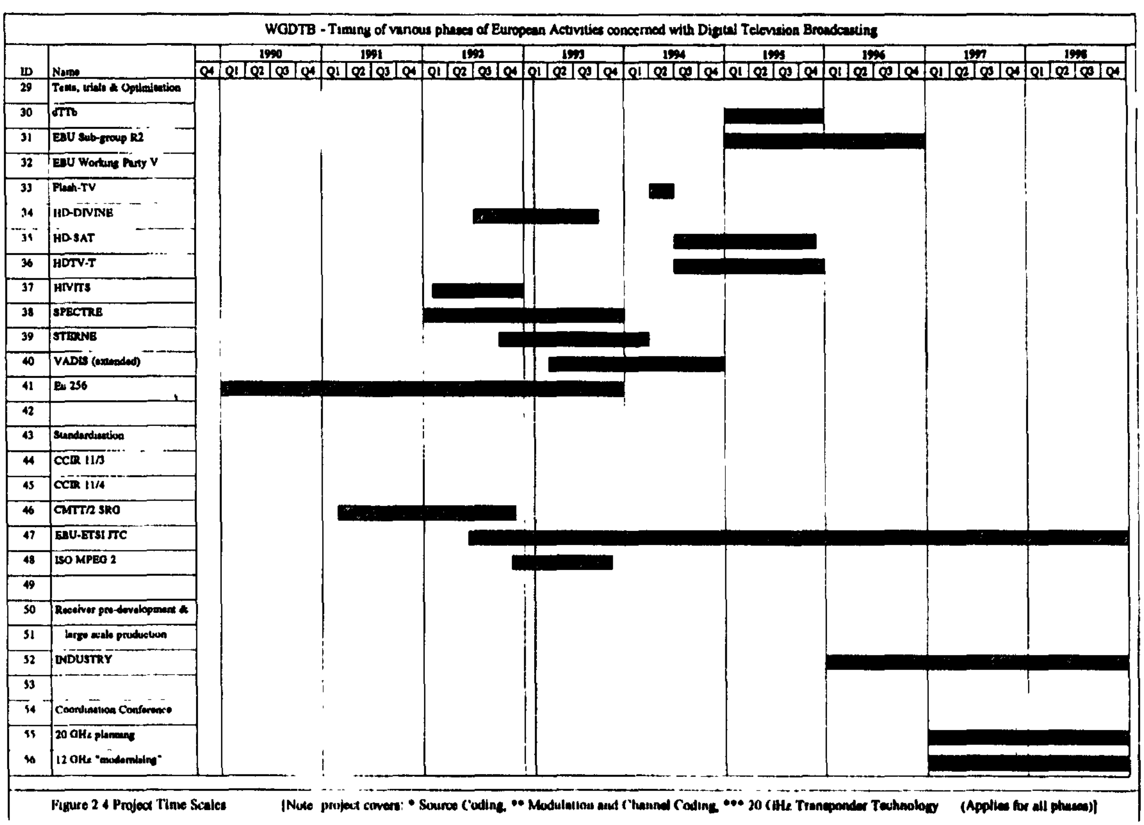




*Figure 2.4: Project time scales*

## 2.4. General constraints on terrestrial broadcasting resulting from existing services

### 2.4.1. Coverage philosophy

Most countries in Europe have, at least for public broadcasting, adopted the philosophy of attempting to provide television to a very large proportion of their populations, rather than just covering the population centres. The result is that, in most countries, more than 98% of the population can receive one or more programmes with a sufficiently high field strength to permit reception without needing to use special high-gain antennas or low-noise pre-amplifiers. The number of programmes receivable is usually three or four. There are countries where only one or two programmes are available, but this is more a matter of national resources than of any planning constraints. There is nothing special about the value of 98%, and many countries aim for 99% or more, although it has to be accepted that the problems and the number of relay stations increase rapidly as the percentage coverage becomes higher.

The above type of coverage is usually referred to as "national", but there is a related variant, called "regional". This recognises that different parts of a country may need different programmes.

In both cases, however, an attempt is made to cover populated areas right to the borders of a country or region within a country. This inevitably means that there are coverage overlaps in the border areas, and it then becomes necessary to continue to protect the additional services against interference in the same way that the existing service is protected. This places special constraints on planning in or near these border areas.

A further planning concept is that of "local" or "private" services. These are primarily intended to cover only the more densely populated areas, with no regard to the interference they may experience in the regions between the commercially-attractive densely populated areas. Of course, the planning of these local services must also ensure the continued protection of the national or regional services in all parts of their coverage areas. It is not uncommon for the maximum coverage of a fully developed set of local stations in a given country to be in the region of 70 to 80% of the population. However, there are differences between countries and between different local programmes chains in a given country.

An important result of these planning philosophies is that a large number of transmitters is needed to provide coverage. Fig. 2.5 shows the number of transmitters using each of the UHF channels in Europe overall (actually in mid-1990, although the numbers have not changed much since then). In many countries, there are restrictions on channels 61 to 69 (they are used by other services) and this causes the discontinuities which can be seen. There are also restrictions for channels 34 to 37, and channel 38 is used by Radio Astronomers.

One of the differences between European and US coverage philosophy is that the primary services in Europe have national or regional coverage aims, while in the US, the term "local" is more appropriate, regardless of the size of the coverage area of the population contained.

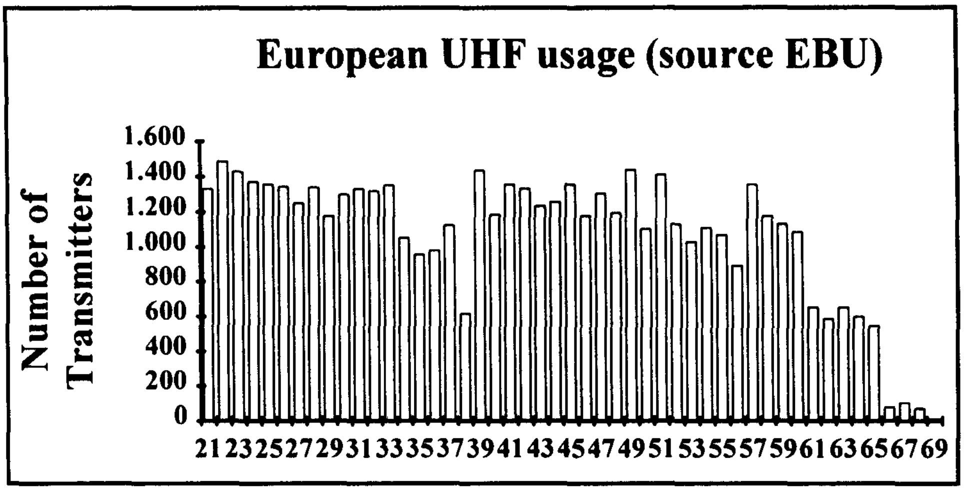



*Figure 2.5: Number of transmitters for each UHF channel*

The primary source of interference in any European country is co-channel. One primary aim of planning is to keep the impairment caused by co-channel interference at:

- no worse than CCIR Impairment Grade 4 for any interference which is continuous;
- no worse than Grade 3 for interference which is present for only 1% of the time (5% in a few countries).

The latter requirement is usually the more difficult to meet.

Few problems are experienced with other interference mechanisms and they now offer few constraints on planning, either for the choice of channels at a given station, or much more importantly, for the choice of channels for overlapping coverages from adjacent stations. The last point is particularly important. Within Europe, there are extensive coverage overlaps from adjacent stations in most countries and, indeed, any channel relationship in overlap areas can be found except for co-channel.

Although the directional properties of a domestic receiving antenna help to reduce the levels of unwanted signals, it is very important to note that where the programmes from adjacent stations are different, viewers will use both, usually with separate antennas (but not always !). In the case where the overlapping services use adjacent channels, the field strength of the wanted signal can be more than 20 dB below that of the unwanted signal. This has important implications in the case where additional (for example, digital) adjacent channel transmissions are proposed from a given transmitting station.

There have traditionally been some planning constraints regarding the channels to be used at a given station. For example, adjacent channels, channels subject to local oscillator interference, or image channels have tended to be avoided[2] in the past

2 These channels to be avoided are usually called "taboo channels" or "related channels". Note that the "taboo" classification of these channels refers only to their non-use at the same transmitting site. They may be more appropriately referred to as "related channels".

(although not in all countries). As the performance of television receivers has improved, these constraints have been removed.

The last constraints to go has been the use of adjacent channels at a given site (although this is now happening), partly because of the difficulties of obtaining:

- enough suppression of spurious components in the lower adjacent channel (a major problem with high efficiency klystron transmitters);
- enough isolation between transmitters to avoid inter-modulation problems.

However, the use of separate, but nearby, sites for adjacent channel transmissions has proved successful. The effect for the viewer is that he can use a single receiving antenna, provided that the transmitting sites are fairly close together, and that he experiences little or no picture impairment, provided that the field strengths of the signals are within a few dB of one another.

### 2.4.3. Looking for frequencies

New techniques of channel coding like COFDM allow the implementation of single frequency networks and permit much more efficient use of the frequency bands than the existing analogue systems. However, such networks can only be placed in a band reserved solely for this service and there is, at present, no broadcasting band (or other band) clear of any occupation.

In these conditions, is it possible to envisage an allocation of a new band for digital television broadcasting?

If we look below 3GHz, which is a reasonable limit for a good terrestrial transmission, frequency bands are used by numerous services and notably by land mobile systems which have, and will grow in, importance. Moreover, other new broadcasting systems are searching for new allocations, and WARC 92 has just allocated a band for Digital Audio Broadcasting between 1452 and 1492MHz.

In this context, it is not likely to be possible to obtain a new band for digital terrestrial television broadcasting. It is thus necessary to make the new broadcasts in the bands currently allocated for television broadcasting, which seems to be logical, since the new service will take the place of the analogue PAL and SECAM services in the long term.

One possibility is to use the "related" channels, which were previously avoided at a given transmitting site. This would require the radiated power to be kept low. This is necessary to avoid co-channel interference to service areas of other nearby analogue transmitters, which use the related channels.

The maximum effective radiated power of a digital service will, therefore, almost certainly be limited by the constraint of not causing co-channel interference to existing analogue services. This may lead to digital transmitter radiated powers of the order of 20 to 30dB less than the existing analogue transmitters from the same site.

The exact level of coverage will depend on the modulation method used for the digital service. It will, for example, be less for an HDTV service using 64 QAM than a standard definition service based on QPSK. Potential coverage is difficult to quantify until the protection ratios and minimum field strength figures for the digital system are decided.

If the coverage figures for "related" channel transmissions should prove inadequate to provide market viability at a quality level which is sufficient to encourage viewers to migrate towards digital reception, then it will be necessary to find channels which can be dedicated to digital use.

### 2.4.4. Examples of existing situations

In the United Kingdom, planning for analogue television in Bands IV and V has provided four services, each covering 99.3% of the UK population. Generally all four services are radiated from each transmitter site. The channels are grouped in the form N, N+3, N+6, N+10, for Band IV and lower Band V, and N, N+4, N+7, N+10 for upper Band V. These groupings were chosen to avoid reception problems from adjacent channel, image channel, and local oscillator interference.

In Germany, the VHF/UHF bands (I, III and IV/V) are used by analogue television services. Full area coverage with the public programmes is provided, which requires the operation of roughly 290 high power transmitters and in addition 8000 fill-in stations. Moreover about 200 private stations are operated in areas with a high population density. Channels 61-69 are not used by analogue television services. If this part of the spectrum can be made available for broadcasters, it may offer a good chance for a start with digital terrestrial television.

In Scandinavia, the introduction of digital terrestrial services seems to be viable without changing current PAL broadcast networks. The presumption made in the preliminary studies is that each of the current networks should have the possibility to be "simulcast" with the same coverage area in a digital format. The single frequency option is interesting, particularly the concept of Local Single Frequency Networks giving the possibility of rather sharply defined regions for regional broadcasts. In the short term, nation-wide SFNs will be difficult, due to the existing PAL transmitter networks.

In France, bands I, III, IV and V (up to 830 MHz) are presently used to broadcast 6 networks. Band III is shared with a land mobile system (Radiocom 2000) and no possibility exists in that band at present. The networks have 112 main stations and 3290 re-broadcast stations. The percentages of coverage are respectively 99.9% for networks 1, 2, 3 and 87, 77 and 66% for networks 4, 5 and 6. Studies on the possibility of introducing digital television in the UHF band are now being carried out on the adjacent channels of the three first networks.

In Spain, five programmes (2 public and 3 private) with national coverage, and 1 or 2 programmes with regional coverage, are at present provided using bands I, III, IV and V (channels 2 to 4, 5 to 11 and 21 to 65). The two public networks reach 98 and 96% coverage, and the three private 80%. Although all television stations at present allocated in bands I and III must emigrate to the UHF bands before the beginning of 2000, leaving them free for other services, studies have confirmed that there will be a certain number of free and taboo channels available to start with digital broadcasting, giving the possibility to simulcast the 5 current PAL programmes with national coverage.

In Italy, the channel occupancy in the VHF/UHF bands is very close to spectrum saturation. Planning criteria, derived from the Stockholm Plan, make use of taboo channels with protection ratios updated to the improved performance of new generation receivers. It is unlikely to be possible to introduce Single Frequency Networks (SFN) on a nation-wide basis, at least in the initial phase. Instead, there are opportunities of finding frequency assignments on a local basis in urban areas. By careful selection of the transmitter locations, a service level of about 50% of the population resident in towns with more than 50,000 inhabitants seems possible, for at least one digital Television/HDTV network.

## 2.5. Interrelationship with other distribution media

### 2.5.1. General

Video and audio signals will be distributed via a variety of media by the time digital terrestrial television broadcasting (DTB) is introduced. These media, including DTB, cannot always be transparent, which implies that an interrelationship is important. Most of the signals will be either 625/50/2:1 or 1250/50/2:1, YUV coded, have colorimetric and other scanning parameters conforming to CCIR Rep. 624 and Rec. 709, and until counter arguments arise, it seems reasonable to base the DTB system on the same parameters.

### 2.5.2. Satellite

Due to the characteristics of this medium, the modulation parameters of satellite systems may be different to terrestrial systems. For analogue satellite signals, little commonality between the HDMAC and DTB source decoder can be expected. What remains is the multiplex and conditional access architecture. Although there are no obvious arguments in this early stage of development, it seems reasonable to maximise the commonality by selecting for DTB an ECM/EMM access control system, as used, for example, in Eurocrypt. At the moment there are no plans in Europe for digital satellite video services. However, if a project with this purpose did start, it would be clearly beneficial to have identical source decoders, if the desired quality levels of such a future satellite service were not significantly different to the range offered by DTB.

### 2.5.3. Cable

#### 2.5.3.1. Modulation and source coding

A number of issues that are valid in selecting the modulation and source coding scheme are not so important for cable (SFN, co-channel interference, power, linearity, graceful degradation, portable reception). Consequently optimising digital transmission over cable may lead to different source coding and modulation strategies (although this is not to be desired). A compromise between receiver complexity and efficiency of cable usage has to be found, and this needs attention by the project management dealing in these areas.

#### 2.5.3.2. Multiplexing

The multiplex and scrambling architecture of DTB must allow for the insertion of local signals at cable head-ends into the DTB signal. In cable head-ends it must be possible to interface with satellite and DTB services at the multiplex level.

### 2.5.4. B-ISDN

See 2.5.3.

### 2.5.5. Cassettes and discs

#### 2.5.5.1. Cassettes

It is impossible to predict today how future digital video consumer recording and playback equipment will be designed. The following relates to a digital consumer recorder and still has a large degree of uncertainty. Possibly a future digital consumer recorder will have (apart from analogue) 2 digital interfaces: one at baseband (Rec. 656-like) and one that enables virtually transparent recording of a 25 to 27Mbit/s bit stream. Recording errors have typically a bursty character in spite of the 2D error-correction. A reliability indicator can be provided. The BER will depend among other things on the tape quality. Values between $10^{-7}$ and $10^{-9}$ are to be expected.

'Trick' modes like reverse play, fast forward etc., are not possible with this latter interface. To make these possible, the bit stream has to be reduced to below 20Mbit/s and additional electronics will be required. Nevertheless it seems very attractive to record and play-back DTB signals via this interface, because a cascading of source encoding and decoding is avoided this way. The maximum bit-rate between 25 and 27Mbit/s is below the rate desired by the WGDTB for HDTV, but it does conform to what the dTTb project considers to be realistic today.

#### 2.5.5.2. Disks

Compared to tape, the advantage of a disc is the random access option. The disadvantage is that compared to tape, the active area is considerably smaller. These properties make disc suitable for interactive products like CDI and photo-CD. On the other hand there will probably be no, or only a minor, interrelation with DTB.

### 2.5.6. Multimedia

Picture, sound, voice I/O, text, computer power etc. combined with a degree of imagination can lead to very interesting services and products. One example could be that, the decoding programme is transmitted together with the bit stream. The only thing that needs to be standardised and probably already is de-facto standardised in the multi-media future is the operating system. Interrelation with DTB is currently unpredictable.

# 3. POTENTIAL SERVICES BASED ON DTB

## 3.1. General concepts

A digital terrestrial broadcasting system might be developed in a wide number of variants, depending on whether, for example, it is intended for HDTV broadcasting, or for the simultaneous transmission of several conventional programmes within one channel.

The system necessarily will include the following basic elements (see Fig. 3.1) although some of them may be combined to form one physical device. One or several source coding systems would be used, each of them involving the coding of a video signal and one or several sound signals. To these digitally coded signals, digital data are added. These signals would be multiplexed in a service component multiplexer. The resulting digital, signal is protected with an external error correction code (e.g. Reed Solomon).

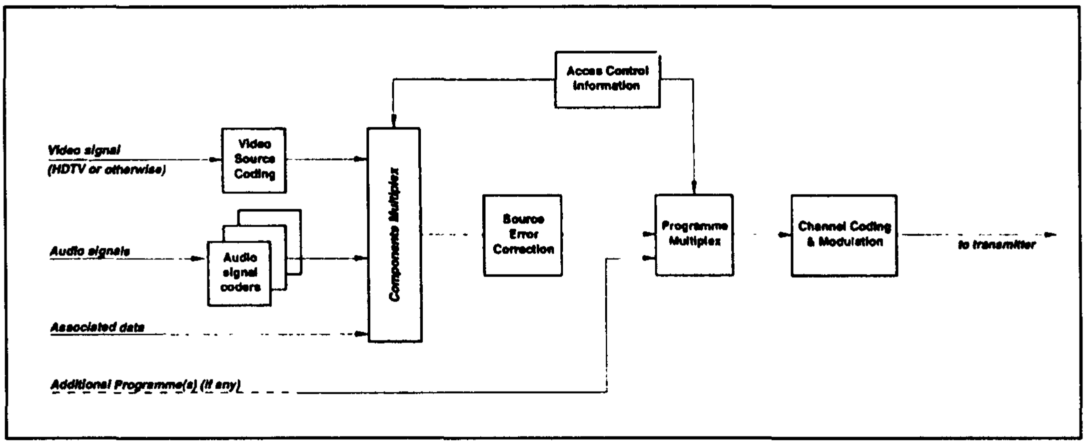



*Figure 3.1: Basic structure of digital terrestrial television system*

Several such programmes may be, in turn, multiplexed into one channel. Access control data may be included at either multiplex level or at both levels. The resultant data signal may then be channel coded (using, for example, a convolutional code), and used to modulate the carrier (or carriers).

## 3.2. Introduction

The wish to use terrestrial broadcast frequencies for carrying a digital television presents the service designer with a fundamental trade-off. This is how best to share available channel capacity between the ruggedness of the signal required to reach a particular class of receiver (which may be portable, mobile or permanently connected to a roof-top antenna), and the bit-rate which may be reliably delivered to it.

Overall channel capacity is, however, dependent upon carrier-to-noise ratio, and this may be increased by increasing the power of the radiated signal. Thus the ultimate restriction on the bit-rate which may be conveyed is imposed by the maximum radiated power permissible within the available spectral environment. This will be set to prevent interference into other broadcast services under poor propagation conditions. Studies of spectrum planning on a Europe-wide level are currently being undertaken, within several European activities.

For a given channel bandwidth and carrier-to-noise ratio, the service designer can use a range of possible modulation and error protection strategies to achieve the ruggedness required to serve the desired class of receiver. This ruggedness requirement is most conveniently expressed (reciprocally) as an efficiency which describes the ratio between usable bit-rate and usable channel bandwidth (bit/s/Hz).

Having determined that a certain bit-rate will reach the target receivers, the service designer must then consider the picture resolution which this bit-rate is able to support while maintaining a consistent image quality. This involves a knowledge of the bit-rate compression obtainable with recent algorithms and the statistical distribution of television scene content. The bit-rate available may be high enough for the designer to have more than one option in the way that he allocates it to television services. He may, for example, be able to convey either one high definition or two conventional definition services. Such options could even remain open, to allow a flexible, switched operation, on a programme-by-programme basis, provided that the receiver is designed appropriately.

An analysis of the wide range of possible service options may, at first sight, seem impractical because of the large number of possible permutations involved. Fortunately, the problem becomes more manageable, because the classes of receiver which require the most rugged signals (portable and mobile) are also those where lower definition pictures are acceptable. The limited set of recognised display resolutions and qualities then becomes a convenient discrete variable, against which to analyse the advantages and disadvantages of the various delivery options.

The WGDTB took, for this analysis, four qualities of picture, each of which is considered in the following sub-sections. It should be noted that the contents of sub-sections 3.3-3.6, which analyse these, are intended for the purposes of discussion, and do not represent system proposals of the Working Group. Rather, it was discussions based upon these considerations which led the Working Group to its conclusions on flexible and hierarchical systems. These four qualities of picture are broadly classified as follows:

**HDTV quality** (Section 3.3) (High Definition Television) where the potential exists for the delivery of a picture which is subjectively identical to the interlaced HDTV studio standard (1440/1250/50/2:1). Quality shall remain consistent with this for a given percentage of television programme material (where this percentage is in the high nineties but is yet to be identified).

**EDTV quality** (Section 3.4) (Enhanced Definition Television) where the potential exists for the delivery of a picture which is subjectively indistinguishable from the 4:2:2 level of CCIR Rec. 601 (720/625/50/2:1). This quality shall be maintained for a given percentage of programme material (which will be in the high nineties but is yet to be identified).

**SDTV quality** (Section 3.5) (Standard Definition Television) where the quality is approximately equivalent to that of current PAL or SECAM. This equivalent quality may be achieved from pictures sourced at the 4:2:2 level of CCIR Rec. 601 and subjected to processing as part of the bit-rate compression. The result should be such that when judged across a representative sample of programme material, subjective equivalence to PAL/SECAM is achieved.

**LDTV quality** (Section 3.6) (Limited Definition Television) where the quality is equivalent to that obtainable from the MPEG-1 system which operates on a source resolution approximately ¼ of the 4:2:2 level of CCIR Rec. 601. This quality is considered by some to resemble that of VHS (albeit only for a proportion of programme material).

There is interest in conveying digital television pictures to more than one class of receiver simultaneously, for example, a rugged SDTV representation to portables, and an HDTV representation of the same material to stationary domestic receivers. This approach may be efficiently achieved by employing a hierarchical picture compression scheme, where the higher resolution signal is reconstructed by first decoding and exploiting the rugged lower resolution version. These possibilities are discussed in Section 3.7.

## 3.3. Terrestrial HDTV for stationary receivers with roof-top antennas

### 3.3.1. Features of the HDTV service

A new standard for terrestrially delivered television is expected to have a lifetime of many tens of years, and to operate in an environment where many other digital transmission media will be conveying to the viewer a range of high definition visual services. The consequences of this are two-fold:

a) like other media, the system should be capable of carrying high definition images if it is to meet with the expectations of the users over this timespan, and

b) as much system commonality or compatibility must be sought between the techniques employed for terrestrial broadcasting and those for other media, since this will result in simpler signal translation between media (e.g. terrestrial to cable) and a more economically attractive common receiver for the consumer.

This latter consideration may not only suggest that HDTV source coding for other media should be taken account of in the terrestrial system, but also rather that, since the terrestrial application is technically the most challenging, it should be terrestrial HDTV which actually sets the compatibility constraints for other media to follow. This could also be the case in modulation. OFDM methods, for example, are widely regarded as highly effective for terrestrial broadcasting, and could be considered for other media too, in the interest of commonality.

Most popular strategies for the introduction of DTB into bands of frequencies currently occupied by analogue services, rely on the "simulcasting"[3] of services in both digital and analogue forms, for some period of time. These strategies then assume that, because of some other unique incentive which will be provided by the digital service, viewers will change their receivers to allow them to watch these services. When sufficient migration has taken place, the analogue services could then be discontinued. There are several ways in which such an incentive can be provided, but many believe that this is best achieved by conveying the digital services in HDTV.

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[3] Simulcasting here means broadcasting the same programme content in two independent channels.

### 3.3.2. HDTV service characteristics

For an HDTV service, the following target characteristics have been identified by the Working Group:

**Image** The image quality should be essentially transparent (<12% on the Double Stimulus Continuous Quality Scale) to the HDI studio standard for [99.7%] of the content of typical television programmes.

**Audio** Up to 5 audio channels with CD/DAB quality with additional commentary channels should be available.

**Data** A minimum capacity of 64kbit/s should be available, with a higher rate for subscriber addressing, teletext, etc., provided by the dynamic allocation of multiplex capacity. This may be possible when not all available bit-rate is consumed in providing adequate picture quality for some television scenes, and the system should be capable of providing a mean capacity of at least 1Mbit/s.

The total bit-rate necessary for these services, largely governed by that of the video requirement, is thought to be of the order of 30Mbit/s. To meet the relatively stringent requirement for image quality given above, and to include sufficient error protection overhead to be workable in the terrestrial environment, this bit-rate must be considered to be ambitiously low (i.e. it assumes the development of extremely effective digital compression techniques). At the present time however, there is insufficient experimental evidence upon which to base a more realistic estimate.

Those participants in the WGDTB having a special interest in HDTV transmission expressed the requirement that flexibility should be provided at the level of the multiplex, to enable the conveying of alternative broadcast services. In particular, that provision should be made for the broadcaster to convey a number of SDTV services, perhaps to cater for specialist programming outside of peak viewing times. It is considered important that when such multi-programming flexibility exists, it should be under the control of a single broadcaster and should not consist of SDTV services from a number of different broadcasters. This is because:

a) the HDTV broadcaster would not have the freedom to change easily the number and balance of HDTV/SDTV services with time and,

b) networking would otherwise have to be provided between the sources and DTB service multiplexer leading to a number of technical difficulties, particularly if statistical multiplexing of data were to be considered.

It is considered essential for coping with propagation variation within the service area, that in the presence of errors, the quality of picture and sound services, degrade gracefully. It is recommended that for systems employing hierarchical or scalable quality levels, that a degradation hierarchy should be established which has levels in common. A detailed discussion of this is given in section 3.7. Relative failure of sound and vision services should be determined through subjective tests. The traditional approach where sound fails after vision is thought to be worthy of investigation.

### 3.3.3. Frequency spectrum requirements for an HDTV system

Considering a terrestrial broadcast channel with a bandwidth of 8MHz, the service bit-rate calls for a modulation efficiency of 4bit/s/Hz. This could be met by the application of a 16-QAM or 32-QAM modulation scheme, either on a single carrier; or better, on the carriers of an OFDM multiplex.

If the system is to operate in terrestrial channels where high power analogue broadcasting is not possible for interference reasons, it will be necessary to 'condition' the spectrum of the transmission. Such conditioning is important to make the signal rugged to co-channel and adjacent channel interference from existing analogue services, and involves the practice of not transmitting useful data at frequencies where energy in the interfering analogue services is concentrated. Studies, for example, in the SPECTRE project, have resulted in a system design where the bandwidth which is sacrificed to achieve this conditioning amounts to 1.5MHz. Of this, 0.796MHz provides protection against co-channel interference and 0.704MHz provides protection against adjacent channel interference. This suggests that the effective bandwidth may be well below the total channel bandwidth.

For viewers close to the transmitter, where the influence of co-channel interference is weaker, or in applications where the channel employed is not shared with analogue services (up to some considerable distance), it would be possible to convey useful data in the 0.796MHz. Although this would be at the expense of some flexibility in a common receiver, it must be recognised that similar flexibility will have to be incorporated anyway, if the receiver is to operate efficiently in the all-digital environment remaining after analogue services have ceased.

A further decrease in available capacity will result from the introduction of guard intervals into the OFDM symbols to enable improved performance if operated in a single frequency network (SFN). At present there are no considered analyses of networking situations available, upon which to base an informed choice of guard interval (which is influenced by the symbol length itself, or the number of OFDM carriers employed in a system). For roof-top antenna reception it only needs to be a small proportion of the total symbol time (for example, a few % in the 16-QAM version of SPECTRE).

Broadcast power levels have already been briefly discussed and must represent a careful compromise between the requirement to deliver services with the chosen modulation to the larger majority of the service area in a simulcast shared-spectrum environment, and the requirement not to interfere significantly with the existing analogue services. The use of fill-in (relay) transmitters or a network of low power transmitters operating in a SFN could help to provide improved coverage of signals with high order modulation. However, the economics of providing a number of such low power transmitters have yet to be shown to be viable.

### 3.3.4. Prerequisites for implementation of an HDTV service

It is useful to consider what aspects of technology, economics, standardisation and regulation remain to be addressed before HDTV could be delivered to stationary receivers.

#### 3.3.4.1. Technology

| | |
|---|---|
| **Studio** | Studio standards for HDTV are well established, as is the technology for performing all the studio processing required for programme production. There is still however, a requirement for increased practicability (rugged and/or lightweight equipment, for example) but developments are continuing to be made in this area. |
| **Contribution networks** | Coding equipment for 140Mbit/s transmission of HDTV has already been demonstrated by RACE HIVITS, Eureka 256 and others. Many transmission media are now capable of reliably handling this data capacity. |
| **Primary distribution** | In principle a HDTV contribution codec could be operated as a distribution codec at a lower bit-rate. However, a more attractive option may be to employ the secondary distribution (emission) coding for this purpose. The need for transcoding would then be obviated. |
| **Secondary distribution** | The digital coding and modulation methods for emission are covered above, and are still the subjects of considerable study. The requirements imposed by the terrestrial UHF/VHF channel need to be better characterised, as do the linearity requirements of the transmitter. |
| **Receivers** | At present the technical and economic requirements of receivers are not characterised. Some important issues are flexibility (to operate in different networks, to have automatic tuning, to have co-channel interference conditioning for different European analogue standards, etc.), complexity (implementational constraints on VLSI a few years hence), target price and availability of large-screen HDTV LCDs. One suggestion to move forward in this area, being adopted in the dTTb project, is to create an engineering model or reference receiver and also (not so far adopted) to create a 'user's view' model, or concept receiver, in order to help clarify the requirements. |
| **Home recording** | Much work needs to be done to characterise the requirements for a domestic VCR. If it is to be digital, consideration must be given to transcodability from the emission coding algorithm in order to minimise accumulated distortion. |

#### 3.3.4.2. Economics

Investment in HDTV programme production, transmission infra structure and domestic receivers will be very large. Even after the technology is established and available, both broadcasters and receiver manufacturers will need to be confident of a return on these investments before plans to implement services will be made. The introductory strategy whereby an existing analogue service is simulcast in digital HDTV, does however the have major economic advantage for the introduction of DTB, that an entirely new programme service does not have to be launched. The success of an entirely new service would be extremely sensitive to market penetration of receivers during its start-up phase and would probably represent quite a gamble on income from advertising or subscriptions.

At this time little more can be done than to identify factors which contribute to these costs so that a more informed economic analysis can be performed later. Such factors are: the prior or simultaneous availability of HDTV on other media (additional incentive for HDTV production and domestic display purchase), the choice of introductory strategy for terrestrial digital services (simulcast/new programming, use of 'taboo' channels/new spectrum, use of existing transmitter masts/new dense low power networks), the flexibility to provide multi-channel SDTV outside of peak viewing hours, and the availability of affordable large-screen displays.

#### 3.3.4.3. Standardisation

Although source formats for HDTV are well understood and reported within standards bodies, no standards yet exist for the digital coding and multiplexing associated with the networking, emission, or domestic storage of digital HDTV. By considering the very large number of approaches and parameters involved in both video and audio coding, the magnitude of the task becomes apparent. This magnitude is however compounded by the need to ensure that transcodability between standards with minimal accumulation of distortion, is possible.

In practice this will mean that standards must share some degree of commonality or compatibility and may well have to be considered simultaneously.

#### 3.3.4.4. Regulation

In the US, regulations are planned to use digital terrestrial `taboo' channels for the simulcasting of existing services. Furthermore, in the US, a date may be set for the termination of terrestrial analogue broadcasts. Spectrum availability differs between different European countries, which would make it difficult to impose a unique date by Directive here. The Working Group has reached no conclusion on whether any regulation should be encouraged, either at a national or European level. However the topic has received debate.

It has been generally noted that the provision of flexibility for a UHF/VHF channel to be used for different services at different times of the day will require adjustment to regulatory processes in most European countries.

### 3.3.5. Possible date of availability of an HDTV service

It is believed that the date at which digital terrestrial HDTV broadcasting will start is more likely to be governed by issues of economics and spectrum availability than technology. The influence that the costs of large-screen displays, receivers and programme subscriptions is perceived to have on the desirability of the consumer to view HDTV, governs the investment that the broadcaster will make in HDTV equipment for the studio. At present it is not possible to predict when the economics will be right to begin HDTV broadcasting. Many broadcasters, however, recognise the need to plan the technology now to allow for HDTV transmission, because failure to do so could result in the establishment of SDTV technology which would not be upgradable, and in the long-term would not meet the needs of the viewer.

One estimate is that for the full receiver implementation of an HDTV system (of the kind discussed in the WGDTB), 0.3 μm integrated circuit technology must be available. This is not expected before 2003. The development of an affordable HDTV display requires considerable further work and at present no definite time can be foreseen for its availability.

As examined in sub-section 3.3.4, much of the technology for sourcing and networking the services already exists, but considerable effort remains to be expended on the specification of the emission system, receiver development and DVCR design. These developments must take into account the need to define a system which has all the features necessary to satisfy the broadcasters and viewers for several tens of years, since further basic changes to the transmission standard would be very difficult.

No precise estimates have yet been made of the likely development or standardisation time of the emission, reception or recording elements of the broadcast system because studies of many aspects of the technology are still in their infancy. Several exciting concepts have been proposed to enhance the possible features offered by terrestrial HDTV, but their viability remains to be demonstrated.

### 3.3.6. Costs

For the broadcaster, the questions of capital system costs, transmission costs and HDTV programme production costs are all fundamental considerations and at the present time it is not possible to say which will be the most influential on service start-up and operation. Sub-section 3.3.4.2 identified some of the unknowns associated with this, perhaps the greatest of which is whether terrestrial broadcast will be the first medium to deliver HDTV to the home. The probability of success in starting HDTV broadcasting on any medium will be enhanced by not being the first medium to offer HDTV, since the take-up of receivers may not be rapid. A simulcast strategy towards the introduction of such services will however ensure that services can be watched at some picture resolution, and therefore be financially supported to some extent.

Some initial costings relevant to terrestrial HDTV are becoming available now. Many European broadcasters now have experience of the costs of HDTV production (through involvement with Vision 1250, for example) and some cost estimates are being worked on, for example, in the dTTb project for an OFDM-based transmitter and a model receiver. Because of its low power operation, a digital transmitter will result in a cost reduction compared to high power analogue transmitters, in terms of capital, operational and maintenance costs.

Some initial estimates of the cost of an encoder have been made by LER based on a quantity of about 20 systems and 1992 prices. These estimates show that for an HDTV-sourced system, the video and audio compression are thought to occupy 22 circuit boards (230 x 160mm) at a cost of about 85kECU (including case and PSU). The multiplexer is thought to occupy 4 boards at a cost of about 15kECU and the modulator is thought to occupy 10 boards at a cost of about 40kECU, both costs include case and PSU.

Receiver and DVCR costs are at present unknown, as are the dominant factors in determining these costs. It cannot be assumed that initially the cost of an HDTV receiver will be largely that of the display, with VLSIs forming a relatively small contribution. The question of how flexible the receiver must be, in terms of its ability also to handle PALplus, SECAM or HD-MAC, must be analysed. Some introductory strategies will require multiple standard reception.

Even after the consumer has invested in the reception equipment, how much will he be prepared to pay in order to watch the programmes? This will be a further critical dimension affecting the success of the system.

### 3.3.7. Applications of the service to other media

The success of a new digital broadcasting scheme will depend upon an early consideration of the relationship with other media. Present cable systems, for example, are closely tied to terrestrial broadcasting technology, and it will be important to ensure that the modulation schemes adopted for the terrestrial channel are also suitable for cable. This issue is being studied, for example, within the dTTb project.

There are parameters of a terrestrial system design which would not be incorporated in a system designed specifically for cable; however, in the interests of receiver commonality, many such parameters should remain common. In cable systems it may prove useful to demodulate and remodulate at the cable head end, if a narrower channel spacing is required than in the terrestrial environment. To some extent this could be done by removing the unused spectrum which provides for interference conditioning in the terrestrial environment, but it must be remembered that after the cessation of analogue broadcasting this conditioning would be omitted.

Digital satellite broadcasting may employ different modulation methods from terrestrial broadcasting; but, at the very least, commonality at the picture coding level should be strived for. The requirements of digital coding for recorded media differ significantly from those of transmission media, and it may be that close commonality cannot be achieved here. Transcodability with minimal accumulated distortion has, however, been identified as a vital criterion.

It has been suggested that for the purposes of digital source coding a generic scheme should be adopted which can be adapted to specific media of interest. While this is a praiseworthy concept, it is increasingly being recognised in many fields of digital communications that optimal systems can only be designed by breaking down the traditional boundaries between source coding, channel coding and modulation and considering them together. Several of the concepts discussed in this report, for example graceful degradation, require an intimate interworking between all parts of the system.

For other considerations of interrelationships with different media see sub-section 2.5.

## 3.4. Terrestrial EDTV for stationary receivers with roof-top antennas or portables with small antennas

### 3.4.1. Features of the EDTV service

A key feature of this level of service is that it can serve portable receivers while at the same time providing a higher quality (with 16:9 aspect ratio) than any current terrestrial system. The particular advantage of portability is seen by some as being a vital feature of a terrestrially delivered service, since this cannot be practically achieved by any other delivery medium. It is also a vital service to retain, since the public has become used to being able to obtain some level of service from a portable receiver and it is not envisaged that such a service could be taken away. Many viewers, in certain parts of Europe, do not have access to roof-top antennas and must rely on portable reception.

In the case of the features offered by HDTV reception (3.3), it was stated that introductory strategies relied on the provision of some incentive with digital services to entice the consumer to buy a digital receiver during a period of analogue/digital simulcast. For the same strategy to operate here, it must be hoped that the incremental advantage of the display quality offered, over that of analogue systems, is sufficient to provide this encouragement. In assessing such quality some consideration must be given to the `absence of noise' advantage that is often claimed for digital systems over analogue systems. It remains to be seen, however, what the viewer's reaction to occasional digital coding impairments will be.

### 3.4.2. EDTV service characteristics

For an EDTV service, the following target characteristics have been identified by the Working Group:

**Image** The image quality should be essentially transparent (<12% of the double stimulus continuous quality scale) to the 4:2:2 level of CCIR Rec. 601. This quality should be maintained for a given percentage of television programme material (where this percentage is in the high nineties, but is yet to be identified).

**Audio** At least one stereo service with CD/DAB quality with additional commentary channels should be provided.

**Data** A minimum capacity of 64kbit/s should be available, with a higher rate for subscriber addressing, teletext, etc., provided by the dynamic allocation of bit-rate during the transmission of non-critical television scenes. A mean capacity of between 0.5 and 1Mbit/s should be achievable by this means.

The total bit-rate necessary for these services is thought to be of the order of 11.25Mbit/s and evidence suggests that this is currently achievable within the bounds of quality required.

Graceful degradation to cope with variation of reception conditions is thought to be desirable, and consideration should be given to the use of common hierarchical levels in the event that a system having hierarchical quality is adopted. Relative failure of sound and vision services should be determined through subjective tests. The traditional approach whereby sound fails after vision is thought to be worthy of investigation.

### 3.4.3. Frequency spectrum requirements for an EDTV system

In a terrestrial channel of 8MHz bandwidth, the service bit-rate calls for a modulation efficiency of the order of 1.5bit/s/Hz which probably corresponds well with the level of ruggedness required for portable reception. Such a scheme might, for example, employ QPSK modulation at 2bit/s/Hz with the remaining capacity allocated to error protection.

Overall spectral efficiency depends upon the level of conditioning applied in order to achieve protection against interference, and the possibility that the symbol period is extended to give a guard interval. Some initial thoughts on this have been given in sub-section 3.3.3.

The broadcast power level, as with any digital terrestrial service, represents a careful compromise between the service coverage (here to portable receivers) and interference into existing analogue services. Portables in urban areas however, may be well served by a dense network of low power transmitters operating as a SFN. The economics of the provision of such a network needs examination before it can be relied upon as a delivery mechanism.

### 3.4.4. Prerequisites for implementation of an EDTV service

#### 3.4.4.1. Technology

The technology required to implement the professional parts of the broadcast and networking systems exist. The principles required to achieve the level of source compression required are well-known and could, in principle, be extended to hierarchical operation. Transmission standards for contribution coding of 4:2:2 material are now set, and commercial codecs and networks are available. Experience from existing trials of DTB and DAB suggests that the modulation and channel coding technology is established, although effort is required to develop a complete broadcast specification which encompasses multiplexing of all the service components. Mass production of the receiver

ICs will only begin when the confidence of a standard is available.

### 3.4.4.2. Economics

In the event that 16:9 aspect ratio production is already in operation, a system employing conventional resolution sources would call for no additional investment in the studio. Networking is also moving rapidly in the direction of digital components, within the budgets available to most broadcasters.

The key features for the success of EDTV services will therefore depend upon the cost of a domestic digital receiver and the willingness of viewers to pay for the programmes. Receiver costs must be as low as possible, perhaps no more than 10% more expensive than (equivalent) analogue receivers, particularly if a migration towards digital reception is to be encouraged in a simulcast environment.

It must be remembered that the incremental quality between existing analogue and EDTV digital may not, in itself, be a sufficient incentive for the consumer to change receiver. If the coverage of portable reception can be increased, interest in the service may be raised by marketing, since the public perception of portable reception today is not of a high quality service. Appropriate styling for high portability is also expected to increase the consumer-attractiveness of such receivers.

### 3.4.4.3. Standardisation

Although the technology for the required level of compression for an EDTV service is well established, there are at present many collaborative projects (Vadis, MPEG, dTTb, etc.) working on detailed systems, but with slightly different objectives. When the problem of agreeing a single approach between these projects, is compounded by the slightly different requirements of each European country, an even more difficult situation arises.

As was stated in the consideration of HDTV system standardisation, the need for transcodability and interworking between different media is likely to mean that standards for more than one medium will need to be considered simultaneously. Such an agreement is likely to more difficult to achieve than any so far attempted in the worlds of broadcasting and telecommunications.

### 3.4.4.4. Regulation

The part of the WGDTB group which examined the EDTV system felt that policy statements regarding the future direction of television services in Europe would need to be made. Such statements should define a certain digital system as being the successor to PAL and SECAM and should also define, for each European country, the strategy which must be adopted in order to phase out analogue services.

### 3.4.5. Possible date of availability of an EDTV service

Although the technology exists to implement parts of a system in small quantities today, it is not expected that a standardised and operational service could begin before 1999.

### 3.4.6. Costs

Some very provisional estimates of costs for a professional encoder have been made by LER based upon an estimated quantity of 20 systems. For a 4:2:2-sourced system, the video and audio compression are thought to occupy 8 circuit boards (230 x 160mm) at a cost of about 30kECU (including case and PSU). The multiplexer is thought to occupy 4 boards at a cost of about 15kECU and the modulator is thought to occupy 10 boards at a cost of about 40kECU, both costs include case and PSU.

Cost estimates for the receiving function are currently unknown; but, at present prices for large scale production, silicon costs are roughly 15ECU/cm². In order to account for miniaturisation trends with the passage of time, it has been suggested that chip area decreases by a factor of 30 each decade should be taken as a rule of thumb.

### 3.4.7. Applications of the service to other media

Comments made in sub-sections 2.5 and 3.3.7 are also relevant to applications involving EDTV.

## 3.5. Terrestrial SDTV to stationary receivers with roof-top antennas and portable receivers with small antennas (and if possible to mobile receivers)

### 3.5.1. Features of the SDTV service

The main feature offered by the delivery of SDTV is that more services can be conveyed in one existing UHF/VHF channel than at present, while portable reception remains possible. The exact number of services will depend on the class of receiver which it is aimed to serve. For stationary receivers with roof-top antennas, it may be possible to carry 4 SDTV services in an existing UHF/VHF channels, while for reception on portable receivers, 2 services is a more practical objective. The level of error protection necessary to cover mobile receivers is such that only one SDTV service could be provided. At present there is insufficient experimental evidence to show how rugged a portable reception option can be, and some level of limited `mobility' may be indeed possible.

In terms of the incentive necessary to encourage viewers to buy new digital receivers during a simulcast period, it must be an increase in the number of television services available which would create this demand.

Consumer electronics companies (e.g. Philips and Thomson) are convinced that the availability of a portable service having a quality equivalent to that delivered at present, would provide the incentive for consumers to acquire additional sets.

### 3.5.2. SDTV service characteristics

For an SDTV service, the following target characteristics have been identified by the Working Group:

**Image** The image quality should be equivalent to PAL or SECAM when judged across a representative sample of programme material and with an aspect ratio of 16:9. One estimate is that this corresponds to a source sampled with about 420pels by 575 lines.

**Audio** One stereo channel should be provided with a quality equivalent to that of FM radio. It is suggested that Musicam (ISO/IEC MPEG) operated at a rate below 128kbit/s/channel will provide this quality level.

**Data** A minimum capacity of 64kbit/s should be available, with a higher rate for subscriber addressing, teletext, etc., provided by the dynamic allocation of bit-rate during the transmission of non-critical television scenes. A mean capacity of 0.5Mbit/s should be achievable by this means.

The total bit-rate necessary to provide the SDTV service is thought to be in the range 5 - 6.5Mbit/s/service.

Flexibility for 1 EDTV service instead of 2 SDTV services to portables has already been identified in sub-section 3.4.2.

Graceful degradation in the presence of transmission errors is considered essential for any service which is to be delivered to portables. It is suggested that, of the discrete levels involved in the degradation hierarchy, one is common with the quality offered by an LDTV service.

### 3.5.3. Frequency spectrum requirements for SDTV services

The requirement for spectral efficiency and bandwidth will depend upon whether mobile reception of the service is required. For portable reception, an efficiency of about 1.5Mbit/s/Hz would have to be achieved in order to fit two SDTV services into a single UHF/VHF channel. Such a scheme might employ QPSK modulation at 2bit/s/Hz, with the remaining capacity allocated to error protection.

For mobile reception, an efficiency < 1bit/s/Hz must be sought, resulting in only one service per UHF/VHF channel. A number of possible schemes might be considered for achieving this. One approach is to employ QPSK modulation while allocating a significant overhead for the use of error protection.

### 3.5.4. Prerequisites for implementation of SDTV services

#### 3.5.4.1. Technology

**Studio** SDTV standards for the studio are established for 4:3 and are being developed for 16:9 production.

**Networks** Contribution and primary distribution coding, and networking for PAL exists. Further implementation would have to be undertaken by most broadcasters in order to achieve component networking.

**Secondary** Algorithms for picture and sound compression for the emission system exist, but channel coding and modulation still require development. The demodulator and decoder required for implementation in a consumer receiver remain to be developed.

**Home** A DVCR has yet to be developed for operation with received data at 5 - 6.5Mbit/s. This could have to be transcoded to a higher rate in order to maintain the same quality with intraframe coding. As stated for other services, care should be taken to ensure that any transcoding is achieved with minimal accumulated distortion. Transcoding to a rate of 1.5Mbit/s could be performed; but the quality achievable, once intraframe-coded, must be regarded as extremely low. Computer back-up systems, employing 'Video 8' cassettes, are available on the market with a capacity corresponding to 3 hours of LDTV.

#### 3.5.4.2. Economics

No significant investment in new production equipment would need to be made for an SDTV service. Additional networking and transmitters would however have to be provided. The simulcast of existing analogue services must be performed on some of the available channels if migration to all-digital working is to occur, so the incentive for the purchase of digital receivers must be the presence of some new SDTV services. For the broadcaster however, the economics of the start-up of such new services will probably be less favourable than launching a new service on an existing satellite or cable system, because there will be no existing base of digital receivers.

Consumer electronics companies believe that the increased quality available on portable receivers will result in the rapid sale of a sufficient number of such sets that income for the broadcaster from additional advertising and subscriptions will make a service launch economically attractive.

#### 3.5.4.3. Standardisation

At present no standards exist for the digital coding required to convey SDTV services, however MPEG-2 is expected to provide an SDTV/EDTV quality coding standard by 1993 or 1994. The WGDTB however wishes to ensure that appropriate provision is made in the development of such a standard for flexible or hierarchical operation. No standard is available for the multiplexing of service components or for the modulation associated with emission. These are expected from, for example, the dTTb project in 1994, but would then have to progressed in the EBU/ETSI JTC, CCIR, etc. in the normal way.

#### 3.5.4.4. Regulation

Regulations for defining the use of the UHF spectrum in a transition period from analogue to all-digital broadcasting have yet to be established, and will differ from one European country to another. The participants of the WGDTB who assessed SDTV service provision felt that the Working Group should determine whether any necessary regulation should be at national or European level, and should then provide the guidelines necessary to enable progress towards a system which is eventually harmonised across Europe.

The flexible use of a channel to provide one EDTV or several SDTV services will also need to be clarified by some regulatory or legislative processes in most European countries.

### 3.5.5. Possible date of availability of SDTV services

As discussed above, almost all the technology required to launch terrestrial SDTV services exists with the exception of the emission standard and receiver standard. Taking account of the industrial development time required for standardisation and consumer products (receivers and VCRs) a service could begin as early as 1997 or 1998. As before, however, the date for actual availability is more likely to be determined by the will to launch a new commercial venture, in the context of competition from other services and media, than it is to be determined by availability of the technology.

### 3.5.6. Costs

Cost estimates are similar to those given in sub-section 3.5.6.

### 3.5.7. Applications to other media

Comments made in sub-sections 2.5 and 3.3.7 are also relevant to applications involving SDTV.

## 3.6. Terrestrial LDTV services for mobile receivers

### 3.6.1. Features of the service

The key feature of LDTV (Limited Definition TV) delivery is that it is rugged enough to be received in moving vehicles while offering a number of services within a single UHF/VHF channel. The trade-off is that the data-rate conveyed is low, and only able to support a picture resolution of ¼ that of CCIR Rec. 601. No market studies have been conducted to identify the desirability of such services by the consumer. It is unlikely that alone such services could provide the incentive required for a migration towards digital television, but the possibility exists (given available spectrum) for these services to be provided in addition to other DTB services, in order to serve the mobile customer.

### 3.6.2. LDTV service characteristics

**Image** The image quality should be capable of conveying the CIF (Common Intermediate Format) image format (360pels by 288 lines) with minimal degradation. The performance expected is that of the MPEG-1 algorithm.

**Audio** One stereo service should be provided with a quality which is equivalent to that of FM radio. It is thought that this could be provided by MUSICAM/ IEC/ISO MPEG coding operating at a rate below 128kbit/s/channel.

**Data** A minimum capacity of 64kbit/s should be available, with a higher rate for subscriber addressing, teletext, etc., provided by the dynamic allocation of bit-rate during the transmission of non-critical television scenes. A mean capacity of between 0.128Mbit/s should be achievable by this means.

The total bit-rate required for the delivery of an LDTV service is 1.5Mbit/s, where 1.15Mbit/s is allocated to the video. Recent studies on picture quality at low bit-rates suggests that the MPEG-1 algorithm is not completely transparent to material sourced at the CIF, but that significant distortion is introduced on critical pictures. The response of the viewer to such quality variation is not currently known.

Neither graceful degradation nor hierarchical quality levels are considered necessary for LDTV services because of their ruggedness. Flexibility can however be provided with 'sound radio' services, one LDTV service being equivalent to perhaps 5 or 6 'sound radio' services.

### 3.6.3. Frequency spectrum requirements for LDTV services

The efficiency of the system as a carrier of bits within a given bandwidth will be low, because of the overhead necessary to make the signal rugged. An efficiency of less than 1bit/s/Hz is needed, resulting in the ability to convey 3 LDTV services within a single UHF/VHF terrestrial channel.

Transmission power and loss of capacity with interference conditioning applies as with other DTB services, here however the use of single frequency networking may be essential to achieve the national coverage without retuning, and at an acceptable power level.

### 3.6.4. Prerequisites for implementation of LDTV services

#### 3.6.4.1. Technology

The technology for LDTV services is available. Current studio production equipment, and contribution and primary distribution codecs already convey qualities in excess of what is required for LDTV. The MPEG-1 source coding could be used for emission, and a suitable modulator could be that used for DAB. A domestic digital recorder is still required, but the bit-rate necessary is not thought to be high enough to cause any significant technical difficulties.

#### 3.6.4.2. Economics

The economics of mobile television services remain to be established and it is considered questionable whether the allocation of an 8MHz UHF channel to 3 LDTV services makes economic sense. To serve mobile receivers, a national single frequency network is desirable and this may not be possible in all European countries, especially during the transitional period from analogue to digital (when only `taboo' channels may be available). Even where it is feasible, a nationally clear frequency will be an extremely valuable resource which could be better employed carrying other DTB services.

It is hard to see the availability of rugged mobile services providing the unique incentive for migration from analogue to digital reception since the concept of simulcasting requires at least an equivalent quality digital service to the analogue one which it seeks to replace. Perhaps the introduction of digital mobile services should be considered only after analogue broadcasting has ceased and spectrum replanning for all-digital use of the spectrum is being implemented.

#### 3.6.4.3. Standardisation

The required standards of MPEG-1 and DAB could be expected to be complete in 1993 for submission to the ETSI/EBU JTC.

#### 3.6.4.4. Regulation

As with other services, flexibility of operation to provide different services within a single channel will require adjustment to legislative or regulatory practices in European countries.

### 3.6.5. Possible date of availability of LDTV services

It is thought that the technology required to implement LDTV emission, receivers and domestic recorders is available today. The standardisation of such a system would take several years however.

### 3.6.6. Costs

Beyond the costs of implementing a new television service using existing technology, only the special requirement for a single frequency network with associated low power transmitters has been identified.

Receivers must be available at a price which makes them attractive at the time of the launch of an LDTV service. No estimates of the costs of such receivers is available at present.

### 3.6.7. Applications of the service to other media

Applications of a LDTV service to other media are likely to be very limited because of the low picture quality, which is constrained by the particular problems of mobile reception. Compatibility with some storage media such as laser disc, where there is also a capacity limitation, may be sought. This could provide an additional source of entertainment for the mobile television viewer.

## 3.7. Consequences of flexibility and scalability on different services

### 3.7.1. Introduction

In Section 3.1 to 3.6 the desirability of a wide range of service options ranging from HDTV quality to LDTV quality was identified, together with the need for flexibility to reconfigure the available bit-rate between different service options on a programme-by-programme basis. Furthermore, in the same Section it was recommended that, in order to cope with varying propagation conditions, the quality of picture and sound services should degrade gracefully as the error-rate increases.

Whilst in principle both of these requirements can be met in a digital system, this flexibility must in practice be purchased at some cost, compared with a system which would target only <u>one</u> quality objective, and which was designed to operate under a fixed set of propagation conditions. This cost may be expressed in several ways including increased bandwidth, increased transmitter power, or decreased picture quality for any given option. In addition, this flexibility will almost inevitably increase the complexity, and perhaps the cost, of the decoders.

Two broad approaches to achieving the desired flexibility have been identified:

a) Multicast: Here the same programme is broadcast in different "bit-streams" (or sub-channels) each of which is independent. Each bit stream conveys a certain resolution level, and is appropriately protected against errors by channel coding, selected to give the required level of ruggedness.

b) Hierarchical: A single bit stream is broadcast which contains, embedded within it, information needed for all service options and levels of ruggedness. The bit-stream is used, totally or partially, for decoding purposes, depending on the level of performance required and the error-rate which exists on the channel.

Hierarchical systems or multicast systems are considered terminologically to be the two sub-categories of multi-layer systems.

The basic tools to be worked out in a multi-layer scheme largely consist of scalability and flexibility. They are defined in the following sections.

### 3.7.2. Definition of terms

The block diagram of the transmitter part of a hierarchical system is illustrated on Fig. 3.2. It includes 3 modules: source encoding, multiplexing, and channel coding/ modulation. The basic techniques implemented in these different modules are as follows:

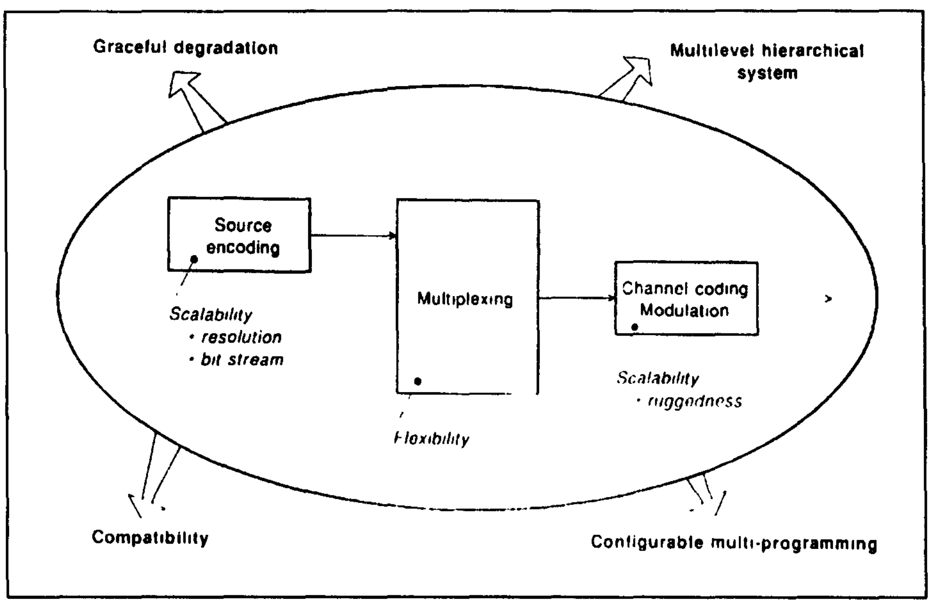



*Figure 3.2: The hierarchical approach: issues and tools*

a) Scalability (in the source encoder) *[9]*

Scalability here is the capability for a source encoder to organise encoded data in order to allow decoding at several standards and at several quality levels. Two different kinds of scalability can be identified:

- resolution scalability,
- bit stream scalability,

Resolution scalability means that the video can be decoded at different resolutions or sizes directly. This is distinct from decoding a full size image and then condensing it. Resolution scalability may be spatial, temporal or both.

Bit stream scalability means that some of the bits in the bit stream can be neglected, but decoding of a useful picture is still possible. This implies that the bit stream is formatted in such a way that bits within can be ignored. Bit stream scalability may be obtained by:

- scaling spatial resolution,
- scaling temporal resolution,
- scaling coding noise.

These types of scalability can be correlated to a greater or lesser extent, and in some cases can be even identical. As far as future digital TV/HDTV broadcast systems are concerned, two basic requirements related to scalability exist:

- compatibility,
- graceful degradation.

These are explained in more detail later in this report.

b) Flexibility

The flexibility is the capability of broadcasting, in a channel, one or several programmes belonging to a (limited) set of configurations. The flexibility is performed at the multiplex level, and requires a flexible multiplexing scheme, supported by appropriate signalling facilities.

c) Scalability (in channel coding and modulation)

The purpose of sophisticated channel coding and modulation is to improve the ruggedness of a bit stream with techniques adapted to the characteristics of the transmission channel. The ruggedness scalability is the capability for those techniques which improve "bad channel" transmission characteristics (i.e. mobile, portable) to be also efficient when used for a "good channel" (i.e. fixed) out of its operational limits. In other words in technical terms, the ruggedness scalability should allow the exchange of protection in a Rayleigh channel against noise on Ricean fading channel.

d) Hierarchical system

The concept of hierarchy, from a service point of view, has to be specified according two axis:

- the picture resolution,
- the class of receiver (fixed, portable, mobile).

The two hierarchical systems which have been proposed for the most serious study are as follows.

- The first one would have:
  - three levels of picture quality: HDTV, EDTV and SDTV. Note that there are only two different scanning standards since EDTV and SDTV have the same number of lines.
  - two classes of receiver: fixed and portable.

- The second one would have:
  - three levels of picture quality: EDTV, SDTV, LDTV,
  - three classes of receiver: fixed, portable and mobile.

The hierarchy in terms of picture resolution (or quality) makes use of the fact that information relating to low resolution pictures is nested in the bit stream describing a higher resolution picture. This enables, for instance, a conventional definition TV receiver to decode only the low resolution information from HDTV broadcast data. This can be described as (HDTV to TV) downward compatibility *[2]*.

A hierarchy dealing with several classes of receiver is called hereafter hierarcrchy of services, and is defined as follows.

The hierarchy of services consists in broadcasting, in the same channel, one programme to different types of terminal: fixed, portable and mobile. The hierarchy of services can be supported by the hierarchical coding, but it must also operate ruggedness scalability with different types of channel coding (or modulation) adapted to fixed, portable or mobile channel characteristics *[4]*.

If, as an example, a three level hierarchy were to be realized with EDTV, SDTV and LDTV members of this hierarchy, the following proposal would apply. The basic assumption could be that individual members were aiming at specific receiving conditions, as follows:

LDTV -> mobile

SDTV -> portable

EDTV -> fixed

where the global bit-rate of the channel is split into:

a - LDTV encoded data

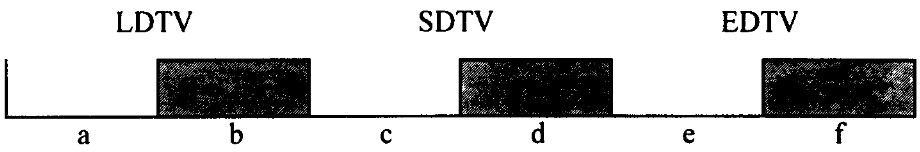
b - "mobile" channel coding on "a"

c - SDTV complement encoded data

d - "portable" channel coding on "c"

e - EDTV complement encoded data

f - "fixed" channel coding on "e"




It can be seen in Fig. 3.3 that in the hierarchy of services, efficient use is made of the LDTV and SDTV components of the video signal to increase the area of fixed and portable reception (with a reduced quality). This possibility is worth examination as a basis for "graceful degradation".

e) Upward and downward compatibility

Upward and downward compatibility (Fig. 3.4) refer to the broadcast systems, in which different picture formats are used for the encoder and decoder.

Upward compatibility means that a higher resolution receiver is able to decode pictures from the signal transmitted by a lower resolution encoder (e.g. TV-signals in HDTV-receiver). In this case two methods of display are possible:

- The lower resolution picture is displayed as a screen 'window'.
- The lower resolution picture is upconverted to a full size picture. However, this up-conversion need not be part of the decoding process.

Downward compatibility means that the lower resolution receiver is able to decode pictures from the signal, or parts of the signal (embedded bit stream) transmitted by a higher resolution encoder. Two methods of downward compatibility can be distinguished:

- The decoder reconstructs the entire picture at a lower resolution
- The decoder reconstructs a part (window) of the incoming picture. For broadcast applications this case would be of no value.

It should be noted that the data rate of the embedded bit stream is not necessarily fixed.

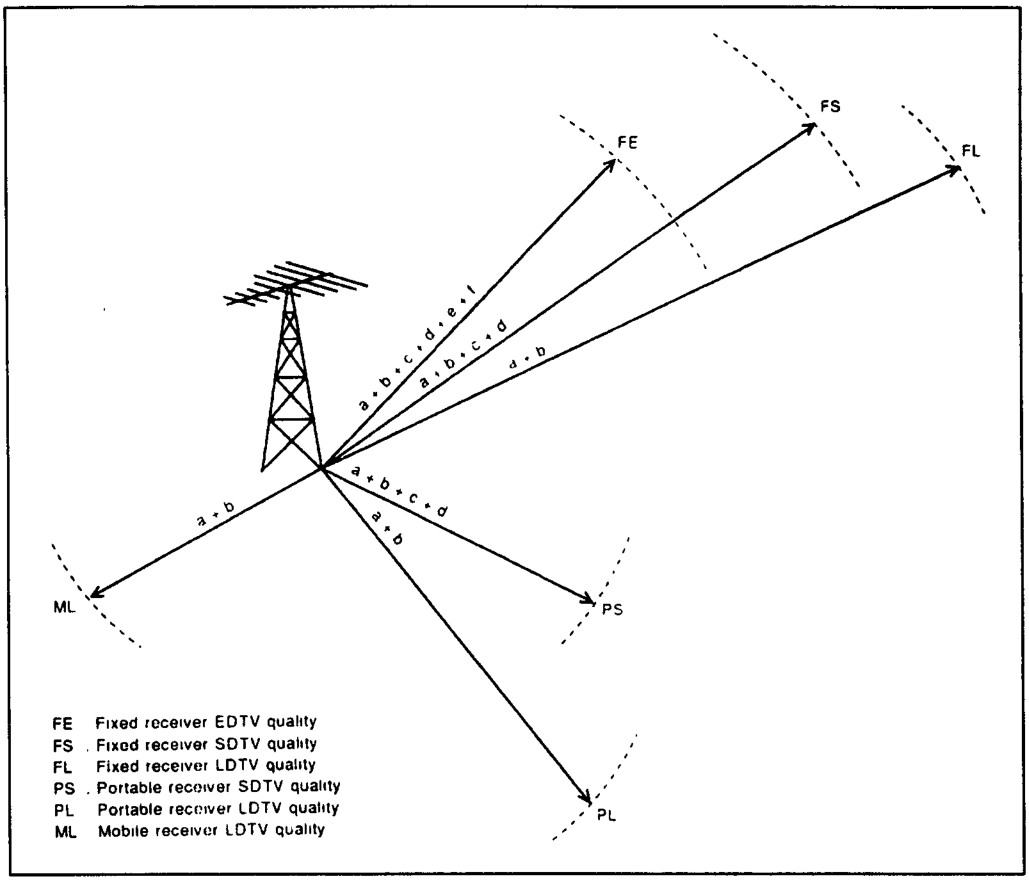



*Figure 3.3: Receiver type quality and coverage area*

| Transmitter | Receiver |
|---|---|
| HDTV ——————————>HDTV | |
| SDTV ——————————>SDTV | |

*Figure 3.4: An illustration of upward/downward compatibility*

f) Graceful degradation

Graceful degradation *[3]* means that some level of picture quality is available in adverse receiving conditions (e.g. in buildings or at the borders of the service area). This can be achieved by implementing different layers of error robustness into the broadcast signal.

These different layers of error robustness can be obtained by using a multi-resolution modulation (non-uniform modulation) combined with hierarchical coding or different modulation schemes and/or different types of FECs for different proportions of the signal. Therefore graceful degradation requires bit stream scalability, and the data rate in the different protection layers has to be fixed. Three layers would be used in a system as shown in Fig. 3.5. These layers have progressively higher robustness and represent spatial or spatial temporal low pass information of the video signal.

Even if resolution scalability is not mandatory for graceful degradation, it seems relevant to use the multi-resolution levels as intermediate steps in the graceful degradation process. With the multi-resolution levels as backbone, it is technically possible to refine the graceful degradation mechanism with intermediate levels based on a bit stream scalability mechanism (Fig. 3.6).

g) Configurable multi-programming

The configurable multi-programming facility would consist in broadcasting, in one channel, several programme configurations at different hours of the day (for example: 1 multi-resolution programme (HDTV + EDTV + SDTV) at prime time, and several SDTV programme channels in off-peak hours).

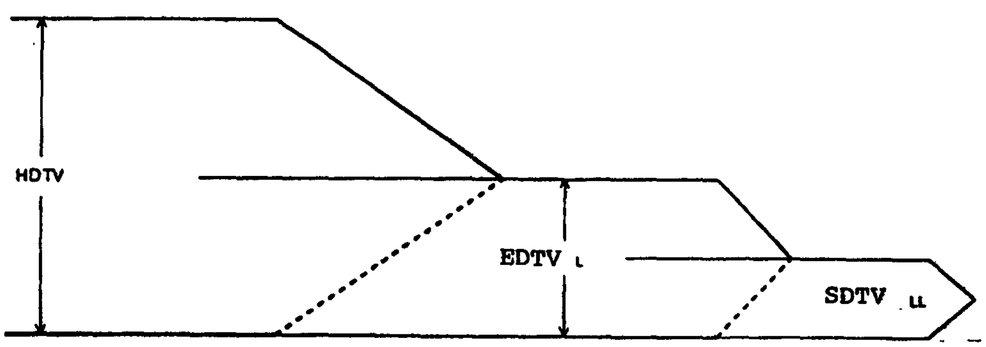



*Figure 3.5: Partioning of data rates for HDTV signals*
*L LL = spatial/temporal low frequency parts with high protection*

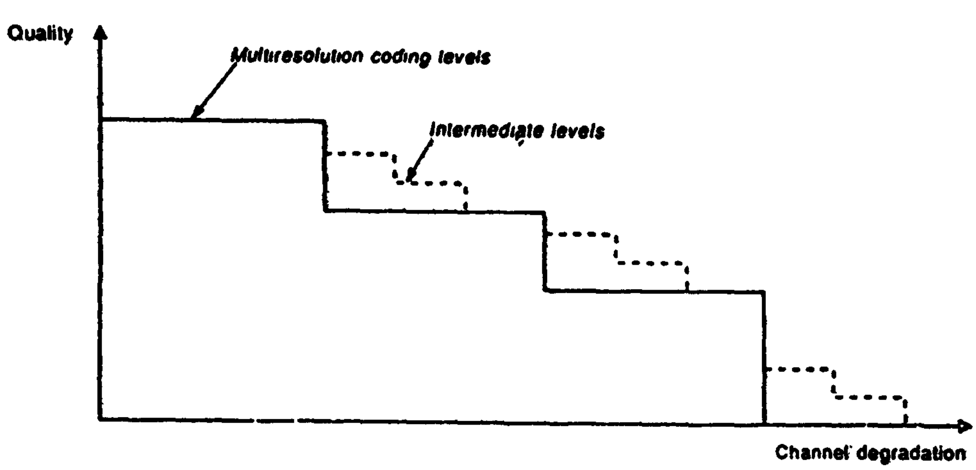



*Figure 3.6: Resolution scalability and graceful degradation levels*

### 3.7.3. Comparison between hierarchical coding and multicast

Appendix 4 to this report outlines concepts for the implementation of hierarchical coding.

#### 3.7.3.1. Assumptions for comparison

For the purpose of arriving at a useful comparison between the hierarchical and multicast options, it is necessary to make a number of assumptions. It is important to stress, however, that these are only assumptions, and that they will need to be fully justified. It is vital that these decisions, which will affect the services offered to viewers of the new digital system, should not be made by default.

The assumptions made in this analysis are that viewers have access to a number of options, which are as follows.

a) Viewers have at least one option that has a high level of displayed picture quality.

b) Viewers have at least one option that is available at low cost.

c) Viewers have at least one option that is highly rugged.

For the purposes of this document, it will be assumed that option (c) does not require the provision of graceful degradation, and that any graceful degradation associated with options (a) or (b) can be combined with the lower levels of the service hierarchy. Thus the member of members of the total combined hierarchy can be limited to three at most (and two, if options (b) and (c) can be combined).

#### 3.7.3.2. Available technical parameters - multicast

It will be assumed that OFDM techniques are employed, with suitable guard intervals to give adequate immunity to multipath effects. The calculations assume that 16 QAM is used for the high quality option, with QPSK being used for the high-ruggedness option, and we ignore, for the time being, any other options. We assume also that the error protection coding and spectral conditioning (to optimise the compatibility of the digital signal with existing PAL/SECAM signals sharing the same UHF channels) are common to the two options. The ratio of the channel bandwidths of the two options therefore depends on the ratio of their bit-rates after picture coding (given the spectral efficiency of the modulation). Assuming a value of about 4:1 for this ratio gives an approximate ratio of 2:1 between the bandwidths used for the two services. This will be reduced slightly if account is taken of the need to transmit the associated sound channels (common to both the HDTV and 625-line services), and also if 32 QAM is used instead of 16 QAM for the HDTV service.

Thus if the bandwidth required by the HDTV service transmitted on its own is "R", the bandwidth required by the multicast HDTV and 625-line services will be between 1.5R and (say) 1.7R. If no commonality can be achieved between options (b) and (c), so that a three-level system must be transmitted, then the bandwidth must be increased still further. The magnitude of this increase will depend, once again, on the requirements specified for options (b) and (c). If less than 625-line quality is acceptable for either of these two services, then it might be possible to constrain the further increase in bandwidth within reasonable bounds. In particular, the use of the MPEG-1 coding algorithm would mean a bit-rate of only 1-1.5 Mbit/s. If, however, full 625-line pixel density quality is needed for both options, then the overall bandwidth required for a service with three levels of hierarchy could be expected (by extending the above arguments) to be between about 2R and 2.6R.

### 3.7.3.3. Available technical parameters - hierarchy

At present, the bit-rate reduction efficiency of hierarchical coding methods is not as high as that of the best single-resolution algorithms (e.g. motion-compensated hybrid DCT). This may be due at least in part to the fact that the predictions made for the different levels do not complement each other. Thus, a significant proportion of the bit-rate allocated to the high quality picture is used to undo the compromises that were accepted for the equivalent 625-line version. Further development of such layered algorithms is therefore necessary, and a promising approach is the study of suitable sub-band algorithms.

For the purposes of this document, it will be assumed that the bit-rate required by the 625-line service [option (c)] is similar to that for the equivalent multicast option. Taking account of the comments noted above, however, the picture quality may not be quite as high. The bit-rate for the option (a) service should be less than the equivalent multicast option, thus allowing an overall saving to be achieved.

Assuming the same modulation systems as outlined earlier (16 QAM for the HDTV service, QPSK for the rugged option), if the layered coding algorithm were to achieve 100% efficiency, the bandwidth required by the option (c) service would be 2 x 0.25R[4] = 0.5R, and the additional bandwidth required for the option (a) service would be 0.75R. The combined bandwidth for both services will become 1.25R. As before, the effect of the use of 32 QAM for the HDTV picture would be to increase the proportion of the bandwidth devoted to the high ruggedness picture; the combined bandwidth for both services will then become about 1.4R (although, of course, the value of R will be lower in this case by virtue of the higher-level modulation system). Making an allowance of 20% for the lower efficiency currently achieved by layered coding algorithms, these two figures become 1.5R and 1.7R respectively. Introducing a third level to the hierarchy could be expected to result in a smaller increase in bandwidth than for the equivalent multi-cast option. A typical value that might be expected for the overall bandwidth of the three-level hierarchy could then be, for example, between 1.6R and 1.9R.

### 3.7.3.4. Preliminary analysis: bandwidth cost for hierarchical approach

The simplest scenario for future digital television services would be the use of an HDTV "stand-alone" system. This would also be the most economic use

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4 The HDTV to 625-line ratio is presumed to be 4:1 in this hierarchical scheme. This is not completely reached today as the 2:1 ratio is the present state of the art but this may not be unrealistic in the future.

of RF spectrum. It might, however, not be possible to achieve all of the desired services. In particular, this route would appear to rule out the possibility of a low-cost option (unless the existing PAL and SECAM transmissions are used for this purpose). The bandwidth required for such a "stand-alone" digital HDTV system would be, by definition, R.

The next simplest scenario is the use of a "multi-cast" of different digital services. This is, in principle, not an efficient use of the available of spectrum, although in practice the inefficiency might not be worse than that of other more complex hierarchical alternatives, especially if the number of levels in the hierarchy is fairly limited. It would lead to relatively simple circuitry in all receivers, since each receiver need only decode the data for its appropriate service, and it would provide minimum constraint on the development of future enhanced services (since it would be possible to turn unused options "off" without affecting the other levels of the hierarchy). The bandwidth required for such a "multi-cast" system would be between 1.5R and 1.7R for a two-level hierarchy, and between 2.0R and 2.6R for a three-level hierarchy.

The third option would be the use of "layered" coding to achieve a hierarchical system. This is the option about which the least is known at the present time. A two-level hierarchy might require a bandwidth of between 1.5R and 1.7R; a three-level hierarchy might require a bandwidth of between 1.6R and 1.9R. It should be stressed that these figures have been derived in an extremely subjective way, and are subject to a high degree of uncertainty.

### 3.7.4. Conclusion

In comparing a flexible system which includes the desired features of multiple resolution services, graceful degradation, and the facility to reconfigure the bit-stream to convey multiple services, three factors have to be taken into account:

i) the increased source bit-rate needed to convey the multiple resolution picture,

ii) the additional error-protection, or more robust modulation system, needed to convey the lower resolution pictures with the required level of ruggedness for the intended operating conditions and/or to provide graceful degradation,

iii) the overhead imposed by a flexible multiplex.

The last factor, that of the flexible multiplex is, in principle, relatively small compared with the other two factors.

The cost, in terms of increased bandwidth requirement, increased transmitter power, or reduced picture quality or ruggedness for the various service options depends upon the proportion of channel resources (bit-rate and transmitter power) allocated to each service option and to the particular source-coding, channel coding and modulation techniques employed. All of these are still under study. However, based upon very preliminary analysis of some example systems the following sample results are indicated:

a) For a two-level system using the multicast approach to deliver high-quality pictures to fixed receivers and standard-quality pictures to portable receivers, the overhead for this compared with a single resolution/ruggedness service amounts to between 30-40% of the available channel resources.

b) For a three-level multicast system the corresponding overhead amounts to more than 50%.

c) For a two-level hierarchical approach, in which the required resolutions are both conveyed embedded in the source bit stream by using scalable source coding techniques the overhead is similar to that needed for the multi-cast approach. However, alternative source coding and modulation techniques still under development may reduce the overhead in this case to 10-20%.

d) For a three-level hierarchical approach using scalable source coding the overhead in the example system considered amounted to up to 50% of the overall channel resources. However, as with the two-level hierarchical approach alternative techniques still under study may reduce the overhead to 10%-20%, as in the two-level case.

These results encourage continuing the investigations of hierarchical systems, but it is important that an evaluation is made of the overall merits and demerits of such an approach. This evaluation should be done through the measurement of the benefits/costs ratio. The benefits are: performance, better quality of services, better use of investments for all the partners playing a role in the setting up of a new broadcast system, easier service introduction. The costs are related to: bit-rate, receiver complexity, loss of picture quality in comparison with a dedicated system, and R&D effort needed.

### 3.7.5. References

1. WGDTB 1018: "Hierarchical multi-resolution coding for digital broadcasting of television"
2. WGDTB 1020: "A hierarchical multi-resolution coding algorithm based on subbands"
3. WGDTB 1034: "Proposal for a hierarchical HDTV system to be developed for DTB in Europe"
4. WGDTB 1035: "Comments on hierarchical coding and hierarchy of services"
5. WGDTB 1044: "Requirement for MPEG-2 based video codecs to be used in digital video broadcast services"
6. WGDTB 1049: "Possible parameters for a hierarchical system for digital television"
7. WGDTB Temp 7
8. "Coding of motion picture and associated audio" ISO/IEC JTCI/SC29/WG11.
9. dTTb Annual Review Report.
10. Doc. TG-CMTT/2-SRG-086: "Provisional conclusions of SRG on the basis of demonstrations of hierarchical compatible systems"

## 4. DTB SYSTEMS PROPOSED BY THE WORKING GROUP

The WGDTB has analysed potential options for future digital terrestrial television in Europe.

From this work the group proposes a study path to be followed, in a coordinated way, by European digital terrestrial television projects.

The working hypothesis of the current discussion was that a single current terrestrial television channel (with about 7.5MHz of available bandwidth) would be the basic unit of the new future digital television services. If a wider bandwidth were possible (for example one and a half current TV channels) there would be consequences for the practicality and potential quality of the system. However, there is insufficient planning evidence that this would be feasible at the moment, and all that follows assumes that only a single channel is available.

To allow the following proposals to be understood, some short-form definitions are given first. These are also given earlier in the text.

**HDTV quality** (Section 3.3) (High Definition Television) where the potential exists for the delivery of a picture which is subjectively identical to the interlaced HDTV studio standard. Quality shall remain consistent with this for a given percentage of television programme material (where this percentage is in the high nineties but is yet to be identified).

**EDTV quality** (Section 3.4) (Enhanced Definition Television) where the potential exists for the delivery of a picture which is subjectively indistinguishable from the 4:2:2 level of CCIR Rec. 601. This quality shall be maintained for a given percentage of programme material (which will be in the high nineties but is yet to be identified).

**SDTV quality** (Section 3.5) (Standard Definition Television)where the quality is approximately equivalent to that of current PAL or SECAM. This equivalent quality may be achieved from pictures sourced at the 4:2:2 level of CCIR Rec. 601 and subjected to processing as part of the bit-rate compression. The result should be such that when judged across a representative sample of programme material, subjective equivalence to PAL/SECAM is achieved.

**LDTV quality** (Section 3.6) (Limited Definition Television) where the quality is equivalent to that obtainable from the MPEG-1 system which operates on a source resolution approximately ¼ of the 4:2:2 level of CCIR Rec. 601. This quality is considered by some to resemble that of VHS (albeit over a relatively low proportion of programme material).

Three types of receiving situation can be foreseen for future digital services: the circumstance in which a fixed rooftop aerial is used and the receiver is stationary, the circumstance where the receiver is portable but not mobile (i.e. does not need to be viewed on the move), and the circumstance where the receiver is mobile.

As will be seen, in the following proposals, HDTV and EDTV receivers are assumed to need to work with rooftop aerials. SDTV receivers are assumed to work in a portable environment, and LDTV receivers are assumed to work in a mobile environment.

## 4.1. First proposal: a multilevel approach based on HDTV with additional reconfigurability (P1)

### 4.1.1. Description of the proposed system

Basic system elements are given in Fig. 4.1.

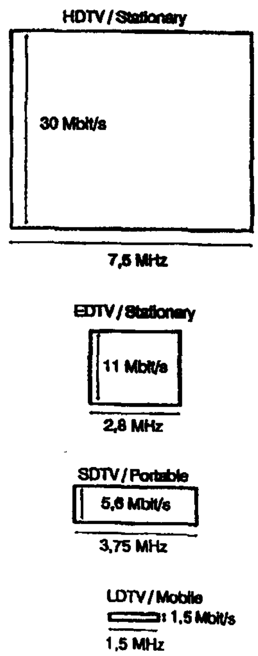


*Figure 4.1: Potential digital services, with source data rates and bandwidths*

The first proposal of the WGDTB (Fig. 4.2) is a three-level system which should be studied, and if practical, developed. The system should allow, within one terrestrial channel, simultaneous reception at HDTV quality via fixed rooftop aerials, reception at EDTV quality via fixed rooftop aerials, and reception at SDTV quality via set-top (or built-in) aerials. In the last case, SDTV reception is designed for portable receivers.

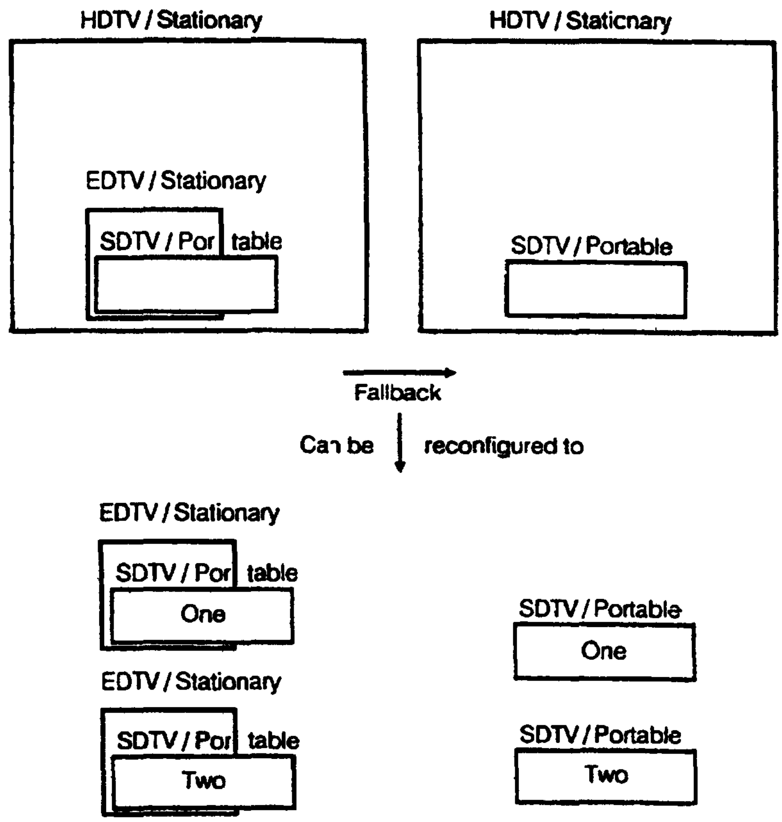


*Figure 4.2: First proposal*

The group recognizes that this objective may not be technically possible and a fall-back needs to be defined. In this case, the target should be a two-level hierarchical system which allows simultaneously, stationary reception at HDTV quality via fixed rooftop aerials, and portable reception at SDTV quality via set-top (or built-in) aerials. In this latter case, SDTV is designed for portable receivers.

Although many European broadcasters believe that any new digital system must provide HDTV quality to be useful, they accept that the idea of allowing different configurations in a system is worth evaluating. Lower quality multi-channel broadcasting may be appropriate in some parts of the world. Furthermore, it may be possible to use an EDTV/SDTV standard common to other transport media.

The WGDTB therefore believes that studies should be made of the cost and other implications associated with allowing the system to be reconfigurable.

If the three-level system mentioned above proves practical, the alternative configuration should be to carry, in the same channel, two independent EDTV services, receivable on stationary receivers via rooftop aerials, which are also simultaneously receivable in a portable environment at the SDTV level, via a built-in or set-top aerial.

If it proves necessary to concentrate on the two level hierarchy mentioned above, the alternative configuration should be to allow, in the same channel, two SDTV quality level services, receivable in a portable environment.

### 4.1.2. Probabilities for the practicality of the first proposal

There are three key technical areas or technologies which are important for the potential realization of the different elements of the first proposal:

- channel coding
- source coding
- spectrum availability

Spectrum availability will depend to some extent on circumstances in individual European countries, but the requirement for spectrum would apply for any proposal. The proposal is based on conventional 8MHz channel spacing, and may potentially be implemented such that single frequency networks (SFN) are possible. It is not expected to be incompatible with the current terrestrial broadcasting plan. The proposal is also based on the assumption that sufficient power can be radiated in the spectrum available, to convey signals having the order of modulation required, while avoiding interference to existing analogue services. For high-order modulation schemes this may impose a significant limitation to practicality.

Channel coding, especially the modulation of the digital signal on to the carrier and the demodulation in the receiver, is one of the most critical technologies for this proposal. The practicality of an appropriate modulation scheme will be one of the main prerequisites for the embedding of services aimed simultaneously at rooftop aerials and portable reception, as well as for the reconfigurability. The main areas of concern are:

- the practicality of the high-level modulation scheme needed to transmit HDTV and EDTV signals with a spectral efficiency (bit rate/bandwidth) of approximately 4 bit/s/Hz or more.
- the necessary difference between the modulation schemes for the reception by stationary receivers with rooftop aerials and that for portable reception. The latter needs to be very rugged (2bit/s/Hz).
- the need to keep the format of transmitted signals of the normal service and the reconfigured services compatible so that transmitters and receivers will be able to handle both.

There are several potential approaches which may be used for the proposal: Quadrature Amplitude Modulation (QAM) techniques with non-uniform levels, Orthogonal Frequency Division Multiplex (OFDM) with different modulation schemes for different carriers, or Spread Spectrum Code Division Multiplex

(SS-CDM) with different weightings for different data streams.

Source coding and the correct multiplexing of the different levels of image information for the different service qualities (HDTV/EDTV/SDTV) is the third critical aspect of the first proposal. Whether or not it will be possible to make HDTV, EDTV and SDTV quality images available from the one received signal in such a way that the overhead data rate will mean an acceptable quality reduction of the HDTV image is not clear. Further, the electronics, particularly in the (lower priced) EDTV receiver, has to be inexpensive. The SDTV signal will need to be embedded or simultaneously broadcast, and will have to be usable by inexpensive portable receivers.

If the MPEG-2 system is designed with appropriate objectives, it could be used for DTB.

Source and channel coding would be interrelated in such a way that reception under good carrier-to-noise-ratio (CNR) conditions enables HDTV images to be displayed, and reception with low CNR automatically reduces the image quality to SDTV.

### 4.1.3. The appeal of the first proposal to the different segments of the marketplace

#### 4.1.3.1. Consumers

From the consumer's point of view there could be seen to be four potential selling features for a DTB system:

- better (HDTV/EDTV) images and high-quality sound in the living room,
- more programmes,
- portable reception with stable quality,
- mobile reception with stable quality.

The first proposal would fulfil the first three of the four.

Additional data capacity can be included in the DTB signal multiplex, such that new or enhanced auxiliary services may be offered.

On the other hand some of the technical features for future television services conceived to date are not part of the first proposal, e.g., mobile reception, 3-D images, interactive television. Within the marketing module of activity, in a possible European DTB-project, further investigations will need to be made of the relative attractiveness of such services.

#### 4.1.3.2. Broadcasters

Public broadcasters are interested in television systems that are able to deliver services to virtually all the viewers in their service areas. These systems should be able to deliver to the viewers in a transparent way any image and sound quality, that are generated within their studios. The system should be capable of transmitting HDTV images, and should be able to distribute images at the existing quality levels to all the receivers in use, irrespective of their display standard. No existing terrestrial programme maker/broadcaster would welcome a system that would effectively increase the number of competitive programmes his viewers could receive. Finally, the change from PAL or SECAM to DTB has to be affordable and has to be based on a sound business plan.

The first proposal offers to the broadcaster a new distribution standard for terrestrial broadcasting and cable, and to portable television receivers. Due to the multi-level approach, all existing source quality levels can (in the three-level-proposal) be delivered to the viewer. Thus, in the first proposal, a universal approach has been designed. The transmission of multiple programmes is also possible, in principle, in the reconfigured system. It therefore can be controlled, and is an optional ingredient of the first proposal.

On the other hand, a DTB system, of whatever form, will make the installation of new transmitter equipment systems necessary, which will inevitably be costly. The business opportunity for existing broadcasters may only be in the fact that, at some point in time, the only way to reach a part of their viewers is via DTB.

#### 4.1.3.3. Industry and dealers

The main value to industry and dealers in developing and promoting DTB is the interest of their consumers to buy new equipment. Therefore, as long as 4.1.3.1 is taken care of, and over-sophistication of the standard and the related costs in the receivers is avoided, the first proposal should be in the interests of both industry and dealers.

#### 4.1.3.4. Regulators and network operators

The main interest of both parties is expected to be in making the most efficient use of the frequency spectrum. Either the prospect of more programmes in the part of the frequency spectrum currently allocated to television broadcasting, or a reduction of that part of the spectrum in the longer term, is the most attractive element of DTB for them.

The first proposal, with all its flexibility, will give regulators and network operators the necessary tools to optimize their services or their spectrum allocation.

### 4.1.4. A possible introduction strategy

The introduction possibilities for the first WGDTB proposal depend mainly on the additional frequency spectrum that can be allocated to DTB, in the period before today's services in PAL and SECAM can be discontinued. It is probably true that unless at least five to ten programme services are available from the start of the service, irrespective of the technical quality, DTB may be insufficiently attractive to succeed.

If the spectrum requirement needed can be met, different countries in Europe and different broadcasters/programme companies could use the first proposal according to their own needs. It may be assumed that DTB services will be operational by the end of the century. At that point in time the studio standard used in nearly all European production facilities is expected to be either 625 lines/components or HDTV.

Many broadcasters will want to use DTB either to:

- start full HDTV-quality broadcasting at the very beginning of DTB-transmissions, since all these programmes will be able to be received by all DTB-receivers at their respective quality levels, i.e., HDTV, EDTV or SDTV, or,

- broadcast their existing programmes in "simulcast" mode[5] (effectively at EDTV quality) and upgrade their studio facilities to HDTV later on, possibly tracking the market penetration of DTB receivers for HDTV without interruption of the DTB service, or they will want to include parts of their programming in HDTV, according to their specific situation, or,
- start the terrestrial broadcasting of an additional programme (for example one that up to that point in time, was distributed only via satellite) in EDTV-quality.

In certain countries, where regulators and/or network operators are interested in broadcasting the existing and/or more programmes terrestrially, the use of DTB may start based on the "reconfigured" version of the first proposal, i.e., with two EDTV programmes within one of the existing 8MHz-channels.

The first proposal thus offers several introduction scenarios to fit respective needs in certain countries, or of certain partners in the marketplace.

## 4.2. Second proposal: a multilevel approach based on EDTV and SDTV and mobile. LDTV reception with additional reconfigurability (P2)

### 4.2.1. Description of the proposed system

A second proposal was also thought worthy of study by a number of members of the WGDTB (Fig. 4.3). This is a dual configuration system which will allow either of the configurations below.

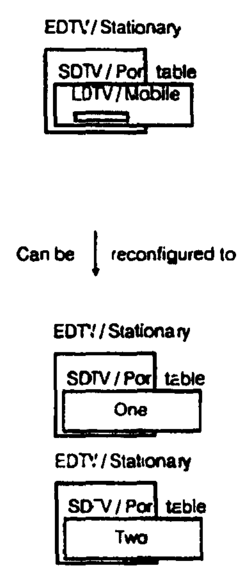


*Figure 4.3: Second proposal*

- Simultaneous reception of a single service at EDTV quality with a FIXED receiver via a rooftop aerial, SDTV quality in a PORTABLE environment and LDTV quality in the MOBILE environment.

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[5] In this context simulcast means broadcasting the same programme content simultaneously in two different channels.

- Simultaneous reception of two independent services at an EDTV quality level in a FIXED environment and at SDTV quality in a PORTABLE environment.

### 4.2.2. Probabilities for the practicality of the second proposal

By and large, the same arguments apply as described in 4.1.2 for the first proposal. Again spectrum allocation cannot be considered a unique factor for the second proposal. Channel coding again is expected to be one of the most critical areas. The second proposal includes the transmission to three different types of receiving conditions; stationary, portable and mobile, and three different types of modulation schemes are expected to be necessary. In some respects, the second proposal is even more critical, in terms of a technical challenge, than the first proposal.

Source coding and the correct multiplexing is just as critical as for the first proposal. But it should be noted that the second proposal is based on a three level hierarchy and that a fallback may be less attractive. Since mobile reception is the main difference in the target of the second, compared to the first, proposal, a fallback towards a two-level hierarchy would imply, that only SDTV and LDTV, or EDTV for fixed and SDTV and LDTV for mobile and portable reception, would be part of the system. Both the appeal to the market of this approach and the possibility to reconfigure such a fallback system need study.

### 4.2.3. The appeal of the second proposal to the different participants in the marketplace

The second proposal differs from the first proposal only in one point: it does not include HDTV, instead it includes LDTV for the mobile receiver. A reconfiguration is included, and after reconfiguration the first and the second proposal are identical. In the following, therefore only the arguments relative to the difference between proposals 1 and 2 will be presented.

#### 4.2.3.1. Consumers

Up till now, mobile reception of TV programmes has not been an important or possible part of the existing viewing habits. Whereas it seems safe to forecast that mobile reception in trains and buses will become commonplace after the introduction of the first proposal, the potential market development for individual mobile reception (for example, for wrist-watch television, etc.) is not yet predictable.

#### 4.2.3.2. Broadcasters

The second proposal, since it does not include HDTV as an option, does not seem to be of interest for most of the European national broadcasters.

#### 4.2.3.3. Industry and dealers

If a trend towards individual television viewing in moving vehicles were to become an important trend of the future, the potential sales volume of small sets may be enormous. On the other hand, small receivers in the past often have become a major business for non-European companies after a relatively short time.

#### 4.2.3.4. Regulators and network operators

It is assumed that the appeal of the second proposal is similar to that of the first proposal and would lie mainly in the use of the system to reduce spectrum occupancy for the broadcasting of a given number of programmes.

### 4.2.4. A possible introduction strategy

Since the second proposal is not likely to be of interest to national broadcasters, the introduction strategy will have to rely on the initiative of industry, network operators and regulators. Since again it is assumed that television production facilities will have been converted to 625 line/components or HDTV by the time of the market introduction of the second proposal, the signal quality will most probably be sufficient for EDTV transmissions.

Network operators therefore would prepare the introduction of the second proposal by installing transmitter systems, for example along motorways. Industry could then introduce receivers into the marketplace. From the above-mentioned reasons the programmes available for the new service most probably will be the existing ones. Special non-broadcast services could be introduced additionally that could be specifically targeted to a mobile audience, e.g., traffic reports in the form of live views of certain critical parts of the motorway system, information about city traffic, etc.

## 5. A MECHANISM FOR THE PRACTICAL IMPLEMENTATION OF THE PROPOSALS OF THE WGDTB

The following is a description of a mechanism, which will allow the evaluation and, when the feasibility can be proven, the realization of both proposed systems within the shortest timeframe possible. It is assumed, that one single mechanism will be sufficient, to develop and evaluate both proposed systems in parallel. It is assumed as well, that of the activities of the "European Launching Group for Digital Video Broadcasting", the following elements of the infra-structure can be taken for granted:

- Steering Board
- Technical Project Manager
- Project office
- R&D module of activity
- Pre-Standardization and Standardization module of activity
- Policy Support module of activity
- Marketing Model and Implementation Strategies module of activity.

Due to the description of each of the above-mentioned elements within the text proposed for an MOU on Digital Video Broadcasting, only the work within the R&D module will have to be considered below.

## 5.1. Overview of the possible approaches

The process of designing and testing DTB systems in accordance with the first and the second proposal of the WGDTB will consist of several steps. The activities of ISO-MPEG would be used, if the system has been appropriately designed, for a source coding algorithm for the image and sound. Channel coding and modulation schemes will need to be designed and tested. They could in turn influence the selection of the source coding algorithm. As a next step a decision will be possible on the practicality of the proposed systems or the fallback mode (in the first proposal). A complete chain including encoder, transmitter, receiver, decoder will have to be designed. Field tests will be needed to finalize the system design. The choice among more than one system proposals may be necessary.

Generally speaking a number of approaches may be envisaged that could be used to organize this overall process, namely:

a) creation of a specific stand-alone project for this purpose,

b) use of one of the existing research projects (see paragraph 2.3 of this report) in Europe any after modifications necessary,

c) redirection of existing research projects, possibly merging them to only one,

d) use of as many of the existing research projects as possible with little or no modification and some added R&D-capacity, to create an integral project with project leadership.

For the above, a) seems to be too inefficient at this stage and b) as well as c) will not be possible because of the individual situations of each of the projects, d) is considered the practical solution.

## 5.2 "Euro-DTB[6]" or "DTB 2000"

"Euro-DTB" or "DTB 2000" are possible names of a project consisting of:

- a Coordination Group (like the WGDTB, and possibly even the existing WGDTB) incorporating representation from the existing European research projects,
- a Project Manager,
- a Project Office as a focus of information on project plans,
- possible additional projects, set up to complement the existing projects.

Euro-DTB would have as its first purpose to act as a platform for communication among the existing European research projects with the main goal, to help concentrate all ongoing work within these projects on the first and second proposal of the WGDTB. It is hoped that communication to a large extent will help to streamline the overall activities.

The separate technical module of activity is meant to include the "glue" research and

6 (The term "Euro" might have a psychological disadvantage in interesting other parts of the world in the system developed).

development necessary to convert to outputs of the existing projects to an integral result according to the first and second proposals of the WGDTB. In addition, the evaluation and testing of the resulting systems could be performed within this module. Last but not least, the technical liaison necessary for all pre-standardization and standardization work should be concentrated here, such that the above-mentioned module according to the proposed MOU may communicate with only one grouping in this respect. A funding mechanism for this technical module may have to be found.

The overall coordination should be in the hands of the Technical Project Manager according to the proposed MOU.

### 5.3. The next steps

If the Launching Group accepts this document at its December meeting, the next step should be a rapid first meeting of the Coordination Group described in 5.2, even if the Project Manager is not yet appointed. This should investigate in detail the technical contents of the existing European projects in the light of the first and second proposal of the WGDTB. The Coordination Group should be charged with the task of encouraging a concentration of these projects towards the agreed target systems. The Coordination Group should then propose the contents of the separate technical module and give a cost estimate, and proposed schedule, for the overall activities within the R&D module of activity, according to the proposed MOU. This could be presented to the Steering Board in early 1993.

### 5.4. The element of competition

In arriving at a final system the following elements need to be considered:

- In the most successful commercial hi-tech organizations, several development teams are set up to produce independent solutions for new systems. The competition stimulates the teams to be creative etc., and is thought to produce the best final products, although there are always 'loosers' in such a process, who have to be prepared to accept this.
- In the US the digital terrestrial system is being developed by direct competition between different projects implemented in hardware. This is not by choice however, but is due to the nature of the US industry and legislative process.
- In the Eureka 95 project, a number of proposed algorithms, each developed by different groups, were evaluated by comparative tests. However, in this case the comparisons were made of system simulations, and not hardware. This is certainly a less expensive approach that comparative hardware evaluations, but the results are less conclusive. Sometimes the difference in performance between simulation and hardware can be greater that the differences between systems.

Taking all these lessons into account the WGDTB current view is therefore that as a first step, full system requirements should be drawn up, against which simulations can be judged. Those systems that pass this first test should then be subject of comparative evaluations in hardware form, and the final European system chosen on the basis of the resulting rank order, using a previously agreed and weighted range of factors (quality, silicon area, flexibility, etc.).

### 5.5. Approximate provisional timetable

(Subject to appropriate funding being available)

1993: agreement on system requirements, decisions, actions and weightings, methods of evaluation, and test sequences.

1994: evaluation of potential candidates by simulation (or hardware if available).

1995: formal comparative evaluations of hardware systems.

1996: final system selected, field trials, and refinements.

1997: finalization of specification.

# APPENDIX A

## Launching Group "Digital Video Broadcasting in Europe"
## Working Group on Project Description
## Terms of Reference

The Working Group is mandated, to investigate the following issues, and to finalize a report to the launching group not later than Dec. 31st, 1992.

1. Types of digital terrestrial television services, which could be offered in view of state-of-the-art technological possibilities. Topics to be covered are:
   - Possible dates of availability
   - Quality levels (picture, sound)
   - Features of possible services
   - Frequency spectrum requirements
   - Costs (production, contribution, transmission, reception, recording)
   - Prerequisites for implementation (technology, economics, regulations, standardisation)
   - Possible interrelationship with other distribution media (CATV, DBS...)
   - Hierarchical interdependence of possible services
   - Existing R&D activities, that are being carried out in Europe relevant for the field
2. Research, development and preparatory work needed, to realize the above mentioned services, that is not covered by ongoing national and European projects.

   The report to the launching group should additionally include:
3. a proposed definition of the overall technical and operational goals for digital terrestrial television services in Europe to be included in a European Memorandum of Understanding (MOU),
4. a description of the relevant activities in other parts of the world and a comparison between the proposed definition and the strategies elsewhere,
5. a comprehensive annex to be attached as a technical annex to the European MOU, describing the most important results of the work.

# APPENDIX B

## WGDTB MEMBERSHIP

| | | | |
|---|---|---|---|
| Prof. Dr. U | REIMERS, Chairman | NDR | Germany |
| Mr M. | ANNEGARN | PHILIPS | Netherlands |
| Mr P. | APPELQUIST | SVT | Sweden |
| Mr R. | BEDFORD | RA | United Kingdom |
| Mr R. | BOYER | THOMSON | France |
| Dr I. | CHILDS, Vice-Chairman | BBC | United Kingdom |
| Mr H. | COHEN | MICE | France |
| Mr P. | COSTANZO | FRANCE 2 | France |
| Dr M. | COMINETTI | RAI | Italy |
| Dr R. | GLÜCKSMANN | BMPT | Germany |
| Mr K. | HACKER | DBT | Germany |
| Mr D. | IBANEZ | RETEVISION | Spain |
| Dr U. | KRAUS | THOMSON | Germany |
| Mr N. | LODGE | ITC | United Kingdom |
| Mr B. | MARTI | CCETT | France |
| Mr J. | PONCIN | FRANCE TELECOM | France |
| Mr A. | REEKIE | CCE | Belgium |
| Mr D. | SAUVET-GOICHON, Vice-Chaiman | TDF | France |
| Dr R. | SCHÄFER | H. HERTZ Inst. | Germany |
| Dr M. | SILVERBERG | GRUNDIG | Germany |
| Dr D. | UHLENKAM | NOKIA | Germany |
| Dr H. | WILKENS | IRT | Germany |
| Mr D. | WOOD, Secretary | EBU SPG | |

# APPENDIX C

## FURTHER DETAILS OF US PROPOSALS

## SOURCE ENCODING

HDTV systems in the US are proposed for different scanning formats: 1050/59.94/2: in the case of ADTV and Digicipher, and 787.5/59.94/1: for DSC HDTV and CCDC HDTV. All systems have a 16:9 picture aspect ratio, but resolutions are different, dependent on clock rates and scanning formats.

Table A3.1 shows horizontal and vertical resolution for luminance and colour-difference signals of the four systems.

Video source encoding schemes of almost all systems are based on hybrid motion compensated DCT coding, as in H.261 and MPEG standards. Irrelevant and redundant temporal picture information is eliminated by subtracting reconstructed and motion compensated macroblocks of previously encoded pictures from those of the incoming pictures, the residue is transformed by Discrete Cosine Transform on an 8 x 8 block basis. DCT coefficients are quantized and variable length coded (VLC). The resulting variable data rate is then buffered to achieve a constant data rate at the output of the video encoder. Rate control is achieved by change of the quantization according to feedback of the bit buffer status and by pre-analysis functions.

Fig. A3.1 gives an overview of the encoder functionality. Differences between the four systems exist in the prediction strategy for motion compensation, in the pre-analysis functions, in the rate control and quantization, as well as in the bit coding.

The video encoding strategy of ADTV is presently conforming to the MPEG-1 standard and will be adjusted to MPEG-2, as soon as the standardization process of MPEG-2 has settled the major prediction issues for interlaced images. DSC, Digicipher and CCDC do not use bi-directional prediction as in MPEG, but have similar procedures otherwise. For quantization, DSC uses a block adaptive scheme, which is controlled by an elaborate pre-analysis function. This pre-analysis measures expected and necessary bit-rates, related to a model of the human visual system and identifies the respective quantization strategy.

Further differences, i.e. in motion vector search range and macroblock size, are listed in Table A3.1.

For audio coding, all systems support four independent channels or two stereo pairs. ADTV uses the MUSICAM algorithm (MPEG compliant) while DSC and Digicipher use the Dolby proposal AC-2. CCDC uses a technique developed at MIT. All audio encoding techniques result in a similar total data rate of 0.5Mbit/s, approximately.

### Transport protocol

Formatting procedures were not published by all proponents. For ADTV, an Asynchronous Transfer Mode (ATM) packet-like structure was chosen, which includes the MPEG data in a frame, consisting of sync, identification, and FEC bytes. The structure is shown in Fig. A3.2, illustrating the contents of a transport cell and its MPEG adaptation header. As a means to allow graceful degradation and adaptation to the channel requirements, the MPEG data in ADTV is split into high (HP) and standard (SP) priority information. For this, a prioritization procedure is used, to allocate header, vector and DCT coefficient information, together with the most important coefficient data in the HP data stream, while the rest is carried in the SP channel. Fig. A3.3 illustrates the breakpoint adaptation in the prioritization (analysis) buffer.

In DSC, a transport format was chosen, which resembles the field transmission structure of conventional NTSC. This leads to a two-dimensional format including sync and error correction bytes. The similarity to NTSC timing allows for special mechanisms to control interference issues.

All systems employ Reed-Salomon-Coding in conjunction with interleaving techniques for error correction. Parameters of these are given in Table A3.1.

**Channel coding and modulation**

Channel modulation systems of the four proponents differ, but all of them achieve a modulation efficiency of 3-4bit/s/Hz.

The Digicipher variants are based on 16-QAM, with a possibility of using 32-QAM in clean channel environments. No special means are provided in the modulation scheme to cope with NTSC co-channel interference.

In ADTV, the 32-QAM/16-QAM alternative is used again, but the data is split into two portions, which are modulated on to separate carriers. Carrier positions are shown in Fig. A3.4. This shows the resulting Spectrally Shaped (SS)QAM. In this scheme, the high priority data (as distinguished from the low priority data through the prioritization process described earlier) is modulated on to a carrier in the range of the Nyquist slope of a conventional NTSC receiver. The result is high protection of the NTSC signal against HDTV interference. On the one hand, the most valuable information in the HDTV channel is also better protected against NTSC co-channel interference, due to the low energy content of NTSC in the low range of the Nyquist slope.

Further protection in both directions is achieved by a guard band between the HP and SP subchannels in the area of the NTSC picture carrier, as Fig. A3.4 illustrates. The standard priority information is transmitted with lower energy in the range of higher NTSC frequencies, where is gives less disturbance in case of interference. In addition to the advantage of reduced co-channel impairments, the two-carrier system of ADTV provides for a simple approach to graceful degradation in the fringes of coverage areas. In those areas the SP information is lost first, due to its lower energy, but the HP information is still sufficient to yield a lower resolution picture.

In the Zenith/AT&T proposal, a dual transmission mode is used for graceful degradation and for bad channel conditions. Under best conditions, the digital data is sent in 4-VSB mode, with adverse channel situations, the encoder can switch in a 2-VSB mode adaptively or permanently. However, the 2-VSB mode reduces the symbol length to 1 bit only, as Table A3.1 shows. Against co-channel interference the DSC system uses a comb filter approach. Since its time base is similar to NTSC, the modulated data can be submitted to a recursive pre-coder which provides for a spectral shaping similar to a multiple frame comb filter, with nulls at the NTSC picture carrier and close to the colour and sound subcarriers. At the decoder, a similar, but FIR, filter is used in conjunction with a modulo interpreter and slicer, which determine the 4 valid data levels out of 7 possible ones.

All systems provide for echo cancellation, in the range 2-24μs.

**Transmission aspects**

All HDTV proponents have specified similar thresholds for C/N performance: about 10-12dB are necessary for the low performance mode (HP signal only in ADTV, 2-VSB only in DSC, 16-QAM only in the Digicipher systems). For full performance, about 16dB C/N are needed in all four systems, as Table 1 shows.

For ATTC admission, the proponents had to provide figures on coverage and interference calculations. Fig. A3.5 illustrates the respective numbers in Table A3.1 on the example of the ADTV system for mutual interference of HDTV and NTSC signals. Conditions such as effective radiated power (ERP), antenna heights, transmitter distance, and NTSC grade A and B service contours can be read from Fig. A3.5, which also shows coverage areas and the areas which will be effected by interference under these circumstances. Numbers for the four systems relate to slightly different basic conditions but indicate similar performance nevertheless. An exception is given by the NTSC-to-HDTV interference figure in ADTV. A larger HDTV area free from NTSC interference is obtained by the Spectrally Shaped QAM system.

US-Proposals for Terrestrial HDVT-Transmission

| Proposal | ATRC (ADTV) | Zenith/AT&T (DSC) | GI (DigiCipher) | ATVA (CCDC) |
|---|---|---|---|---|
| Input Format | 1050 / 59.94 / 2 | 787.5 / 59.94 / 1 | 1050 / 59.94 / 2 | 787.5 / 59.94 / 1 |
| active H*V Y | 1440 x 960 | 1280 x 720 (sq.p.) | 1408 x 960 | 1280 x 720 (sq.p.) |
| per frame C | 720 x 480 | 640 x 360 | 352 x 480 | 640 x 320 |
| Video Data compression | hybrid (MPEG1)<br>bidirectional | hybrid<br>unidirectional | hybrid<br>unidirectional | hybrid<br>unidirectional |
| Pre-Analysis | yes | DFD + DCT + percep-<br>tual weigth calc. | none | none |
| Film mode | | Buffer optimization | field drop | |
| Frame types | I, P, B | P | P (+I blocks) | I, P |
| Transform | DCT 8x 8 | | DCT 8 x 8 | DCT 8 x 8 |
| Motion estimation | full search, + 1/2<br>pel compute | DCT 8 x 8<br>hierarchical block<br>matching, 1/2 pel | full search, integer<br>pel | spatio-temp. con-<br>straint(gradient),<br>1/2 pel |
| MB size | 16 x 16 | | 32 x 16 | 16 x 16 (8x8) |
| MV range | 95 x 95 | 8 x 8 (32 x 16) | | 32 x 16 |
| Quantization | weighting uniform | 96 x 80 | weighting uniform | weigthing |
| Bit assignm. strategy | by buffer status<br>feedback | indiv.f. each coeff.<br>by pre-analysis<br>(HVS model) + buf-<br>fer status feedback | by buffer status<br>feedback | fixed bits per frame |
| Audio encoding | MPEG (Musicam) | AC-2 | AC-2 | MIT-AC |
| Error Protection | RS + interleaving<br>(148, 127)+TC 9/10 | RS (167, 147)<br>+ interleaving | RS (116, 106)/(155,<br>145) + interleaving<br>+ TC 3/4 - 4/5 | RS (168, 158)<br>RS (126, 116) |
| Data Rate (Mbps) total | 24.0/16.0 | 21.52/10.76 | 24.39/19.51 | 19.89 (26.43) |
| video | 18.5/13.8 | 17.0/8.5 | 17.49/12.59 | 18.88 |
| audio | 0.504 ( 4ch.) | 0.504 (4ch.) | 0.504 (4ch.) | 0.755 |
| data | 0 256 | 0.412 | 0.252 | 0.252 |
| FEC + S | 4.74 (ca. 20%) | 3.604 (ca.17%) | 6.144 (ca. 25%) | 6.54 |
| Modulation | Twin 32 or 16<br>QAM | 4-VSB/2-VSB | 32-/16-QAM | 32 QAM/16 QAM |
| symbol rate (MSPS) | | 10.76 | 4.88 | 5.2867 |
| spectral shaping | 3.84 + 0.96 | Comb filter | BPF | Baseband FIR |
| echo eq. range | Twin - QAM | | -2....24 μs | -2....24μs<br>(single echo) |
| System Threshold Noise (C/N) (dB) | 16.1 (SP)/11.1(HP) | 16(4-VSB)/<br>10(2-VSB) | 16.5/12.5 | 15.7 (32 QAM)/<br>11.7 (16 QAM) |
| Transmission Power (dBk) | 26.3 | 22.5 | 19.5 | 23.9 (32 QAM)<br>20.4 (16 QAM) |
| Coverage Area (miles) | 54.5/55.5 | 53 (4VSB)/<br>59 (2VSB) | 61 | 53.1 (32 QAM)<br>53.5 (16 QAM) |
| Radius/Dist. (Interf. area) | | | | (32 QAM (16<br>QAM) |
| HD->NTSC | 40.3/115 (14.1%) | 42/112 (12.7%) | 41.5/100 | 40.5 /114.7<br>(40/103) |
| NTSC-HD | 53.1/115 (0.9%) | 45/112 (5.9%) | 42/100 | 45.5/114.7 (46/103) |
| HD->HD | 45/125 (2.3%) | 48/125 (2.9%) | 47/100 | 45.3/102.9 (43/89) |
| Graceful Degradation | via data priority | via bi-rate coding | | |

*Table A3.1: US-proposal for terrestrial HDTV-transmission*

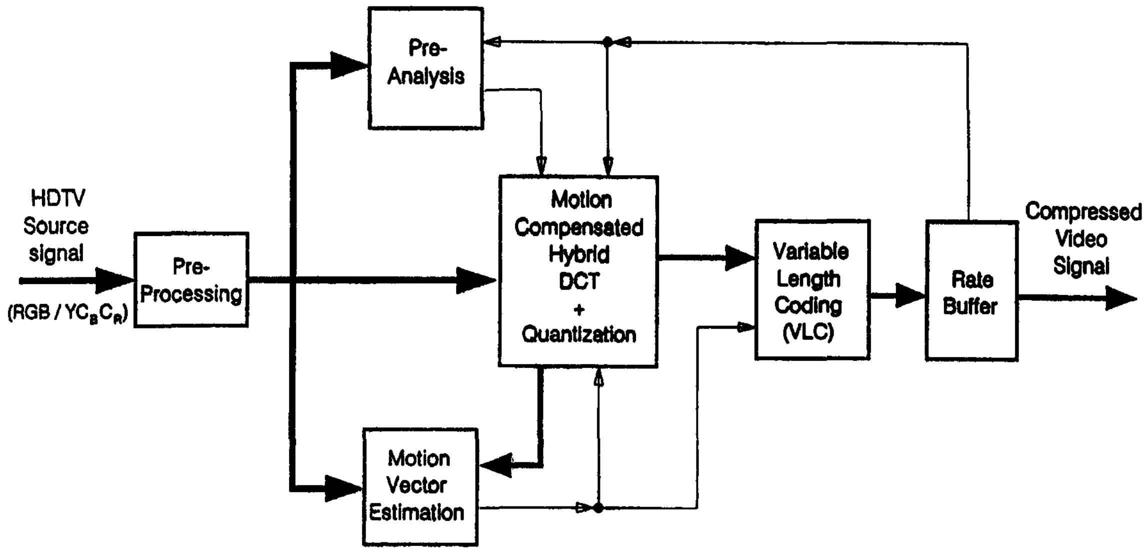


*Figure A3.1: Generalized functions of source encoder*

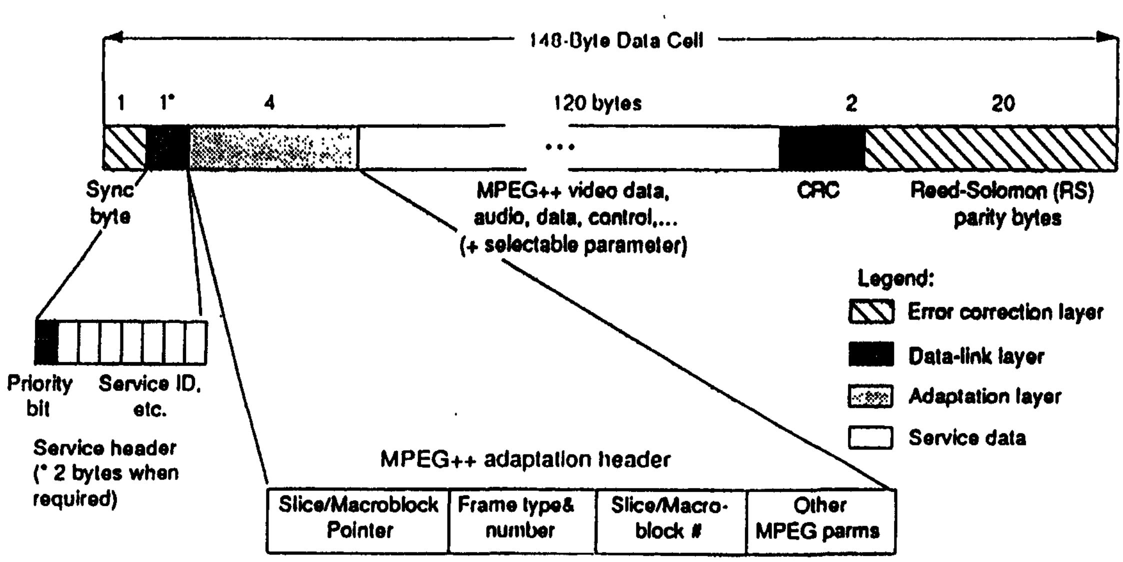


*Figure A3.2: Transport cell data format (ADTV)*

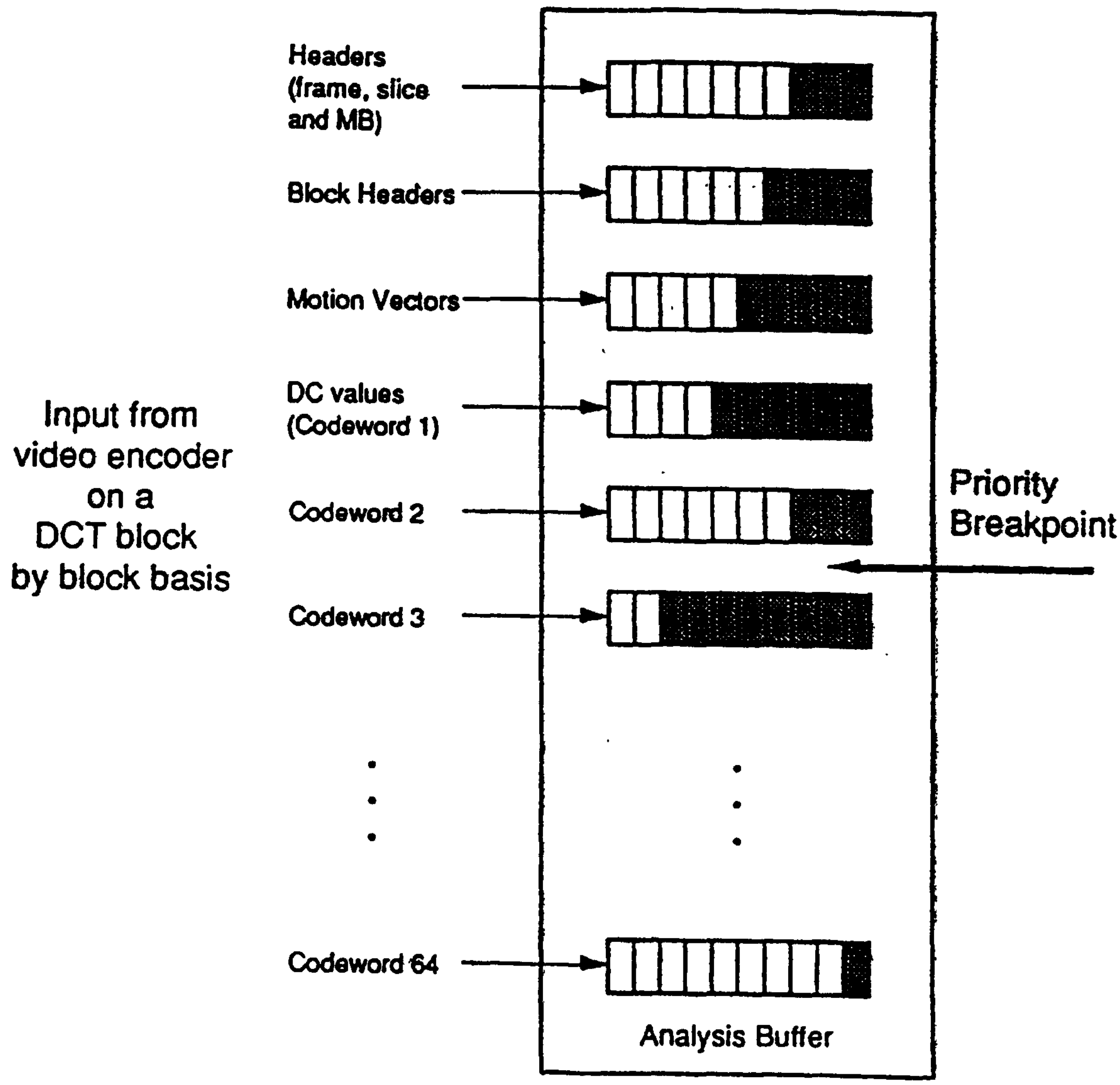


*Figure A3.3: Dynamic priority assighnement for HP/SP layer (ADTV)*

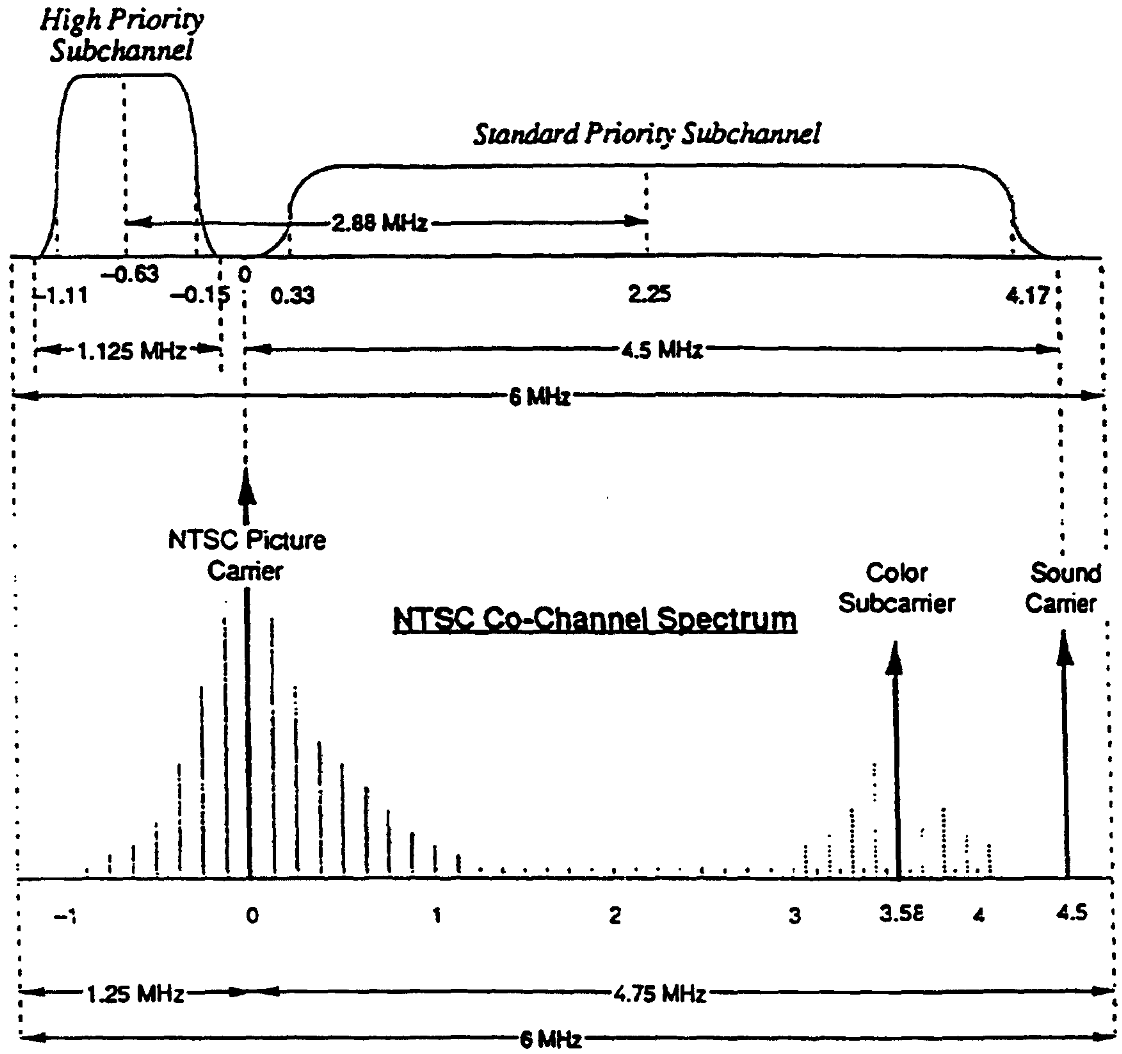


*Figure A3.4: SS-QUAM spectrum in relationship to NTSC (ADTV)*

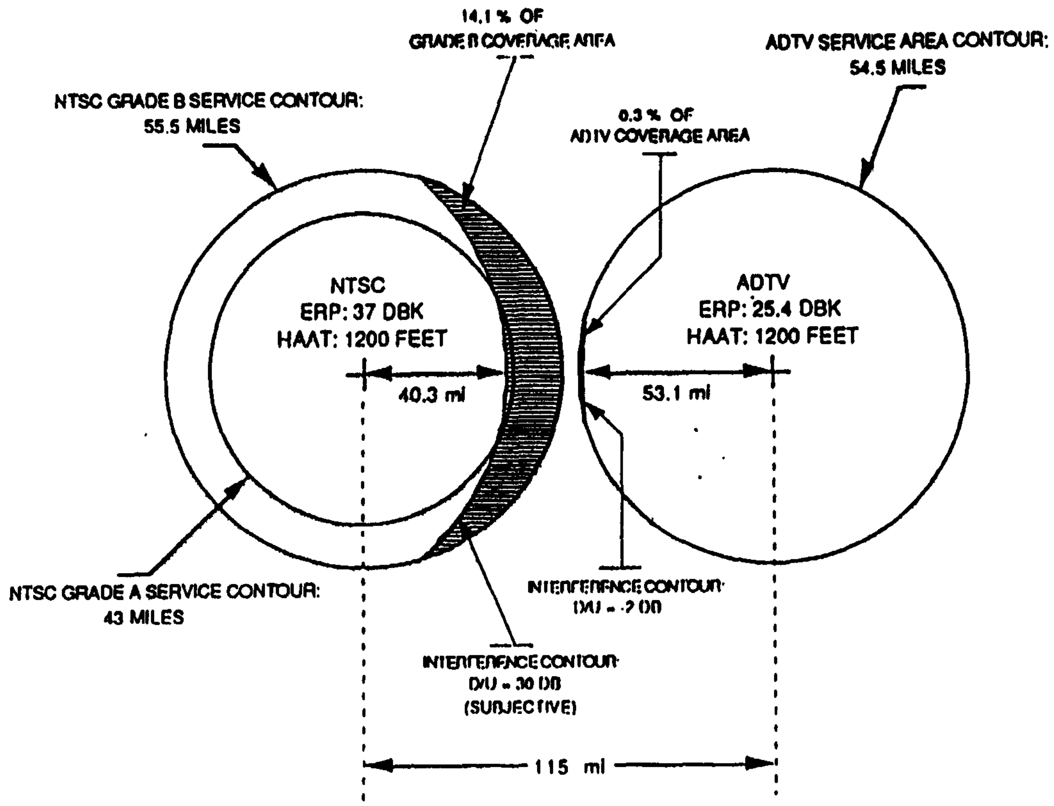


*Figure A3.5: Coverage and interference areas with NTSC and ADTV co-channel transmission*

# APPENDIX D

## CONCEPTS FOR THE IMPLEMENTATION OF HIERARCHICAL CODING

## 1. SOURCE CODING

At the source coding level, resolution and bit stream scalability are required for a hierarchical system. While resolution scalability can be achieved with either block or frequency scanning techniques, bit stream scalability requires the use of frequency scanning techniques.

Some recent investigations have shown that frequency scanning is better than block scanning (2dB S/N ratio at 8Mbit/s). On the other hand, the cost of resolution scalability can be roughly evaluated at less than 1dB in S/N ratio lost in quality for the same bit-rate (including the overhead specific to resolution scalability) compared to a stand-alone scheme. In other words, the overcost of scalability means a bit-rate increased by approximately 12% for the same quality level.

## 2. CHANNEL CODING AND MODULATION

### 2.1. Channel coding

The purpose of the channel coding is to protect the bit stream against the impairments due to the transmission channel. According to the class of receiver which is considered, several channel models are used:

- AWGN (Average White Gaussian Noise).
- Rice channel for fixed receivers.
- Rayleigh channel for portable and mobile receivers.

The available techniques are:

- Time interleaving.
- Frequency interleaving, both of them aiming at scattering bursts of errors.
- Forward Error Correction.
- Trellis coded modulation, which requires a strong relationship between channel coding and modulation.

Their parameters are adapted to the channel characteristics in order to obtain the best efficiency. For implementing a hierarchical system, a possible solution is to apply different channel coding techniques to parts of the bit stream dedicated to different classes of terminals. The requirements on the hierarchy of services lead to consideration of scalability in the channel coding mechanisms which should allow a fixed receiver to easily decode, even in very bad conditions, the bit stream dedicated to a portable and mobile receiver. This is the ruggedness scalability.

## 3. MODULATION

In the modulation process, (Fig. A4.1) the type of constellation contributes to the improvement of the ruggedness of the bit stream against signal-to-noise ratio reduction. Theoretically, 6dB are gained each time the number of points of the constellation is divided by four. Unequal protection can this be performed by using several types of constellation for different priority bit streams.

- In sharing the bandwidth in "frequency slots" each being associated with a type of constellation: QPSK, 16QAM, 64QAM...
- In sharing the frame of symbols in several sub-frames having each a type of constellation. A temporal multiplex of different modulated symbols is obtained.
- In using a multi-resolution scheme.

## 4. MULTIPLEX FLEXIBILITY

Various methods to build a multiplex are known and applied for contribution and distribution transmissions of digital signals. Several of these techniques could be combined to form a multiplex scheme.

### Fixed frame

The partitioning of the digital data stream is determined and known to any receiver. Examples using this technique are teletext, RDS, CCIR Recs. 721 and 723. These schemes offer rugged recovery of the frame header using little overhead for the transmission of header information. Due to the fixed structure of the multiplex, there is little room for changes or extensions for individual components.

### Data packets

Data packets of equal length and with fixed position within the multiplex are transmitted. This method is used for the data transmission in the D2-MAC system. As the position of the packet headers is known to the receiver, rugged recovery is ensured. New services can be included in a compatible way. On the other hand a large amount of data is required to transmit and protect the packet headers.

### Packets and packs

Packets of data from a number of elementary bit streams are multiplexed together in a flexible way. Each elementary bit stream contains coded data of only one source (e.g. video, audio or data). The packets have individual headers unique to each source. Packets are grouped in packs which in turn have their own headers. This method is proposed for the transmission of ISO MPEG data streams.

Methods of providing high flexibility for the dynamic allocation of resources to source data streams, together rugged transmission in transmission channels with errors, remain to be studied in detail.

### Asynchronous Transfer Mode (ATM)

For the transmission in the future B-ISDN, the method known as Asynchronous Transfer Mode is considered, which also makes use of packets of fixed length, but in combination with variable transmission rates where the packets are not synchronized to each other. These techniques offer different levels of flexibility. In any such system, a signalling mechanism dealing with the following problems would have to be implemented:

- To relay the channel coding requirements of the different parts of the service components from the source to the channel encoder.
- To inform the receiver of the location, size and degree of protection of the service components.

## 5. SOME ISSUES TO BE INVESTIGATED ASSOCIATED WITH MULTI-LAYER SYSTEMS

- The S/N margin (in dBs) between a fixed, portable and mobile channel, to verify whether this margin can be covered by available channel coding and modulation techniques.
- The user requirements on the S/N ratio margins between fixed receivers with EDTV, SDTV and LDTV quality, to verify whether they can be met by ruggedness scalability.
- The complexity of a compatible receiver compared to a non-compatible receiver.
- The area where graceful degradation should operate, taking into account the service requirements (i.e. the distance between a 100% quality reception and a 0% quality reception).
- Evaluation of the source coding algorithm in terms of percentage of failure time.
- The characteristic of the portable channel and the specification of adapted channel coding techniques.
- The definition of the methods of operation and performance of FEC dedicated to source coding and FEC for channel coding.
- The investigation of Unequal Error Protection codes.
- performance and complexity.
- The specification of the level of flexibility required by:
  - multi-programming,
  - fixed or variable bit-rate of the data stream resulting from the resolution scalability.
- Comparision of picture quality and bit-rate reduction efficiency achieved by hierarchical and multi-cast coding systems.

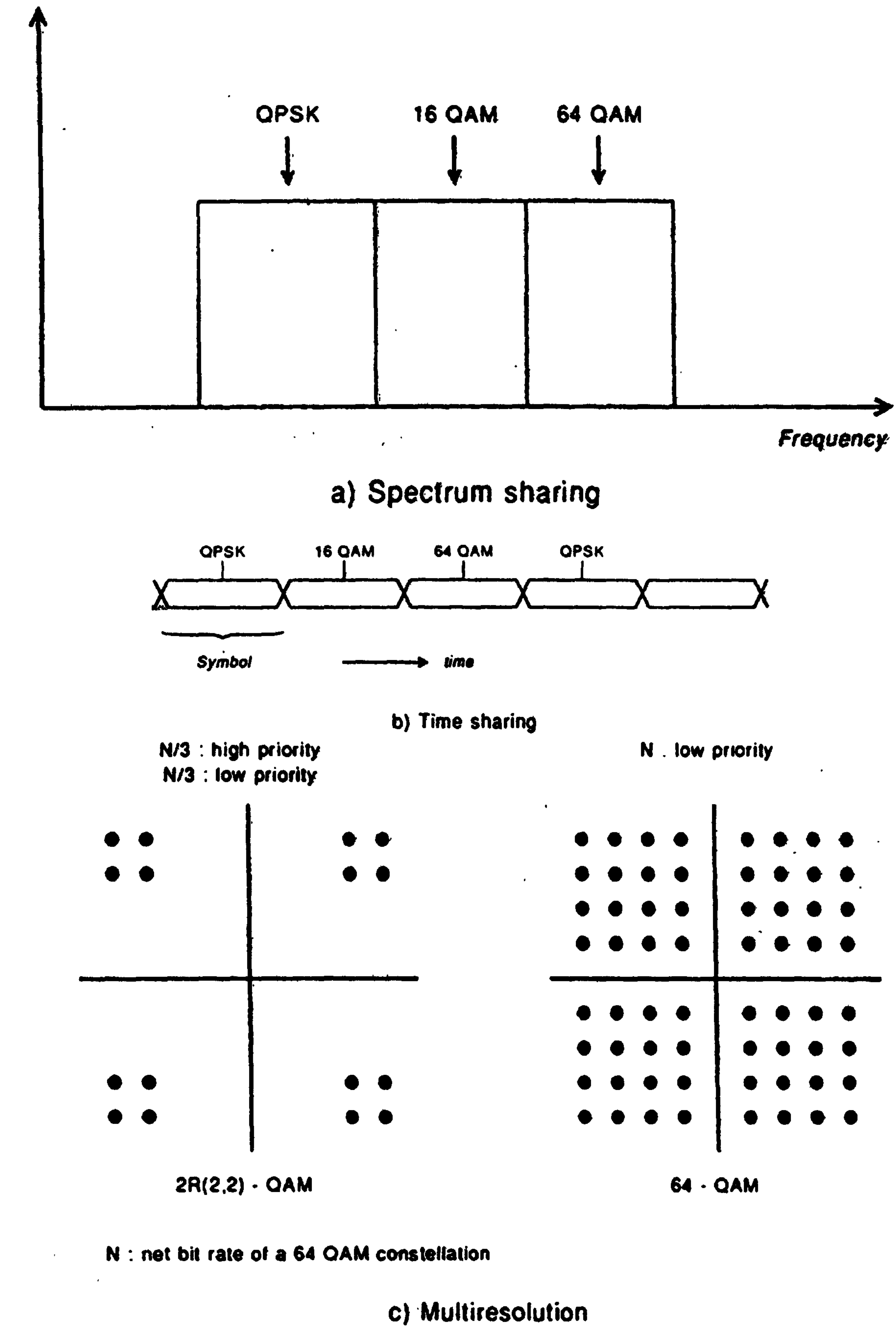


*Figure A4.1: Multi-priority modulation*

Chapter 8

# Memorandum of Understanding for the development of harmonised digital video broadcasting services in Europe

The European Group for Digital Video Broadcasting

## PREFACE

Section 2.1.2. of the Communication to the Council describes in some detail the work of the European Group for Digital Video Broadcasting, formerly known as the European Launching Group for Digital Video Broadcasting. This chapter reproduces the MoU agreed and signed by 85 organisations on 10 September 1993, together with a list of signatories as at that date.

# MEMORANDUM OF UNDERSTANDING FOR THE DEVELOPMENT OF HARMONISED DIGITAL VIDEO BROADCASTING (DVB) SERVICES IN EUROPE

**The signatories:**

considering:

- that the future of terrestrial, cable and satellite television services is important for the European public, European broadcasters, operators and for the manufacturers who are part of the European consumer electronics industry;
- that DVB offers the prospect of early European services by satellite and cable with some commonality with later more unified terrestrial systems, avoiding the differences between PAL and SECAM plus variants within these systems. There are also prospects of a closer unification with systems being developed for NTSC countries;
- that the standardization by the Motion Picture Experts Group (MPEG) in the specification of a single global standard for source coding and multiplexing, applicable to satellite, cable and terrestrial transmission, due to be finalized in November 1993;
- that Europe's enhanced systems (PALplus and D2-MAC) offering 16 by 9 format are of value to prepare and facilitate the market for future services in the image format of 16:9;
- that the future technology for programme production, editing, transmission network and home display will be digital;
- that digital systems, combined with compression technology, are already finding application feeding cable head ends;
- that a digital system, when combined with compression technology and modern modulation techniques, may be introduced into the current terrestrial frequency bands without disturbing existing PAL, SECAM or NTSC services providing that access to these terrestrial frequency bands is made available;
- that digital television systems could eventually offer various advantages in comparison with current analogue services in terms of quality and more efficient use of the spectrum, for example, digital modulation techniques, under certain conditions, can allow adjacent terrestrial transmitting stations to operate on the same channel frequency;
- that some of the DVB techniques may also be applicable to the future multi-media environment;
- that the European Directive on the adoption of standards for satellite broadcasting of television signals requires the Commission to submit, if necessary, before the end of 1994 proposals to the Council on a comprehensive policy of standardisation of HDTV and that the Commission has indicated the wish to take full advantage for this purpose of any results coming out of this MoU;
- that European equipment manufacturers, broadcasters, operators and research centres have already started co-operating within the framework of European programmes on digital television such as EUREKA 625 (VADIS), RACE 1018 (HIVITS), EUREKA 256, RACE 2082, dTTb, RACE 2075, RACE FLASH TV, HDSAT, and various related national projects;

- that unless European broadcasters, operators and manufacturing industry develop and implement a digital television system in an appropriate timescale, they may lose to others the potential use of this technology, its application in the broadcasting spectrum and their own competitive position;
- that, considering the size of the workload in this field, it is important to make sure that tasks and costs are adequately allocated between the organisations which are likely to invest in the development of digital television, that their objectives reflect ongoing development of market and technological realities and that no time is lost in the progress of their work;

---

agree the following Articles and Annex:

**ARTICLE 1**

The purpose of this Memorandum of Understanding is to create in Europe a framework for a harmonious and market driven development of digital television via cable, satellite and terrestrial broadcasting, - the DVB project.

This shall be realized through:

- promoting and contributing to the definition of technical standards for digital television and their widespread utilisation/adoption.
- facilitating the introduction of new services using those standards, which may include studies on associated matters such as frequency planning and conditional access.
- facilitating the closest possible coordination between pre-competitive R&D and standardisation.

**ARTICLE 2**

This MoU may be signed by an entity/administration, a group of entities or European organisation entitled to membership of the European Telecommunications Standards Institute or the European Broadcasting Union, who are established in countries covered by either of those two bodies, who commit themselves to the purposes of the MoU and commitment themselves to actively contribute to the work of at least one of the modules.

**ARTICLE 3**

The deliverables from this MoU are intended to contribute to the following objectives:

a) a draft European Standard(s) or for the end of 1993 in order to be able to begin services in 1995, meeting the needs of the market for digital video broadcasting via satellite and cable and incorporating the global MPEG-2 standard for source coding and multiplex to the extent possible;

b) a draft European Standard(s) for terrestrial digital video broadcasting for the end of 1995;

c) through the appropriate arrangements, draft European Standards for the relevant aspects of the related receiving equipment for (a) and (b) above;

d) a technology base within Europe that positions European enterprises to fully exploit the market on a competitive basis for digital video broadcasting technology;

e) a contribution to official policy frameworks that removes obstacles to a market led and consumer-orientated introduction of a digital video broadcasting service in Europe;

f) a flow of information between broadcasters, satellite operators, cable network operators, terrestrial operators and manufacturers that provides the confidence to enter into various commercial agreements for the exploitation of DVB;

g) to facilitate the flow of information and cooperation with other parts of the world.

**ARTICLE 4**

The initial organisational structure shall comprise the following:

- Technical Module
- Satellite/Cable Commercial Module
- Terrestrial Commercial Module
- Steering Board
- General Assembly

**ARTICLE 5**

The Technical Module provides technical expertise and is open to the technical experts of all signatories.

The Technical Module also provides a forum for the coordination of R&D activities. It shall register details of all R&D projects wishing to come within the scope of this MoU.

The Technical Module works according to requirements set down by the relevant Commercial Module. It delivers specifications for one or more Standards via the Steering Board to the recognised standards setting entities, notably the EBU/ETSI Joint Technical Committee and its relations with CENELEC. It provides a conduit to other relevant standardisation activities including MPEG for the purpose of meeting the objectives of this MOU.

The Technical Module may be asked by the Steering Board to study the frequency planning needs of DVB;

The Technical Module shall endeavour to reach consensus including the use of indicative voting but if this is not possible in a timely way it shall put the options together with the minority opinions to the relevant Commercial Module and to the Steering Board as appropriate.

**ARTICLE 6**

The Satellite/Cable Commercial Module is open to senior managers from any signatory making significant financial investment in the implementation of services/products targeted to come to market by 1995.

The Satellite/Cable Commercial Module shall provide a focus for all non-technical aspects needed to successfully implement digital video broadcasting services by satellite/cable means by 1995.

The Satellite/Cable Commercial Module shall provide the definition of service requirements, priorities and time scale requirements to the Technical Module (and other relevant *ad hoc* groups) and upon completion of the specifications, shall report to the Steering Board their endorsement of the results.

The Satellite/Cable Commercial Module shall endeavour to reach consensus including the use of indicative voting but if this is not possible in a timely way it shall put the options and the minority opinions to the Steering Board.

## ARTICLE 7

The Terrestrial Commercial Module is open to senior managers from any signatory planning to make significant financial investment in the implementation of services/products.

The Terrestrial Commercial Module shall provide a focus for all non technical aspects needed to successfully implement digital video broadcasting services by terrestrial means.

The Terrestrial Commercial Module shall provide the definition of service requirements, priorities and time scale requirements to the Technical Module (and other relevant *ad hoc* groups) and upon completion of the specifications, when agreement between the two modules is achieved, shall report to the Steering Board their endorsement of the results.

The Terrestrial Commercial Module shall endeavour to reach consensus including the use of indicative voting but if this is not possible in a timely way it shall put the options and the minority opinions to the Steering Board.

## ARTICLE 8

The organisational elements under this MoU shall aim to create a common vision of Europe's future digital distributive electronics highways. The MPEG-2 standard shall be taken as the reference point. All distributive media shall be considered and all quality levels for which a market demand is identified shall be covered, including high definition TV.

Each module shall be responsible for its own organisational arrangement, providing always that such arrangements may not conflict with the MoU. Each shall propose for approval by the Steering Board their terms of reference and rules of procedure. They may offer to the Steering Board a nomination for the Chairman of their Module.

## ARTICLE 9

The Steering Board is composed of:-

- a maximum of 34 elected representatives, ensuring a balanced representation of views from broadcasters, operators, manufacturers and administrations. A representative of the EC, EBU, ACT, ECCA and EACEM shall be *ex officio* members of the Steering Board; without voting rights. Only those signatories indicating their intention to contribute with resources and activities for the benefit of the DVB project will be eligible to stand for election;

- In addition the Steering Board may co-opt additional members without voting rights to ensure an adequate spread of interests including geographical balance;
- The chairmen of the modules and any *ad hoc* groups shall be *ex officio* members of the Steering Board; without voting rights.

The Procedure for the appointment of the first Steering Board is set out in Annex 1.

The Steering Board shall be responsible for:

- the policy direction of the overall Digital Video Broadcasting Project (DVB);
- coordination, priority setting and management of the DVB project;
- advice to public authorities including the EC Commission on regulatory needs to facilitate the aims and objectives of the MOU. For this purpose a 2/3 majority shall be required to approve a matter. The Chairman shall not vote except in the case of a tied vote where he/she shall have the casting vote. If there are minority or dissenting views these shall be transmitted as well to the EC Commission and other relevant parties.
- amending the working structure within the MOU as required from time to time including establishing *ad hoc* groups;
- electing a Chairman who will hold office for two years and who may be reelected;
- establishing its own rules of procedures, including voting rules. For this purpose a 3/4 majority shall be required to approve a matter. The Chairman shall not vote except in the case of a tied vote where he/she shall have the casting vote;
- appointing the Chairmen of subordinate bodies taking into account any nominations from those bodies and approving their terms of reference and internal rules;
- dealing with any procedural disputes For this purpose each member of the Steering Board (including coopted and *ex officio* members) shall have one vote. A simple majority shall be required to approve a matter. The Chairman shall not vote except in the case of a tied vote where he/she shall have the casting vote;
- appointing a Project Manager as may be required;
- keeping all signatories informed of the work of the Steering Board, Modules and any *ad hoc* groups
- preparing any proposals for amending the MoU to be put to the signatories for approval.

The EBU shall provide a "Project Office" support to the Steering Board for any Project Management.

**ARTICLE 10**

1. Ordinary meetings of the General Assembly shall be convened once a year, at which the Assembly shall consider the report of the Steering Board, adopt the accounts for the past year and approve the budget for the next year.

2. Every two years, at an ordinary meeting, the General Assembly shall appoint the members of the Steering Board for the next two years. Existing members will be eligible for re-appointment.

3. Extraordinary meetings of the General Assembly may be convened by the Steering Board and shall be convened on a proposal of at least 1/3 of the signatories.

**ARTICLE 11**

The activities under this MoU shall be funded in one of three ways:

i) Costs of individual participation to be met by the organisations of the participants. This shall include elected officials.

ii) The EC Commission shall contribute to the funding for any Project Management approved by the Steering Board, up to a ceiling to be fixed each year by the budgetary authorities of the Community.

iii) All other expenses from a membership fee set out each year by the Steering Board within a ceiling not too far removed from the initial membership fee.

The initial membership fee shall be 10,000 ECU with no reduction for those joining later in a particular year. The Steering Board may waive the membership fee where a number of members already paying their fee also wish to be represented by an association.

The membership fee account shall be administered by the European Broadcasting Union based on the decisions of the Steering Board.

**ARTICLE 12**

All signatories shall be entitled to receive all output documents from all modules of activities and the minutes of the Steering Board. The Steering Board shall determine its policy on the confidentiality of papers and similarly each of the modules of activity shall do likewise for its papers. This procedure shall also take into account the confidentiality agreements of the various contributing projects.

**ARTICLE 13**

This MoU shall come into operation when at least 12 organisations according to Article 2 from at least 4 different countries have signed it. (The actions to be taken to convene the first General Assembly and for signing the MoU are set out in Annex 1).

**ARTICLE 14**

A signatory may withdraw from the MoU at any time by giving 1 month's notice in writing to the Chairman of the Steering Board. Only those members who comply with the principles of Article 2 may maintain membership.

**ARTICLE 15**

Any changes to the MoU shall be prepared by the Steering Board and put to the vote at a meeting of the General Assembly. A proposed change shall be adopted where 2/3 or more of the signatories vote and where 2/3 or more of the votes cast are in favour.

## ARTICLE 16

The Steering Board shall provide guidance on any questions of interpretation of the MoU.

## ARTICLE 17

This MoU shall come into force as provided under Article 13 and shall continue in force until December 31st, 1997; provided that such term may be extended by the resolution of the General Assembly.

## ARTICLE 18

Nothing contained in this MoU shall be construed as creating a partnership, joint-venture, agency, trust or other association of any kind, each signatory being individually responsible only for its obligations as expressly set forth in this MoU. No signatory shall act or represent or hold itself out as having authority to act as an agent or partner of any other signatory, or in any way bind or commit any other signatory to any obligations.

***

## ANNEX 1

### PROCEDURE FOR CONVENING THE FIRST GENERAL ASSEMBLY AND ELECTING THE FIRST STEERING BOARD.

The German Federal Ministry of Posts and Telecommunications shall be entrusted to convene the first General Assembly. It shall do this on 10th September 1993 or a date soon after.

Prior to opening the General Assembly the MoU shall first be open for signature.

Subject to the requisite number of signatures as set down in Article 13, the first General Assembly shall then be opened and chaired by a representative of the German Federal Ministry of Posts and Telecommunications.

The Chairman shall call for nominations from those present who have signed the MoU for membership of the Steering Board. The first Steering Board shall comprise representatives in four constituencies in the following numbers:

| | | |
|---|---|---|
| Broadcasters (public and private) | - | 12 |
| Infrastructure providers (satellite, cable or network operators) | - | 8 |
| Manufacturers | - | 8 |
| Governments/national regulatory bodies | - | 6 |

Where for any constituency the number of nominations exceeds the above numbers then the Chairman of the GA shall proceed with an election. The election in any constituency shall be confined to those present from that constituency. Each eligible signatory shall arrange in preference order those who have been nominated. In the event of a tie between two or more signatories the procedure shall be repeated just for those who have tied for a place on the Steering Board.

The General Assembly shall be closed when its agenda has been completed. The elected Steering Board shall then be convened on the same day and location and elect its Chairman from amongst its membership and conduct such other business as the Steering Board deems essential.

Once the Steering Board has elected its Chairman then all future signed copies of the MoU shall be received by the Steering Board Chairman. Future General Assemblies shall be convened and chaired by the Chairman of the Steering Board.

ANNEX 2

**MEMORANDUM OF UNDERSTANDING FOR THE DEVELOPMENT OF HARMONISED DIGITAL VIDEO BROADCASTING (DVB) SERVICES IN EUROPE.**

*I herewith declare on behalf of my Company/Organisation/Administration our participation in the Project on Digital Video Broadcasting on the basis of the Memorandum of Understanding as from 10 September 1993.*

**Broadcasters**

| | |
|---|---|
| BBC | UK |
| Betatechnik | D |
| BSkyB | UK |
| Canal+ | F |
| Carlton (ITV) | UK |
| Channel 4 | UK |
| CLT | LUX |
| CNN | UK |
| Corporacio Catalana | E |
| Danmark Radio | DK |
| EBU | CH |
| Eurosport | Pan European |
| Filmnet | NL |
| France TV | F |
| GRF [1] | F |
| MTV | UK |
| Premiere | D |
| Pro 7 | D |

[1] Groupement Français de Radiodiffusion Français dans l'UER.

| | |
|---|---|
| RAI | I |
| RTI | I |
| RTL | D |
| RTL 4 | LUX |
| SAT 1 | D |
| Sogeable | E |
| Sveriges TV | S |
| Thames TV | UK |
| TF 1 | F |
| TV 5 (Bruges Group) | F |
| TV Plus | NL |
| VPRT | D |
| ARD (WDR) | D |
| Yleisradio | SF |
| ZDF | D |
| ACT | B |
| M 6 | F |
| RTVE [2] | E |

[2] Radio Television de Espana.

## Manufacturers

| | |
|---|---|
| Alcatel-Sesa | E |
| Amstrad | UK |
| Cambridge Industries | UK |
| Eurodec | F |
| Fuba | D |
| General Instr. | UK |
| Grundig | D |
| Hewlett Packard | Europe |
| JVC Ltd. | UK |
| Löwe Opta | D |
| News Datacom | UK |
| Nokia | CH |
| Pace | UK |
| Panasonic | D |
| Pesa | E |
| Philips | NL |
| Rohde & Schwarz | D |
| Seleco S.p.A. | I |
| Sony Europe | D |
| Tandberg TV | N |
| Thomson-CE | F |
| Tonna Electronique | F |
| Toshiba Consumer Electr. | UK |

**Network Operators**

| | |
|---|---|
| Cable TV Association | UK |
| Eutelsat | Europe |
| Ecca | B |
| France Telecom | F |
| Hispasat | E |
| Le Cable | F |
| National Transcommunication | UK |
| TDF | F |
| PTT Telecom | NL |
| Retevision | E |
| DBP Telekom | D |
| Tele Norway | N |
| CCETT | F |
| HD Divine AB | S |
| Nozema | NL |
| SES | LUX |
| TDF | F |
| Tele Danmark Cable TV | DK |
| Teracom | S |
| France Telecom Cable | F |

**Regulatory Bodies**

| | |
|---|---|
| DTI | UK |
| ITC | UK |
| Ministry for P&T | D |
| Ministry for Communications and Tourism | DK |
| Dirección General de Telecomunicaciones (MOPT) | E |
| Ministry for Industry, P&T and Foreign Trade | F |
| P.T. Ministry | I |
| Ministry of Transport and Communications | NL |