International Telecommunication Union

DTTB Handbook

Digital terrestrial television broadcasting in the VHF/UHF bands

Version 1.02

Edition 2002

Radiocommunication Bureau

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OVERALL INTRODUCTION

In view of their many advantages, digital television systems are destined to replace the analogue television systems which have been used for more than half a century to provide sound and vision services to countless millions of people around the world. The ITU-R has decided to provide guidance to engineers responsible for the implementation of digital terrestrial television broadcasting (DTTB) in the form of a single handbook which combines material dealing with both systems and planning aspects of this new, exciting and yet highly complex topic. The result is a rather large volume and anyone interested in the subject of digital television should find something in it which is both informative and useful.

It is not to be expected that all existing analogue television services will be replaced by digital services overnight. Rather, it will take several, perhaps many, years for this to happen. It is hoped that this handbook will continue to provide information and help throughout the interesting years which lie ahead.

PART 1

SYSTEMS PART

PART 1

TABLE OF CONTENTS

PART 2

(See page 127)

CHAPTER 1

INTRODUCTION

1.1 Scope

This part of the Handbook provides tutorial information and an overview on the subject of digital terrestrial television broadcasting (DTTB) systems. It describes a system designed to transmit high quality audio and video services over a single 6, 7, or 8 MHz broadcasting channel and provides a tutorial on the technologies that support the Recommendations developed by the former Task Group 11/3 during the period 1992 through 1996. It also provides a summary of the state of development in systems specifications and plans for service implementation up to the end of 1998.

1.2 Background

The majority of established broadcasters use terrestrially-based emission systems operating in the VHF/UHF frequency bands. The issue of delivery of high-definition television (HDTV) picture signals and associated sound services within a single 6, 7, or 8 MHz VHF/UHF channel resulted in a review of the application of digital coding techniques in terrestrial transmission.

The migration from a television service dependent primarily on the application of analogue technologies to one that is based on digital technologies has been evolving over the past thirty years. This television service migration is part of a natural outgrowth of the convergence of the television, telecommunications, and computer arts and sciences through the shared use of digital technology.

The input and output signals of television systems, at the camera and at the receiver, respectively, are inherently analogue. Thus, the question "why digital?" is a natural one.

While signal degradations in the analogue signal are cumulative and the characteristics of the degradations make them difficult to distinguish from the video signal, the ability to regenerate a digital pulse train exactly renders the digital signals theoretically immune to impairments from external sources. Digital bit streams can be interleaved within a single channel. This interleaving process allows for the emission, transmission, storage, or processing of ancillary signals along with the video and associated audio. Further, compression techniques based on redundancy reduction can be applied to digitized video and audio services allowing the possibility of transmitting one HDTV service or multiple standard services in an existing broadcasting channel.

The arrival of second and third generation component and composite digital video tape recorders, switchers, animated graphics and special effects machines and agreement on a serial digital signal interface by 1990, have hastened the move to implementation of the all-digital production facility. Digital production and use of digital tape recorders moved the broadcaster's practice on multi-generation editing from five generations of post-production editing using analogue technology to tens of generations using digital technology. The application of digital techniques has reduced camera set-up time from hours to near-instantaneous. Digital library systems made the location of recorded media transparent to the user. Computer control of the entire process penetrated deeply into the programme generation and distribution facility bringing with it precise control and function repeatability. [1]

The only domains in broadcasting left solely to the analogue world had been interplant transmission and final transmission to the consumer. These last barriers were overcome in the early 1990s with the application of digital compression technology, generally constructed on the application of discrete cosine transform (DCT) based encoders and the use of quadrature amplitude modulation (QAM) and related multi-level modulation techniques. [2]

By 1990, efforts within North America to find a means of transmitting a HDTV image within the existing 6 MHz bandwidth, UHF television channel focused on the use of digital data compression and modulation schemes to meet system requirements. Practical feasibility demonstrations of different systems in North America were quickly followed by similar demonstrations in Europe and in the Asia-Pacific region.

By mid-1991, reports of work being done in the United States, in the Nordic countries, in the United Kingdom, France, Italy, Japan, and in other parts of the world, showed that bit-rate reduction schemes on the order of 60:1 could be successfully applied to both HDTV source images and conventional television source images. The results of this implied that HDTV images could be transmitted in a relatively narrow-band channel in the range of 15-25 Mbit/s and that conventional television services could be offered at rates ranging from 1.5 Mbit/s to 12 Mbit/s depending upon the service quality goals. Using standard, proven modulation techniques it would be possible to transmit either a single HDTV programme or multiple conventional television programmes within the existing 6, 7, and 8 MHz bandwidth channels provided for in the VHF and UHF television bands.

In the period between 1991 and 1995, the development of related standards with common system elements for digital satellite, cable and terrestrial broadcasting had been undertaken worldwide. ITU Recommendations developed by the former Task Group 11/3 addressed the common elements of the digital terrestrial television broadcasting system. Specifications for digital satellite and cable broadcasting services were then in the final stages of approval in several areas of the world and are reflected in ITU Recommendations and regional standards. Broadcast services compliant with these standards were also in operation in several parts of the world. Specifications for digital terrestrial television broadcasting having common system elements with those for satellite and cable were also in an advanced state, and these were expected to be completed in 1996.

By 1996, plans for the introduction of digital terrestrial television broadcasting services were in an advanced state in a number of countries.

These advances in communication technology brought digital transmission of television services to a practical reality. The commonly held view is that the application of digital technology to television sciences provides higher picture quality and sound quality than conventional analogue terrestrial television transmission, and at the same time, increases the efficiency of the use of the spectrum by allowing multiple programme services to be broadcast in current single-programme channels.

For digital television services to be successful, there must be a consensus on standards in the areas of source and channel coding, modulation methods, content identification, and error protection and correction. In addition, it is important to consider harmonization with other media.

References

[1] BARON, S. An Overview of the DTTB Model. ITU/SMPTE Tutorial on Digital Terrestrial Television Broadcasting. SMPTE 1994, ISBN 0-940690-24-1, p. 1-5.

[2] Recommendation ITU-R BT.798 – Digital television terrestrial broadcasting in the VHF/UHF bands.

CHAPTER 2

AN OVERVIEW OF THE DTTB MODEL

2.1 The challenge

The application of digital technology to broadcasting provides three primary advantages:

– consistent service quality with improved immunity from noise and near-error-free, perfect picture and sound propagation within the range of performance;

– lower operating costs through the use of compression technology and improved system reliability; and

– increased programme diversity, the ability to provide multiple services in an existing single broadcasting service channel.

The application of digital technology to the television sciences encompasses a number of separate technical disciplines and processes:

– the development of picture, sound and data compression schemes that are compatible with the needs of a digital emission system and provide appropriate levels of system performance;

– identification of picture, sound, and data multiplexing, modulation and channel coding characteristics that meet system requirements;

– understanding the spectrum and planning aspects of digital services including area coverage for different reception and environmental conditions; and

– the ability to provide a digital emission system in the terrestrial VHF/UHF bands allowing for possible simultaneous transmission with existing analogue television services.

The process of digitizing conventional 525-line or 625-line television images results in a video data stream on the order of 270 Mbit/s [1] [2] [3]. The process of digitizing HDTV images results in a video data stream on the order of 1 200 Mbit/s [4]. The technology available in 1992 appeared to be able to support the transport of digital data streams in terrestrial television channels or to efficiently utilize the data space on a satellite transponder at the rate of approximately 3.5 to 4.0 bits/Hz of channel bandwidth. A 6, 7 or 8 MHz channel could then be expected to support a data stream of approximately 20 Mbit/s. This implied compression of the data representing the original source images at a rate as high as 60:1 to accommodate the need to provide HDTV services. The data stream must also provide for the transport of associated audio and ancillary data services such as captioning, programme identification, etc.

2.2 The ITU DTTB model

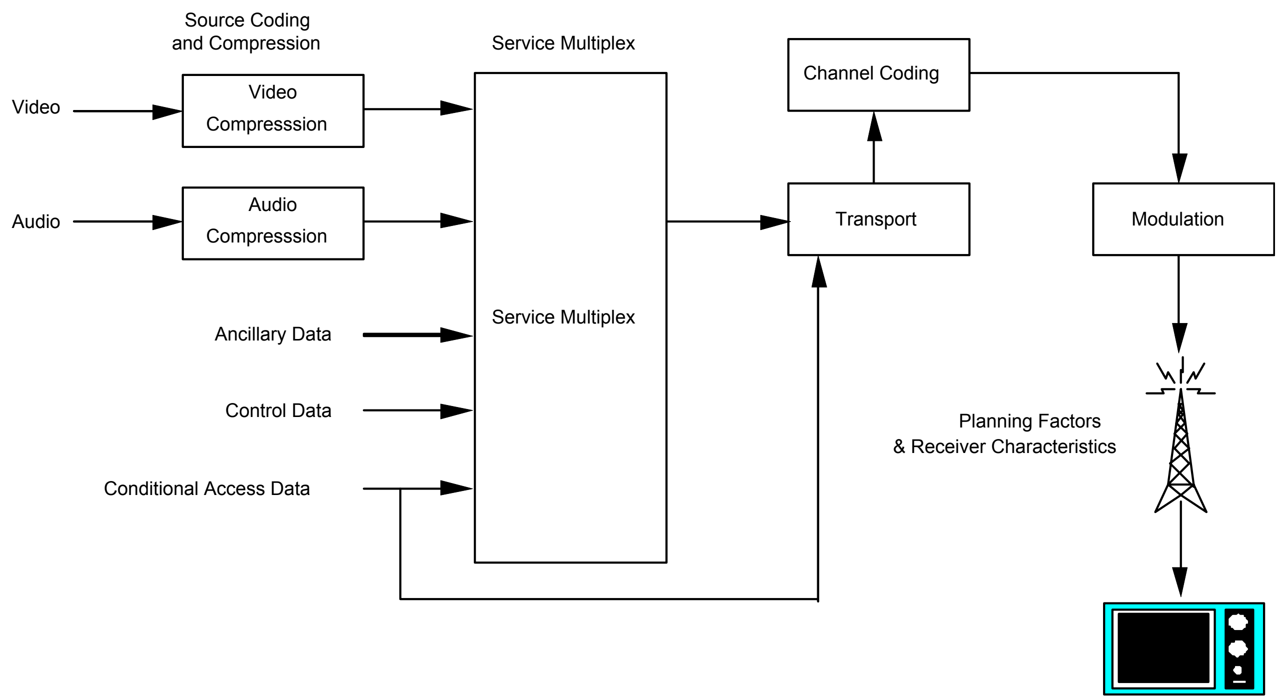


FIGURE 2.1

DTTB system model

Within the International Telecommunications Union, Radiocommunications' Sector (ITU-R), the former Task Group 11/3 was organized and charged in January 1992 with responding to the urgent question concerning digital terrestrial television broadcasting [5]. The former Task Group 11/3 established a model of a digital television broadcasting system and used the model as the basis of its investigations. The model was divided into four subsystems (reference Fig. 2.1):

– source coding and compression;

– service multiplex and transport;

– the physical layer (modulation scheme); and

– planning factors (including both the transmission and receiver planning factors) and implementation strategies.

"Source coding" refers to bit-rate reduction methods also known as data compression and error protection techniques that are appropriate for application on the video, audio, and ancillary digital data streams, The term "ancillary data" includes control data, including conditional access control, and data associated with the programme audio and video services such as closed captioning. "ancillary data" can also refer to independent programme and data services.

The “service multiplex and transport” refers to the means of dividing the digital data stream into “packets” of information, the means of uniquely identifying each packet or packet type, and the appropriate means of multiplexing the video data stream packets, the audio data stream packets, and the ancillary data stream packets into a single data stream. Interoperability or harmonization between digital media such as terrestrial broadcasting, cable distribution, satellite distribution, recording media, and computer interfaces must be a prime consideration in developing an appropriate transport mechanism.

The “physical layer” refers to the means of using the digital data stream information to modulate the transmitted signal. The discussion of modulation techniques includes channel coding and error protection techniques using both single carrier and multiple carrier schemes.

“Planning factors and implementation strategies” include discussions of strategies appropriate for the introduction and implementation of digital terrestrial television broadcast service taking into account existing broadcast services. The plans for any such strategies must recognize the interference characteristics of the over-the-air media and the practical limitations imposed at the receiver.

References

[1] Recommendation ITU-R BT.601 – Studio encoding parameters of digital television for standard 4:3 and wide-screen 16:9 aspect ratios.

[2] Recommendation ITU-R BT.656 – Interfaces for digital component video signals in 525-line and 625-line television systems operating at 4:2:2 level of Recommendation ITU-R BT.601 [Part A].

[3] Recommendation ITU-R BT.1200 – Target standard for digital video systems for the studio and for international programme exchange.

[4] Recommendation ITU-R BT.709 – Parameter values for the HDTV standards for production and international programme exchange.

[5] Question ITU-R 121/11 – Digital terrestrial television broadcasting.

CHAPTER 3

VIDEO AND AUDIO SOURCE CODING

3.1 Definitions

3.1.1 Source and channel coding

Classical communications theory (i.e. based on the work of Shannon) shows that, under certain assumptions, it is possible to separate operations involving data compression and generation of signals for transmission so that they can be dealt with and optimized independently. This is where the concepts of separate source coding and channel coding arise.

3.1.2 Source coding

Source coding involves only characteristics of the source. That is, the communications channel characteristics have no influence on the source coding. Source coding exploits the inherent redundancy in the source signal to reduce the amount of data to be transmitted. This data compression stage may be loss-less or, in the case of video and audio signals, it could introduce some signal degradation. Any operation that looks at the characteristics of the source signal and takes advantage of them for the purposes of data reduction is source coding.

3.1.3 Progressive scanning

Progressive scanning in a raster-based sequence of images simplifies, to some extent, the filtering and interpolation used to convert among formats with different numbers of scan lines, different numbers of samples per line, and different temporal sampling (i.e., picture rate). Since the MPEG-2 algorithm can process complete pictures, progressively-scanned sources can be accommodated and a 24 frame/s film mode can be provided.

3.1.4 Square pixels

For computer graphics, equal geometric spacing among horizontal samples on a line and among samples displaced vertically is desirable for simple rendering of objects that may be transformed after creation. Picture elements (pixels) that exhibit equal horizontal and vertical geometric spacing are termed square pixels.

3.2 Benefits

Digital Television gives many benefits in terms of quality and flexibility, but in its raw form occupies a much greater bandwidth than today's analogue signals. A DTTB service must be able to offer 4:3 and 16:9 aspect ratio component pictures and at a minimum be capable of handling a source resolution of 720(h) $\times$ 480(v) samples per frame (Recommendation ITU-R BT.601) as provided for in Recommendation ITU-R BT.1208.

In the absence of transmission errors, the picture quality would be that offered by the low bit rate redundant data reduction coding. Such quality is not constant, but is highly dependent on the particular content of the picture material being coded. At the end of 1995, work, therefore, continued on methods of assessing picture sequence criticality in order to develop techniques for determining the service quality of low bit rate coded pictures.

3.3 Low bit rate video coding and service quality

A conventional and HDTV studio signal is compressed with image coding for a lower data transmission rate and transmitted with digital modulation over a conventional VHF/UHF channel, with a bandwidth of 6, 7, or 8 MHz.

Apart from image information, capacity is also required for audio, data services like teletext and forward error correction coding (FEC). An example of bit rates for various services is as follows:

Video	24 Mbit/s	(motion-compensated hybrid DCT coding)
Audio	approx. 400 kbit/s	(5 mono audio channels)
Data	64 kbit/s	(undefined content)
FEC	2 Mbit/s	(Reed-Solomon, such as RS (224,208) or RS (227,207)).

3.4 Examples of video scanning standards

a) *Spatial formats* (samples/line × lines/frame)

1920 × 1152, 1920 × 1080, 1920 × 1035, 1440 × 1152, 1280 × 720, 960 × 576, 720 × 576, 720 × 480, 704 × 480, 640 × 480, 352 × 240

b) *Temporal formats* (frames/second)

23.98, 24, 25, 29.97, 30, 50, 59.94, 60

Interlaced or progressive scanned pictures

3.5 Video compression and coding [1] [2] [3]

3.5.1 Introduction

The digital terrestrial television broadcasting (DTTB) system is designed to transmit high quality video and audio over a single 6, 7, or 8 MHz terrestrial channel. Modern digital transmission technologies can deliver a maximum of between 17 Mbit/s and 20 Mbit/s to encode video data within a single 6, 7, or 8 MHz terrestrial channel. This means that encoding a HDTV video source whose resolution is typically five times that of the conventional television (NTSC, PAL or SECAM) resolution requires a bit-rate reduction by a factor of 50 or higher. To achieve this bit-rate reduction, there is worldwide agreement on the use of MPEG-2 video coding. In order to meet the requirements of the many applications and services envisioned, the DTTB system must accommodate both progressive and interlaced scanned pictures across a broad range of spatial and temporal resolutions. Video compression may represent the severest challenge to the DTTB system.

3.5.2 Introduction to MPEG

The Moving Picture Experts Group (MPEG) is an international group formed under the auspices of the ISO and IEC. It is formally known as ISO/IEC JTC 1/SC 29/WG 11.

MPEG's original terms of reference were to provide a "generic coding method of moving picture images and of associated sound for digital storage media having a throughput of up to about 10 Mbit/s. The coding method to be defined is expected to have applications in many other areas as distribution and communication".

The development of standards was split into two phases – MPEG-1 and MPEG-2.

MPEG-2 was later extended to encompass HDTV (also loosely referred to as MPEG-3). MPEG-1 commenced in 1988 and was concerned with compressed video at bit-rates around 1.5 Mbit/s. This was appropriate for mass storage devices such as CD-ROMs and transmission on 1.554 and 2.048 Mbit/s PDH digital channels. MPEG has registered the committee document of this standard as ISO/IEC 11172.

Development of the MPEG-2 standard commenced in July 1990. The aim was to define a standard for the coded representation of audio-visual information providing broadcast quality at data rates up to 15 Mbit/s, based on the Recommendation ITU-R BT.601 digital television standard. In November 1991 MPEG carried out a program of formal subjective tests on a total of 32 video coding algorithms from Europe, North America and the Far East. Following this evaluation a test model algorithm was defined. It uses a hybrid-DCT approach (refer to section on digital compression techniques) and provides flexibility for further improvements.

At the March 1993 MPEG meeting in Sydney and the July meeting in New York the MPEG “profiles” and “levels” specifications were essentially finalized. Table 3.1 provides a brief definition of the five Profiles and lists the pixel resolutions that characterize the four Levels. It also indicates the maximum bit rates applicable to the valid profile/level combinations. From a broadcaster’s perspective the standard will accommodate:

– both interlaced and progressively scanned pictures;

– 4:2:0 and 4:2:2 picture sampling schemes;

– a variety of picture resolutions (up to theoretically 16 000 pixels × 16 000 lines) and including all the commonly used field/frame rates in broadcasting applications;

– coding “scalability”. Briefly this feature allows a standard definition (SDTV) or limited definition (LDTV) decoder to extract the information it requires from a higher level HDTV bit stream. One transmission can then serve all the different definition decoders.

It is expected that most of video requirements will be met by the main profile/main level specification. Note this does not provide for 4:2:2 sampling.

At the New York meeting it was decided to initiate the development of a MPEG-4 standard for very low bit rate coding of video and audio with the objective of producing a draft specification by 1997.

MPEG works in close liaison with other standardization bodies, particularly ITU-T, ITU-R and SMPTE. The former Radiocommunication Task Group 11/3 on Digital terrestrial television broadcasting took an active interest in MPEG standards.

The most significant point to note is that MPEG standards are not precise hardware implementation standards but rather generic descriptions of how the compressed set of video, audio and data signals will be multiplexed into a stream of digital packets for transmission. This standardization of the coding will in turn allow the decoder function to be standardized. In this sense the standard “presumes” the use of certain encoder hardware functions. It is therefore quite possible for different manufacturer’s implementations of MPEG encoders to display differing picture quality.

3.5.3 Digital compression techniques

All current television systems contain redundant information, that is information which is not required to faithfully convey the picture between two points in a network. A modest degree of compression can be effected by simply removing this information before transmission. As this does not affect the picture quality it is referred to as a “lossless” compression technique. For example most of the sync information can be removed from a PAL/NTSC video signal.

However to obtain higher compression ratios, techniques have to be employed which do affect picture quality, albeit by a very small degree. These are characterized as “lossy” methods. The particular lossy methods which are utilized in the MPEG and similar types of compression systems are described in this section. The descriptions relate to progressively scanned pictures, however it should be noted that MPEG-2 allows for the coding of both progressive and interlaced pictures.

TABLE 3.1

MPEG-2 profiles and levels

Levels	Profiles				
	Simple Main without B-frames 4:2:0	**Main** B-frames 4:2:0	**SNR scalability** 4:2:0	**Spatial scalability** 4:2:0	**Professional** 4:2:2
High 1920 x 1152	x	80 Mbit/s	x	x	100 Mbit/s
High -1440 1440 x 1152	x	60 Mbit/s	x	60 Mbit/s	80 Mbit/s
Main 720 x 576	15 Mbit/s	15 Mbit/s 90% of users	15 Mbit/s	x	20 Mbit/s
Low 352 x 288	x	4 Mbit/s	4 Mbit/s	x	x

×: invalid combination.

3.5.4 Inter-frame prediction coding and motion compensation

A powerful method of reducing the information bit rate is to derive a prediction of the picture element (pixel) in question from the previous picture frame. The difference between the actual picture pixel value and its predicted value is then transmitted.

In most pictures the difference (error) value will be small as there is a significant degree of commonality (temporal redundancy) between successive frames. As will be explained later, transmission of a small range of values for most of the time allows the bit rate to be markedly reduced. In the decoder the same prediction process or algorithm recreates the prediction value and the transmitted difference value is added to this to derive the original pixel amplitude.

To improve the prediction process, a macroblock of 16×16 pixels in the current picture is compared with all the 16×16 blocks over a defined search area in the previous picture. The block which provides the best match is selected and is subtracted from the current block.

This matching process minimizes the difference values transmitted and, in particular, compensates for movement of objects within the picture. It is referred to as motion compensation. The vector value which defines the relative spatial relationship of the "best fit" block to the current block – refer Fig. 3.1 – is coded and transmitted to the decoder.

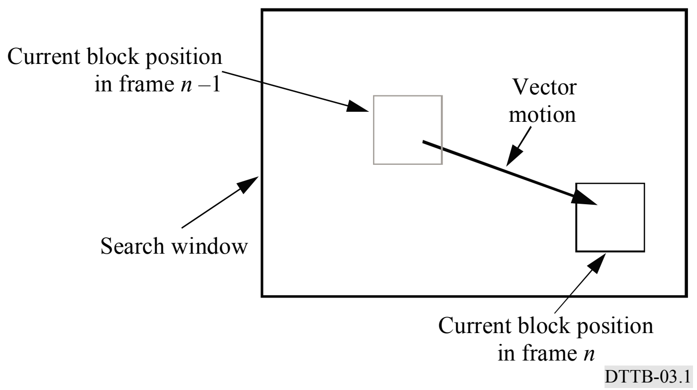


FIGURE 3.1

Motion compensation

The block diagram shown in Fig. 3.2 provides the essential functional elements necessary for predictive coding.

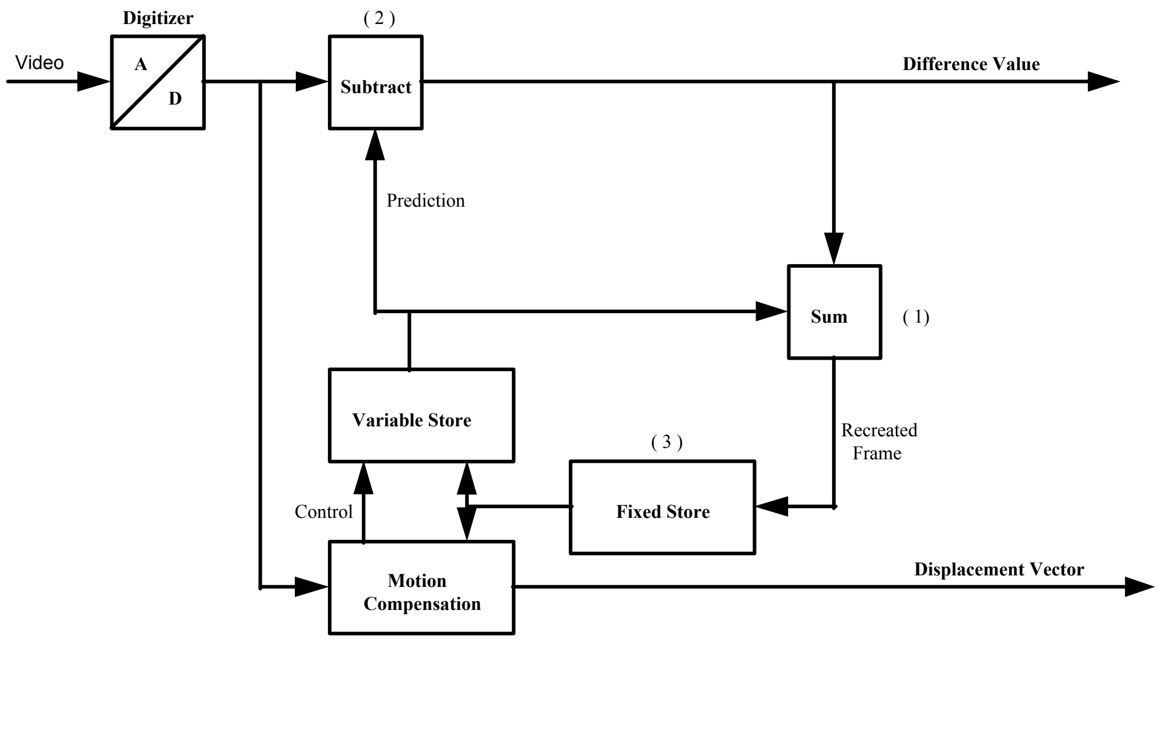


FIGURE 3.2

Inter-frame predictive coding

The fixed store holds the previous frame; the variable store is used for block matching. The summing unit (1) replicates the inverse action of the decoder i.e. inverse action of the differencing unit (2). By including it in the encoder's feedback loop the encoder is able to track and correct for picture discrepancies between the encoding and decoding functions.

Although this description assumes the prediction is formed from the immediate past frame, both MPEG-1 and MPEG-2 allow the prediction to be based on a frame occurring several frames before the current one (refer to § 3.5.11).

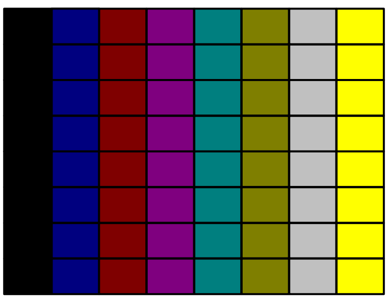
3.5.5 Intra-frame coding

To start off the encoding process the fixed store (3) is initially filled with "null" values. The current frame is then directly coded without reference to a predicted frame. This establishes a reference for the decoder. It is normal practice to transmit such an intra-coded reference frame to the decoder from time to time to prevent the possible accumulation of any prediction or transmission errors.

3.5.6 Discrete cosine transform (DCT) coding

The method used in the MPEG coder for transform coding of video is the discrete cosine transform (DCT) method. DCT converts a block of typically 8×8 pixels from the two-dimensional spatial domain to the frequency domain – hence the term transform coding.

In Fig. 3.3 a grey scale a) is represented by its amplitude values b) and then transformed to frequency coefficients c). Horizontal frequency terms increase from left to right, vertical frequency terms from top to bottom. Hence the upper left-hand corner represents the zero-frequency or DC (average) term, the lower right-hand the highest frequency term.



a)

0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5
0	12.5	25	37.5	50	62.5	75	87.5

b)

43.8	-40	0	-4.1	0	-1.1	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0

c)

FIGURE 3.3

DCT coding

Note that the transform process in itself does not result in any bit reduction as the same number of bits per coefficient is required for the transformed block as the original block. The key to the process is that the transformed frequency coefficients are more suitable for subsequent bit-rate reduction techniques. In particular the trend for the transformed picture blocks to contain zero or near zero coefficient values – there are 60 "0's" in the grey scale example – can be used to advantage.

In practice in the MPEG video encoder the DCT is applied to the picture frame after it has been subjected to inter-frame prediction coding. Hence the amplitude values before transforming are generally small and this further enhances the trend for the transformed block to contain small coefficients. As a further generalization the block matching (motion compensation) process is closer for low frequency picture content than high frequency detail. Hence the high frequency DCT coefficients can be expected to be larger in amplitude as they represent the difference due to the inexact matching. The same comment does not apply when the input to the DCT is an intra-frame coded picture as no motion compensation is employed. Coding is effected in 8×8 pixel blocks.

3.5.7 Coefficient quantization

In any pulse-code-modulation (PCM) process the input signal is sampled on a repetitive basis and the sampled values are assigned code values corresponding to their amplitudes. To minimize any distortion, the quantization step, i.e. the change in input signal amplitude to move from one code value to the next, must be small. For example in high quality audio, 16-bit coding (65 536 steps) is commonly used. If higher distortion can be tolerated then the number of steps can be reduced.

In video it is well known that the eye is less sensitive to high frequency detail and hence the high frequency DCT coefficients can be more coarsely coded, i.e. fewer quantizing steps, than low frequency coefficients without any perceptible loss of picture quality. This is carried out by dividing the coefficients by a value, "n", greater than one and rounding the result to nearest integer (in a digital sense). The weighting factor, n, varies according to the position of the coefficient in the block with higher frequency coefficients attracting larger values of n.

The computing of the "quantizing matrix" which contains the values of n for a given picture block also takes into account:

– whether luminance or chrominance information is being processed – the eye's response changes between the two;

– whether the block comes from a inter- or intra-frame coded picture – as indicated in § 3.5.6, the distribution of coefficient amplitudes differs between the two;

– the location of the block within the picture and the picture content – some blocks need to be coded more accurately than others; this is particularly true of blocks corresponding to very smooth gradients where slight inaccuracies become noticeable.

In addition to this frequency-dependent quantization it is possible to further reduce the number of quantizing steps needed to describe the range of DCT coefficient values by using a non-linear quantizing law i.e. amplitude dependent. Referring to Fig. 3.4 it is seen that large value coefficients are more coarsely coded than small value ones. The quantizer output codeword length is thus reduced relative to the input. Also all values within the dead band are set to zero.

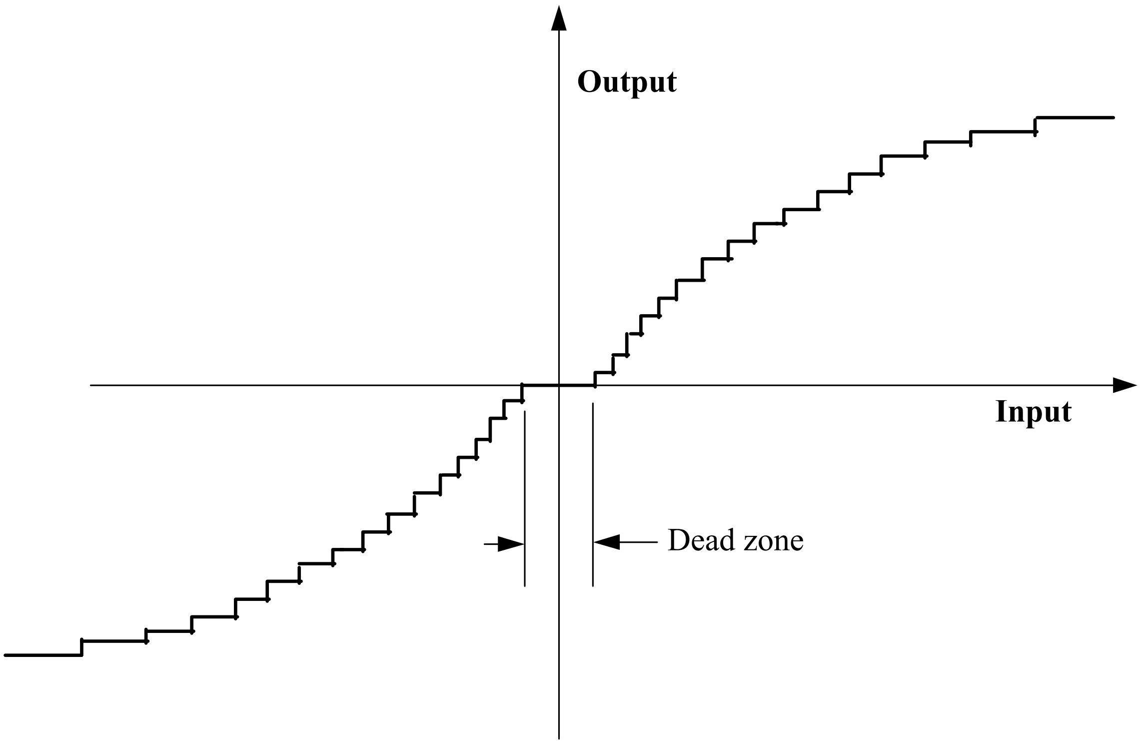


FIGURE 3.4

Non-linear quantizer characteristic

When coding complex pictures it may be necessary to change the quantization matrix values for every DCT block and the MPEG standards allow for this. Obviously for the decoder to keep track of what the encoder is doing, any changes to the matrix must be transmitted to it.

In summary the quantizing strategy implemented in a typical MPEG video encoder can be very complex, however it is one of the keys to obtaining good picture quality at modest bit rates. Different approaches to the quantizing strategy by manufacturers could result in different levels of performance.

3.5.8 Run length coding

As already discussed the effect of the various coding techniques is to reduce most of the coded values that need to be transmitted to a value of zero or a near-zero value. In practice when the processed DCT coefficients are read out of their store in serial form the output bit stream can be expected to contain strings of “0”s. The likelihood of this occurring can be improved by reading out the store in “zig-zag” fashion as depicted in Fig. 3.5.

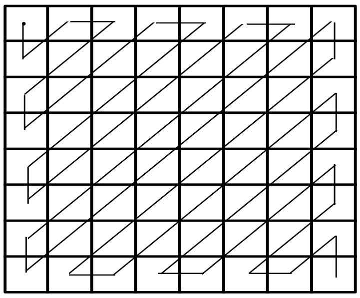


FIGURE 3.5

Scanning of 8 × 8 pixel block

This process groups the low- and mid-frequency coefficients (which are more likely to have zero values) together by reading out the store in terms of ascending frequency coefficients. In addition to this zig-zag scanning MPEG-2 allows for an alternative method.

Rather than transmitting the string of contiguous “0”s that typically results when the store is read out the run length coder sends a unique codeword in place of the string. As this codeword is shorter than the run of “0”s it represents the coding bit rate is reduced.

3.5.9 Variable length coding

Variable length coding (VLC) takes advantage of the fact that certain coded values are going to occur more often than others after the picture frame has been subject to prediction, transform and quantization coding. In particular these processes will give rise to a predominance of near-zero DCT coefficients (after quantizing). If frequently occurring values are assigned short length codewords and infrequently occurring ones transmitted using longer codewords an effective bit-rate reduction will be obtained.

As an analogy if English text was being transmitted, “a, e, i” would be sent with short length codes, whereas “z” would be sent using a long codeword. A good example of this is Morse code.

VLC is also referred to as Entropy coding. Note that in itself VLC is a lossless coding technique.

3.5.10 MPEG video encoder

Referring to Fig. 3.6, the feedback loop which simulates the decoder now includes the inverse quantizer and DCT processes. Following the RLC and VLC units the motion compensation vector information is multiplexed into the bit stream. As the codewords are of variable length a buffer has to be employed to allow the bit stream is transmitted at a uniform rate. To prevent the buffer overfilling or emptying a feedback loop provides an additional control input to the quantizer. If the buffer is nearing its capacity the quantizer is instructed to code the coefficient values more coarsely i.e. reduce the number of bits needed to describe the range of values. Conversely if the quantizer is near empty the quantizer can add dummy codewords.

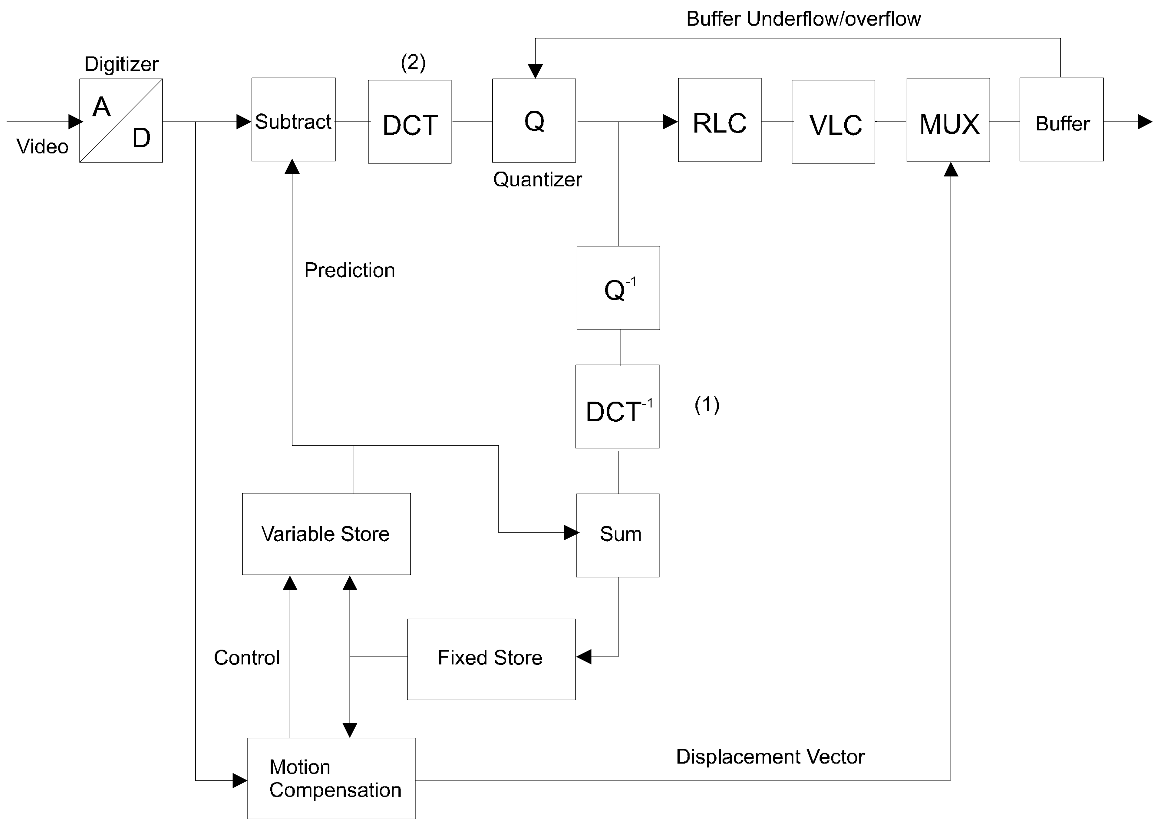


FIGURE 3.6

Basic MPEG video encoder

3.5.11 I, B & P-frames

In MPEG parlance the intra-frame coded pictures (refer to § 3.2) when transmitted are referred to as I-frames and the inter-frame predicted pictures (§ 3.1) are referred to as P-frames. As already mentioned an I-frame is always initially sent to provide a reference for the decoder with P-frames subsequently sent. Additionally MPEG provides for "bidirectional predicted" frames to be sent, interspersed between the I- and P-frames. These are referred to as B-frames. This is depicted in Fig. 3.7.

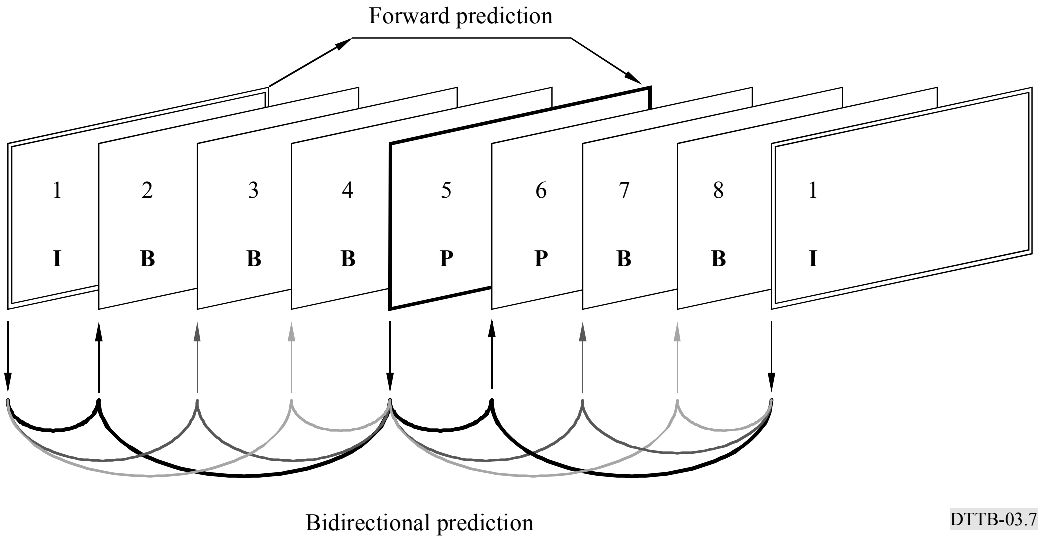


FIGURE 3.7
I, B and P-frames

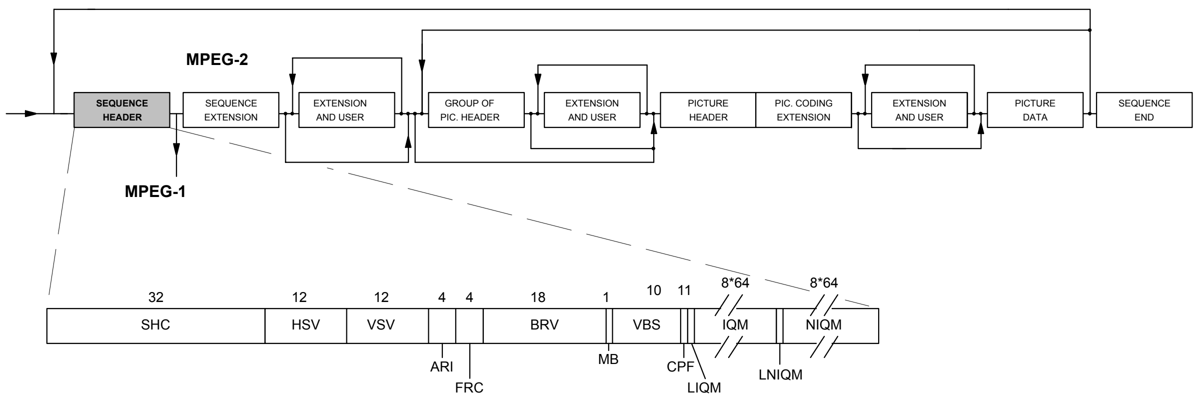
The predicted frame (5) is derived from the intra-frame (1) initially sent, i.e. frame (1) becomes the "previous frame" and frame (5) the "current frame" in the description contained in § 3.5.4.

In this example three B-frames are sent between the I- and P-frames. Frames (2), (3) and (4) are interpolated from both the past frame (1) and the future frame (5). (Looking into the "future" can be done by storing all the frames before processing.) Block matching (motion compensation) occurs using picture information from both frames (1) and (5). One of the advantages of bidirectional interpolation is that the future frame can provide information about a scene change that may not have been present in the past frame. Since B-frames can be derived in the decoder without the frame as such being sent by the encoder the information rate is reduced (higher compression). The disadvantage of using B-frames is the additional processing complexity and memory requirements necessary, particularly in the cost-sensitive decoder.

I, B & P-frames are also called I, B & P-pictures.

3.6 MPEG-2 video bit stream

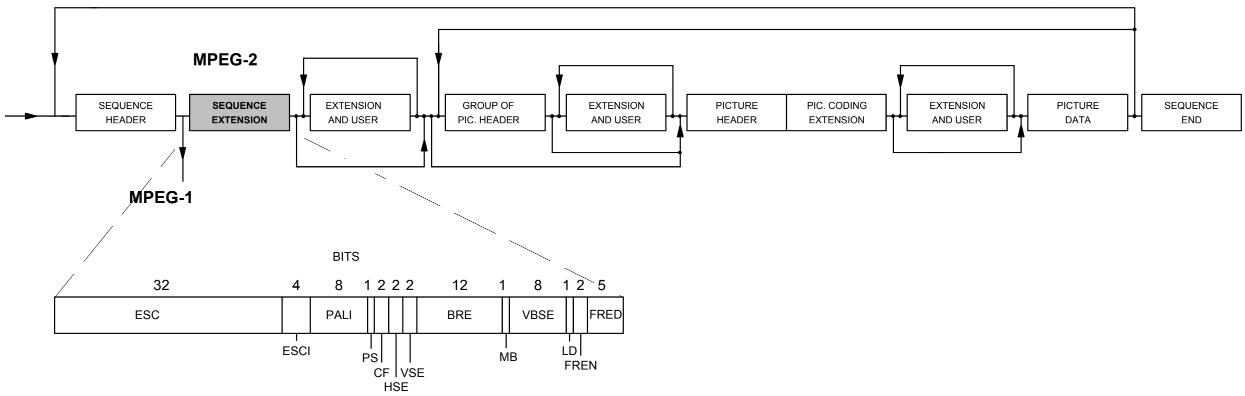
This section and Figs. 3.8 to 3.13 provide detailed information about the structure and content of the MPEG-2 video bit stream.



SHC – Sequence_header_code (32 bits)
HSV – Horizontal_size_value (12 bits)
VSV – Vertical_size_value (12 bits)
ARI – Aspect_ratio_information (4 bits)
FRC – Frame_rate_code (4 bits)
BRV – Bit_rate_value (18 bits)
MB – Marker_bit (1 bit)
VBS – Vbv_buffer_size_value (10 bits)
CPF – Constrained_parameter_flag (1 bit)
LIQM – Load_intra_quantizer_matrix(1 bit)
IQM – Intra_quantizer_matrix (8*64) bits
LNIQM – Load_non_intra_quantizer_matrix(1 bit)
NIQM – Non_intra_quantizer_matrix (8*64) bits

FIGURE 3.8

Sequence header



ESC – Extension_start_code (32 bits)
ESCI – Extension_start_code_identifier (4 bits)
PALI – Profile_and_level_indication (8 bits)
PS – Progressive_sequence (1 bit)
CF – Chroma_format (2 bits)
HSE – Horizontal_size_extension (2 bits)
VSE – Vertical_size_extension (2 bits)
BRE – Bit_rate_extension (12 bits)
MB – Marker_bit (1 bit)
VBSE – Vbv_buffer_size_extension (8 bits)
LD – Low_delay (1 bit)
FREN – Frame_rate_extension_n (2 bits)
FRED – Frame_rate_extension_d (5 bits)

FIGURE 3.9

Sequence extension

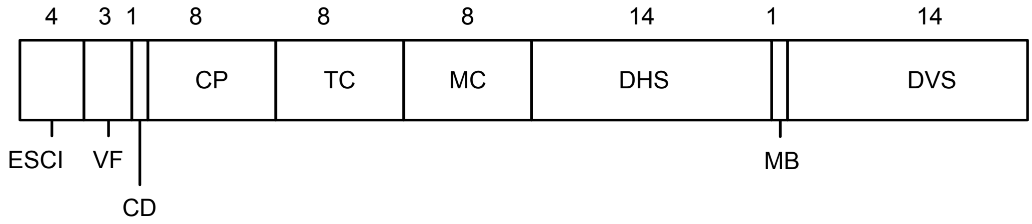
Extension and user data

This description relates to the first “extension & user data” block encountered in the bit stream.

Extension data

Extension start code (32 bits)
Sequence display extension
Sequence scalable extension
Quant matrix extension
Copyright extension
Picture display extension
Picture spatial scalable extension
Picture temporal scalable extension

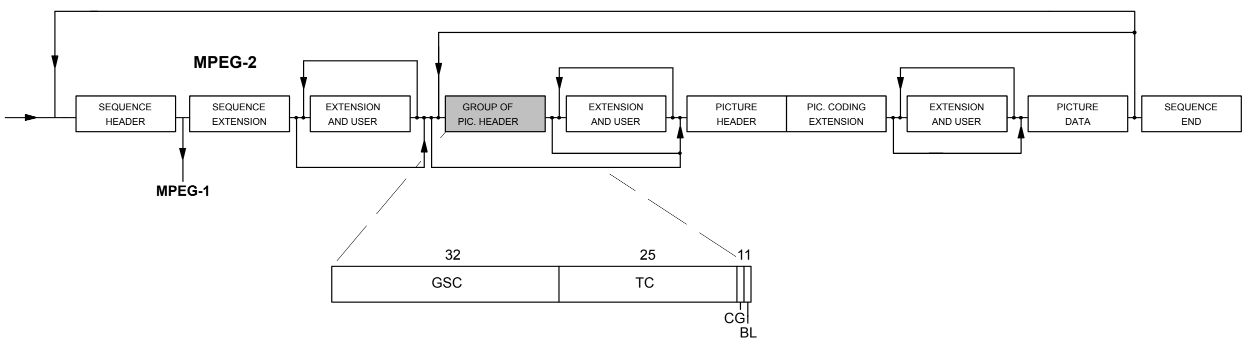
User data



ESCI – Extension_start_code_identifier (4 bits)
VF – Video_format (3 bits)
CD – Colour_description (1 bit)
CP – Colour_primaries (8 bits)
TC – Transfer_characteristics (8 bits)
MC – Matrix_coefficents (8 bits)
DHS – Display_horizontal_size (14 bits)
MB – Marker_bit (1 bit)
DVS – Display_vertical_size (14 bits)

FIGURE 3.10

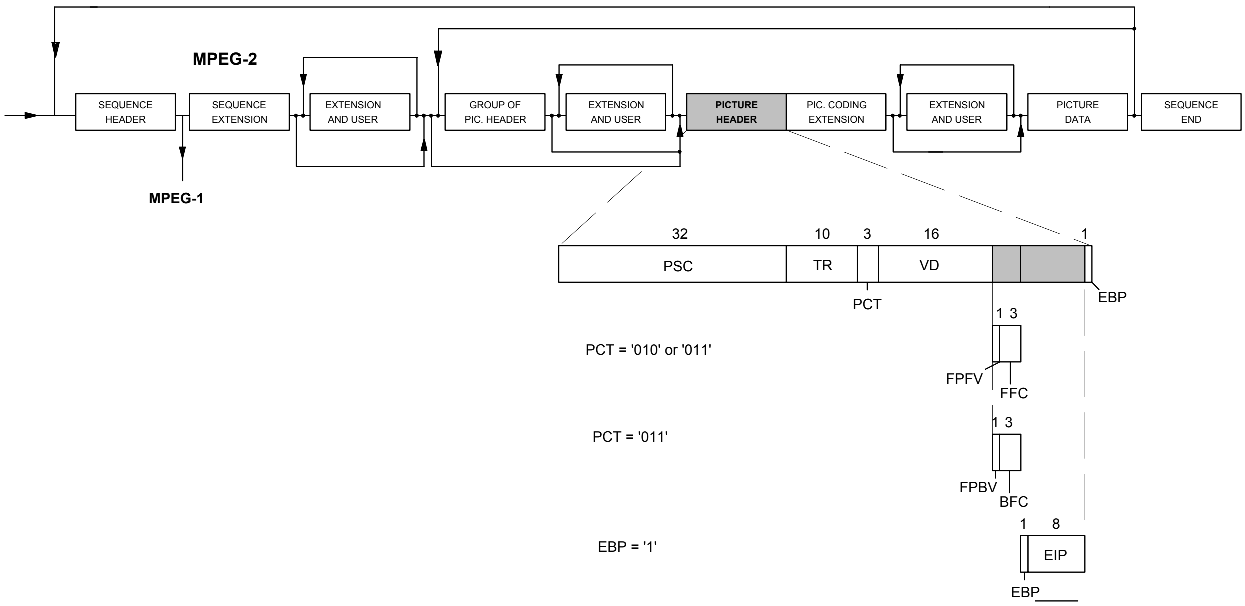
Sequence display extension



GSC – Group_start_code (32 bits)
TC – Time_code (25 bits)
CG – Closed_gop (1 bit)
BL – Broken_link (1 bit)

FIGURE 3.11

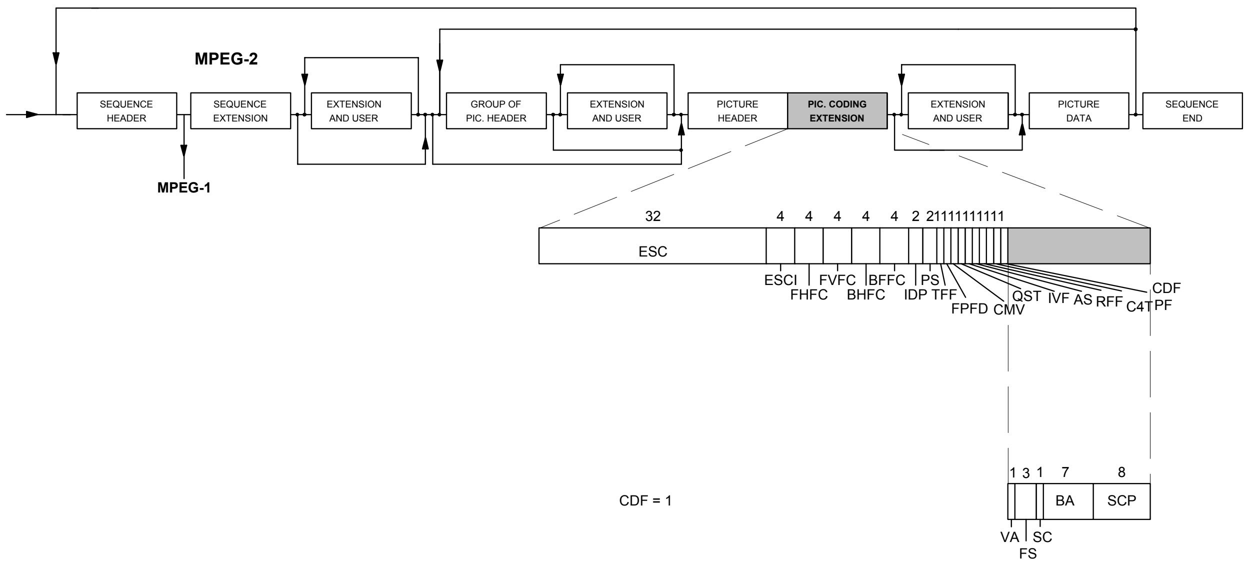
Group of pictures header



PSC – Picture_start_code (32 bits)
TR – Temporal_reference (10 bits)
PCT – Picture_coding_type (3 bits)
VD – Vbv_delay (16 bits)
FPFV – Full_pel_forward_vector (1 bit)
FFC – Forward_f_code (3 bits)
FPBV – Full_pel_backward_vector(1 bit)
BFC – Backward_f_code (3 bits)
EPB – Extra_bit_picture (1 bit)
EIP – Extra_information_picture (8 bits)

FIGURE 3.12

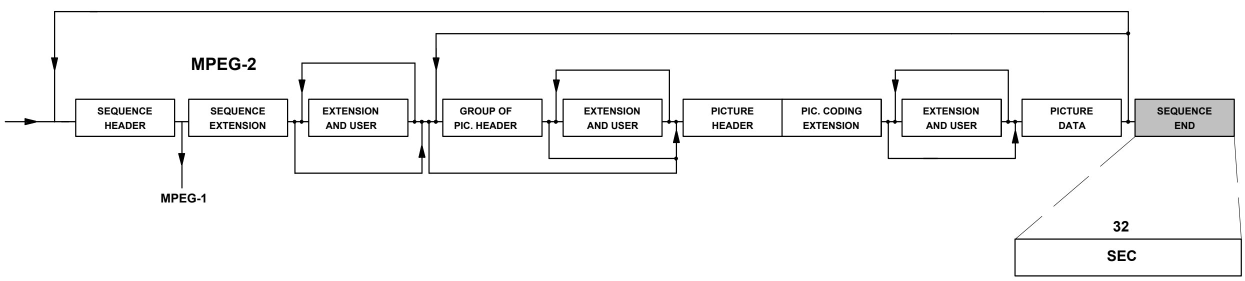
Picture header



ESC – Extension_start_code (32 bits)
ESCI – Extension_start_code_identifier (4 bits)
FHFC – Forward_horizontal_f_code (4 bits)
FVFC – Forward_vertical_f_code (4 bits)
BHFC – Backward_horizontal_f_code (4 bits)
BVFC – Backward_vertical_f_code (4 bits)
IDP – Intra_dc_precision (2 bits)
PS – Picture_structure (2 bits)
TFF – Top_field_first (1 bit)
FPFD – Frame_pred_frame_dct (1 bit)
CMV – Concealment_motion_vectors (1 bit)
QST – Q_scale_type (1 bit)
IVF – Intra_vlc_format (1 bit)
AS – Alternate_scan (1 bit)
RFF – Repeat_first_field (1 bit)
C4T – Chroma_420_type (1 bit)
PF – Progressive_frame (1 bit)
CDF – Composite_display_flag (1 bit)
VA – V_axis (1 bit)
FS – Field_sequence (1 bit)
SCBA – Sub_carrier_burst_amplitude (1 + 7 bits)
SCP – Sub_carrier_phase (8 bits)

FIGURE 3.13

Picture coding extension



SEC – Sequence_end_code (32 bits)

FIGURE 3.14

Sequence end

3.7 Audio compression and coding

3.7.1 Introduction

The Compact Disc has made digital audio popular. Its 16-bit PCM format is an accepted audio representation standard although its bit rate of 706 kbit/s per monophonic channel is rather high. In audio production resolutions up to 24-bit PCM are in use. EBU/AES interface specifications allow for a 16 to 24-bit resolution and 32, 44.1 or 48-kHz sampling frequency. Lower bit rates are mandatory if audio signals are to be transmitted over channels of limited capacity or are to be stored in storage media of limited capacity. Earlier proposals to reduce the PCM rates have followed those for speech coding. However differences between audio and speech signals are manifold since audio coding implies higher values of sampling rate, amplitude resolution and dynamic range, larger variations in power density spectra, differences in human perception, and higher listener expectations of quality. Unlike speech, we also have to deal with stereo and multichannel audio signal presentations.

New coding techniques for high quality audio signals use the properties of human sound perception by exploiting the spectral and temporal masking effects of the ear. The quality of the reproduced sound must be as good as that obtained by 16-bit PCM with 44.1 or 48 kHz sampling rate. If, for a minimum bit-rate with reasonable complexity of the codec, no perceptible difference between the original sound and the reproduction of the decoded audio signal exists, the optimum has been achieved. Source coding systems, have been shown to allow a bit-rate reduction from 768 kbit/s (16 bits at 48 kHz) down to about 100 kbit/s per monophonic channel, while preserving the subjective quality of the digital studio signal for critical signals. This high gain in coding is possible, because the quantizing noise is adapted to the masking thresholds and only those details of the signal are transmitted which will be perceived by the listener.

Recommendation ITU-R BS.1115 addresses two-channel low bit-rate audio coding to be used for digital sound broadcasting applications. For emission applications, ISO/IEC 11172-3 (MPEG-1) Layer II at 128 kbit/s for single channel, and at 256 kbit/s for two-channel configuration is recommended. For contribution and distribution links, ITU-R recommends the use of MPEG-1 Layer II at data rates of 180 kbit/s per channel, or 120 kbit/s per channel if no further cascading is used.

Multichannel audio is of interest in DTTB. At present, multichannel audio is known primarily from the cinema. But even in consumer applications, multichannel has been used for the last few years, e.g. Dolby-Surround with home-TV and VCRs. With the introduction of Advanced or High Definition Television (ADTV, HDTV) with its improved resolution and increased picture size, giving an impression similar to a cinema, an improved audio performance is desired. A way to achieve an improved realism is to use more than two audio channels. Subjective assessments [4] indicate that the switch from mono (1/0) to stereo (2/0) is equivalent to one grade improvement on the ITU-R 5-point quality grading scale; from stereo (2/0) to three channel (3/0) an additional one grade of improvement; from three channel (3/0) to surround sound (3/2) an additional one half grade improvement.

Recommendation ITU-R BS.775 – Multichannel stereophonic sound system with and without accompanying picture, specifies the use of the 3/2 multichannel audio system (left, centre, right; left surround, right surround). The advantage of this system is a large listening area, but a disadvantage is the need for a higher transmission bit-rate. Recommendation ITU-R BS.1196 recommends that DTTB systems should use for audio coding the International Standard specified in ISO/IEC IS 13818-3 or the North American Standard specified in ATSC A/52. With the application of the coding systems recommended by the ITU-R, an economical way for storage or transmission of the multichannel audio is available. Besides the applications with ADTV and HDTV, a lot of multimedia applications, which become more and more popular for consumers, will introduce multichannel audio, if the data-rates can be handled in an economical way.

3.7.2 Characteristics of a DTTB audio system

A suitable sound system for television broadcasting should meet several basic requirements and provide a number of technical/operational features.

3.7.2.1 3/2 Stereo presentation

As regards stereophonic presentation, Recommendation ITU-R BS.775 identifies a centre channel C and two surround channels Ls, Rs, in addition to the basic left and right stereo channels L, R, as the reference sound format. It is referred to as "3/2-stereo" (3 front/2 surround channels), shown in Fig. 3.15 and requires handling of five channels in the studio, storage media, contribution, distribution, emission links, and in the home.

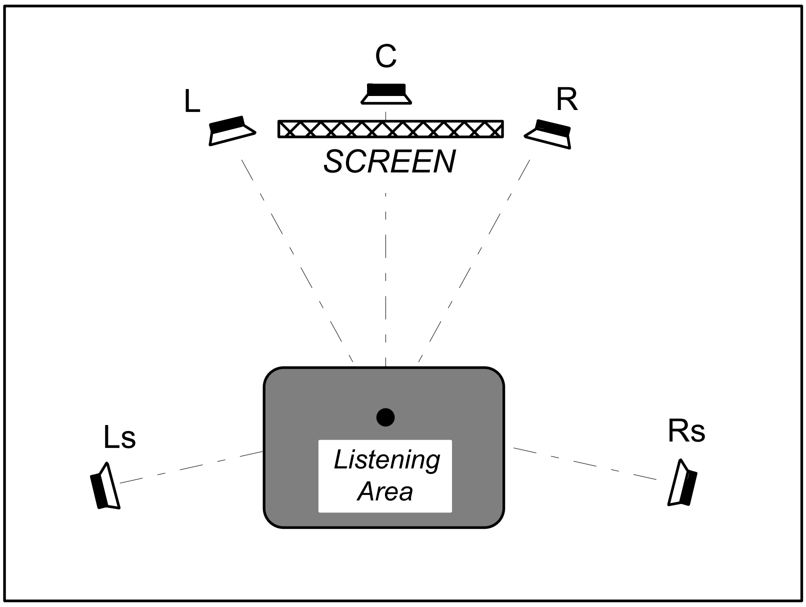


FIGURE 3.15

3/2-stereo reference loudspeaker arrangement

For sound applications with picture accompanying the sound, the three front channels ensure sufficient directional stability and clarity of the picture related frontal images, according to the common practice in the cinema. The 3/2-stereo format has also been found to be the optimum compromise for audio-only applications and an improvement of two-channel stereophony. The addition of one pair of surround channels to the three front channels allows improved realism of auditory ambiance.

3.7.2.2 Low frequency enhancement channel

According to the Recommendation ITU-R BS.775 the 3/2-stereo sound format should provide one optional low frequency enhancement (LFE) channel in addition to the full range main channels with the LFE channel being capable of carrying signals in the frequency range 20 Hz to 120 Hz. The purpose of this channel is to enable listeners, who choose, to extend the low frequency content of the programme in terms of both frequency and level. In this way it is the same as the sub woofer channel used in the digital film sound format, and thus optimum compatibility with film sound material would be ensured in this aspect.

3.7.2.3 Downward compatibility

A hierarchy of sound formats providing a lower number of channels and reduced stereophonic presentation performance (down to 2/0-stereo or even mono) and a corresponding set of downward mixing equations are recommended by Recommendation ITU-R BS.775 to provide downward compatibility. The hierarchy and recommended coefficients for 3/2 configuration are shown in Fig. 3.16. Useful alternative lower level sound formats are 3/1, 3/0, 2/2, 2/0, 1/0. These may be used in circumstances where economic or channel capacity constraints apply in the transmission link or where only a lower number of reproduction channels is desired.

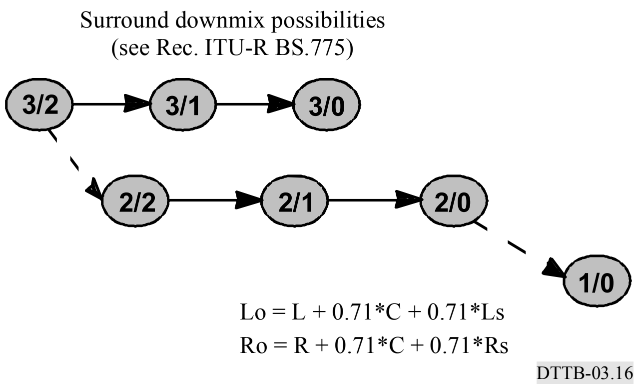


FIGURE 3.16

Down mix from 3/2 down to 1/0 for a future multichannel audio system

3.7.2.4 Backward compatibility

In the case that an existing two-channel DTTB service is extended to multichannel, and compatibility with existing two channel receivers is required, Recommendation ITU-R BS.775 identifies two ways in which this backward compatibility could be realized. The multichannel service may be provided simultaneously with the two-channel service (simulcasting operation). The alternative is that the transmitted left and right channels convey compatible signals, downmixed (matrixed) from the multichannel signals. In addition to the stereo channels, additional channels can be transmitted that carry appropriate signals, which allow retrieval of the original multichannel set of signals by dematrixing. The advantage of the latter method is that less additional data capacity is required to add the multichannel service.

3.7.2.5 Associated services and configurability

In addition to the main multichannel service, associated services may be required.

In some areas multilingual services may be of benefit. This can be accomplished in various ways. For example complete multichannel mixes can be transmitted for each language. Alternatively, an individual dialogue channel for each language may be transmitted in addition to a common multichannel music and effects mix.

Additional sound services may include those for the hearing impaired and for the visually impaired. For the hearing impaired, a clean dialogue channel (ie. no music/effects) is advantageous. For the visually impaired, a descriptive channel would be needed.

Optimum exploitation of the available bit-rate for multichannel stereo performance and sound quality on the one hand and bilingual programmes or associated services on the other depends on the application, on the type of programme, etc. For this reason a number of alternative sound channel/service/quality level configurations is beneficial.

3.7.3 Overview of the DTTB audio system

As illustrated in Fig. 3.17, the DTTB audio subsystem comprises the audio encoding/decoding function and resides between the audio inputs/outputs and the transport subsystem. The audio encoder(s) is (are) responsible for generating the audio elementary stream(s) which are encoded representations of the baseband audio input signals. The flexibility of the transport system allows multiple audio elementary streams to be delivered to the receiver. At the receiver, the transport subsystem is responsible for selecting which audio streams(s) to deliver to the audio subsystem. The audio subsystem is responsible for decoding the audio elementary stream(s) back into baseband audio.

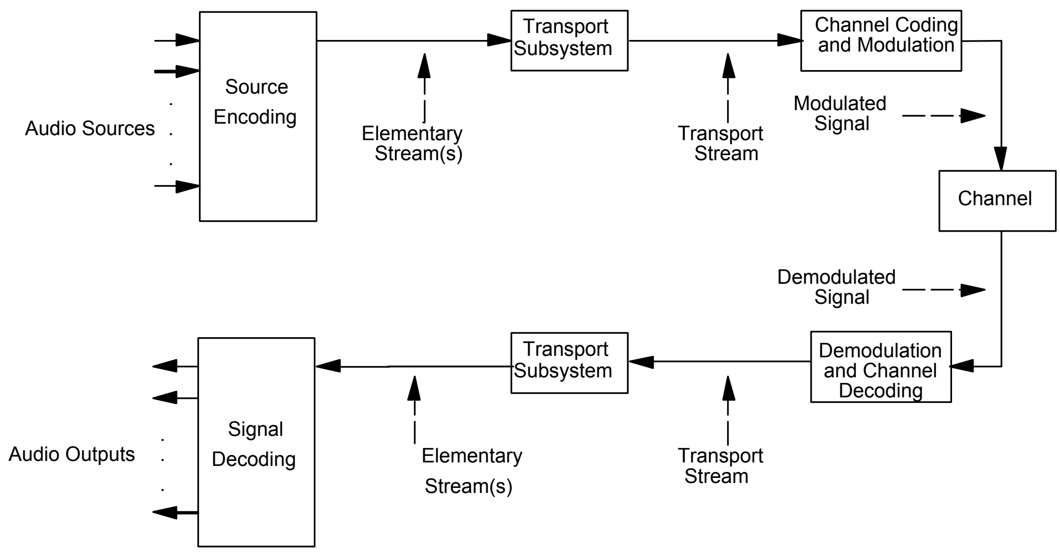


FIGURE 3.17

Audio subsystem within the digital television system

An audio program source is encoded by a digital television audio encoder. The output of the audio encoder is a string of bits that represent the audio source, and is referred to as an *audio elementary stream*. The transport subsystem packetizes the audio data into PES packets which are then further packetized into a transport stream. The transmission subsystem converts the transport packets into a modulated RF signal for transmission to the receiver. At the receiver, the received signal is demodulated by the receiver transmission subsystem. The receiver transport subsystem converts the received transport packets back into an audio elementary stream which is decoded by the digital television audio decoder. The partitioning shown is conceptual, and practical implementations may differ. For example, the transport processing may be broken into two blocks; one to perform PES packetization, and the second to perform transport packetization. Or, some of the transport functionality may be included in either the audio coder or the transmission subsystem.

Additional audio sources, such as multilingual channels are incorporated in the main audio elementary stream in ISO/MPEG-2 coding, they are conveyed by additional elementary streams in AC-3 coding.

Audio encoder interface

The audio system accepts baseband audio inputs with channelization consistent with Recommendation ITU-R BS.775 – Multi-channel stereophonic sound system with and without accompanying picture.

Sampling frequency

The system conveys digital audio sampled at a frequency of 48 kHz, locked to the 27 MHz system clock. Sampling frequencies of 44.1 and 32 kHz may also be supported. Auxiliary services at one half of these frequencies may also be supported in the MPEG-2 system.

Resolution

In general, input signals should be quantized to at least 16-bit resolution. The audio compression system can convey audio signals with a resolution of more than 16 bits.

3.7.4 Overview and basics of audio compression

A major objective of audio compression is to represent an audio source with as few bits as possible, while preserving the level of quality required for the given application. The challenge in providing a bit-rate reduced sound service is to code the signal in a manner in which the errors that are introduced are inaudible to humans. The ISO/IEC MPEG-2 Layer II and AC-3 systems both use a sub-band representation of the audio signal in order to take advantage of the frequency masking properties of the human hearing system. The frequency spectrum of the audio signal is separated into sub-bands by the use of a sub-band or transform filter bank. This results in a representation of the audio signal by sub-band samples (MPEG-2) and by frequency coefficients (AC-3).

The sub-band signals may be quantized because the resulting quantizing noise will be at a similar frequency, and relatively low signal-to-noise ratios (SNRs) are acceptable due to the psychoacoustic phenomenon of masking. A psychoacoustic model of human hearing determines what actual SNR is acceptable in each sub-band. A bit allocation operation distributes the available bits among the sub-bands in accordance with the required SNR. The sub-band values are quantized to the precision indicated by the bit allocation operation and formatted into the audio elementary stream. The basic unit of encoded audio is the audio access unit (or frame) which consists of a fixed number of sub-band samples. Each frame of audio is an independently decodable entity. Knowledge of the bit allocation allows the decoder to unpack and de-quantize the sub-band signals. The synthesis filterbank is the inverse of the analysis filterbank, and converts the reconstructed sub-band signals back into a linear PCM signal.

3.8 The ISO/IEC IS 13818-3 (MPEG-2) Layer II system

3.8.1 Introduction

The selection of an appropriate audio source coding scheme for use in digital television is of vital importance and should be considered very carefully from a number of viewpoints. The choice of coding scheme has an important impact on the achievable quality, the required bit rate and the complexities of the encoders and decoders.

ISO/IEC Standard IS 13818-3 describes in Layer II a digital surround sound system commonly referred to as MPEG-2 audio. It has been adopted in Recommendation ITU-R BS.1196 for the coding of multichannel audio signals. It is based upon and compatible with the two channel coding system described in ISO/IEC Standard 11172-3, commonly referred to as MPEG-1 audio, which is recommended for two-channel sound contribution, distribution and emission purposes in Recommendation ITU-R BS.1115. Layer II is the most widely employed of the three layers defined in MPEG-1, due to its optimum combination of coding gain and encoder/decoder complexity.

The MPEG-2 Audio System is a compatible extension of MPEG-1, extending and enhancing its technical and operational capabilities by the inclusion of additional sampling frequencies and the capability for multichannel coding. Up to five channels may be carried (in addition to up to seven commentary channels) plus an additional low frequency enhancement channel (often referred to as "5.1-channel"). A two-channel downmix of the multichannel audio signal is generated in the MPEG-2 encoder and carried by the MPEG-2 bit stream. The multichannel audio signal is obtained by decoding the MPEG-2 bit stream with an MPEG-2 decoder. As far as two-channel decoding of an MPEG-2 multichannel bit stream is concerned, there are two options in accordance with the desired flexibility of the two-channel downmixing:

– an MPEG-1 decoder may be used; then the downmix as selected and generated in the encoder is obtained;

– a simplified MPEG-2 multichannel decoder, incorporating a downmixing stage, allows for the generation of any downmix to fit the particular user's needs, independent of which downmixing was employed in the encoder.

The second option has a moderately higher complexity than the first; flexibility and complexity go hand in hand.

Features of the MPEG-2 multi-channel audio system:

– The MPEG-2 audio system has been thoroughly tested and proven in accordance with the test procedures of Recommendation ITU-R BS.1116;

– the MPEG-2 audio system has been designed as part of the complete MPEG-2 multiplexing system, including video, audio and data bit streams, with defined synchronization and buffering requirements for ISO/IEC 13818-2 coded video and ISO/IEC 13818-3 coded audio streams;

– when carried in the MPEG-2 transport bit stream, the MPEG-2 audio system provides efficient cross-referencing among the system features (such as announcements, programme types, language types, programme delivery control, etc.);

– the MPEG-2 audio stream is robust against bit errors and is capable of providing error concealment in the receiver in order to improve subjective audio quality under adverse reception conditions, thus enabling a system design in which the failure point will be after that of other signal components (e.g. video, data) sharing the same multiplex;

– unequal error protection may be applied, which may improve the performance of the system under some error conditions;

– the MPEG-2 audio system is designed to be compatible with the existing analogue Dolby ProLogic decoders.

By the use of the MPEG-2 coding system, a multichannel service can be overlaid on a two-channel, MPEG-1 service operating in accordance with Recommendation ITU-R BS.1115. In comparison to simulcasting of separate two-channel and multichannel services, this may lead to a lower required total data capacity and to improved ease of operation.

3.8.2 Principal user features of ISO/IEC 13818-3 Layer II

This section summarizes features of ISO/IEC 13818-3 Layer II multichannel audio coding. In particular, those items are addressed which in a digital terrestrial television broadcasting system are relevant to service providers, broadcasters, network operators and consumers.

3.8.2.1 Coherent MPEG-2 system design

The MPEG-1 audio and MPEG-2 audio standards are both an integral part of a set of standards, MPEG-1 and MPEG-2 respectively. Both MPEG-1 and MPEG-2 contain, next to the parts that define the audio coding and video coding, the parts “systems”, “conformance” and “technical report”.

The systems part defines, among other things, how multiple MPEG audio and MPEG video streams can be multiplexed into one bit stream, and how synchronous playback can be realized. This includes the definition of a time stamp mechanism and a decoder buffer model. The multiplexed stream also contains additional information on the video and audio streams which is desired to be easily accessible at system level, such as language and audio type (hearing impaired, visually impaired, ...), whether the audio and video are exactly locked and whether variable bit rate has been used for the audio.

The conformance part defines procedures to test validity of MPEG bit streams and procedures to test conformance of a decoder implementation to the standard. It also defines minimum implementation accuracy requirements for an MPEG audio decoder.

Technical Report 13818-S is in preparation which contains the source code of a software encoder and decoder, written in C-language. This is intended to help interested parties to get quickly accustomed with the MPEG Video and MPEG Audio standard, to speed up implementation work, and to facilitate testing of implementations.

The MPEG standards also contain a list of companies and institutions that claim to possess Intellectual Property Rights relevant to those standards.

To summarize, MPEG audio is an integral part of a set of standards that not only defines the coding, but also addresses other issues relevant for implementing a complete system.

3.8.2.2 The generic multichannel coding system

The MPEG-2 audio Layer II coding system provides a hierarchy of sound formats from full 5.1 digital surround sound down to a lower number of audio channels and reduced stereophonic presentation performance, with or without additional multilingual/commentary channels. There are two ways to deal with the alternative sound formats: either on the transmitter site which means that only a reduced sound format will be encoded and transmitted, or on the receiving site which means that the lower hierarchy sound formats are obtained by an appropriate downmix in the MPEG-2 Layer II decoder.

MPEG-2 audio Layer II supports any common sampling frequency, i.e. 32, 44.1 and 48 kHz, and a word-length up to 24 bits for the PCM audio input/output signal. The multilingual/commentary channels can be operated either at the same or at half of the sampling frequency of the main surround sound programme.

3.8.2.3 Interoperability and compatibility

The broadcaster may want to consider compatibility to existing systems in order to minimize the number of transcoding or recording stages. This might, for example, be relevant in the selection of the coding scheme to be used for contribution, distribution or commentary links, for simultaneous broadcasting of audio and television services, in the use of pre-encoded source material, or for archiving. In this respect, both ISO/IEC 11172-3 Layer II (which can always be enhanced to ISO/IEC 13818-3 multichannel coding in a compatible way) and ISO/IEC 13818-3 Layer II are reported to be widely employed by a variety of applications.

The existing applications may require forwards and backwards compatibility with MPEG-1 audio. Additionally, the existence of Dolby ProLogic decoders, allowing for analogue surround sound, requires compatibility with the ProLogic decoders.

The MPEG-2 Layer II multichannel coding was developed to be compatible with:

– existing single and dual/stereo channel applications using MPEG-1 Layer II;

– multichannel systems using the Dolby Surround system.

In addition, for those applications where backwards compatibility is not required, MPEG-2 Layer II, allows for a non-matrixed mode. This provides an even higher coding gain due to the fact that, in this mode, certain constraints can be dropped.

The flexible matrix concept of MPEG-2 Layer II allows for these different choices. A control signal which is carried in the multichannel header of the MPEG-2 Layer II bit stream indicates to the decoder which dematrixing procedure is to be applied to reconstruct the full multichannel audio signal, typically consisting of five discrete audio channels, in the decoder.

Compatibility with MPEG-1 audio

One of the basic features of the MPEG-2 audio coding standard is backwards/forwards compatibility with the existing sound format. The backwards compatibility to two-channel stereo may be a strong requirement for many service providers who may provide in the future high quality digital surround sound. Following Recommendation ITU-R BS.1115, there is already a widespread use of MPEG-1 audio Layer II decoders which support mono and stereo sound.

Backwards compatibility of MPEG-2 audio Layer II means that an existing two-channel MPEG-1 audio Layer II decoder should decode properly the basic 2/0-stereo information from the multichannel bit stream.

Forwards compatibility of MPEG-2 audio Layer II means that a multichannel decoder is able to decode properly an MPEG-1 Layer II bit stream in single channel, dual channel, stereo or joint stereo mode.

An ISO/IEC 13818-3 multichannel bit stream can be decoded by either an ISO/IEC 11172-3 or an ISO/IEC 13818-3 decoder, according to the required reproduction properties. This is advantageous not only with respect to the multichannel enhancement of existing services, but also with respect to the price: MPEG offers the basic two-channels service with minimum complexity by using one of the simple ISO/IEC 11172-3 decoder ICs to retrieve the encoded down mix.

Operability with Dolby ProLogic

The compatibility with the Dolby ProLogic decoder is another important feature of MPEG-2 audio Layer II. Dolby ProLogic decoders are widely used in particular for video/audio applications such as stereo television, CD-i, Video CD, Computer multimedia and DVD, and also for audio only applications, e.g. DAB.

A two-channel Dolby Surround signal Lt/Rt can be successfully conveyed by using MPEG-1 Layer II. This is already used, e.g. on Video CD.

Thanks to its flexible matrixing technique, MPEG-2 Layer II provides compatibility to a Dolby ProLogic decoder, i.e. a Dolby Surround signal can be obtained by decoding the MPEG-2 Layer II bit stream using an MPEG-1 decoder to which a ProLogic decoder is connected. All 5.1 discrete audio channels can be reconstructed by an MPEG-2 multichannel decoder, which has to use a special dematrix procedure which is inverse to the Dolby Surround encoding in the encoder.

A simplified MPEG-2 decoder, incorporating a downmixing stage followed by a two-channel synthesis filter-bank, may be used to reconstruct a Dolby Surround compatible downmix from an MPEG-2 multichannel bit stream, irrespective of the downmix selected and employed in the MPEG-2 encoder.

3.8.2.4 Bit rates

MPEG-2 audio Layer II allows for a wide range of bit rates from 32 kbit/s up to 1 066 kbit/s, including all 15 bit rates up to 384 kbit/s defined in the MPEG-1 standard. Compatibility with MPEG-1 is realized by optionally splitting the MPEG-2 audio frame into two parts:

Part A: The Primary Bit Stream is MPEG-1 compatible and contains the beginning of the MPEG-2 specific information in an area which the MPEG-1 decoder regards as ancillary data.

Part B: The Extension Bit Stream contains the remainder of the MPEG-2 specific information. The size of the Extension Bit Stream can be varied in units of bytes with a maximum length of 2 047 bytes.

If the total bit rate does not exceed 384 kbit/s, the whole of the MPEG-2 specific information can be included in the Primary Bit Stream; the Extension Bit Stream is not required. More details including corresponding figures are given in ISO/IEC 13818-3 and in Recommendation ITU-R BS.1196, Annex 1, § 3.1.2.

The wide range of bit rates allows for applications which require a low bit rate and high audio quality, e.g. if only one coding process has to be considered and cascading can be avoided. It also allows for applications where higher data rates, i.e. up to about 180 kbit/s per channel could be desirable if either cascading or post processing has to be taken into account.

Variable bit rates

MPEG-2 allows for variable bit rate. This can be of interest in ATM transmission or storage applications, e.g. DVD. It can also be worthwhile in a broadcasting environment when multiple independent audio and video streams are sharing the same constant channel capacity. Since the use of an inadequate bit rate may lead to various audible artefacts, a constant-capacity service has to allocate the bit rate at which the most critical material can still be conveyed at the desired level of audio quality.

Variable bit rate encoding makes use of the fact that some audio sequences contain less relevant information than others and that the instantaneous bit rate demand may vary considerably. For typical television material, the ratio between the highest and the average required bit rate can be considerable: for example, the ratio has been observed to be about a factor of two for much movie material. As a result, the use of variable instead of constant bit rate coding can lead to a significant reduction of the required capacity.

3.8.2.5 Cascaded coding

Experiments carried out by a specialists group of ITU-R have shown that a coding process can be repeated nine times with MPEG-1 Layer II without any serious subjective degradation, if the bit rate is high enough, i.e. 180 kbit/s per channel. If the bit rate however is only 120 kbit/s, up to 3 coding processes may occur (see Recommendation ITU-R BS.1115).

Due to the similar nature of the coding, ISO/IEC 13818-3 Layer II multichannel coding is expected to perform similarly with multiple cascaded coding (non-matrixed contribution and distribution, matrixed emission).

3.8.2.6 Error resilience

The degradation characteristic of the audio information due to transmission losses should ensure that the audio quality would be always higher than the video quality at any level of channel degradation. Inherent insensitivity to bit-errors and the structure of the encoded bit stream of MPEG-2 Audio Layer II allow efficient channel coding techniques providing a high error resilience using only a low amount of redundancy for error protection.

DAB application of MPEG-1 audio Layer II demonstrates that suitable channel coding with minimum channel bit rates combined with concealment can provide:

– graceful degradation at the fringe edge of the coverage area for single and burst errors;
– improved sound quality for portable or mobile reception in areas of poor reception;
– intelligibility of dialogue in the case of dropped image due to high bit error rate.

For technical details see § 3.8.3.7 and the DAB specification.

3.8.2.7 Editing

Digital audio material can be stored (prerecorded, archived etc.) and edited in coded form. ISO/IEC 13818-3 Layer II coded audio bit streams are easily accessible in units of audio frames. Editing can best be performed on frame boundaries. Due to the smearing effect of the sub-band filtering employed in the decoder (refer to § 3.8.3.3), the editing points will exhibit a soft-transition in the decoded audio. Thus, annoying clicks are avoided.

The editing time resolution depends upon the frame duration. For ISO/IEC 13818-3 Layer II, the frame duration is fixed for a given audio sampling frequency. This enables a well-defined deterministic approach for the computation of the most suitable editing point.

3.8.2.8 Low frequency enhancement (LFE) channel

MPEG-2 audio system provides an LFE channel, according to Recommendation ITU-R BS.775. The purpose is to enable listeners to extend the low frequency content of the reproduced programme in terms of both frequency and level. It is the same as used for digital film sound systems. It carries high level, low frequency sound effects which are intended to be fed to special subwoofer loudspeakers. In that way the magnitude of the low frequency content of the main channels is restricted so that the main speakers are not required to handle these special effect signals.

The low frequency enhancement channel is not used for the entire low frequency content of the multichannel sound presentation. It is an option, at the reproduction side, and thus carries only the additional LFE information. The main channels carry normal low frequency sounds, and they are sufficient on their own if the effects are not desired by the user.

3.8.2.9 Loudness and dynamic range

Dynamic range control (DRC)

The dynamic range of an audio programme signal is the range between the highest and the lowest useful programme signal level. The MPEG-2 audio standard enables the audio signals associated with the television programme to be coded with a wide dynamic range; usually the full dynamic range of the source programme material. This means that the viewer may reproduce the sound accompanying the picture with a realistic dynamic range, free of the artefacts associated with fast-acting dynamic compression systems.

In many cases the source programme dynamic range may be much larger than is required in the domestic environment, for example if there is a high level of background noise, if the viewer wishes to listen to the audio component of the programme as the background to some other activity or simply if a reduced dynamic range is preferred. The means to reduce dynamic range at the receiver is then a useful facility.

The MPEG-2 Layer II audio coding system can provide an embedded dynamic range control system which allows a common encoded bit stream to deliver programming with a dynamic range appropriate for each individual listener. The gain adjustments necessary for dynamic range control can be made unobtrusively if the broadcaster uses an advanced dynamic range controller, such as used for the digital audio broadcasting (DAB) system. At the broadcaster's premises a DRC signal is generated, which describes the audio gain riding to be applied in the receiver, as a succession of values. These DRC data are transmitted in a coded form together with the audio signal. It is important:

– that the audio signal is transmitted with its original programme dynamic, without any pre-compression, in order to provide maximum quality if desired by the user;

– that the same gain adjustments are applied to all channels of a multichannel sound presentation, in order to avoid spurious displacements of phantom sound sources due to the DRC gain changes.

The DRC data may be incorporated in the MPEG-2 Layer II ancillary data field as programme associated data (PAD). In the receiver, the regenerated DRC data may be used to control the audio gain in order to match the dynamic range to the requirements of the listener and to improve audibility in difficult conditions.

In addition, a self-contained dynamic range control system may be implemented in the MPEG-2 Layer II decoder which does not need DRC data from the encoder. This ensures that dynamic range control is provided in any case, for example, if the programme provider does not send DRC data. The system is based on scale factor weighting. The received scale factors are used as sound level information and weighted according to the desired static and dynamic compression characteristic. Satisfying results are achievable without significant increase of decoder complexity and without additional delay.

Loudness normalization

It is important for the digital television system to provide uniform subjective loudness for all audio programmes. Consumers find it very annoying when audio levels fluctuate between broadcast channels (observed when channel hopping), or between programme segments on a particular channel (commercials much louder than the entertainment).

Loudness normalization need not be dealt with any differently than it is for analogue or for linear PCM digital audio.

Recommendation ITU-R BS.645 defines the terms "permitted maximum signal level" and "alignment level". The sound programme signal should be controlled so that the level indicated on a peak programme meter does not exceed the permitted maximum signal level; the instantaneous peaks will be greater. The alignment level is 9 dB below the permitted maximum signal level.

EBU Technical Recommendation R68-1992 specifies that in digital audio production equipment, the alignment level should be 18 dB below the maximum signal coding level.

The input to the audio encoder is linear PCM digital audio and should be in accordance with the above recommendations.

3.8.2.10 Multilingual capability

The MPEG-2 Audio standard provides a variety of configurations for multilingual services. These can be implemented using the syntax of a single MPEG-2 audio stream (including extension bit stream for bit rates over 384 kbit/s). A simple way to offer a main programme in 2- or 3-channels with an alternative language programme in 2-channel is to use the "second stereo programme configuration". This is indicated in the surround field of the MC_header in the MPEG-2 Layer II audio frame.

In addition to a main programme (which may be 5.1 multichannel), there are up to seven "multilingual channels" available. These can be single channels of commentary or dialogue, alternative two-channel stereo presentations, or an alternative multichannel presentation. The multilingual language channels can be sent at the same sampling rate as the main programme, or at half this sampling rate. The lower sampling rate is useful when reduced audio bandwidth is acceptable, for example in the case of commentary channels for sporting events.

Alternatively, more than one audio stream may be sent in the MPEG-2 transport stream. Each audio stream can have its language indicated in an ISO_639_language_descriptor. The audio_type field in this descriptor can indicate "clean effects" i.e. language independent.

3.8.2.11 Programme Associated Data (PAD)

ISO/IEC 13818-3 Layer II offers the capability to carry ancillary data inside the coded bit stream. This data channel is used for transmission of the so-called programme associated data (PAD), in order to enable a number of audio related services. For example, the PAD can comprise indication of associated services (see § 3.8.2.12), music/speech, foreground/background, audio related text information for the display, etc.

The capacity of the PAD channel can be allocated at will, and even be varying over time. The operation of an ISO/IEC 13818-3 audio decoder does not depend upon the programme associated data contents. Also, ISO/IEC 13818-3 does not prescribe the formatting of the PAD. As a result, an appropriate data system can be tailored to each application, independently from the basic audio coding capability.

Programme associated data are carried in one block per frame. The resulting synchronization of PAD and audio data can be exploited to provide time-critical audio-related services. An example is the dynamic range control information which can be used in a suitable decoder to adjust the gain in order to change the dynamic range of the decoded audio (see also § 3.8.2.9).

3.8.2.12 Associated services and configurability

The DTTB audio system offers sound channel configurability. It provides the capability to decode simultaneously a number of channels which may be allocated to different sound services using different sound formats. Optimum sound presentation performance for the main audio service on the one hand and a maximum number of additional associated services on the other can be achieved economically by providing alternative sound channel configurations.

The flexible use of sound channels is fully supported by MPEG-2 Layer II. The ISO_639_language_descriptor indicates the language as well as the associated service type (music/effects, dialogue, hearing impaired, visual impaired, commentary, others) for each MPEG audio stream in the multiplex.

The system is capable of combining individual services. For example, there may be (optional) associated services which are intended to be combined with the main audio service, such as hearing impaired, visually impaired, commentary. There may be other associated services which are intended to form the main audio service, in particular the final mix of the music/effects service with the specific language of the dialogue service.

Multilingual services

There are basically two concepts for the provision of bilingual or multilingual performance for the main capabilities.

1. *Final mix before transmission*

   For each language or service, a different bit stream is provided. This method leaves full artistic freedom. Any mix of music, effects, and dialogue for an individual language can be performed, fully independent from operational constraints. The configuration (two-channel stereo, 5.1 channel, according to Recommendation ITU-R BS.775) and/or bit rate may be different for each language. The disadvantage is however that for *N* languages *N* times the bit rate for one language would be needed. The bit rate could in some cases be reduced, e.g. by transmitting only the main service in 5.1 channel, and alternative languages in two-channel stereo.

2. *Final mix in the receiver*

   The music/effects pre-mix (music/effects service without dialogue) may be transmitted in full 5.1 channel digital surround, accompanied by at least one dialogue channel per language. The final mix of both is performed in the receiver.

   The additional bit rate required for each additional language is only that needed for one dialogue channel. The disadvantage is of course the artistic limitation: this does not allow unrestricted placement of the dialogue and inclusion of dialogue reverberation in other channels. However, for certain programme types (e.g. sports) the final combination of dialogue and music/effects service is attractive due to the bit rate savings.

The various methods to convey multilingual services are particularly well supported by the MPEG Audio and MPEG Systems standard. The transmission of independent data streams for each language is fully supported by the MPEG systems layer. The ISO 639 language descriptor indicates the language and the format for each MPEG audio stream in the multiplex. The preference of the user for a certain language and/or audio type, stored in the receiver memory, together with the information in the language descriptor, can be used to select automatically one of the audio streams available in the MPEG multiplex.

Typically, the dialogue would be mixed in the centre channel, but alternatively a receiver could offer the user the possibility to pan the dialogue across any channel. Furthermore, the dialogue service could be performed not in the monophonic format but alternatively in the stereophonic format (2/0-stereo or 3/0-stereo), offering enhanced realism at the expense of higher bit rate. Dialogue can be panned and its reverberation can be distributed across the front channels.

The provision of the music/effects service together with the dialogue service can be done very efficiently. Each MPEG-2 audio stream can carry up to seven additional (monophonic) languages in the same bit stream. The advantages of this method are that only one audio decoder is required (it is not necessary to access different bit streams for different languages) and that the dialogue is inherently synchronized with the corresponding music/effects portion. An important additional advantage is that the transmission of several dialogue channels within one bit stream together with the music/effects has an averaging effect on the bit rate demand: the probability that all the audio channels have a peak demand in bit rate at the same time is low, and thus the total required bit rate for base sound and *N* dialogue channels will be lower than the bit rate for the base sound plus *N* times the bit rate for one dialogue channel. For example, during those times when dialogue is not present, the dialogue service requires only a very low bit rate, and it may be used for other purposes.

Service for the hearing impaired

Special programme sound for the hearing impaired may be provided as a part of a DTTB service. Some listeners may find it hard to comprehend dialogue when music or effects are also present in the programme mix at a significant level. The hearing impaired service aims to provide more intelligible dialogue. In one form, a hearing impaired service consists of a single channel of dialogue only (which may have been processed, for example to reduce its dynamic range). This signal could be used on its own or together with the main programme sound with the balance between the two adjusted by the listener for optimum effect.

The hearing impaired signal would typically be mixed into the centre channel, or be delivered to a discrete output (which, for instance, might feed a set of open-air headphones worn only by the hearing impaired listener).

Alternatively, the hearing impaired service could be a separate complete programme in two-channel or multichannel stereo. In this case it would consist of a remix of the original programme with greater emphasis on dialogue and removal of unnecessary and distracting sounds.

The service for the hearing impaired can be provided by using one or more of the "multilingual channels" of MPEG-2 Audio. In this way the service is carried in the same audio stream as the main programme sound. In this case the corresponding audio_type field in the MPEG-2 Transport would have the value 0x02 to indicate a hearing impaired service.

Service for the visually impaired

A special commentary for the visually impaired may be provided as a part of a DTTB service. In this type of service, a narrator describes the visual content of the scene so that the programme can be better enjoyed by people who are blind or who have impaired vision. With care, this kind of narration can be made to fit around the existing dialogue of the programme. This service would usually be provided as a single audio channel to be added to the existing programme sound.

An alternative would be to provide a separate complete programme in two-channel or multichannel stereo for the visually impaired. This might be necessary if the existing programme sound needed modification in order for the narration to be included.

A service for the visually impaired can be provided by using one or more of the "multilingual channels" of MPEG-2 audio. In this way the service is carried in the same audio stream as the main programme sound. Alternatively, the service can be provided as a separate audio stream. In this case the corresponding audio_type field in the MPEG-2 Transport would have the value 0x03 to indicate a visually impaired service.

Commentary services

The commentary service conveys optional programme commentary which may be added to any loudspeaker channel by the listener. Typical uses for the commentary service might be optional added commentary during a sporting event, or different types or levels of commentary available to accompany documentary or educational programming.

When commentary service(s) are provided, the receiver may notify the listener of their presence. The listener should be able to call up information (probably on-screen) about the various available commentary services, and optionally request one of them to be selected for decoding along with the main service.

Emergency service

The emergency service is intended to allow the insertion of emergency announcements. The normal audio services do not necessarily have to be replaced in order for the high-priority message to get through. The transport demultiplexer will give first priority to this type of audio service. When the audio decoder receives an emergency service, it may stop reproducing any main service being received and only reproduces the emergency service. This service can particularly be operated using the ISO_639_language_descriptor provided in the ISO/IEC 13818-3 Layer II system.

3.8.3 MPEG-2 Layer II technical details

This section summarizes principal technical elements of ISO/IEC 13818-3 Layer II multichannel audio coding. A more extensive guide on technical details of the ISO/IEC International Audio Coding Standard can be found in the corresponding ISO/IEC "Technical Report" and in Annex 1 of the Recommendation ITU-R BS.1196.

3.8.3.1 Compatibility matrixing

Backwards compatibility implies the provision of compatibility matrices in the multichannel encoder using appropriate down mix coefficients to create the compatible stereo signals Lo and Ro. The inverse matrix to recover the five separate audio channels has to be applied in the MPEG-2 multichannel decoder. The basic matrix equations used in the encoder to convert the five input signals L, R, C, Ls and Rs into the five transport channels T0, T1, T2, T3 and T4 are according to Recommendation ITU-R BS.775.

Concerning the syntax, the compatibility is realized by exploiting the ancillary data field of the ISO/IEC 11172-3 audio frame for the provision of additional channels. The "variable length" of the ancillary data field gives the possibility to carry the complete multichannel extension information. A standard two-channel MPEG-1 audio decoder just ignores this part of the ancillary data field.

The details are explained in the Recommendation ITU-R BS.1196, Annex 1.

3.8.3.2 Forming frames

The encoded multichannel audio signal is structured in frames corresponding to 1 152 PCM audio input samples. Thus the frame length depends on the sampling frequency:

Sampling frequency	32 kHz	44.1 kHz	48 kHz
Frame length	36 ms	26.1ms	24 ms

The MPEG-2 audio frame consists of two parts: the primary part and the optional extension part. If the total bit rate for the multichannel audio signal does not exceed 384 kbit/s, the whole information of the encoded signal may kept within the primary bit stream.

The compatible stereo signal is located in the audio part of the MPEG-1 compatible frame. The extension channels (centre, surround, low frequency enhancement, multilingual) are located in the ancillary data field of MPEG-1. This field starts with the multichannel header providing specific multichannel audio information. The header is followed by the multichannel CRC-field. It is a mandatory 16-bit CRC-word for error detection starting with the first bit of the multichannel header and ending with the last bit of the scale factor select information field. The multichannel CRC-field is followed by the multichannel composite coding status info providing information about dynamic crosstalk, multichannel prediction and transmission channel switching. This is followed by multichannel audio data (bit allocation information, scale factor select information, predictor coefficients, delay compensation for prediction, scale factors, sub-band samples).

The extension frame, carrying part of the multichannel audio data, maybe added to the primary frame, allowing for a total bit rate exceeding the MPEG-1 upper limit of 384 kbit/s. The extension frame starts with a sync-word, followed by a 16 bit CRC-word and a field indicating the number of bytes in the extension frame. The multichannel audio data, overflowing from the primary part, follows. Optionally, ancillary data can be carried at the end of the primary part of the MPEG-2 Audio frame. Detailed figures presenting the structure of the MPEG-2 Audio frame are given in Recommendation ITU-R BS.1196, Annex 1, § 3.1.2.

3.8.3.3 Sub-band filtering

To reduce the amount of information inherent to a PCM audio signal, the MPEG Layer II technique performs a perceptual frequency shaping of the noise resulting from the quantization and coding process. It involves a time/frequency transformation based on a sub-band splitting of the input PCM audio signal by means of a polyphase filter bank. The spectrum of the broad band audio signal is divided in 32 sub-bands of equal bandwidth and one sample per sub-band is produced for each set of 32 PCM input samples. This highly optimized sub-band analysis utilizes a 512-tap prototype filter that is modulated in the frequency domain such as to obtain the 32 desired sub-band filters. This approach, described in the ISO/IEC 11172-3 standard, allows for an equivalent structure using a polyphase filtering and a fast discrete cosine transform. The MPEG standard has been studied by many authors who have proposed very fast implementations of the filter-bank.

This filter bank, optimized in delay and computational load, gives a good trade-off between temporal and frequency resolution, naturally allowing for a reliable handling of any kinds of audio signals, be they stationary or transient (no so-called “pre-echo” distortions encountered).

3.8.3.4 Bit allocation

Each channel of the audio signal is divided into frames of 1 152 samples, which are transformed into 1 152 sub-band samples (36 per sub-band).

This signal simultaneously passes through a psycho-acoustic model that determines for every frame, a dynamic masking curve in the frequency domain. This curve is used to derive the upper limit (also called masking threshold) for the energy of the noise which can be injected in each of the sub-bands of the audio channels during the information reduction process (e.g. quantization of the sub-bands samples) without yielding any audible degradation.

The number of bits allocated to a block of 36 sub-band samples is directly related to the masking threshold of that sub-band for the channel under consideration. A straightforward way of producing such a curve is to analyse each audio component by one of the psycho-acoustic models given in ISO/IEC 11172-3. This ensures the high level of quality which has already been assessed many times during the international ISO/MPEG standardization process.

The use of more sophisticated psycho-acoustic models and allocation is permitted by the ISO/IEC 11172-3 and ISO/IEC 13818-3. This allows for further improvement of the coding efficiency, thanks to the progress made in the fields of psycho-acoustic modelling research, without losing compatibility with existing decoders.

The bit allocation description information is transmitted to the decoder every frame. The coding of this information was optimized according to the long-term statistical distribution of the quantizers over the sub-bands.

3.8.3.5 Scaling and quantization

The number of steps used to quantize the samples in a given sub-band (expressed by the bit allocation) is calculated dynamically according to signal-to-mask ratios, relative to the desired bit rate and to normalized tables of possible quantization per sub-band. This process is performed every frame.

The sub-band samples are encoded by a simple block uniform quantizer. A scale-factor is derived for every set of 12 sub-band samples. These samples are normalized in order to fit the quantizers characteristics. The scale-factors and the coded number of steps used for a given sub-band are transmitted to the decoder. The number of quantization levels may cover a range from 3 to 65 535, with a possibility of not transmitting a sub-band signal altogether.

3.8.3.6 Joint stereo coding

According to binaural models it is possible to determine largely that portion of the stereophonic signal which is irrelevant with respect to the spatial perception of the stereophonic presentation. The stereo-irrelevant signal components are not masked; however, on the other hand, they do not contribute to the localisation of sound sources. Thus, stereo-irrelevant components of any channel may be reproduced via any loudspeaker, without affecting the stereophonic impression, and crosstalk is permissible for certain time intervals in certain regions of the spectrum.

The “intensity stereo” and “dynamic crosstalk” modes exploit this effect. If either of these is enabled for a certain sub-band group, the bit allocation and coded sub-band samples are not contained in the bit stream, and they have to be copied from the transmitted sub-band samples of the corresponding transmission channel. The scale factor select information and the scale factors which will be used for the re-scaling of the sub-band samples are however contained in the bit stream.

3.8.3.7 Error concealment

The audio frame structure of MPEG-2 audio Layer II is closely related to the structure of MPEG-1 audio Layer II which is used for DAB. The following measures in the encoder have been found to provide appropriate concealment strategies:

Frame-CRC (First cyclic redundancy code, specified by ISO/IEC 13818-3)

– A 16-bit parity check word can be used for error detection of the main audio information within the encoded bit stream, i.e. ISO-header, bit-allocation and scale factor select information. If the frame is indicated as non-reliable or non-decodable, a number of measures are possible in the decoder;

– Simple muting of the complete audio frame resulting in improved reception compared with a very annoying impairment due to incorrect reception of main audio information;

- If a substantial error occurs only in the header or side-information of the extension channels (centre and surround channels), either simple muting or replacement by the compatible stereo signal can be applied. The replacement ensures a minimum degree of annoyance;
- Frame replacement by previous correct decoded frames results in improved audio quality, for most of the signals much better compared to simple muting.

Scale factor CRC (Second cyclic redundancy code)

Additionally to the CRC check word defined by the MPEG-2 audio standard, which is used for detecting errors within the significant side information of an MPEG-2 audio Layer II frame, another CRC check may be applied for the detection of errors within the three MSBs of the scale factors. The four (typical) CRC check words are inserted in the MPEG-2 audio Layer II bit stream just in front of the two last bytes of the ancillary data field. Each check word is associated with a group of adjacent sub-bands.

The bits included in the CRC check are the 3 MSBs of all scale factors of the sub-band group, according to their order in the bit stream. The method for the calculation of the CRC words is the same as for the CRC word defined by ISO/IEC 11172-3 for the side information of one MPEG audio Layer II frame.

Reliability-Information

If conventional or punctured convolution codes are used for the error protection, an additional reliability information is derived in the channel decoder. This will give more information on the violated data and would support the adaptation of concealment to the error situation.

3.8.4 Conclusion

The ISO/IEC 13818-3 (MPEG-2) Layer II audio coding system, in conjunction with the ISO/IEC 13818-1 (MPEG-2) Systems Layer, provides a very flexible and efficient audio service for digital terrestrial television broadcasting. It also provides compatibility for receivers conforming to the Recommendation ITU-R BS.1115 for two-channel sound broadcasting.

3.9 AC-3 system description

3.9.1 Introduction

The first implementation of the AC-3 system was developed for the motion picture industry for use on 35 mm motion picture film prints. This system was introduced to the market in 1991. Shortly thereafter, this system was proposed for use as a digital television sound system. The system continued to evolve until its final standardization by the United States Advanced Television Systems Committee in 1994. In this standards proceeding the requirements for DTTB audio services were considered. The combination of the AC-3 Standard (ATSC A/52) and the Digital Television Standard (ATSC A/53) specify all of the audio characteristics for a complete DTTB service. After being tested by the FCC Advisory Committee, the AC-3 system was included in the DTTB system recommended to the FCC for use in the United States.

In 1995, AC-3 became an ITU-R recommended coding system for the DTTB application (see Recommendation ITU-R BS.1196). As standardized, the AC-3 system is incompatible with the (closely related) sound coding system used in the cinema. (Differences are being maintained as to form a protective barrier to motion picture copyright infringement.) The experience gained in the cinema application of multichannel digital sound coding has been of great benefit to the final development of the standardized AC-3 coder.

The philosophy behind the AC-3 coder is to perform a straightforward coding of the individual audio channels. In order to achieve the highest coding gain, the system does not use any form of matrixing. Provision of a downmix version of the multichannel sound signals is considered to be a proper function of the decoder, which rather than reproducing a single predefined 2-ch downmix, can produce a downmix suitable for a listener with mono, stereo, or matrix surround reproduction equipment. This philosophy involves some penalty in decoder complexity, but allows the flexibility in that the downmix may be adjusted to the particular needs of each user. The 2-ch downmixing AC-3 decoder is of lower complexity than the multichannel AC-3 decoder.

3.9.2 AC-3 technical details

3.9.2.1 Forming audio blocks

The process of converting the audio from the time domain to the frequency domain requires that the audio be blocked into overlapping blocks of 512 samples. For every 256 new audio samples, a 512 sample block is formed from the 256 new samples, and the 256 previous samples. Each audio sample is represented in two audio blocks, and thus the number of samples to be processed initially is doubled. The overlapping of blocks is necessary in order to prevent audible blocking artefacts. New audio blocks are formed every 5.33 ms. A group of 6 blocks are coded into one AC-3 sync frame.

3.9.2.2 Window function

Prior to being transformed into the frequency domain, the block of 512 time samples is windowed. The windowing operation involves a vector multiplication of the 512 point block with a 512 point window function. The window function has a value of 1.0 in its centre, and tapers down to almost zero at its ends. The shape of the window function is such that the overlap/add processing at the decoder will result in a reconstruction free of blocking artefacts. The window function shape also determines the shape of each individual filterbank filter.

3.9.2.3 Time division aliasing cancellation transform

The analysis filterbank is based on the fast Fourier transform. The particular transformation employed is the oddly stacked time domain aliasing cancellation (TDAC) transform. This particular transformation is advantageous because it allows the 100% redundancy which was introduced in the blocking process to be removed. The input to the TDAC transform is 512 windowed time domain points, and the output is 256 frequency domain coefficients. The frequency resolution of the filterbank is 93.75 Hz.

3.9.2.4 Transient handling

When extreme time domain transients exist (such as an impulse or a castanet click), there is a possibility that quantization error, incurred in coarsely quantizing the frequency coefficients of the transient, will become audible due to time smearing since the quantization error within a coded audio block is reproduced throughout the block. It is possible for the portion of the quantization error which is reproduced prior to the impulse to be audible. Time smearing of quantization noise may be reduced by altering the length of the transform which is performed. Instead of a single 512 point transform, a pair of 256 point transforms may be performed, one on the first 256 windowed samples, and one on the last 256 windowed samples. A transient detector in the encoder determines when to alter the transform length. The reduction in transform length prevents quantization error from spreading more than a few milliseconds in time, which reduces its audibility.

3.9.2.5 Coded audio representation

The frequency coefficients which result from the transformation are converted to a binary floating point notation. The scaling of the transform is such that all values are smaller than 1.0. An example value in binary notation (base 2) with 16-bit precision would be:

$$0.0000\ 0000\ 1010\ 1100_2$$

The number of leading zeroes in the coefficient, 8 in this example, becomes the raw exponent. The value is left shifted by the exponent, and the value to the right of the decimal point (1010 1100) becomes the normalized mantissa to be coarsely quantized. The exponents and the coarsely quantized mantissas are encoded into the bit stream. The dynamic range of the exponents is sufficient to handle the dynamic range of 24-bit PCM audio signals.

3.9.2.5.1 Exponent coding

Some processing is applied to the raw exponents in order to reduce the amount of data required to encode them. First, the raw exponents of the 6 blocks to be included in a single AC-3 sync frame are examined for block-to-block differences. If the differences are small, a single exponent set is generated which is usable by all 6 blocks, thus reducing the amount of data to be encoded by a factor of 6. If the exponents undergo significant changes within the frame, then exponent sets are formed over blocks where the changes are not significant. Due to the frequency response of the individual filters in the analysis filter bank, exponents for adjacent frequencies rarely differ by more than ±2. To take advantage of this fact, exponents are encoded differentially in frequency. The first exponent is encoded as an absolute, and the difference between the current exponent and the following exponent is then encoded. This reduces the exponent data rate by a factor of 2. Finally, where the spectrum is relatively flat, or an exponent set only covers 1-2 blocks, differential exponents may be shared across two or four frequency coefficients, for an additional savings of a factor of 2 or 4.

The final coding efficiency for exponents is typically 0.39 bits/exponent (or 0.39 bits/sample since there is an exponent for each audio sample). Exponents are only coded up to the frequency needed for the perception of full frequency response. Typically, the highest audio frequency component in the signal which is audible is at a frequency lower than 20 kHz. In the case that signal components above 15 kHz are inaudible, only the first 75% of the exponent values are encoded, reducing the exponent data rate to <0.3 bits/sample.

The exponent processing changes the exponent values from their original values. The encoder generates a local representation of the exponents which is identical to the decoded representation which will be used by the decoder. The decoded representation is then used to shift the original frequency coefficients to generate the normalized mantissas which are quantized.

3.9.2.5.2 Mantissas

The frequency coefficients produced by the analysis filterbank have useful precision dependent on the wordlength of the input PCM audio samples, and the precision of the transform computation. Typically this precision is on the order of 16-18 bits, but may be as high as 24 bits. Each normalized mantissa is quantized to a precision between 0 and 16 bits. The goal of audio compression is to maximize the audio quality at a given bit rate. This requires an optimum (or near-optimum) allocation of the available bits to the individual mantissas.

3.9.2.6 Bit allocation

The number of bits allocated to each individual mantissa value is determined by the bit allocation routine. The identical core routine is run in both the encoder and the decoder, so that each generates the identical bit allocation. The AC-3 bit allocation has a time resolution of one block time (5.3 ms), and a frequency resolution of one filterbank sub-band (94 Hz).

3.9.2.6.1 Backward adaptive

The core bit allocation algorithm is considered backwards adaptive, in that some of the encoded audio information within the bit stream (fed back into the encoder) is used to compute the final bit allocation. The primary input to the core allocation routine is the decoded exponent values, which give a general picture of the signal spectrum. From this version of the signal spectrum, a masking curve is calculated. The calculation of the masking model is based on a model of the human auditory system. The masking curve indicates, as a function of frequency, the level of quantizing error which may be tolerated. Subtraction (in the log power domain) of the masking curve from the signal spectrum yields the required SNR as a function of frequency. The required SNR values are mapped into a set of bit allocation pointers (baps) which indicate which quantizer to apply to each mantissa.

3.9.2.6.2 Forward adaptive

The AC-3 encoder may employ a more sophisticated psycho-acoustic model than that used by the decoder. The core allocation routine used by both the encoder and the decoder makes use of a number of adjustable parameters. If the encoder employs a more sophisticated psycho-acoustic model than that of the core routine, the encoder may adjust these parameters so that the core routine produces a better result. The parameters are inserted into the bit stream by the encoder and fed forward to the decoder.

In the event that the available bit allocation parameters do not allow the ideal allocation to be generated, the encoder can insert explicit codes into the bit stream to alter the computed masking curve, and thus the final bit allocation. The inserted codes indicate changes to the base allocation, and are referred to as delta bit allocation codes.

3.9.2.7 Rematrixing in 2/0 mode

When signals which have been coded as Left, Right, are decoded by a Dolby Prologic Surround decoder, there are signal conditions (predominant Centre (= L + R), or Surround (= L – R)) which cause coding artefacts to be reproduced by different loudspeakers than those which are reproducing the predominant signal. This can result in the unmasking of artefacts. The audibility of these artefacts can be reduced by operating the coder at a higher bit rate than would otherwise have been acceptable for 2-ch stereo reproduction.

When the AC-3 coder is operating in a two-channel stereo mode, an additional processing step is inserted in order to enhance interoperability with Dolby Surround 4-2-4 matrix encoded programs. The extra step is referred to as **rematrixing**. This technique reduces the need for AC-3 signals to be encoded with an increased bit rate when they might be ProLogic decoded.

The signal spectrum is broken into four distinct rematrixing frequency bands. Within each band, the energy of the Left, Right, Sum, and Difference signals are determined. If the largest signal energy is in the Left or Right channels, the band is encoded normally. If the dominant signal energy is in the Sum or Difference channel, then those channels are encoded instead of the Left and Right channels. The decision as to whether to encode Left and Right, or Sum and Difference is made on a band-by-band basis and is signalled to the decoder in the encoded bit stream.

3.9.2.8 Coupling

In the event that the number of bits required to encode the audio signals transparently exceeds the number of bits which are available, the encoder may invoke coupling. Coupling involves combining the high frequency content of individual channels and sending the individual channel signal envelopes along with the combined coupling channel. The psycho-acoustic basis for coupling is that within narrow frequency bands the human ear detects high frequency localization based on the signal envelope rather than the detailed signal waveform.

The frequency above which coupling is invoked, and the channels which participate in the process, are determined by the AC-3 encoder. The encoder also determines the frequency banding structure used by the coupling process. For each coupled channel and each coupling band, the encoder creates a sequence of coupling coordinates. The coupling coordinates for a particular channel indicate what fraction of the common coupling channel should be reproduced out of that particular channel output. The coupling coordinates represent the individual signal envelopes for the channels. The encoder determines the frequency with which coupling coordinates are transmitted. When coupling is in use, coupling coordinates are always sent in block 0 of a frame. If the signal envelope is steady, the coupling coordinates do not need to be sent every block, but can be reused by the decoder until new coordinates are sent. The encoder determines how often to send new coordinates, and can send them as often as every block (every 5.3 ms).

3.9.3 Bit stream syntax

3.9.3.1 Sync frame

The audio bit stream consists of a repetition of audio frames which are referred to as AC-3 sync frames. Shown in Fig. 3.18, each AC-3 sync frame is a self contained entity consisting of synchronization information (SI), bit stream information (BSI), 1536 samples of encoded audio, and a CRC error check code. The sync frame may be considered an audio access unit. Within SI is a 16-bit sync word, an indication of audio sampling rate, and an indication of the size of the sync frame. Sample rates of 32 kHz, 44.1 kHz, and 48 kHz are supported. The frame size may be fixed in every frame for fixed bit rate operation, or made dynamically variable to support variable bit rate operation. Supported bit rates range from 32 kbit/s to 640 kbit/s. Unused data capacity at the end of the frame may be used to carry program associated data or aux data. The formatting of this data is not specified.

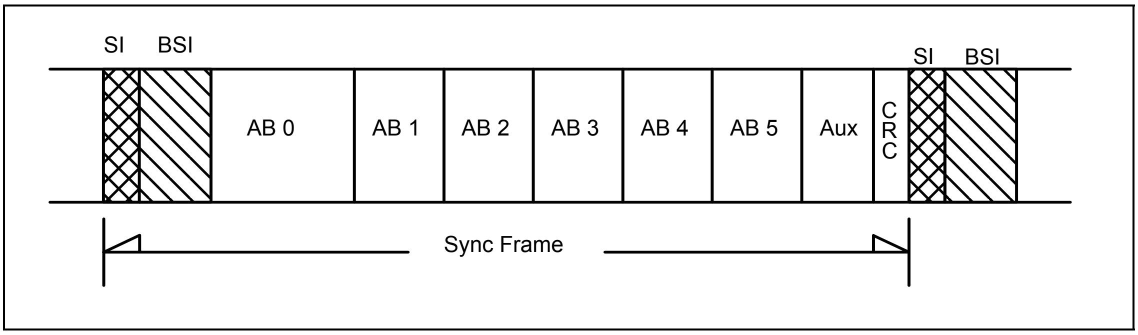


FIGURE 3.18

AC-3 synchronization frame

3.9.3.2 Splicing, insertion

Encoded AC-3 bit streams may be edited. The ideal place to splice encoded audio bit streams is at the boundary of a sync frame. If a bit stream splice is performed at the sync frame boundary, the audio decoding will proceed without interruption. The windowed overlap/add process in the decoder synthesis filterbank will provide a very smooth transition across the splice point. If a bit stream splice is performed randomly, there may be an audio interruption. The frame which is incomplete will not pass the decoder's error detection test and this may cause the decoder to mute. The decoder will not find sync in its proper place in the next frame, and will enter a sync search mode. Once the sync code of the new bit stream is found, synchronization will be achieved, and audio reproduction may begin once again. The outage may be on the order of two frames, or about 64 ms. Due to the windowing process of the filterbank, when the audio goes to mute there will be a gentle fade down over a period of 5.3 ms. When the audio is recovered, it will fade up over a period of 5.3 ms. Except for the approximately 64 ms of time during which the audio is muted, the effect of a random splice of an AC-3 elementary stream is relatively benign.

3.9.3.3 Error detection codes

Each AC-3 sync frame ends with a 16-bit CRC error check code. The decoder may use this code to determine whether a frame of audio has been damaged or is incomplete. Additionally, the decoder may make use of error flags provided by the transport system. In the case of detected errors, the decoder may try to perform error concealment, or may simply mute. A second 16-bit CRC is located within the AC-3 sync frame and allows a decoder to be implemented with an input buffer size of only 2/3 of a frame.

If additional information about data errors is available (from the ECC system) then the audio decoder can consider reproducing errored data. For instance, if it is known that a byte containing only mantissa data is in error, reproduction of that incorrect mantissa value would likely be preferable to performing a frame repeat or mute. If a critical byte of information is in error, then a concealment or mute would have to be performed. The majority of the data in the bit stream is mantissa data.

3.9.3.4 Multiplexing AC-3 in MPEG-2 transport streams

The flexibility of the MPEG-2 systems layer allows non-MPEG defined audio bit streams to be conveyed. Specifications on how the AC-3 elementary stream may be carried can be found in Appendix 1 of Annex 2 of Recommendation ITU-R BS.1196 and Recommendation ITU-R BT.1300. The specifications allow systems to be implemented with proper synchronization and buffering, and with all necessary descriptors. For any bit rate, an AC-3 elementary stream may be conveyed as a single stream through the MPEG-2 multiplex, requiring the assignment of only a single packet ID (PID) value. Receivers may access any single sound service with a single PID filter.

3.9.4 Loudness and dynamic range

3.9.4.1 Loudness normalization

It is important for the digital television system to provide uniform subjective loudness for all audio programmes. Consumers find it very annoying when audio levels fluctuate between broadcast channels (observed when channel hopping), or between programme segments on a particular channel (commercials much louder than the entertainment). One element which is found in most audio programming is the human voice. Achieving an approximate level match for dialogue (spoken in a normal voice, not shouting or whispering) amongst all audio programming is a desirable goal. The AC-3 audio system provides syntactical elements which make this goal achievable.

Since the digital audio coding system can provide more than 100 dB of dynamic range, there is no technical reason for dialogue to be encoded anywhere near 100% as is commonly done in analogue television. However, there is no assurance that all programme channels, or all programmes or programme segments on a given channel, will have dialogue encoded at the same (or even similar) level. Lacking a uniform coding level for dialogue (which would imply a uniform headroom available for all programs) there would be inevitable audio level fluctuations between programme channels or even between programme segments. In the extreme case, it would be possible for commercial messages to have a sales pitch encoded at the same level as the highest level sound effects in a major motion picture.

Encoded AC-3 elementary bit streams are tagged with an indication (the syntactic element dialnorm) of the subjective level at which dialogue has been encoded. Different audio programmes may be encoded with differing amounts of headroom above the level of dialogue in order to allow for dynamic music and sound effects. The digital television receiver (and all AC-3 decoders) are able to use the value of dialnorm to adjust the reproduced level of audio programmes so that different received programmes have their spoken dialogue reproduced at a uniform level. Some receiver designs may even offer the listener an audio volume control calibrated in absolute sound pressure level. The listener could dial up the desired listening-level for dialogue, and the receiver would scale the level of every decoded audio programme so that the dialogue is always reproduced at the desired level. All compliant AC-3 decoders will make use of the dialnorm parameter and perform the intended scaling. This is an inherent feature of the AC-3 Standard which all decoders must implement.

The BSI portion of the sync frame contains the 5-bit dialnorm field which indicates the level of average spoken dialogue within the encoded audio program. The indication is relative to the level of a full scale 1 kHz sinewave. The measurement of dialogue level is done by a method which gives a subjectively accurate value. The measurement of subjective loudness is not an exact science, and new measurement techniques will be developed in the future. A measurement method which is currently available and quite useful is the "A" weighted integrated measurement (L_{Aeq}). This measurement method will be used until a more accurate method is standardized and available in practical equipment. Any new measurement methodology which is developed should be normalized (scaled) so that its results generally match those of the L_{Aeq} method.

It is important for broadcasters and others who deliver encoded audio bit streams to ensure that the value of dialnorm is correct. Incorrect values will lead to unwelcome level fluctuations in consumer homes. The worst-case example of incorrect (or abusive) setting of dialnorm would be to broadcast a commercial message which indicates dialogue at a low level, but which is actually encoded with dialogue at full level. This would result in the commercial message being reproduced at the same level as a full scale explosion in a feature film (>100 dB SPL in some home theatre setups!). If such abuses occur, there may be a demand for regulatory enforcement of audio levels. Fortunately, bit streams which contain an incorrect value of dialnorm are easily corrected by simply changing the value of the 5-bit dialnorm field in the BSI header.

There are two primary methods which broadcast organizations may employ to ensure that the value of dialnorm is set correctly. The first method is to select a suitable dialogue level for use with all programming and conform all baseband audio programmes to this level prior to AC-3 encoding. Then the value of dialnorm can be set to one common value for all programmes which are encoded. Conforming all programmes to a common dialogue level may mean that for some programmes the audio level never approaches 100% digital level (since they have to be reduced in gain), while for other programmes non-reversible (by the receiver) limiting must be engaged in order to prevent them from going over digital 100% (since they had to be increased in gain). Pre-encoded programmes can be included in broadcasts if they have had the value of dialnorm correctly set, and the receiver will then conform the level.

The second method is to let all programming enter the encoder at full level, and correct for differing levels by adjusting the encoded value of dialnorm to be correct for each program. In this case, the conforming to a common level is done at the receiver. This method will become more practical as computer remote control of the encoding equipment becomes commonplace. The database for each audio programme to be encoded would include (along with items such as number of channels, language, etc.) the dialogue level. The master control computer would then communicate the value of dialogue level to the audio encoder which would then place the appropriate value in the bit stream.

In the case where a complete audio programme is formed from the combination of a main and an associated service, each of the two services being combined will have a value of dialnorm, and the values may not be identical. In this case, the value of dialnorm in each bit stream should be used to alter the level of the audio decoded from that bit stream, prior to the mixing process which combines the audio from the two bit streams to form the complete audio program.

3.9.4.2 Dynamic range compression

It is common practice for high-quality programming to be produced with wide dynamic range audio, suitable for the highest quality audio reproduction environment. Broadcasters, serving a wide audience, typically process audio in order to reduce its dynamic range. The processed audio is more suitable for the majority of the audience which does not have an audio reproduction environment which matches that of the original audio production studio. In the case of analogue television, all viewers receive the same audio with the same dynamic range, and it is impossible for any viewer to enjoy the original wide dynamic range audio production.

The AC-3 coding system provides an embedded dynamic range control system which allows a common encoded bit stream to deliver programming with a dynamic range appropriate for each individual listener. A dynamic range control value (dynrng) is provided in each audio block (every 5 ms). These values are used by the audio decoder in order to alter the level of the reproduced audio for each audio block. Level variations of up to ±24 dB may be indicated. The values of dynrng are generated in order to provide a subjectively pleasing but restricted dynamic range. The unaffected level is dialogue level. For sounds louder than dialogue, values of dynrng will indicated gain reduction. For sounds quieter than dialogue, values of dynrng will indicate a gain increase. The broadcaster is in control of the values of dynrng, and can supply values which generate the amount of compression which the broadcaster finds appropriate. The use of dialogue level as the unaffected level further improves loudness uniformity. This control information is an inherent part of the AC-3 elementary stream.

By default, the values of dynrng will be used by the compliant audio decoder. It does not matter that the AC-3 decoder may not be resident in the DTTB receiver. Even if the decoder is in another piece of equipment, the behaviour of the AC-3 decoder to the dynrng control signal will be uniform. The receiver will thus reproduce audio with a reduced dynamic range, as intended by the broadcaster. The receiver may also offer the viewer the option to scale the value of dynrng in order to reduce the effect of the dynamic range compression which was introduced by the broadcaster. In the limiting case, if the value of dynrng is scaled to zero, then the audio will be reproduced with its full original dynamic range. The optional scaling of dynrng can be done differently for values indicating gain reduction (which reduces the levels of loud sounds) and for values indicating gain increases (which makes quiet sounds louder). Thus the viewer may be given independent control of the amount of compression applied to loud and quiet sounds. Therefore, while the broadcaster may introduce dynamic range compression to suit the needs of most of the audience, individual listeners may have the option to choose to enjoy the audio programme with more or all of its original dynamic range intact.

The dynamic range control words may be generated by the AC-3 encoder. They may also be generated by a processor located before or after the encoder. If the dynamic range processor is located prior to the encoder, there is a path to convey the dynamic range control words from the processor to the encoder, or to a bit stream processor, so that the control words may be inserted into the bit stream. If the dynamic range processor is located after the encoder, it can act upon an encoded stream and directly insert the control words without altering the encoded audio. In general, encoded bit streams may have dynamic range control words inserted or modified without affecting the encoded audio.

When it is necessary to alter subjectively the dynamic range of audio programs, the method built into the audio coding subsystem should be used. The system should provide a transparent pathway, from the audio programme produced in the audio post production studio, into the home. Signal processing devices such as compressors or limiters which alter the audio signal should not be inserted into the audio signal chain. Use of the dynamic range control system embedded within the audio coding system allows the broadcaster or programme provider to appropriately limit the delivered audio dynamic range without actually affecting the audio signal itself. The original audio is delivered intact and is accessible to those listeners who wish to enjoy it.

In the case where a complete audio programme is formed from the combination of a main and an associated service, each of the two services being combined may have a dynamic range control signal. In most cases, the dynamic range control signal contained in a particular bit stream applies to the audio channels coded in that bit stream. There are three exceptions: a single-channel visually impaired (VI) associated service containing only a narrative describing the picture content, a single-channel commentary (C) service containing only the commentary channel, and a voice-over (VO) associated service. In these cases, the dynamic range control signal in the associated service elementary stream is used by the decoder to control the audio level of the main audio service. This allows the provider of the VI, C, or VO service the ability to alter the level of the main audio service in order to make the VI, C, or VO services intelligible. In these cases the main audio service level is controlled by both the control signal in the main service and the control signal in the associated service.

3.9.5 Main, associated, and multilingual services

3.9.5.1 Overview

An AC-3 elementary stream contains the encoded representation of a single audio service. Multiple audio services are provided by multiple elementary streams. Audio service multiplexing is performed by the systems layer, not the audio coding layer. Each elementary stream is conveyed by the transport multiplex with a unique PID. There are a number of audio service types which may (individually) be coded into each elementary stream. Each elementary stream is tagged as to its service type using the bsmod bit field. There are two types of *main service* and six types of *associated service*. Each associated service may be tagged (in the AC-3 audio descriptor in the transport PSI data) as being associated with one or more main audio services. Each AC-3 elementary stream may also be tagged with a language code. The number of audio services (main and associated) is limited only by the capabilities of the MPEG-2 multiplex (a maximum of 8 192 streams). The service capabilities described in the sections below are provided by the DTTB system specified by ATSC Standards A/52 and A/53.

Associated services may contain complete programme mixes, or may contain only a single programme element. Associated services which are complete mixes may be decoded and used as is.

They are identified by the full_svc bit in the AC-3 descriptor (see Recommendation ITU-R BS.1196, Appendix 1 of Annex 2). Associated services which contain only a single programme element are intended to be combined with the programme elements from a main audio service.

This section describes each type of service and gives usage guidelines. In general, a complete audio programme (what is presented to the listener over the set of loudspeakers) may consist of a main audio service, an associated audio service which is a complete mix, or a main audio service combined with one associated audio service. The capability to simultaneously decode one main service and one associated service is required in order to form a complete audio programme in certain service combinations described in this section. Audio decoders do not have to accept input data representing audio services which will not be decoded. This allows the size of the memory buffers required in the receiver to be reduced.

3.9.5.2 Summary of service types

The service types which correspond to each value of bsmod are defined in the digital audio compression (AC-3) Standard (Annex 2 of Recommendation ITU-R BS.1196). The information is reproduced in Table 3.2 and the following paragraphs describe the meaning of these service types.

TABLE 3.2

Table of service types

bsmod	Type of service
000 (0)	Main audio service: complete main (CM)
001 (1)	Main audio service: music and effects (ME)
010 (2)	Associated service: visually impaired (VI)
011 (3)	Associated service: hearing impaired (HI)
100 (4)	Associated service: dialogue (D)
101 (5)	Associated service: commentary (C)
110 (6)	Associated service: emergency (E)
111 (7)	Associated service: voice-over (VO)

3.9.5.3 Multilingual services

Each audio bit stream may be in any language. In order to provide audio services in multiple languages a number of main audio services may be provided, each in a different language. This is the (artistically) preferred method, because it allows unrestricted placement of dialogue along with the dialogue reverberation. The disadvantage of this method is that the full bit rate needed to deliver a multichannel service is required to provide a full 5.1-channel service for each language. One way to reduce the required bit rate is to reduce the number of audio channels provided for languages with a limited audience. For instance, alternate language versions could be provided in 2-channel stereo, or mono, with an appropriate lower bit rate.

Another way to offer service in multiple languages is to provide a main multichannel audio service (ME) which does not contain dialogue. Multiple single-channel dialogue associated services (D) can then be provided, each at an appropriate bit rate. Formation of a complete audio programme requires that the appropriate language D service be simultaneously decoded and mixed into the ME service. This method allows a large number of languages to be efficiently provided, but at the expense of artistic limitations. The single-channel of dialogue would be mixed into the centre reproduction channel, and could not be panned. Also, reverberation would be confined to the centre channel, which is not optimum. Nevertheless, for some types of programming (sports, etc.) this method is very attractive due to the savings in bit rate it offers.

Stereo (two-channel) service without artistic limitation can be provided in multiple languages with added efficiency by transmitting a stereo ME main service along with stereo D services. The D and appropriate language ME services are simply combined in the receiver into a complete stereo program. Dialogue may be panned, and reverberation may be placed included in both channels.

Note that during those times when dialogue is not present, the D services can be momentarily removed, and their data capacity used for other purposes.

3.9.5.4 Detailed description of service types

3.9.5.4.1 CM – complete main audio service

The CM type of main audio service contains a complete audio programme (complete with dialogue, music, and effects). This is the type of audio service normally provided. The CM service may contain from 1 to 5.1 audio channels. The CM service may be further enhanced by means of the VI, HI, C, E, or VO associated services described below. Audio in multiple languages may be provided by supplying multiple CM services, each in a different language.

3.9.5.4.2 ME – main audio service, music and effects

The ME type of main audio service contains the music and effects of an audio program, but not the dialogue for the program. The ME service may contain from 1 to 5.1 audio channels. The primary programme dialogue is missing and (if any exists) is supplied by providing a D associated service. Multiple D services in different languages may be associated with a single ME service.

3.9.5.4.3 VI – visually impaired

The VI associated service typically contains a narrative description of the visual programme content. In this case, the VI service is a single audio channel. Simultaneous reproduction of the VI service and the main audio service allows the visually impaired user to enjoy the main multichannel audio program, as well as to follow the on-screen activity. This allows the VI service to be mixed into one of the main reproduction channels (the choice of channel may be left to the listener) or to be provided as a separate output (which, for instance, might be delivered to the VI user via open-air headphones).

The dynamic range control signal in this type of VI service is intended to be used by the audio decoder to modify the level of the main audio program. Thus the level of the main audio service will be under the control of the VI service provider, and the provider may signal the decoder (by altering the dynamic range control words embedded in the VI audio elementary stream) to reduce the level of the main audio service by up to 24 dB in order to assure that the narrative description is intelligible.

Besides providing the VI service as a single narrative channel, the VI service may be provided as a complete programme mix containing music, effects, dialogue, and the narration. In this case, the service may be coded using any number of channels (up to 5.1), and the dynamic range control signal applies only to this service. The fact that the service is a complete mix is indicated in the AC-3 descriptor.

3.9.5.4.4 HI – hearing impaired

The HI associated service typically contains only a single-channel of dialogue and is intended for use by those whose hearing impairments make it difficult to understand the dialogue in the presence of music and sound effects. The dialogue may be processed for increased intelligibility by the hearing impaired. The hearing impaired listener may wish to listen to a mixture of the single-channel HI dialogue track and the main programme audio. Simultaneous reproduction of the HI service along with the CM service allows the HI listener to adjust the mixture to control the emphasis on dialogue over music and effects. The HI channel would typically be mixed into the centre channel. An alternative would be to deliver the HI signal to a discrete output (which, for instance, might feed a set of open-air headphones worn only by the HI listener.)

Besides providing the HI service as a single narrative channel, the HI service may be provided as a complete programme mix containing music, effects, and dialogue with enhanced intelligibility. In this case, the service may be coded using any number of channels (up to 5.1). The fact that the service is a complete mix is indicated in the AC-3 descriptor.

3.9.5.4.5 D – dialogue

The dialogue associated service is employed when it is desired to most efficiently offer multichannel audio in several languages simultaneously, and the programme material is such that the restrictions (no panning, no multichannel reverberation) of a single dialogue channel may be tolerated. When the D service is used, the main service is of type ME (music and effects). In the case that the D service contains a single-channel, simultaneously decoding the ME service along with the selected D service allows a complete audio programme to be formed by mixing the D-channel into the centre channel of the ME service. Typically, when the main audio service is of type ME, there will be several different language D services available. The transport demultiplexer may be designed to select the appropriate D service to deliver to the audio decoder based on the listener's language preference (which would typically be stored in memory in the receiver). Or, the listener may explicitly instruct the receiver to select a particular language track, overriding the default selection.

If the ME main audio service contains more than two audio channels, the D service will be monophonic (1/0 mode). If the main audio service contains two channels, the D service may contain two channels (2/0 mode). In this case, a complete audio programme is formed by simultaneously decoding the D service and the ME service, mixing the left channel of the ME service with the left channel of the D service, and mixing the right channel of the ME service with the right channel of the D service. The result will be a two-channel stereo signal containing music, effects, and dialogue.

3.9.5.4.6 C – commentary

The commentary associated service is similar to the D service, except that instead of conveying primary programme dialogue, the C service conveys optional programme commentary. When C service(s) are provided, the receiver may notify the listener of their presence. The listener should be able to call up information (probably on-screen) about the various available C services, and optionally request one of them to be selected for decoding along with the main service. The C service may be added to any loudspeaker channel (the listener may be given this control). Typical uses for the C service might be optional added commentary during a sporting event, or different levels (novice, intermediate, advanced) of commentary available to accompany documentary or educational programming.

The C service may be a single audio channel containing only the commentary content. In this case, simultaneous reproduction of a C service and a CM service will allow the listener to hear the added programme commentary. The dynamic range control signal in the single-channel C service is intended to be used by the audio decoder to modify the level of the main audio program. Thus the level of the main audio service will be under the control of the C service provider, and the provider may signal the decoder (by altering the dynamic range control words embedded in the C audio elementary stream) to reduce the level of the main audio service by up to 24 dB in order to assure that the commentary is intelligible.

Besides providing the C service as a single commentary channel, the C service may be provided as a complete programme mix containing music, effects, dialogue, and the commentary. In this case the service may be provided using any number of channels (up to 5.1). The fact that the service is a complete mix shall be indicated in the AC-3 descriptor (see ATSC Doc. A/52, Annex A).

3.9.5.4.7 E – emergency

The E associated service is intended to allow the insertion of emergency announcements. The normal audio services do not necessarily have to be replaced in order for the emergency message to get through. The transport demultiplexer will give first priority to this type of audio service. Whenever an E service is present, it is delivered to the audio decoder by the transport subsystem. When the audio decoder receives an E type associated service, it stops reproducing any main service being received and only reproduces the E service. The E service may also be used for non-emergency applications. It may be used whenever the broadcaster wishes to force all decoders to quit reproducing the main audio programme and substitute a higher priority single-channel.

3.9.5.4.8 VO – voice-over

It is possible to use the E service for announcements, but the use of the E service leads to a complete substitution of the voice-over for the main programme audio. The voice-over associated service is similar to the E service, except that it is intended to be reproduced along with the main service. The systems demultiplexer will give second priority to this type of associated service (second only to an E service). The VO service is intended to be simultaneously decoded and mixed into the centre channel of the main audio service which is being decoded. The dynamic range control signal in the VO service is intended to be used by the audio decoder to modify the level of the main audio program. Thus the level of the main audio service will be under the control of the broadcaster, and the broadcaster may signal the decoder (by altering the dynamic range control words embedded in the VO audio bit stream) to reduce the level of the main audio service by up to 24 dB during the voice-over. The VO service allows typical voice-overs to be added to an already encoded audio bit stream, without requiring the audio to be decoded back to baseband and then re-encoded. However, space must be available within the transport multiplex to make room for the insertion of the VO service.

3.9.6 Conclusion

The AC-3 coding system provides both for the basic audio coding, as well as providing a number of features which are useful in a DTTB system. Associated services, delivered using the flexibility of the MPEG-2 multiplex system, provide nearly unlimited flexibility to provide a large number of related audio channels.

3.10 Ancillary Data

DTTB affords the opportunity to augment the basic video and audio service with ancillary digital data services. The flexibility of the MPEG-2 system allows new services to be easily introduced at any time in a completely backward compatible manner. Basic services include Program Subtitles, Emergency Messages, Program Guide information and Teletext.

3.10.1 Teletext

One signal source which can be regarded as data is Teletext which conforms to one of the existing systems described in Recommendation ITU-R BT.653. Systems A, B, C, and D contained in this specification must be capable of operating in a 50 Hz and 60 Hz environment. Because the Teletext signal is digital already it is only required to be packetized by adding a header and additional data. Fig. 3.19 shows an example how a teletext system for DTTB could be arranged.

3.10.2 Programme subtitles

In any television service programme subtitles are an essential feature. There are a number of alternatives for carrying closed caption information. Possibilities include:

– as user data in MPEG-2 video (analogous to the existing Line 21 closed captioning system)

– as private streams in MPEG systems such as using an existing teletext system

– as a registered stream in MPEG systems using the registration descriptor.

3.10.3 Broadcast multimedia services

A DTTB service has the capability of providing multimedia services such as related information services for current TV programmes, navigation services to provide easy programme selection and latest news services with multimedia and hypermedia presentation style. The development of multimedia services in the field of computers and telecommunications has been remarkable. The coding system for the multimedia services are standardised for instance MHEG or Hyper ODA. The interoperability of the multimedia coding system with the standards is necessary to realize common receivers or LSIs. Users can view these multimedia services interactively with TV sets or home computers.

3.11 The MPEG-2 multiplexing structure

Compressed and encoded video and audio signals and ancillary data form compressed elementary streams. This streams are then organised in serialised packets for further storage and transmission.

MPEG-2 specifies three types of streams:

Packetized Elementary Stream (PES)

This is a basic packetized stream for Video, Audio, Data or any other type of stream. A PES packed contains coded bytes from one and only one elementary stream. The PES stream is a logical construction that may be useful within implementations of Program Stream or Transport Stream, however, it is not defined as a stream for interchange and interoperability.

Program Stream (PS)

This is a combination of a number of Packetized Elementary Streams which have a common time base and is used in error free environments. Program stream packets may be of variable and relatively great length. The Program Stream is suitable for applications which may involve software processing.

Transport Stream (TS)

This combines one or more Packetized Elementary Streams with one or more independent time bases into a single stream and is used where the transmission media is prone to errors. Elementary streams sharing a common time base form a program. Transport stream packets are 188 bytes long. The Transport Stream is designed for storage or transmission in lossy or noisy media.

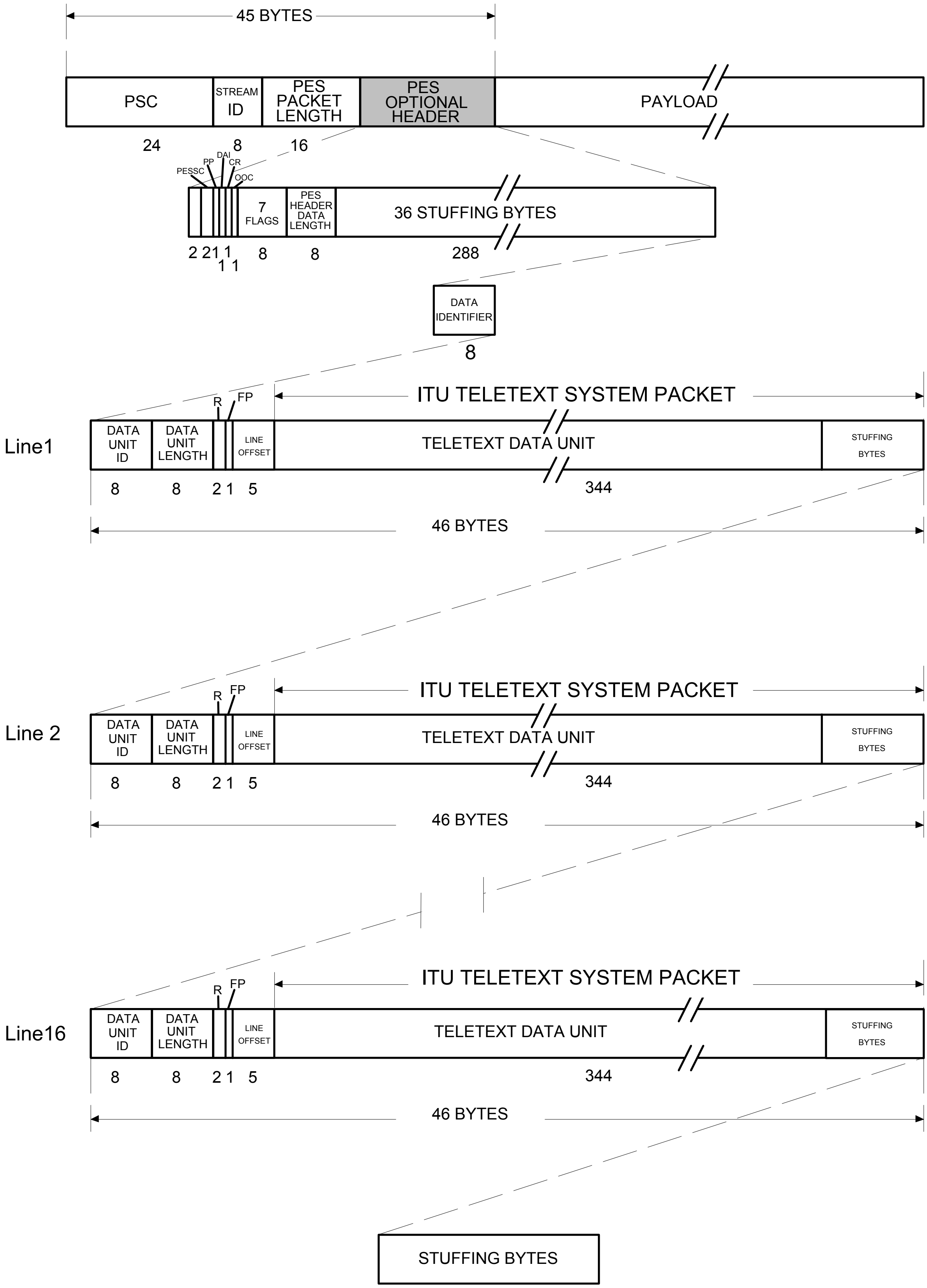
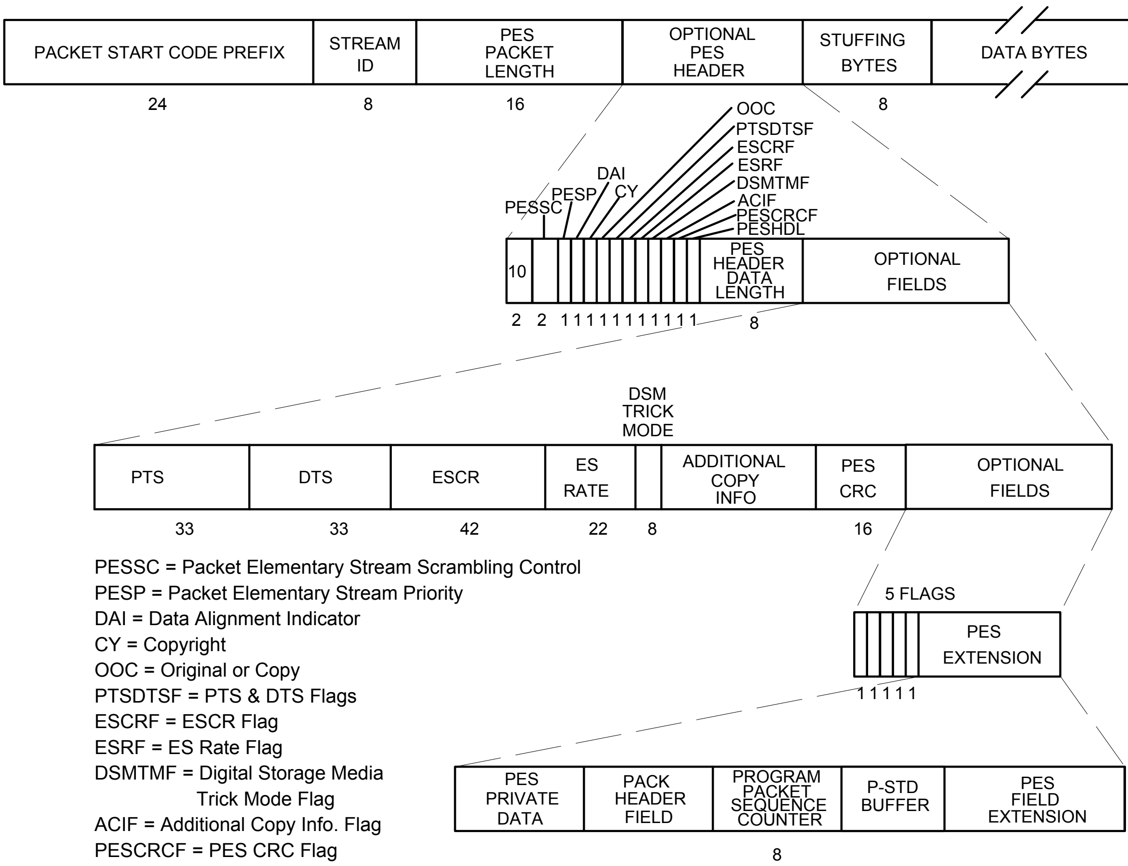


FIGURE 3.19

A teletext packetized elementary stream

3.11.1 Packetized elementary stream

Fig. 3.20 shows the structure of an elementary stream packet.



PESSC = Packet Elementary Stream Scrambling Control
PESP = Packet Elementary Stream Priority
DAI = Data Alignment Indicator
CY = Copyright
OOC = Original or Copy
PTSDTSF = PTS & DTS Flags
ESCRF = ESCR Flag
ESRF = ES Rate Flag
DSMTMF = Digital Storage Media Trick Mode Flag
ACIF = Additional Copy Info. Flag
PESCRCF = PES CRC Flag
PESHDL = PES Header Data Length
PTS = Presentation Time Stamp
DTS = Decoding Time Stamp
ESCR = Elementary Stream Clock Reference
CRC = Cyclic Redundancy Check
P-STD = Program Stream System Target Decoder

FIGURE 3.20

Packet structure

Start code prefix

This has a fixed value of $00 $00 $01 as described above.

Stream ID (identification)

Each type of stream has a particular value:

$BF	Private 2
$C0 – $DF	Audio Stream Number.
$E0 – $EF	Video Stream Number.
$F0 – $FF	Data Stream Number.

Packet length

This gives the length of the packet – the maximum size can be 65 536 bits.

Buffer size

This field can contain the size of the buffer required in the decoder.

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CHAPTER 4

SERVICE MULTIPLEX AND TRANSPORT

4.1 Available structures

4.1.1 ATM

ATM technology was developed to solve the multiplex and transport problems found in the communications world of telephony. As specified in ITU-T Recommendation I.361 this is a structure in which any type of information is encapsulated in cells of 53 bytes. The first 5 bytes (the header) contain the multiplexing information and the last 48 bytes (the payload) contain the user information. In order to ensure end-to-end information and time transparencies, an ATM adaptation layer (AAL) has been defined on top of the ATM layer. Different types of AAL can be specified to cover the full range of services to be supported.

This approach could also be adapted for digital terrestrial television broadcasting (DTTB). A similar packet structure can be adopted, the functionalities and structure of the cell header being optimized for the DTTB environment. The addition of some kind of recurrent frame structure to improve the multiplex performance in poor error conditions may also be needed.

4.1.2 MPEG-2

The Moving Picture Experts Group (MPEG) of ISO/IEC has produced a multiplex structure that could also be used for DTTB. In North America, the transport mechanism for the Advanced Television System Committee Standard is a subset of the MPEG-2 System Transport Stream Syntax.

In Europe, digital multi-programme TV systems have been developed by the DVB project, for Satellite, CATV and SMATV (Satellite Master Antenna Television systems) applications. These systems make use of MPEG-2 video and audio coding methods, as well as of Transport Stream Multiplexing. To achieve the maximum commonality over the various media, MPEG source coding and transport multiplexing methods will be adopted by the DVB project also for the DTTB system under development.

The MPEG-2 system packet multiplex structures were specifically tailored to the needs of broadcast video, audio and data signals with consideration of compatibility with ATM structures included. The MPEG-2 systems packet structure consists of 188 bytes comprised of 4 header bytes and 184 payload bytes. This packet size was designed to be encapsulated within four ATM cells as four 47-byte payloads ($4 \times 47 = 188$) leaving space for 1 ATM AAL byte per ATM cell. The MPEG-2 system can transport data with a lower overhead than the ATM system. Overhead is an important consideration in the highly constrained DTTB environment.

4.1.3 ISDB

The Japanese Administration proposed application of ISDB to digital broadcasting, noting the following inherent characteristics:

– flexibility,
– extensibility,
– interoperability,
– good transmission characteristics,
– easy programme reception,
– conditional access capability, and
– other features, such as low operational cost for broadcasters and simple and low-cost receivers.

Satellite, terrestrial and cable digital broadcasting systems are being developed aiming at the maximally common specification based on the concept of ISDB.

4.2 Multiplexing of video, audio, and data

4.2.1 Introduction

The various component video, audio, and data element bit-streams need to be multiplexed together to form the complex signal ensemble in a DTTB system. In addition, various error protection strategies can be employed at this stage of processing to enhance the ruggedness of the multiplexed data. Randomization of the digital data and data interleaving are possible techniques that can be used so that bursts of channel bit errors can be treated as uncorrelated bit errors by the forward error correction codecs. In addition, synchronization bit sequences can be inserted at intervals to mark these boundaries and provide appropriate video, audio, data/text and control data streams to the processor.

One approach is to arrange transmission bytes in a structure analogous to the line and field structure of existing analogue TV signals. This may also result in a signal structure containing recurring sequences that can be used for synchronization and also as “training signals” for ghost-canceller or channel-equalizer systems.

An alternative approach could be the use of a cell-relay-based data-transport layer that supports the prioritized delivery of video data, thus providing graceful degradation of services under impaired channel conditions. The cell relay might also provide logical synchronization that is essential for reliable delivery of variable-length-coded compressed video in the presence of transmission errors. This data transport protocol also offers service flexibility for a wide mixture of video, audio, and auxiliary data services. The transport processor asynchronously multiplexes the payload data with different priorities into basic transport units called cells. A cell resembles a data packet in conventional packet networks in modern data communication. It has a header and a trailer enclosing a payload area. Each cell has a fixed size and its own error control bits. It is noteworthy that the cell format can be transcodable to B-ISDN (broadband integrated services digital network), thus providing a path for development of future information services.

Use of “headers and descriptors” within the data stream is seen by some as a useful approach that facilitates data processing.

4.2.2 Program versus Transport Stream multiplexing

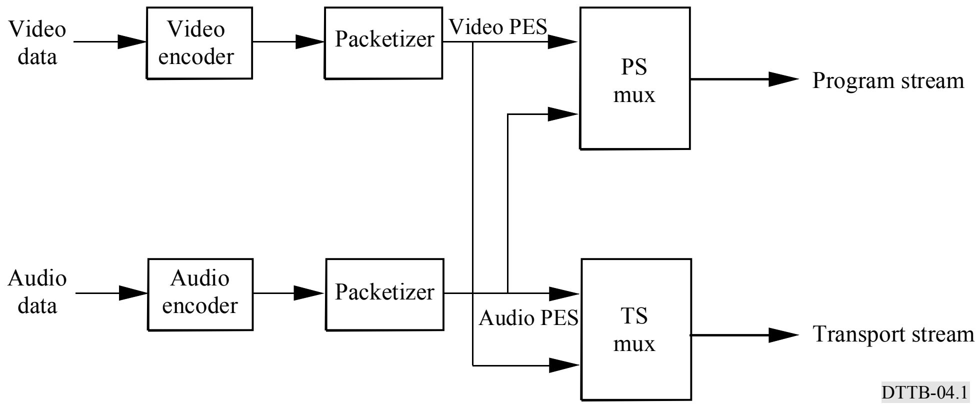


FIGURE 4.1

System level multiplexing approaches

In general there are two approaches to multiplexing elementary bit streams from multiple applications on to a single channel. One approach is based on the use of fixed length packets and the other on variable length packetization. As illustrated in Fig. 4.1, the video and audio elementary bit streams in both cases are generally first formed into variable length **PES (packet elementary streams)** packets (although it should be noted that some applications produce fixed length PES packets). The process of generating the multiplexed bit streams for the two approaches involves a difference in processing only at the final multiplexing stage.

Figure 4.2 provides examples of bit streams for both the program and transport stream approaches to demonstrate their differences. As shown in Fig. 4.2, in the **program stream** approach, PES packets from various elementary bit streams are multiplexed by transmitting the bits for the complete PES packets in sequence, resulting in a sequence of **variable length** packets on the channel.

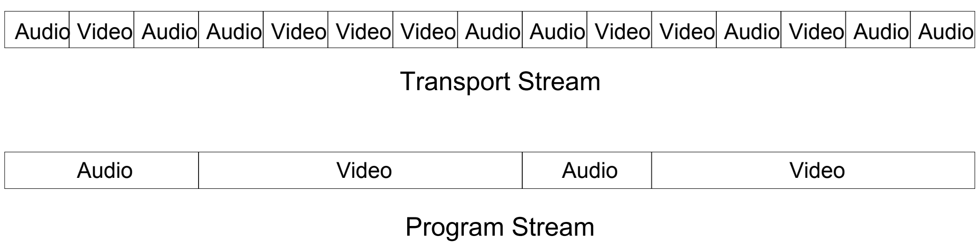


FIGURE 4.2

Packetization approaches

In contrast, the **transport stream** approach, as shown in Fig. 4.2, the PES packets (including the PES headers) are transmitted as a payload of **fixed length** transport packets. Each transport packet is preceded by a transport header which includes information for bit stream identification. Each PES packet for a particular elementary bit stream occupies a variable number of transport packets, and data from various elementary bit streams are generally interleaved with each other at the transport packet layer. Identification of each elementary bit stream is facilitated by the data in the transport headers. New PES packets always start a new transport packet and stuffing bytes are used to fill packets with partial PES data.

The two multiplexing schemes are motivated by different application requirements. Transport streams are appropriate for environments where errors and data loss events are likely, including certain storage media and transmission on noisy channels. Program streams are appropriate for relatively error-free media such as CD-ROMs. Errors or loss of data within PES packets can potentially result in complete loss of synchronization in the decoding process. The program stream approach is used when requirement for compatibility with MPEG-1 is stipulated.

It should be noted, however, that in general, the program and transport data streams both address the same general layers of protocol functionality, and therefore it does not make sense to carry a program bit stream within a transport bit stream or vice-versa. Transcoding between the two formats is feasible and one could build an interface between them.

4.2.3 Advantages of the fixed length packetization approach

The fixed length packetization approach offers flexibility and some additional advantages when attempting to multiplex data related to several applications into a single bit steam.

While digital systems are generally described as flexible, the use of fixed length packets offers a high level of flexibility to allocate channel capacity among video, audio and auxiliary data services. The use of a packet identification field (**PID**) in the packet header as a means of bit stream identification makes it possible to have a mix of video, audio and ancillary data which is flexible and which need not be specified in advance. The entire channel capacity can be reallocated to meet immediate service needs including allocation of the entire bit stream for delivery of data services. This concept is termed **dynamic capacity allocation**.

The ability to dynamically allocate system capacity may be exploited to allow additional elementary bit streams to be added at the input of the multiplexer or to allow these elementary bit streams to be multiplexed at a second stage with the original bit stream. The presence of multiple elementary bit streams in the data channel allows the system to be **scalable**.

The DTTB system was developed with the understanding that there would be future services that could not be anticipated at the introduction of the service. It was, therefore, extremely important that the transport architecture be open-ended. New elementary bit streams could be handled at the transport layer without hardware modifications, by assigning new PIDs at the transmitter and filtering out these new PIDs in the bit stream at the receiver. Backward compatibility was assured when new bit streams were introduced into the transport system since existing decoders would automatically ignore new PIDs. This capability could possibly be used to compatibly introduce newer, higher temporal or spatial resolution services or stereoscopic services by sending augmentation data along with the normal television service data. The presence of multiple elementary bit streams in the data channel and provision for identification of, yet unidentified, future services allows the system to be **extensible.**

Another fundamental advantage of the fixed length packetization approach is that the fixed length packet can form the basis for handling errors that occur during transmission. Error correction and detection processing (which precedes packet demultiplexing in the receiver subsystem) may be synchronized to the packet structure so that one deals in the decoder at the packet level when handling data loss due to transmission impairments. Essentially, after detecting errors during transmission, one recovers the data bit stream from the first good packet. Recovery of synchronization within each application is also aided by the transport packet header information. Without this approach, recovery of synchronization in the bit streams would be completely dependent on the properties of each elementary bit steam. The presence of fixed length packets improves the system's **robustness.**

A fixed-length packet based transport system enables simple decoder bit stream demultiplex architectures suitable for high speed implementations. The decoder does not need detailed knowledge of the multiplexing strategy or the source bit-rate characteristics to extract individual elementary bit streams at the demultiplexer. All the receiver requires is the identity of the packets. The necessary information is transmitted in each packet header at fixed and known locations in the bit stream. The only important timing information is related to bit level and packet level synchronization.

4.2.4 Overview of the transport subsystem

The transport resides between the application (e.g. audio, video, or data) encoding/decoding function and the transmission subsystems. At its lowest layer, the encoder transport subsystem is responsible for formatting the encoded bits and multiplexing the different components of the program for transmission. At the receiver, the decoder transport subsystem is responsible for recovering the bit streams for the individual application decoders and for the corresponding error signalling. (At a higher layer, multiplexing and demultiplexing of multiple programs within a single bit stream can be achieved with an additional system level multiplexing or demultiplexing stage before or after the modem in the transmitter or the receiver.) The transport subsystem also incorporates other higher level functionality related to identification of applications and synchronization of the receiver.

A data transport mechanism based on the use fixed length packets that are identified by headers allows for the identification of a particular application bit-stream (also called an **elementary bit stream**) which forms the payload of the packets. Applications that can be supported include video, audio, data, program and system control information. The elementary bit streams for video and audio can themselves be wrapped in the PES variable length packet structure before transport processing. The PES layer provides functionality for identification, synchronization of decoding, and presentation of the individual application.

Elementary bit steams sharing a common time base can then be multiplexed, along with a control data stream, into **programs**. These programs and an overall system control data stream are then asynchronously multiplexed to form the system bit stream. Programs in this system are analogous to today's conventional broadcast channels. Using this approach, the transport is made flexible in two ways:

1. It permits defining programs as any combination of elementary bit streams. For example the same elementary bit stream could be present in more than one program (e.g. two bit streams with the same audio), a program could be formed by combining a basic elementary bit stream and a supplementary elementary bit stream (i.e. bit streams for scalable decoders), programs could be tailored for specific needs (e.g. regional selection of language) etc.

2. Flexibility at the systems layer allows different programs to be multiplexed into the system as desired, and allows the system to be reconfigured easily when required. The procedure for extraction of programs from within a system is also simple and well defined.

This approach provides other features that are useful for both normal decoder operation and for the special features required in broadcast and cable applications. These include:

– decoder synchronization,

– conditional access,

– local program insertion,

amongst others.

This approach to a bit stream configuration directly addresses issues related to the storage and playback of programs. Although this is not directly related to the DTTB transmission problem, the ability to create programs in advance, store them as a compressed multiplexed bit stream and play them back at the desired time is a desirable feature. The most efficient means of storing programs is in the same format in which they are transmitted, as transport bit streams. The preferred implementation should provide "hooks" to support consumer digital products based on recording and playback of these bit streams, including the use of "trick modes" that are available on current analogue VCRs. It should be noted that the issues related to storage and play back of digitally compressed video bit streams are quite different from those that need to be considered for conventional analogue television systems.

It is desirable that the DTTB transport bit stream be easily carried on other communication systems, and that the DTTB transport bit stream be capable of carrying bit streams generated by other communication systems.

4.3 Higher level multiplexing functionality

The overall multiplexing approach can be described as a combination of multiplexing at two different layers. In the first layer one forms program transport streams by multiplexing one or more elementary bit streams at the transport layer, and in the second layer the program transport streams are combined (using asynchronous packet multiplexing) to form the overall system. The functional layer in the system that contains both this program and system level information is called **PSI** or **Program Specific Information.** This example represents one way of constructing the system, but it is not the only way, and for some architectures, it may not represent the preferred way.

4.3.1 Single program transport multiplex

A program transport bit stream can be formed by multiplexing individual transport packetized elementary bit streams (with or without PES packetization) sharing a common time-base, and a control bit stream that describes the program. Each elementary bit stream, and the control bit stream (also called the elementary stream map) are identified by their unique PIDs in the link header field. The organization of this multiplex function is illustrated in Fig. 4.3. The control bit stream contains the program_map_table that describes the elementary stream map. The program_map_table includes information about the PIDs of the transport streams that make up the program, the identification of the applications that are being transmitted on these bit streams, the relationship between these bit streams, etc.

The transport syntax allows a program to be comprised of a large number of elementary bit streams, with no restriction on the types of applications required within a program. A program transport stream does not need to contain compressed video or audio bit streams. It could contain multiple audio bit streams for a given video bit stream. The data applications that can be carried are flexible, the only constraint being that there should be an appropriate stream_type ID assignment for recognition by a compatible decoder of the application corresponding to the bit stream.

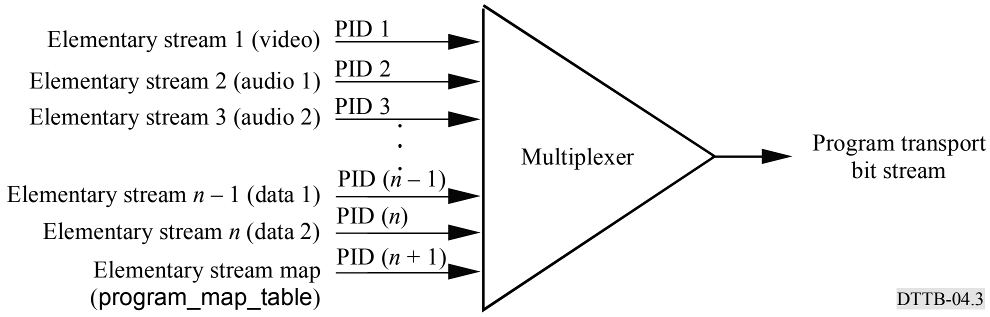


FIGURE 4.3

Programme multiplex stream multiplex

Note that many of the link level functions are carried out independently, without program level coordination for the different elementary bit streams that make up a program. This includes functions such as PID manipulation, bit stream filtering, scrambling and descrambling, definition of random entry packets, etc. The coordination between the elements of a program is primarily controlled at the presentation (display) stage based on the use of the common time base. This common time base is set up by the fact that all elementary bit streams in a program derive timing information from a single clock, the information for which is transmitted via the PCR (program clock reference) on one of the elementary bit streams that constitute the program. The data for timing of presentation is present in the elementary bit streams for individual applications.

4.3.2 System multiplex

The system multiplex allows multiplexing of different program transport streams. In addition to the transport bit streams (with their corresponding PIDs) that define the individual programs, a system level control bit stream with PID = 0 is defined. This bit stream carries the program_association_table that maps program identities to their program transport streams. The program identity is represented by a number in the program_association_table. A program corresponds to what is traditionally called a channel in conventional television systems. The map indicates the PID of the bit stream containing the program_map_table for a program. Thus, the process of identifying a program and its contents takes place in two stages: first one uses the program_association_table in the PID = 0 bit stream carrying the program_map_table for the program, and in the next stage one obtains the PIDs of the elementary bit streams that make up the program from the appropriate program_map_table. Once this step is completed, the filters at a demultiplexer can be set to receive the transport bit streams that correspond to the program of interest.

The system layer of multiplexing is shown in Fig. 4.4. During the process of system level multiplexing, there is the possibility of receiving identical PIDs on different program streams. This poses a problem since PIDs for different bit streams need to be unique. One solution would be to modify the PIDs just before the multiplexing stage. The changes need to be recorded in both the program_association_table and the program_map_table. Hardware implementation of the PID reassignment function in real time is helped by the fact that this process is synchronous at the packet clock rate.

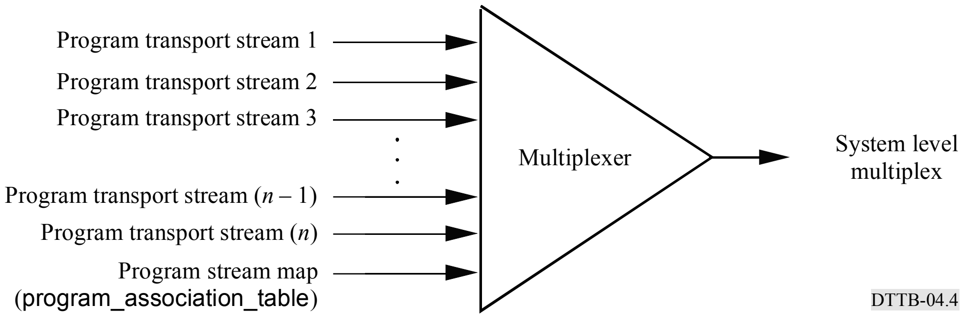


FIGURE 4.4

System level bit stream multiplex

This process can be made scalable by multiplexing multiple system level bit streams on a higher bandwidth channel by extracting the program_association_tables from each system multiplexed bit stream and reconstructing a new PID = 0 bit stream.

Figure 4.5 illustrates one possible implementation approach to extracting elementary bit streams for a program at a receiver, although not necessarily the most efficient approach. In practice the same demultiplexer hardware could be used to extract both the program_association_table and the program_map_table control bit streams. This also represents the minimum functionality required at the transport layer to extract any application bit stream (including those that may be private). Figure 4.6 shows an example of data access flow in the receiver in a different fashion.

It is important to clarify here that the layered approach to defining the multiplexing function does not necessarily imply that program and system multiplexing should always be implemented in separate stages. A hardware implementation that includes both the program and system level multiplexing within a single multiplexer stage can be considered as long as the multiplexed output bit stream has the desired properties.

4.4 The PES packet format

As stated earlier in the text, prior to entering the transport layer, some elementary bit streams will go through PES layer packetization. The PES header carries various rate, timing, and descriptive information as set by the encoder. The PES packet length is described in a field provided for that purpose. The PES packetization interval is application dependent resulting in packets of variable length with a maximum definable size of 2^{16} bytes. If the PES packet length is set to zero, the PES packet can be of any length. A value of zero for the PES packet length can be used only when the PES packet payload is a video elementary stream.

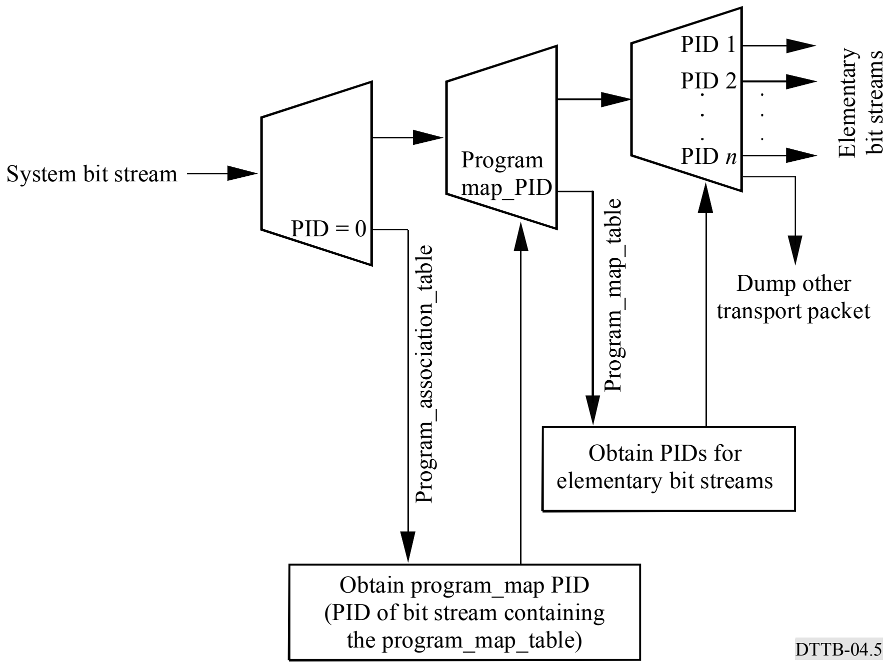


FIGURE 4.5

Transport demultiplexing process

It is useful to have PES packets start on group of pictures (GOP) boundaries when dealing with compressed video and associated audio. The example described in this section represents a subset of the general MPEG-2 description which allows simplification of the receiver. In this example, all data for a PES packet, including the header, are transmitted contiguously as the payload of transport packets. A new PES packet always starts a new transport packet, and PES packets that end in the middle of a transport packet are followed by stuffing bytes for the remaining length of the transport packet.

A PES packet consists of a PES_packet_start_code, PES header flags, PES packet header fields, and a data block payload as shown in Fig. 4.7. The packet payload is a stream of contiguous bytes of a single elementary stream, and for video or audio packets, the payload is a sequence of access units provided by the encoder corresponding to the video pictures and the audio frames.

Each elementary stream is identified by a unique stream_id which is carried by the PES packet. PES packets carrying various types of elementary streams can be multiplexed to form a program or transport stream. The stream_id can take on a number of values indicating the type of data in the payload as shown in Table 4.1.

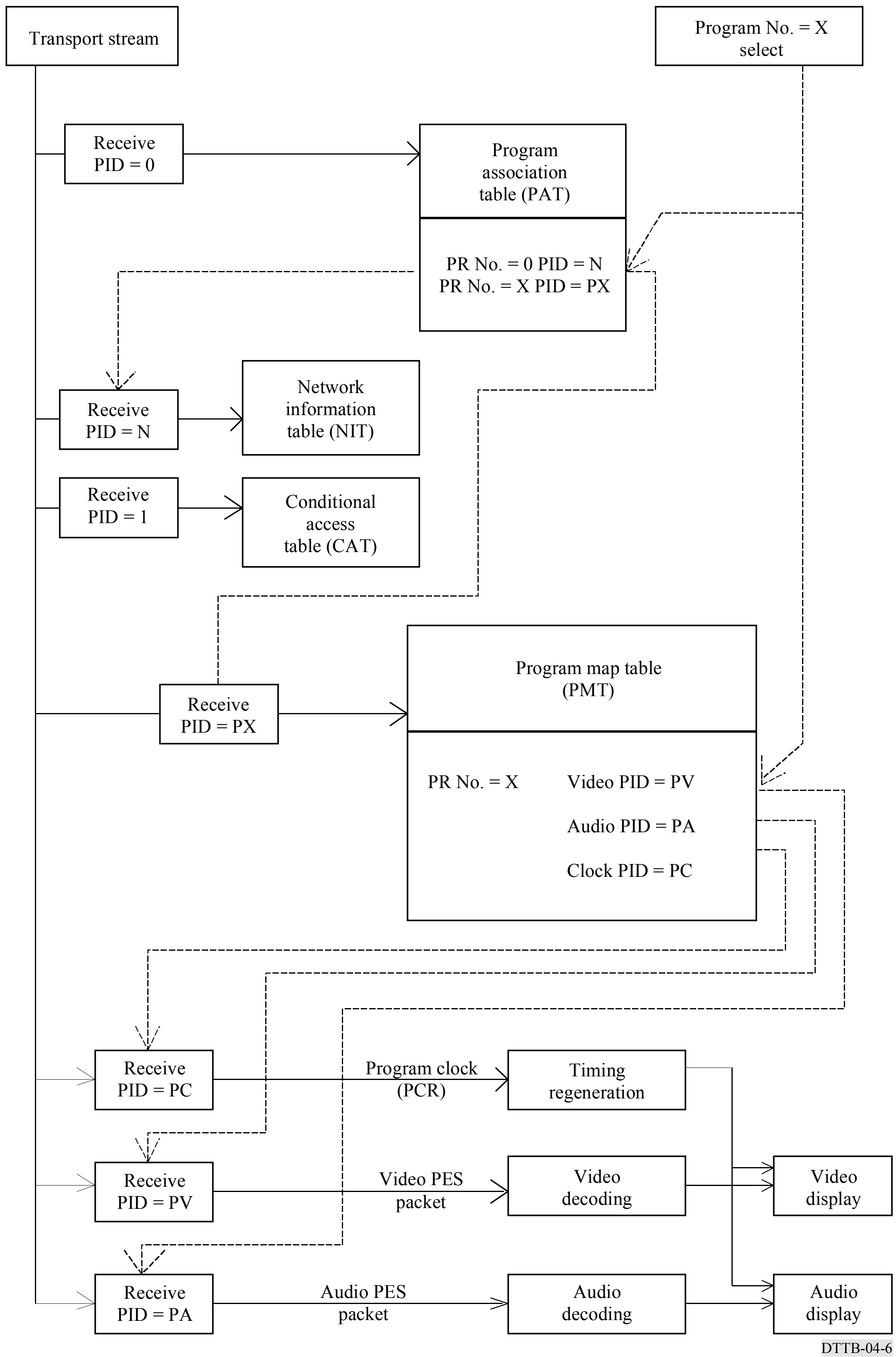


FIGURE 4.6

An example of data access flow in the receiver

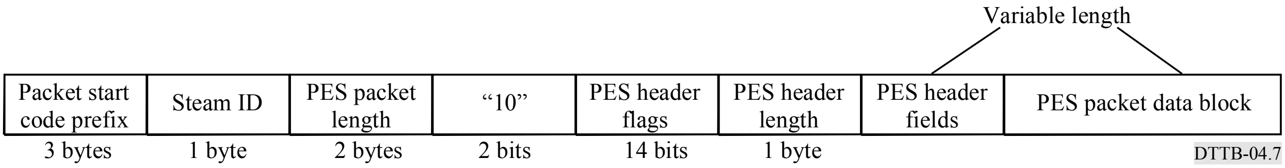


FIGURE 4.7

PES packet structure

TABLE 4.1

PES packet overview

Field	Function/usage
packet_start_code_prefix	Indicates the start of a new packet. Together with the stream_id forms the packet start code. Takes the value 0x00 0001.
stream_id	Specifies the type and number of the stream to which the packet belongs: 1011 1100 – Reserved stream. 1011 1101 – Private Stream 1. 1011 1110 – Padding Stream. 1011 1111 – Private Stream 2. 110x xxxx – MPEG Audio Stream Number xxxxx. 1110 xxxx – MPEG Video Stream Number xxxx. 1111 0000 ECM stream 1111 0001.EMM stream 1111 0010 DSM CC stream 1111 0011 MHEG stream 1111 0100 – 1111 1000 ITU-T Rec. H.222.1 type A – type E 1111 1001 ancillary stream 1111 1010 – 1111 1110 reserved data stream 1111 1111 program stream directory
PES_packet_length	Specifies the number of bytes remaining in the packet after this field. 0x 0000 – this value is only allowed value for video. Audio details to be determined.

The PES header flags for a constrained example of the MPEG-2 system are shown in Fig. 4.8 and described in Table 4.2, and provide indicators of the properties of the bit stream and the existence of additional flags in the PES header.

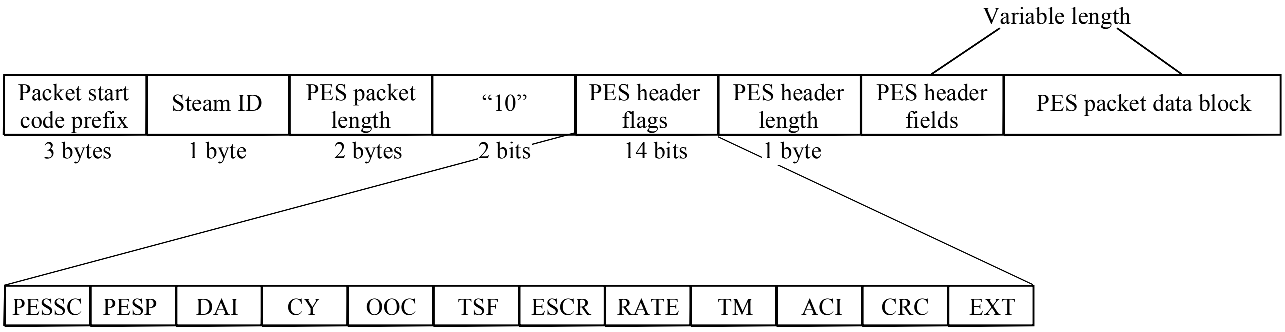


FIGURE 4.8

PES header flags

DTTB-04.8

TABLE 4.2

PES header flags

Flag	Function/usage
PESSC (PES_scrambling_control)	Indicates the scrambling of the PES packet received: 00 – Not scrambled. 01 – User defined. 10 – User defined. 11 – User defined. (In this example, set = 00.)
PESP (PES_priority)	Indicates the priority of this packet with respect to other packets: 1 = high priority; 0 = no priority.
DAI (data_alignment_indicator)	Indicates the nature of alignment of the first start code occurring in the payload. The type of data in the payload is indicated by the data_stream_alignment_descriptor. 1 – Aligned; 0 – No indication of alignment. (Must be aligned for video.)

TABLE 4.2 (*end*)

Flag	Function/usage
CY (copyright)	Indicates the copyright nature of the associated PES packet payload: 1 – Copyrighted. 0 – Not defined.
OOC (original_or_copy)	Indicates whether the associated PES packet payload is the original program or a copy: 1 – Original; 0 – Copy.
TSF (PTS_DTS_flags)	Indicates whether the PTS or PTS and DTS are in the PES header: 00 – Neither PTS or DTS are present in the header. 1x – PTS field is present. 11 – Both PTS or DTS are present in the header. (The PTS flag is set when video data alignment indicator is set. The DTS may be included to signal the decoder of any special requirements. PTS transmissions should be spaced less than 700 ms apart.)
ESCR (ESCR_flag)	Indicates whether the Elementary Stream Clock Reference field is present in the PES header. (In this example, set = 0.)
RATE (ES_rate_flag)	Indicates whether the Elementary Stream Rate field is present in the PES header. (In this example, set = 0.)
TM (DSM_trick_mode_flag)	Indicates the presence of an 8 bit field describing the DSM (Digital Storage Media) operating mode: 1 – Field is present. 0 – Field is not present. (For broadcasting purposes, set = 0.)
ACI (additional_copy_info_flag)	Indicates the presence of the additional_copy_info field. 1 – Field is present. 0 – Field is not present.
CRC (PES_CRC_flag)	Indicates the presence of a CRC field in the PES packet. (In this example, set = 0.)
EXT (PES_extension_flag)	The flag is set as necessary to indicate that extension flags are set in the PES header. Its use includes support of private data. 1 – Field is present. 0 – Field is not present.

The PES header follows the PES_header_length field which indicates the header size in bytes. The size of the header includes all the header fields, any extension fields, and stuffing_bytes. The organization of the PES header is described by the PES header flags and all of the fields of the PES header are optional. Certain applications require particular fields to be set. For example, the DTTB transport of video PES packets requires that the data_alignment_indicator be set. The trick mode flag is generally not set. For digital storage media, retrieval of video requires the opposite conditions to be true. The encoder associated with each application must set the appropriate flags and encode the appropriate fields.

The PES header is shown in Fig. 4.9 and described in Table 4.3.

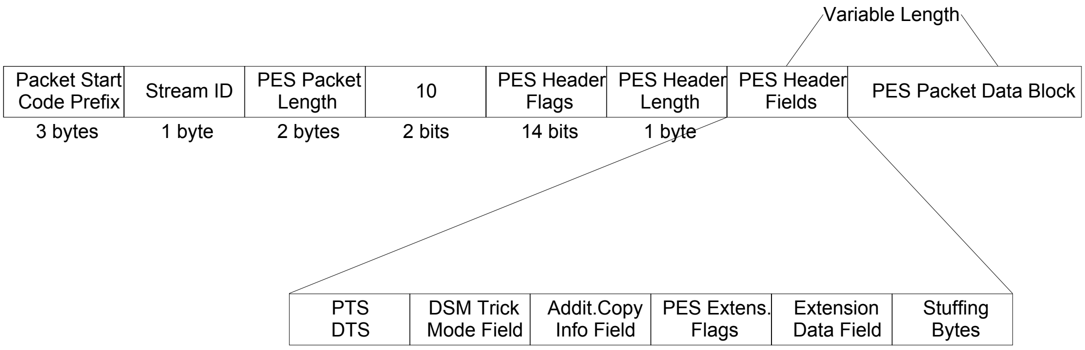


FIGURE 4.9

PES header organization

TABLE 4.3

PES header

field	Function
PTS (presentation_time_stamp) DTS (decoding_time_stamp)	PTS informs the decoder of the intended time of presentation of a presentation unit. DTS informs the decoder of the intended time of decoding of an access unit. An access unit is an encoded presentation unit. When encoded, the PTS refers to the presentation unit corresponding to the first access unit occurring in the packet. If an access unit does not occur in a packet, it shall not contain a PTS. Under normal conditions, the DTS may be derived from the PTS and need not be encoded. A video access unit occurs if the first byte of the picture start code is present in the PES packet payload. An audio access unit occurs if the first byte of the audio frame is present.
DSM_trick_mode	An eight bit field indicating the nature of the information encoded. The field is further partitioned as follows: trick_mode_control (3 bits), field_id (2 bits), intra_slice_refresh (1 bit), and frequency_truncation (2 bits).

TABLE 4.3 (*end*)

Field	Function
trick_mode_control	Indicates the nature of the DSM Mode: 000 – Fast Forward 001 – Slow Motion 010 – Freeze Frame 011 – Fast reverse 1xx – Reserved.
field_id	This identifier is valid for interlaced pictures only and describes how the current frame is to be displayed: 00 – Display field 1 only. 01 – Display field 2 only. 10 – Display complete frame. 11 – Reserved.
frequency_truncation	This field indicates the selection of coefficients from the DSM: 00 – Only DC coefficients are sent. 01 – The first three coefficients in scan order on average. 10 – The first six coefficients in scan order on average. The field is for information purposes only. At times, more than the specified number of coefficients may be sent. At other times, less than the specified number of coefficients may be sent.
intra_slice_refresh	This field indicates that each picture is composed of intra slices with possible gaps between them. The decoder should replace the missing slices by repeating the collocated sites from the previously decoded picture.
field_rep_control	This field indicates how many times the decoder should repeat field #1 as both the top and bottom fields alternatively. After field #1 is displayed, field #2 is displayed the same number of times. This identifier set to "0" is equivalent to a freeze frame with field_id set = "10".

The PES header can contain additional flags if the EXT flag is set. These flags are transmitted in a one byte data field as shown in Fig. 4.10 and are described in Table 4.4. The flags indicate whether further extensions to the PES header exist. In each case the flag is set to "1" if the header field is present.

PES private data flag	pack header field flag	program packet seq.counter flag	STD Buffer Flag	Reserved	PES extension field flag

FIGURE 4.10

PES extension flags field

TABLE 4.4

PES extension flags

Field	Function	Usage
PES_private_data_flag	Indicates whether PES packet contains private data.	As defined.
program_private_sequence_counter_flag	Indicates whether an MPEG-1 systems packet header or an MPEG-2 program stream packet header is present.	As defined.
STD_buffer_flag	Indicates whether the STD_buffer_scale and the STD_buffer_size flags are encoded.	Set = 0 for this example.
PES_extension_field_flag	Indicates the presence of additional data in the PES header.	As defined.

4.5 The packetization approach and functionality

4.5.1 Overview

A DTTB communication system transport bit stream can consist of either fixed length packets or variable length packets. The packetization approach described in this section is based on fixed length packets with a fixed and a variable component to the header field as illustrated in Fig. 4.11.

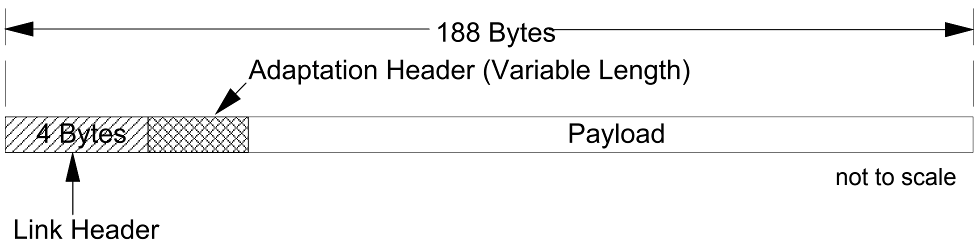


FIGURE 4.11

Transport packet

In this approach, based on MPEG-2 syntax, each packet consists of 188 bytes. The choice of this packet size is motivated by a few factors. The packets need to be large enough so that the overhead due to the transport headers does not become a significant portion of the total data carried. They should also not be so large that the probability of packet error becomes significant under standard operating conditions (due to inefficient error correction). It is also desirable to have packet lengths appropriate to the block sizes of typical, block oriented error correction approaches, so that packets may be synchronized to error correction blocks, and the physical layers of the system can aid the packet level synchronization process in the decoder. Another motive for the particular packet length selection is interoperability with the ATM format. The general philosophy of this approach is to transmit a single DTTB transport packet in four ATM cells.

The contents of each packet and the nature of the data it is carrying are identified by the **packet headers.** The packet header structure is layered and may be described as a combination of a fixed length **link layer** and a variable length **adaptation layer**. Each layer serves a different functionality similar to the link and transport layer functions in the OSI layers of a communications system. This link and adaptation level functionality is directly used for the terrestrial link on which the DTTB bit stream is transmitted. However, these headers could also be completely ignored in a different system (e.g. ATM), in which the DTTB bit stream is just the payload to be carried. In this environment, the DTTB bit stream headers would serve more as an identifier for the contents of a data stream rather than as a means for implementing a protocol layer in the overall transmission system.

The syntax elements of a possible system transport layer bit stream are defined for the purpose of exploring the requirements of such a system. It is understood that while most syntax elements are expected to trigger a response in the transport decoder, all syntax elements need to be recognized at some level of the receiver.

4.5.2 The "link" layer

The link layer is implemented using a four byte header field. Fig. 4.12 shows a possible link layer header with functionality assigned to each bit. Table 4.5 provides a description of each function. The general functions may not all necessarily apply to a DTTB channel, but are useful for providing interoperability (transmitting the same bit stream over other links, including cable links, computer networks, satellite distribution systems, etc.).

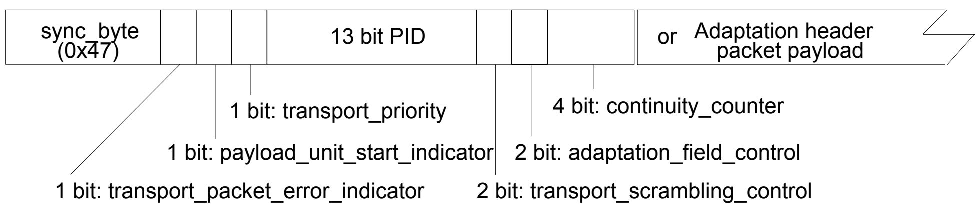


FIGURE 4.12

Link header format

TABLE 4.5

Link header format

Field	Function/usage
sync_byte (Value: 0x47)	Packet synchronization
transport_packet_error_indicator	Indicates if packet is erroneous; 0→no error 1→erroneous packet (Can be used for error signalling from modem to transport demultiplexer. A "1" implies the payload is not to be used.)
payload_unit_start_indicator	Indicates if a PES packet header of the start of a table containing program specific information (PSI) is present in the payload of the packet. The PES packet header always begins the payload of the packet. The starting byte of the PSI table in the packet is indicated using a pointer field. 0→ no PES header or start of PSI table present. 1→ PES header or start of PSI table present.
transport_priority	Priority indicator at input to transmission channels/networks which support prioritization. 0→ lower priority. 1→ higher priority. (In a system that allows packets to be prioritized for transmission either by assignment to a carrier with higher power or to a packet with greater error protection, allows routing to path with appropriate priority.)
PID	Packet Identifier for multiplex/demultiplex.
transport_scrambling_control	Indicates the descrambling key to use for the packet. 00→ not scrambled. 10→ "even" key. 11→ "odd" key. 01→ reserved.
adaptation_field_control	Indicates if an adaptation field follows. 00→ reserved. 01→ no adaptation field, payload only. 10→ adaptation field only, no payload. 11→ adaptation field followed by payload.
continuity_counter	Increments by one for each packet within a given PID and transport priority. If two consecutive transport packets of the same PID have the same continuity_counter value and the adaptation_field_control equals 01 or 11, the two transport packets are considered duplicate. Used at the decoder to detect lost packets. Not incremented for packets with adaptation_field_control of 00 or 10.

Packet synchronization is enabled by the sync_byte which is the first byte in a packet. The sync_byte has the same fixed, preassigned, value for all DTTB bit streams. In some implementations of decoders the packet synchronization function may be done at the physical layer of the communication link (which precedes the packet demultiplexing stage). In this case the sync-byte field may be used for verification of packet synchronization function. In other decoder implementations this byte may be used as the primary source of information for establishing packet synchronization.

An important element in the link header is a 13 bit field called the **PID** (Packet Identifier). This provides the mechanism for multiplexing and demultiplexing bit streams, by enabling identification of packets belonging to a particular elementary or control bit stream. Since the location of the PID field in the header is always fixed, extraction of the packets corresponding to a particular elementary bit stream is very simply achieved once packet synchronization is established by filtering packets based on PIDs. The fixed packet length makes for simple filter and demultiplexing implementations suitable for high speed transmission systems.

Error detection can be enabled at the packet layer in the decoder through the use of the continuity_counter field. At the transmitter end, the value in this field cycles from 0 through 15 for all packets with the same PID that carry a data payload (as will be seen later, the transport allows you to define packets that have no data payload). At the receiver end, under normal conditions, the reception of packets in a PID stream with a discontinuity in the continuity_counter value indicates that data has been lost in transmission. The transport processor at the decoder then signals the decoder for the particular elementary stream about the loss of data. The MPEG-2 specification does allow the continuity_counter to be discontinuous in order to accommodate local insertion of data packets and splicing. As a consequence, the continuity_counter can be discontinuous even in an error-free transmission.

Because certain information (such as headers, time stamps, and program maps) is very important to the smooth and continuous operation of a system, the transport system should provide a means of increasing the robustness of this information to channel errors by providing a mechanism for the encoder to duplicate packets. Those packets that contain important information would be duplicated at the encoder. At the decoder, the duplicate packets are used if the original packet was in error or are dropped. Semantics for identifying duplicate packets are described in the description of the continuity_counter.

The transport format allows for scrambling of data in the packets. Each elementary bit stream in the system can be scrambled independently. One approach to an universal standard would be to specify the descrambling approach to be used but not specify the descrambling key and how it is obtained at the decoder. The key must be delivered to the decoder within a time interval of its usefulness. A portion of the "private" data capacity within the DTTB data stream could be utilized to carry the required conditional access associated data. Two possible solutions would be:

– as a separate private stream with its own PID, or

– a private field within an adaptation header carried by the PID of the signal being scrambled.

The security of the conditional access system can be ensured by encrypting the descrambling key when sending it to the receiver, and by updating the key frequently. There need not be any system imposed limit on the number of keys that can be used and the rate at which these may be changed. The only requirement that might be placed on a receiver to meet the standard is to have an interface from the decryption hardware (e.g., a Smart-card) to the decoder that meets the standardized interface specification.

Information in the link header of a transport packet can describe whether or not the payload in the packet is scrambled and if so, flags the key to be used for descrambling. The Header information in a packet is always transmitted in the clear, i.e., unscrambled. The amount of data to be scrambled in a packet can be made variable depending on the length of the adaptation header. It should noted that some padding of the adaptation field might be necessary for certain block mode algorithms.

Note that the general MPEG-2 transport definition provides the mechanism to scramble at two levels, within the PES packet structure and at the transport layer. Scrambling at the PES packet layer is primarily useful in the program stream where there is no protocol layer similar to the transport to enable this function.

4.5.3 The Adaptation layer

An MPEG-2 derived DTTB system adaptation header uses a variable length field. Its presence is flagged in the link level section of the header. The functionality of these headers is basically related to the decoding of the elementary bit stream that is extracted using the link level functions.

The presence of the **adaptation header field** is signalled in the adaptation_field_control of the link layer as described before. The adaptation header consists of information useful for higher level decoding functions and uses flags to indicate the presence of particular extensions to the field.

The header starts with a fixed length component that is present whenever the adaptation header is transmitted. The format is shown in Fig. 4.13.

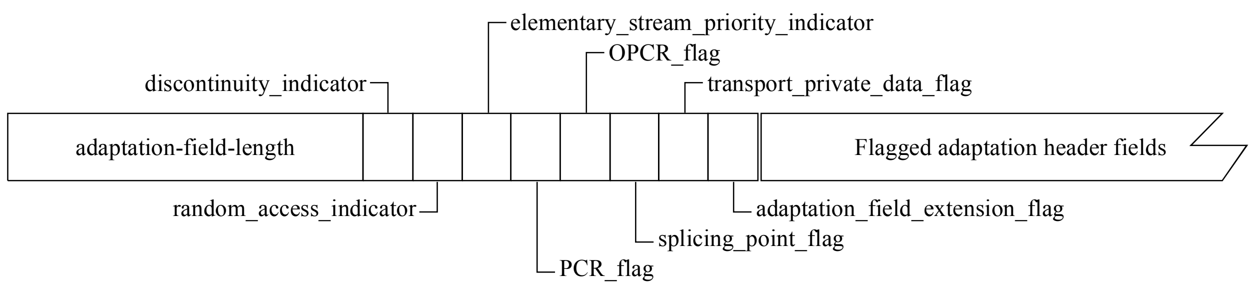


FIGURE 4.13

Fixed-length component of adaptation header

DTTB-04.13

The adaptation_field_length is a one byte field that specifies the number of bytes that follow it in the adaptation header. The adaptation header could include stuffing bytes after the last adaptation header component field. Stuffing bytes are not interpreted at the decoder. In this case, the adaptation_field_length also reflects the number of stuffing bytes. The value in the adaptation_field_length can also be used by the decoder to skip over the adaptation header, and to advance to the data payload when appropriate.

The presence of additional adaptation header fields is indicated by the state of the last five single bit flags shown in Fig. 4.13 where a value of 1 indicates that the indicated field is present. The first three (one-bit) flags do not produce extensions to the adaptation header and are described in Table 4.6.

TABLE 4.6

Field	Function/usage
discontinuity_indicator	Indicates there is a discontinuity in the PCR values that will be received from this packets onwards. This occurs when bit streams are spliced. This flag should be used at the receiver to change the phase of the local clock.
random_access_indicator	Indicates that the packet contains data that can serve as a random access point into the bit stream. One example is to correspond to the start of sequence header information in the video bit stream.
elementary_stream_priority_indicator	Logical indication of priority if the data being transmitted in the packet.

The other components of the adaptation header appear based on the state of the flags.

Synchronization of the decoding and presentation process for the applications running at a receiver is a particularly important aspect of real time digital data delivery systems. Since received data is expected to be processed at a particular rate (to match the rate at which it is generated and transmitted), loss of synchronization leads to either buffer overflow or under flow at the decoder, and as a consequence, loss of presentation/display synchronization. The problems in dealing with this issue for a digital compressed bit stream are different from those for analogue conventional television. In analogue conventional television, information is transmitted for the pictures in a synchronous manner, so that one can derive a clock directly from the picture synchronization information. In a digital compressed system the amount of data generated for each picture is variable (based on the picture coding approach and complexity), and timing cannot be derived directly from the start of picture data. Indeed, there is really no natural concept of synchronization pulses (that one is familiar with in analogue conventional television) in a digital bit stream.

The solution to this issue is to transmit timing information in the adaptation headers of selected packets, to serve as a reference for timing comparison at the decoder. This is done by transmitting a sample of a 27 MHz clock in the program_clock_reference (**PCR**) field, which indicates the expected time at the completion of the reading of the field from the bit stream at the transport decoder. The phase of the local clock running at the decoder is compared to the PCR value in the bit stream at the instant at which it is obtained, to determine whether the decoding process is synchronized. In general, the PCR from the bit stream does not directly change the phase of the local clock but only serves as an input to adjust the clock rate. Exceptions might be during channel change and insertion of local programming. Note that the audio and video sample clocks in the decoder system are locked to the system clock derived from the PCR values. This allows simplification of the receiver implementation in terms of the number of local oscillators required to drive the complete decoding process, and has other advantages such as rapid synchronization acquisition.

The PCR and OPCR fields are described in Fig. 4.14 and Table 4.7.

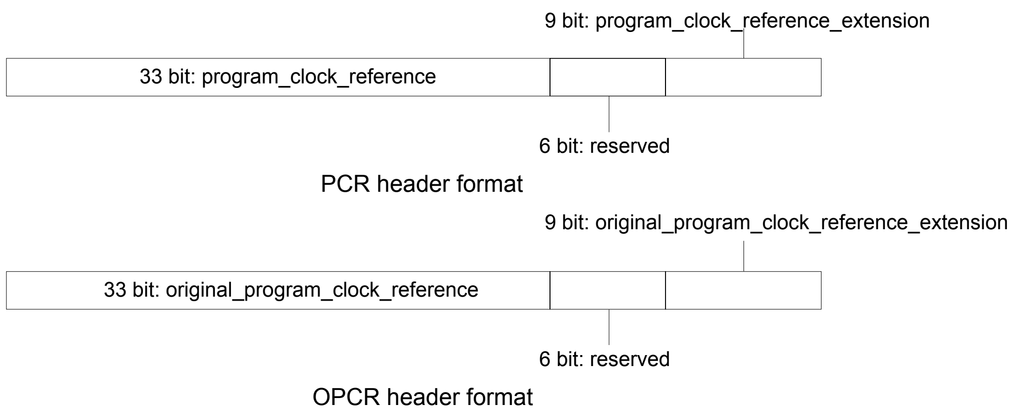


FIGURE 4.14

PCR and OPCR header format

TABLE 4.7

Field	Function/usage
PCR	Indicates intended time of arrival of last byte of the program_clock_reference_extension at the target decoder. Used for synchronization of the system decoding process. This field can be modified during the transmission process (e.g. the PCR will be transmitted at least once every 100 ms).
OPCR	Indicates intended time of arrival of last byte of the original_program_clock_reference_extension at the target decoder for a single program. This field is not modified during the transmission process. (May be used for recording and playback of single programs.)

The total PCR value is based on the state of a 27 MHz clock. The 9 bit extension field cycles from 0 to 299 at 27 MHz at which point the value in the 33 bit field is incremented by one. This results in the 33 bit field being compatible with the 33 bit field used for the 90 kHz clock in MPEG-1. The cycle time of the PCR is approximately 26 h.

The transport_private_data and adaptation_field_extension fields are described in Fig. 4.15 and Table 4.8.

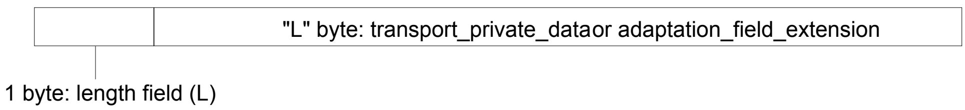


FIGURE 4.15

transport_private_data and adaptation_field-extension header format

TABLE 4.8

Field	Function/usage
transport_private_data	For private data.
adaptation_header_extensions	For future extensions of the adaptation header.

The splice_countdown field is useful for downstream (local) program insertion. The splice_countdown field, described in Table 4.9 is a one byte field that is present if the splicing_point_flag is set.

TABLE 4.9

Field	Function/usage
splice_countdown	Indicates the number of packets in the bit stream with the same PID as current packet until a splicing point packet. The splicing point packet is defined as the packet containing a point in the elementary bit stream from which point onwards data can be removed and replaced by another bit stream. Transmitted as a 2's compliment value. (Use for supporting of insertion of local programming and packets.)

4.5.4 PSIs and the pointer_field

The program_association_table and the program_map_tables that describe the organization of a multiplexed DTTB bit stream are a part of the PSI layer. PSI tables, in general, are transmitted in the appropriate bit stream sequentially without a gap between the tables. This implies that tables need not necessarily start at the beginning of a transport packet and that, therefore, there needs to be an indicator as to where these begin in the bit stream. This functionality is achieved with the pointer_field. The pointer_field is present in the packet if a PSI table begins the in the packet. This event is signalled at the link level by setting the payload_unit_start_indicator to 1. The pointer_field indicates the number of bytes that follow it before the start of a PSI table. As an example, a pointer_field value of 0x00 indicates that a new PSI table begins immediately following it.

The program_association_table is transmitted as the payload of the bit stream with PID = 0 and describes how program numbers associated with program services map on to bit streams containing the program_map_tables for the indicated programs. The program_association_table may be transmitted as multiple program_association_segments with each segment having a maximum length of 1 024 bytes. The program_association_table is described in Table 4.10. The transport decoder can extract individual table segments from the bits stream in whatever order it desires. As shown in Fig. 4.16, each table segment has a fixed length 8 byte header component for table segment identification, a variable length component that depends on the number of entities contained and a 4 byte CRC-32 field.

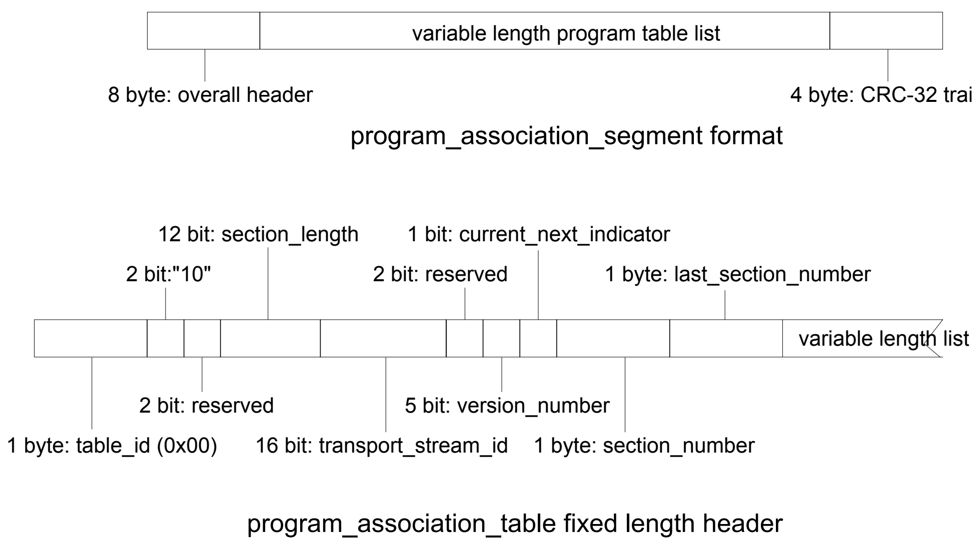


FIGURE 4.16

Program association segment and table header formats

TABLE 4.10

program_association_table header

Field	Function/usage
table_id	1 byte; indicates the nature of the table. 0x00 indicates a program_ association_table.
section_length	12 bits; length of the section of the program_association_table. The length includes all bytes following this field up to and including the CRC. The two most significant bits of the field are set to 00 giving a maximum field value of 1 024. This field allows the transport decoder to skip sections when reading from the bit stream if desired.
transport_stream_id	2 bytes; identification of a particular multiplex from several in the network (may be used in terrestrial applications to indicate service number).
version_number	5 bits; incremented each time there is a change in the program_association_table being transmitted.
current_next_indicator	1 bit; 1 indicates that the map is currently valid. 0 indicates that the map is not currently in use and will be used next.
section_number	1 byte; identifies the particular section being transmitted.
last_section_number	1 byte; section_number for the last section in the program_association_table. Needed to confirm when an entire program_association_table has been received at the decoder.

Reserved bit values are undefined. The 2 bit value "10" following the table_id needs to be received correctly.

The variable length program table list consists of program_count number of fixed length entries corresponding to each program and stuffing_bytes (to make up the program_association_ segment_length). The format for each fixed entry is shown in Fig. 4.17.

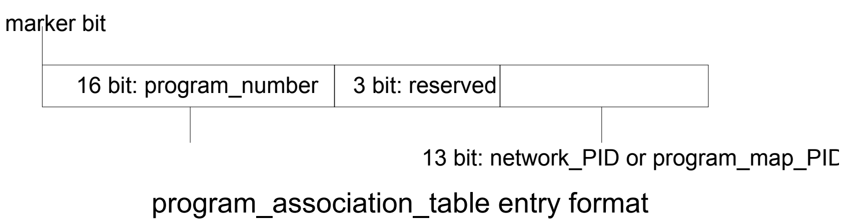


FIGURE 4.17

Program association table entry format

The program identity "0" is reserved for the network_PID (the PID of the bit stream carrying information about the configuration of the entire system). This bit stream is meant to be a private bit stream. For all other program identities, the program_map_PID is the PID of the bit stream containing the program_map_table for the particular program.

The program_association_table ends with a four byte CRC field that contains the results of a CRC calculated over the entire program map segment starting with the segment_start_code_prefix. The CRC is based on the polynomial $x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^{8}+x^{7}+x^{5}+x^{4}+x^{2}+x+1$.

The program_map_table is transmitted as the payload of the bit stream with PID = program_map_PID (as indicated in the program_association_table). The program_map_table carries information about the applications that make up programs. Each program_map_table is transmitted as a single TS_program_map_section. The format for a TS_program_map_section can be described as a combination of an overall header field, fields that describe each program within the table, and a CRC field as shown in Fig. 4.18. The CRC is the same as that used for the program_association_table. Each program_map_PID may contain more than one TS_program_map_section, with each one describing a different program.

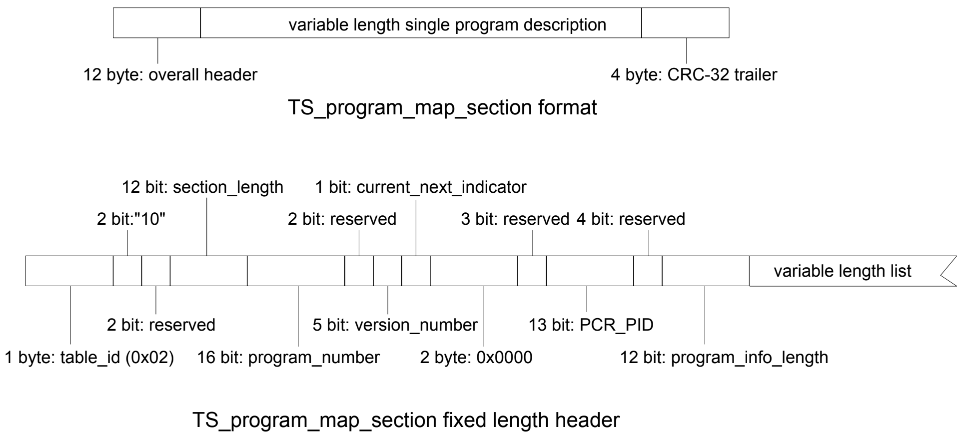


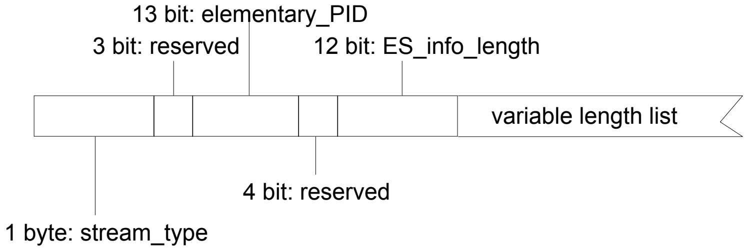
FIGURE 4.18

TS program map formats

The header format for the TS_program_map_section is shown in Fig. 4.18. The format consists of the table_id field contents (0x02); two bytes used to identify the program_number of the program being described; the two bytes following the current_next_indicator are set to “0” since the description of each program is defined as fitting into one section; a 13-bit PCR_PID identifies the PID of the particular packetized elementary bit stream in the program that contains the PCR values for the program; and the program_info_length field indicates the number of bytes of program_descriptors that follow. All other fields have the same format and functionality as found in the program_association_table.

The program description that follows the header consists of the optional, variable length, program_descriptor field (whose length was indicated by the program_info_length field), followed by descriptions of each of the individual elementary bit streams that make up the program.

Each elementary stream description consists of a 5 byte fixed length component and a variable length elementary_stream_descriptor component as shown in Fig. 4.19 and described in Table 4.11.



elementary stream description format

FIGURE 4.19

Elementary stream description

TABLE 4.11

Elementary stream description

field	Function/usage
stream_type	Indicates the application being considered in this elementary stream 0x00 ITU-T/ISO/IEC Reserved 0x01 MPEG-1 video 0x02 MPEG-2 video 0x03 MPEG-1 audio 0x04 MPEG-2 audio 0x05 MPEG-2 private sections 0x06 MPEG-2 PES packets containing private data 0x07 MHEG 0x08 MPEG-2 Part 1, DSM CC 0x09 ITU-T Rec, H.222.1 0x0A – 0x0D MPEG-2, Part 6, Type A – Type D 0x0E MPEG-2 auxiliary 0x0F – 0x07 MPEG-2 reserved 0x80 – 0xFF user private[1]
elementary_PID	Indicates the PID of the transport bit stream containing the elementary bit stream.
ES_info_length	Indicates the length of a variable length elementary_stream_descriptor field that follows.

(1) The stream_type for AC-3 audio is 0x81.

Descriptors are transmitted in the program_descriptor and the elementary_stream_descriptor fields to describe certain characteristics of the program or the elementary bit stream. Each program_descriptor and elementary_stream_descriptor can consist of a number of individual descriptor field elements transmitted sequentially.

A mechanism for indication the presence of descriptors is required in order to use descriptors. This functionality is achieved in the PSI tables described by the length field that precedes the descriptor with a zero value indicating that no descriptor is present. Identification of the descriptor is also required. This is achieved within the descriptor header itself which consists of a one-byte descriptor_tag field followed by a one-byte descriptor_length field that specifies the number of bytes in the descriptor that follows. The set of valid descriptor_tags in the system are defined in the MPEG-2 documentation.

4.6 Features and services

4.6.1 Introduction

A DTTB transport architecture should be flexible and capable of supporting a number of audio, video, and data services through its system multiplex. Data services may be program related or non-program related. SMPTE and others have identified program related data services that could be communicated from the program source that would be helpful to the display of the program. The identified functionalities are seen as desirable for use in the receiver to improve the system performance or to enhance the service for the viewer. Some of the functionalities are viewed as useful in distribution networks to support international program exchange, for scalable service environments or for use in a simulcast implementation scenario.

4.6.2 Audio compression types and language identification

The transport layer syntax allows the definition of a program map which permits identification of individual audio services by their compression algorithm as well as identification of multiple language channels that can be selected by the viewer or by the distribution network. This requirement to identify compression algorithms allows selection of an audio service (monaural, stereo, or surround sound) and bit rate appropriate to the associated program.

4.6.3 Program information

A program service can be provided as an ancillary data service with its own PID. This could take the form of a program guide that is personalized by the service provider. The information required can be supported by a low refresh rate that would not consume a significant amount of the channel bandwidth.

4.6.4 Captioning

Captioning information, like audio associated with the video, must be synchronized with each television frame. Captioning information should be uniquely identified and carried as user data within the video picture layer. However, the value of using PES packets or sections to maintain commonality in processing at the receiver between captioning and other applications should be considered further.

4.6.5 Closed captioning

Closed captioning is a captioning service designed for the hearing impaired. Like general captioning information, closed caption services must be synchronized with each television frame and should be uniquely identified and carried as user data within the video picture layer. However, nothing in the MPEG-2 syntax would prevent closed captioning data being sent in a separate PID, and in some applications this might have some advantage over the carrying the data within the video picture layer. The value of using PES packets or sections to maintain commonality in processing at the receiver between captioning and other applications should be considered further.

4.6.6 Program source and program identification

Program source identification and program identification information has a great many uses. One application is to allow automatic access to programming for recording and delayed playback by the viewer. Program source and program identification should be uniquely identified and carried as an ancillary data service with its own PID.

4.6.7 Conditional access identification

Conditional access systems can be supported by the transport syntax with bits identified in the packet header. Information about the conditional access information including key information should be uniquely identified and carried as private data.

4.6.8 Picture structure information

Some parties interested in implementing DTTB services intend to provide a range of scalable services for use in different reception environments. Compressed and encoded image sequences may also serve as a format for program interchange. The ability of the video syntax to carry the details of the picture sampling structure used in the coded image, including samples per line, lines per frame, frames per second, scanning format (interlace or progressive) and aspect ratio facilitates use of the program material across a broad spectrum of applications.

4.6.9 Colorimetry

Information on the colorimetry characteristics of the encoded video can be supported in the video sequence layer. This includes a description of the colour primaries, transfer characteristics, and the colour matrix coefficients, and allows the receiver device to properly accommodate image sequences derived from sources using different colorimetry.

4.6.10 Colour field identification

Conventional television receivers will dominate the market at the start of DTTB services and will populate the market for many decades thereafter. The advantages of DTTB services may lead to a desire to make these services available to existing conventional (NTSC, PAL, or SECAM) receivers.

Providing colour field information in the video syntax helps the decoder re-encode the image sequence to a conventional service compatible output with reduced artifacts, particularly when the source image sequences were derived from related program material.

4.6.11 Scene changes and clean-insertion points

Automatic scene change detection algorithms may be used in some encoders to improve coding efficiency. Such scene change information, when supported by a production facility, could prove useful to the video encoder at both the compression and transport levels. The information could also prove useful to distribution systems to identify points in the data stream where switching between sources of transmitted bit streams could take place.

There is a further requirement to identify points in the transmitted bit stream other than scene changes where switching between sources of transmitted bit streams or where packet replacement can take place without noticeably disrupting the performance of the receiver. These are termed “clean-insertion” points and are useful for down-stream (local, national, or regional) service providers to modify a cooperative or network service to accommodate it for local use.

4.6.12 Field/frame rate and film pull-down

Systems for use in the 60 Hz environment can be optimized for transmitting film originated image sequences by transmitting the frame rate of the coded bit stream. This allows encoders to maximize coding efficiency by not transmitting redundant fields and signals the decoder the proper order for displaying the decoded pictures. The DTTB frame rate syntax can be supported within the video sequence layer to support frame rates of 23.976 (24 ÷ 1.001), 24, 25, 29.97 (30 ÷ 1.001), 50, 59.94 (60 ÷ 1.001), and 60 Hz as well as an extension for future capabilities.

4.6.13 Pan and scan

4:3 aspect ratio receivers will dominate the market at the start of wide-screen (16:9) aspect ratio services and will populate the market for many decades thereafter. The advantages of wide-screen DTTB services may lead to a desire to make these services available to existing analogue based receivers and other 4:3 aspect ratio display devices.

Pan and scan information could be transmitted as an extension of the picture layer syntax. The pan and scan extension would allow decoders to define a rectangular region which may be panned around the entire coded image, and thereby identify a 4:3 aspect ratio window within a 16:9 coded image.

4.6.14 Random entry into the compressed bit stream

Random entry into the application bit streams such as video and audio is necessary to support functions such as program tuning and program switching. Random entry into an application is possible only if the coding for the elementary bit stream for the application supports this functionality directly. For example, a DTTB video bit stream might support random entry through the concept of Intraframe coding (I-frames that are coded without any prediction, and which can therefore be decoded without any prior information). The beginning of the video sequence header information preceding data for an I-frame could serve as a random entry point into a video elementary bit stream. In general, random entry points should also coincide with the start of PES packets where they are used, e.g. for video and audio. The support for random entry at the transport layer comes from a flag in the adaptation header of the packet that indicates whether the packet contains a random access point for the elementary bit stream. In addition, the data payload of packets that are random access points also start with the data that forms the random access points into the elementary bit stream itself. This approach allows the discarding of packets directly at the transport layer when switching channels and searching for a resynchronization point in the transport bit stream and also simplifies the search for the random access point in the elementary bit stream once transport level resynchronization is achieved.

A general objective is to have random entry points into the programs as frequently as possible, to enable rapid channel switching.

4.6.15 Local program insertion

This functionality is important for down stream switching of packets (inserting local programming such as public service messages or commercials) into an existing bit stream. In general, there are only certain fixed points in the elementary bit streams at which program insertion is allowed. The local insertion point has to be a random entry point but not all random entry points are suitable for program insertion. For example, in addition to being a random entry point, the VBV_delay (video buffer verifier delay) needs to be at a certain system defined level to permit local program insertion.

The VBV_delay information can be computed and transmitted as part of the header data for a picture in the compressed video stream. It thereby defines how full the decoder video buffer needs to be before the bits of the current picture are extracted from the buffer and synchronizes the encoder and decoder processes. This is required to control the memory needed at the decoder for buffering data and to prevent buffer overflow or underflow. Local program insertion also always takes place at the transport packet layer, where the data stream splice points are packet aligned. Implementation of the program insertion process by the broadcaster is aided by the use of a splice_countdown field in the adaptation header that indicates ahead of time the number of packets to countdown until the packet after which splicing and local program insertion is possible. The insertion of local programming usually results in a discontinuity in the values of the PCR received at the decoder. Since this change in PCR is completely unexpected (change in PCR values are usually only expected during program change) the decoder clock could be thrown completely out of synchronization. To prevent this from happening, information is transmitted in the adaptation header of the first packet after the splicing point to notify the decoder of the change of PCR values (so that it can change the clock phase directly instead of attempting to modify the clock rate). In addition, there are constraints on:

– the length of the bit stream that is to be spliced in, to assure that the buffer occupancies at the decoder both with and without the splice would be consistent, and
– the initial VBV value assumed when encoding the bit stream to be spliced in, in order to prevent decoder buffer underflow or overflow.

4.6.16 Individual programme identification

In broadcasting services, two essential functions are necessary: they are, the function of receiving a certain broadcast channel continuously without any action, and the function of automatic reception or recording of an individual programme, simultaneously. Therefore, it is necessary to define a new descriptor, which is called "Event Descriptor", to identify the individual programme, because the programme number corresponds to the programme channel. An example for the descriptor is shown in Fig. 4.20.

descriptor tag	descriptor length	event id
8	8	32

FIGURE 4.20

Structure of the Event Descriptor

4.6.17 Other channel information

In the MPEG-2 systems, each programme can be received only after the PAT and PMT are received, and some delay occurs when one selects or changes the channel. In order to minimize this delay, a self-cross indicator is introduced into the PAT or NIT.

It indicates whether the PAT or NIT is for the information of the transport stream from which a programme is viewed, or for the other transport streams (channels) which can be received. By this function, the information of other channels (streams) can be obtained while viewing some programme, and it provides assistance for channel selection.

CHAPTER 5

PHYSICAL LAYER – CHANNEL CODING AND MODULATION

5.1 Introduction

The transmission of data in digital form has long been known to offer many advantages over analogue transmission. Digital modulation is an outgrowth of the more familiar methods of analogue modulation such as amplitude, frequency and phase modulation. Recommendation ITU-R BT.1306 lists the important parameters of a DTTB modulation system and provides parameter values or ranges of values for those parameters. The Recommendation allows the system designer to tailor system performance to meet a variety of different design constraints. The modulation techniques proposed in the Recommendation are for either single or multiple carrier modulation methods and for different channel bandwidths – 6, 7 and 8 MHz options are available. This chapter explains some of the issues involved in selecting a suitable modulation system for a given application.

5.2 Spectral efficiency

It is generally agreed that to provide a DTTB service that can deliver HDTV or multi-programme SDTV services a bit rate of about 20 Mbit/s (or more) is required. To accommodate such a data rate requires an effective spectrum efficiency of 4 bit/s/Hz for a national 6 MHz system, or 3 bit/s/Hz for national 7 or 8 MHz systems.

Theoretical spectral efficiencies of up to 4 bit/s/Hz can be achieved by 16-QAM, 4-VSB or 16-PSK systems. These modulation methods could be applied either to modulation of a single carrier with a high data rate signal or to modulation of a large number of carriers with low rate data signals. Either single carrier or multi-carrier modulation could be the basis for a worldwide transmission standard.

However, the error statistics of practical terrestrial transmission channels are such as to require the inclusion of forward error correction coding in a practical transmission/modulation system. Considerations related to filter implementation may further reduce effective data rates in practical systems. The result of these considerations is that the net data rate will be less than that predicted from a simple consideration based on theoretical spectrum efficiency and channel bandwidth. With the use of two stage channel codes, a practical implementation can lead to a substantial reduction from gross to net channel data rate. For example, a coding scheme based on use of a 2/3 Trellis Code concatenated with a Reed-Solomon (207,187) code results in a net data-rate only 60% of the gross data-rate.

This has prompted consideration of more complex modulation systems. The added complexity being justified because of the opportunity to provide a required net data rate within a highly error protected channel. As a result designers have been investigating the performance of higher order modulation systems such as 64-QAM or 8-VSB.

Spectral efficiency results not only from the fundamental spectral information "density" in bit/s/Hz of the modulation system within any given channel, but is also influenced very much by the spectrum re-use characteristics of a particular digital system.

Factors affecting spectrum re-use in a given system include:

– the required *C*/*N* of the digital system, which determines the transmitter power levels, which are themselves constrained by the need to protect existing services;
– co-channel and adjacent channel protection ratios for existing and new services;

– the system's capability to provide for single-frequency networks, either locally, regionally or nationally;

– the system's capability to provide for dual-frequency networks.

5.3 Modulation techniques

5.3.1 General considerations

Of the generic modulation systems (*m*-VSB, *m*-QAM, *m*-PSK, *m*-DAPSK) *m*-PSK requires higher transmission power (which may exacerbate channel planning problems) and is therefore not preferred. QAM and VSB modulation systems have similar power requirements and noise performance.

These modulation systems can be applied to either a single carrier modulated at a high data rate or to a large number of carriers modulated at relatively low rates – the multi-carrier approach. Currently most research efforts for DTTB are focused on either a single carrier system using 8-VSB and multi-carrier systems using 16-QAM, 64-QAM or even 256-QAM.

In both cases, the research work builds on experience gained in other fields. Experience with single carrier QAM and QPSK systems has come from applications in the fields of terrestrial microwave and satellite transmission. While experience with multi-carrier systems has come from high frequency modems designed for military and telephone applications but is now being supplemented by experience gained in the development of a digital audio broadcasting system in Europe.

Because of the severe channel impairments that can occur in the VHF and UHF television bands, transmission conditions for DTTB are likely to be significantly more difficult than for satellite or cable transmission.

5.3.2 Single-carrier modulation (SCM)

The modulation method for the single-carrier system proposed for the USA's ATV service, 8-VSB (vestigial sideband), was chosen after comparative testing with QAM since, on an overall technical basis, it provided better service characteristics in a sharing environment with analogue television. This modulation method provides a means for the transmission of a high bit rate eight level baseband signal. In SCM, the effects of multipath are handled by the receiving system, often with an adaptive equalizer, as discussed below.

5.3.2.1 8-VSB modulation

The 8-VSB single-carrier modulation system is as follows: 19.29 Mbits/s are delivered in a 6 MHz channel.

The serial data stream is comprised of 188-byte MPEG-compatible data packets. Following randomization and forward error correction processing, the data packets are formatted into Data Frames for transmission and Data Segment Sync and Data Field Sync are added.

Each Data Frame consists of two Data Fields, each containing 313 Data Segments. The first Data Segment of each Data Field is a synchronizing signal, which includes the training sequence used by the equalizer in the receiver. The remaining 312 Data Segments each carry the equivalent of the data from one 188-byte transport packet plus its associated FEC overhead.

Each Data Segment consists of 832 symbols. The first four symbols are transmitted in binary form and provide segment synchronization. This Data Segment Sync signal also represents the sync byte of the 188-byte MPEG-2 compatible transport packet. The remaining 828 symbols of each Data Segment carry data equivalent to the remaining 187 bytes of a transport packet and its associated FEC overhead. These 828 symbols are transmitted as 8-level signals and therefore carry three bits per symbol. The symbol rate is 10.76 Msymbols/s and the Data Frame rate is 20.66 frames/s.

To assist receiver operation a pilot-carrier is included at approximately 310 kHz from the lower band edge.

System performance is based on the concept of a linear-phase, raised-cosine Nyquist filter response in the concatenated transmitter and receiver. The system filter response is essentially flat across the entire band, except for the transition regions at each end of the band. Due to the vestigial sideband nature of the transmitted signal, the same skirt selectivity on both sides is not required, although this parameter value must be implemented consistently since the receiver must match the transmitter. The roll-off in the transmitter has the response of a linear-phase, root-raised cosine filter.

Additional adjacent channel suppression (beyond that achieved by sideband cancellation) may be performed by a linear phase, flat amplitude response SAW filter. Adjacent channel energy spillage at the IF output needs to be at least 57 dB down from the desired ATV signal power.

5.3.3 Multi-carrier modulation (MCM)

Orthogonal frequency division multiplexing (OFDM) is the most commonly proposed MCM system.

5.3.3.1 OFDM

The OFDM concept is based on spreading the data to be transmitted over a large number of carriers, each being modulated at a low bit rate. In a conventional frequency division multiplex the carriers are individually filtered to ensure there is no spectral overlap. There is therefore no inter-symbol interference between carriers but the available spectrum is not used with maximum efficiency. If however, the carrier spacing is chosen so the carriers are orthogonal over the symbol period, then symbols can be recovered without interference even with a degree of spectral overlap. For maximum spectral efficiency, the carrier spacing equals the reciprocal of the symbol period. The multiplex of carriers may be conveniently generated digitally using the inverse fast Fourier transform (FFT) process.

Preferred implementations of the FFT tend to be based on radix 2 or radix 4 algorithms, or some combination of radix 2 and 4. This preference leads to the number of carriers generated in practical OFDM systems being some power of 2. Example systems are based on 2 048 (2k) carriers and 8 192 (8k) carriers. However, the number of actual carriers transmitted is always smaller than the maximum number possible, as some carriers at either edge of the channel are not used. These unused carriers make a frequency guard-band which allows practical IF filtering. The active carriers carry either data or synchronization information. Any digital modulation scheme may be used to modulate the active carriers, e.g. QPSK, *n*-QAM or *n*-DAPSK where *n* is commonly 16 or 64.

OFDM, due to its multicarrier nature, exhibits relatively long symbol periods, around 224 µs in a 2k system. This long symbol period provides a degree of protection against inter-symbol interference caused by multipath propagation. This protection can, however, be greatly enhanced by use of the guard interval. The guard interval is a cyclic extension of the symbol, in simplistic terms a section of the start of the symbol is simply added to the end of the symbol. The guard intervals for the 2k and 8k systems are 1/32 of the symbol period (7/28 µs) 1/8 of the symbol period (28/112 µs) and 1/4 of the symbol period (56/224 µs) and 1/2 of the symbol period (112/448 µs). As the proportion of the symbol used to make the guard interval is increased the transmission capacity decreases.

However, if a system with a greater number of carriers was used the symbol period would increase and therefore the same proportion of guard interval would give a greater protection in terms of absolute time. For example an 8k system with a symbol period of 896 µs and a 1/4 of symbol period guard interval results in a 224 µs guard interval. However increasing the number of carriers impacts the receiver complexity and the ability to track time-varying channels, so a trade-off is necessary. Figure 5.1 shows how the FFT sampling window, which is equivalent to the symbol period can be positioned within the symbol and guard interval to minimize Inter Symbol Interference (ISI).

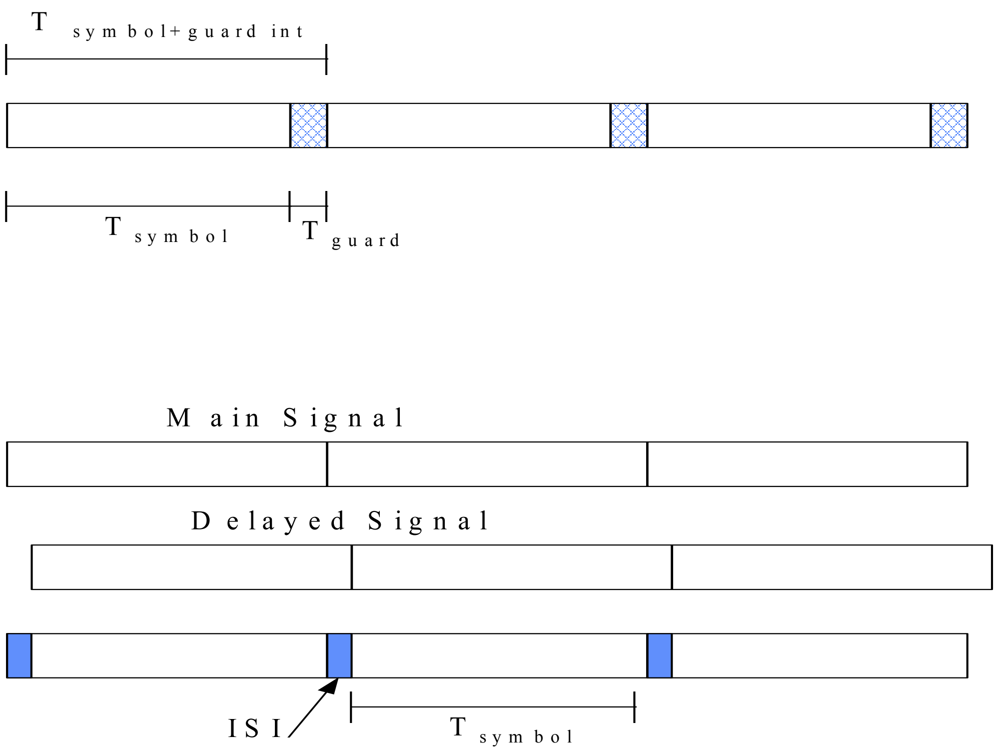


FIGURE 5.1

Guard interval utilization

OFDM when coupled with appropriate channel coding (error correction coding) can achieve a high level of immunity against multipath propagation and against co-channel interference e.g. NTSC, PAL, SECAM. OFDM systems also offer the broadcaster great flexibility as bit rate can be traded against level of protection depending on the nature of the service. For example, mobile reception of the OFDM signal may be possible given due consideration to factors including vehicle speed, carrier spacing, data rate and modulation scheme, whereas, for a service with fixed reception, high order modulation schemes and consequently high data rates could be used.

OFDM signals also allow the possibility of single-frequency network (SFN) operation. This is due to OFDM's multi-path immunity. SFN operation is possible when exactly the same signal, in time and frequency, is radiated from multiple transmitters. In this case at any reception point in the coverage overlap between transmitters, the weakest received signals will act as post or pre-echoes to the strongest signal. However, if the transmitters are far apart the time delay between the received signals will be large and the system will need a large guard interval.

The choice of main parameters for the OFDM system is determined from the requirement for SFN operation.

The carrier spacing in an OFDM system is inversely proportional to the symbol length to achieve orthogonality, for which reason the number of carriers in a channel is determined from the symbol length. In order to obtain a reasonable useful bit rate the maximum guard interval which can be used is approximately 1/4 of the active symbol length. In an SFN, signals from different transmitters arriving outside the guard interval will result in interference.

There are two main possible digital modulation techniques for OFDM systems. The first technique uses *n*-QAM modulation, synchronization signals and "scattered pilots". The second technique uses *n*-DAPSK modulation and some continual pilot carriers. Both systems also carry transmission parameter signalling (TPS) information. The TPS carries information about the transmitted signal e.g. code rate and type of modulation. Figure 5.2 provides a diagrammatic comparison of various modulation system constellations.

Although the use of the guard interval removes the effect of inter-symbol interference under multipath conditions, it can not remove the effect of frequency selective fading. Under these conditions the amplitude and phase of each subcarrier is distorted. If the OFDM receiver is to coherently demodulate the signal it needs to equalize the phase and amplitude of each carrier. This can be done after the FFT using a simple equalizer. This process is known as channel estimation and equalization. Different techniques are proposed to estimate the channel and therefore equalize the signal for QAM and DAPSK. The *n*-QAM system uses a set of scattered pilots in the frequency and time domains with interpolation filtering to estimate the channel response. The *n*-DAPSK system derives the channel response for each carrier from the data using simple recursive filter techniques to estimate and correct phase and amplitude errors. Another technique that does not require the use of pilot carriers is that of filtered decision feedback channel estimation and demodulation. It is an OFDM symbol based fast channel estimation scheme that can quickly adapt to channel variations. Since pilots are not used, higher spectrum efficiency can be achieved.

In the case of frequency selective fading, or when an OFDM signal is subject to analogue co-channel interference some carriers will be affected to a greater extent than others. Figure 5.4 shows that in the case of frequency selective fading, the S/N ratio will be lower on some carriers than on others. In the case of co-channel interference the carriers near the analogue sound and vision carriers will suffer a much greater interfering signal than those elsewhere in the OFDM signal. Channel state estimation is the process of estimating how much each carrier is affected by the combination of frequency selective fading and interference. This channel state estimation information can be passed to the error correction subsystem which can use the information to modify the soft decision information for each recovered bit of data. The Viterbi decoding algorithm as specified in the inner decoder of the error correction subsystem is perfectly suited to making use of the soft decision information generated by the channel state estimation subsystem.

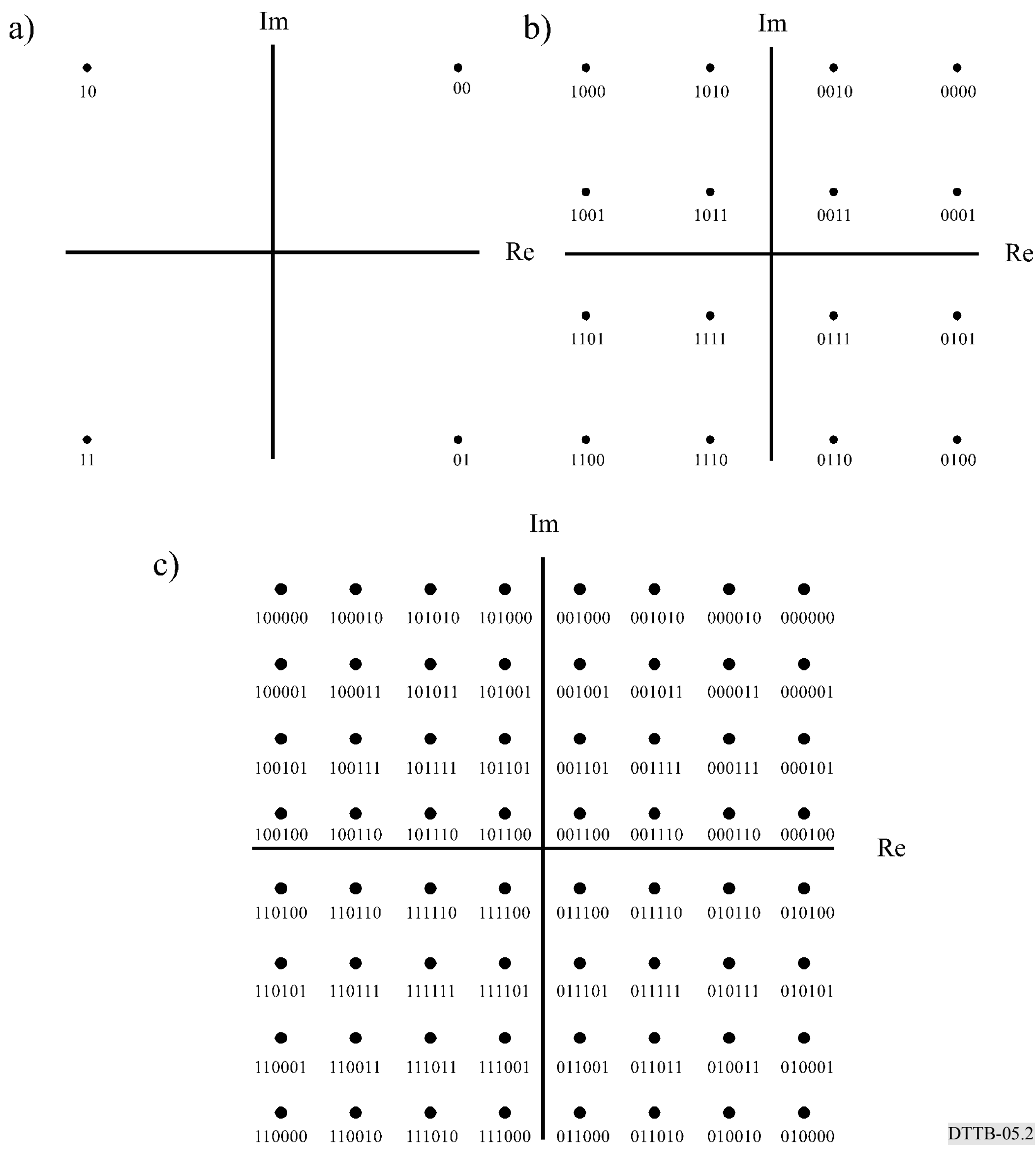


FIGURE 5.2

Comparison of modulation state constellations

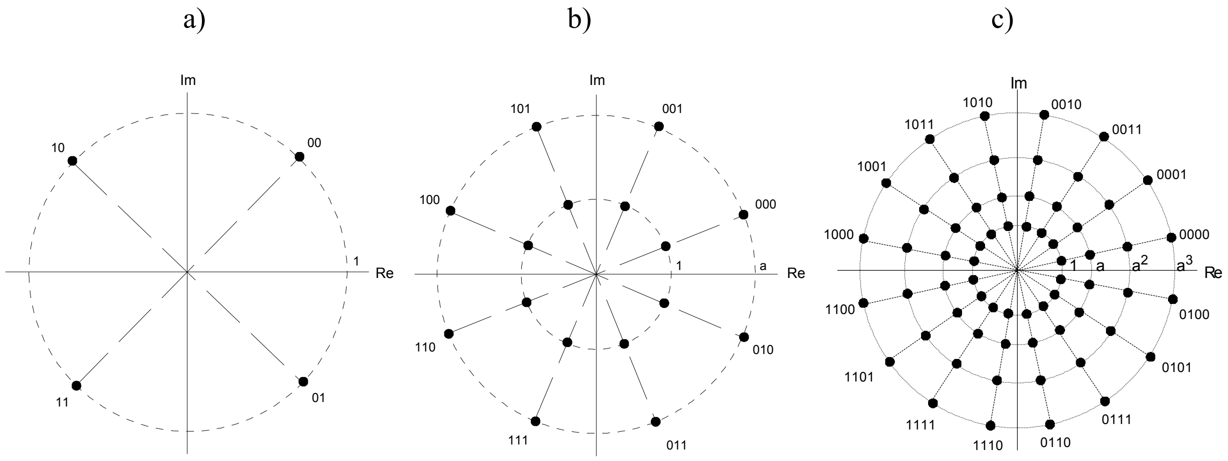


FIGURE 5.3

Constellation diagrams of a) DQPSK, b) 16-DAPSK and c) 64-DAPSK

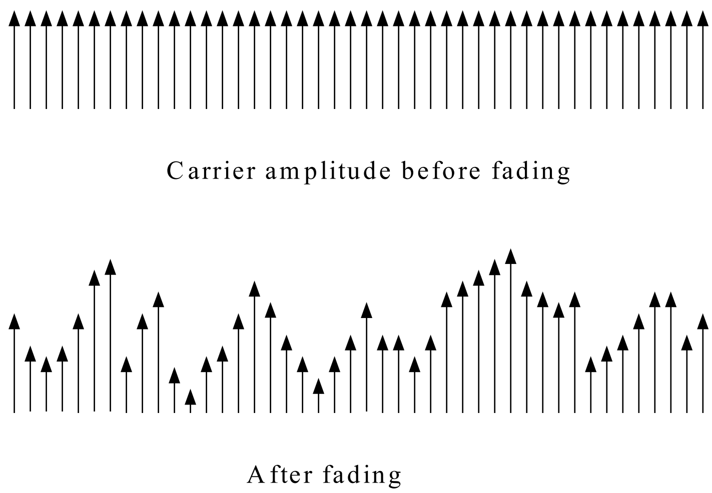


FIGURE 5.4

Effective of frequency selective fading on carrier amplitude

Table 5.1 provides a typical selection of the characteristics of OFDM systems in an 8 MHz channel with appropriate guard intervals. Figures for columns Data rate are in Mbit/s.

TABLE 5.1

System comparison for soft decision decoding
(required *C*/*N* for BER = 2×10^{-4} after Viterbi decoder)

	4-QAM(1)				4-DPSK			
Code rate	Data rate(2)	AWGN channel	Rice channel	Rayleigh channel	Data rate	AWGN channel	Rice channel	Rayleigh channel
1/2	4.7	2.7	3.2	4.6	5.4	5.4	6.0	7.4
2/3	6.3	4.3	4.8	7.0	**7.1**	**7.1**	**7.9**	**10.8**
3/4	**7.1**	**5.3**	**5.9**	**9.7**	8.1	8.1	9.1	13.3
	16-QAM(1)				16-DAPSK			
1/2	9.4	8.2	8.8	10.8	10.8	13.5	14.0	16.2
2/3	12.5	10.5	11.0	14.3	**14.3**	**16.0**	**16.5**	**19.2**
3/4	**14.1**	**11.5**	**12.3**	**16.5**	16.1	17.3	17.7	21.0
	64-QAM(1)				64-DAPSK			
1/2	14.1	13.5	14.1	16.0	16.1	18.4	19.2	21.2
2/3	18.8	15.7	16.4	19.6	**21.5**	**21.5**	**21.8**	**24.3**
3/4	**21.2**	**17.3**	**17.9**	**22.2**	24.2	22.8	23.5	26.8

(1) For the QAM perfect channel estimation had been used.

(2) The data rate for the *M*-QAM is calculated under condition a real channel estimation is used. An overhead of about 12.4% for pilot cells and synchronization symbols has been taken.

By inserting pilot carriers spread in time and frequency, the receiver can use time and frequency interpolation to follow changing channel conditions. Pilot carriers can also be used in the receiver for phase error correction.

5.3.3.2 Band segmented transmission orthogonal frequency division multiplex (BST-OFDM)

It is not necessary that the carriers in an OFDM ensemble be contiguous. It is possible to omit some carriers in an otherwise continuous array so as to minimize interference to or from a distant co-channel analogue signal. An OFDM signal can be segmented and combined in a frequency band while retaining orthogonality. The proposed “band-segmented transmission” (BST) is an example of how the technology can be applied to provide flexibility in frequency usage (and thereby make use of vacant channel slots in an otherwise congested band) and provide extendibility for future systems.

5.4 Channel coding (error correction coding)

Suitably designed channel coding can be used to reduce errors in both single carrier and multi-carrier modulation systems.

For SCM, a training sequence is usually transmitted to assist adaptive equalizer convergence and system synchronization. For MCM reference signals are usually transmitted to obtain channel state information to assist frequency domain equalization and synchronization.

To achieve adequate performance at an ATV threshold point of 15-16 dB carrier-to-noise ratio, a concatenated coding system attaining a BER of 10^{-11} in a Gaussian channel is required. In the concatenated coding approach two levels of forward error correction are employed: an "inner" modulation code and an "outer" symbol error correcting code. Interleavers and de-interleavers are also used to fully exploit the error-correction ability of FEC codes.

The presence of various sources of interference generally requires the use of sophisticated error coding strategies containing large depths of interleaving. Single or concatenated codes could be used for this purpose.

Concatenated error correction coding schemes consist of an inner code, an interleaving scheme and an outer code. All parts of the concatenated coding scheme need to be designed together so as to produce an overall coding system that is well matched to use in the terrestrial channel. For the above reason, it is desirable to treat the concatenated code as one entity and not split the inner and outer codes into source and channel subparts.

At this stage of development Trellis codes are the mostly commonly proposed Inner modulation codes. Code rates of 2/3, 3/4 or 7/8 have been suggested. An alternative may be a more complex turbo code which could provide a lower data rate overhead for a given level of error protection.

In the area of the outer error correcting code, there was an emerging consensus on the use of Reed-Solomon codes. Although different block lengths and correction distances have been suggested by different system proponents it was thought to be realistic to envisage that a range of different Reed-Solomon codes could be processed by a single, appropriately designed, integrated circuit and that this could provide a suitable point for standardization.

For outer coding most systems considered for use in the DTTB environment use the Reed-Solomon method. The system for 6 MHz uses Reed-Solomon at (207,187). The other systems use Reed-Solomon at (204,188). Future applications may utilize other Reed-Solomon structures.

As already stated the presence of various channel impairments requires the use of a sophisticated error coding strategy. However an error coding subsystem has already been specified for the European satellite and cable systems. In order to ensure maximum commonality of receivers the European OFDM system has decided to make use of the same error correction as the DVB satellite baseline system with the addition of an inner frequency interleaver. Therefore a concatenated Viterbi Reed-Solomon strategy is proposed with a between codes interleaver.

The inner interleaver interleaves FFT symbols. It operates on one FFT symbol at a time and is therefore a frequency interleaver only. The interleaver works on a bitwise basis and interleaves bits between the modulated symbols on the OFDM carriers. The purpose of the inner interleaver is to improve the system performance when the channel is subject to frequency selective fading or co-channel interference. The interleaver ought to spread clusters of errors caused by carriers with relatively poor S/N or S/I ratios.

The inner code of the error correction is a convolutional code, as specified in the satellite baseline specification, this code can be decoded using the Viterbi decoding algorithm. The inner code may be punctured to increase available data capacity. The puncturing rates and patterns are as defined in the DVB satellite baseline specification. The use of channel state estimation and soft decision information derived from the received data points can significantly improve the transmission performance. The channel state information can be derived in a number of ways, for example using the amplitude equalization information generated to coherently demodulate each OFDM carrier.

If the capability of the Viterbi algorithm to correct the channel errors is exceeded it will produce bursts of errors. Therefore the outer code must be suited to correcting burst errors. Reed-Solomon (RS) codes have been specified for this task. The particular RS code chosen is a ($k = 188$, $n = 204$) code. RS codes use symbols of 8 bits (bytes). A codeword of length n containing k data bytes and n-k redundant bytes is used. The code rate R is therefore k/n and the code normally provides the capability to correct $t = (n\text{-}k)/2$ errored bytes which in the RS (204,188) case means that up to 8 errored bytes can be corrected.

Since burst errors at the output of the Viterbi decoder will usually affect more than 1 byte, additional interleaving between the inner and outer codes is employed. This interleaver is again as specified in the DVB Satellite baseline specification. It is a convolutional interleaver which interleaves data bytes.

Single error correcting coding schemes may reduce the scale of the interleaving RAM and lead to savings in the cost of decoders. Some block codes have almost the same performance as that of concatenated codes and appropriate decoding LSIs are available.

5.5 Comparisons of early implementations of single- and multi-carrier systems

In SCM, the information bearing data is used to modulate one carrier which occupies the entire RF channel. In MCM, QAM modulated symbols are used to modulate multiple low data rate carriers which are transmitted concurrently.

There are interesting frequency/time-domain dualities between MCM and SCM. MCM can be thought of as a frequency domain technique and SCM as a time domain technique.

One ramification of frequency-time duality is that, to prevent inter-symbol interference for SCM, one must reserve part of the spectrum for pulse shaping (frequency domain), while for MCM one must insert guard intervals (time domain).

For SCM channels with multipath distortion, a training mechanism is usually transmitted to assist adaptive equalizer convergence and system synchronization. An adaptive equalizer and a high-gain directional antenna can also reduce the impact of co-channel DTTB and analogue TV interferences.

For MCM, pilot carriers are usually transmitted to obtain channel state information for frequency domain equalization and synchronization. SCM and MCM have comparable BER performance when the channel noise is additive white and Gaussian.

For an MCM system the use of a guard interval can almost eliminate the intersymbol interference, but it also reduces data throughput. To minimize the loss of throughput, the size of the FFT must be increased. The size of the FFT, however, is limited by digital signal processing speed, cost and receiver phase noise. To compensate for the frequency selectivity of the channel, a 1-tap frequency domain equalizer can be used in combination with soft decision Viterbi decoding using channel state information. The effectiveness of interleaving is also crucial to the performance of the system. The research on optimum codes for high order QAM-OFDM systems is still ongoing.

The performance of both SCM and MCM under combined impairments of noise, co-channel analogue TV interference and strong multipath distortion is yet to be determined.

5.5.1 Impulse interference

For low power impulse interference, multi-carrier systems are more robust to impulse interference, since interference can be averaged over the entire FFT block. On the other hand, a short but high power burst of interference will be expanded by the OFDM process to cause serious interference for a number of symbol periods equivalent to the duration of the impulse across all carriers. This can correspond to a significant number of errors. However it has been reported that field trial results have shown that, with adequate interleaving and error-correction, this type of interference is not a serious problem.

Single carrier systems are sensitive to time domain impulses such as lightning and car ignition interference.

5.5.2 Multipath distortion

In typical DTTB reception situations, multipath propagation caused by reflections or non-homogenities in the propagation medium will cause intersymbol interference to the unprocessed received data stream. Multipath reception will also manifest itself as frequency selective fading within the channel.

For SCM, intersymbol interference, if uncorrected, will result in eye height closure and an increase in the minimum C/I at which the system can operate.

For practical SCM systems an adaptive equalizer (usually a decision feedback equalizer) is used to minimize the effects of multipath distortion. For its operation it requires a training sequence which will slightly reduce data throughput. An adaptive equalizer can also converge without a training sequence by use of a blind equalization technique. Any adaptive equalizer will however increase the system noise threshold when multipath is present. (Adaptive equalizers may also reduce the impact of co-channel and adjacent channel interference.)

Single carrier systems are inherently rugged against frequency selective fading because the fade will only affect a small portion of the bandwidth in which the signal energy is being received.

Multi-carrier systems can be designed to include a “guard interval” which will allow intersymbol interference (due to multipath reception) to be almost eliminated over a wide range of multipath delay durations.

There are two important cases of the use of guard intervals to reduce intersymbol interference in multipath situations. First, where multipath occurs as a result of reflections or inhomogenities in the transmission media. In this case relatively short multipath delays for example, of up about 50 μs might be encountered. Secondly, if on-channel active repeaters are used as part of a single frequency network (SFN) concept, longer multipath delays may be encountered. (The duration of SFN multipath delays will depend on transmitter spacing.)

The disadvantage of using long guard intervals (which may be required when actual existing transmitter network location requirements are considered) is that, for a fixed overall symbol duration, an increased guard interval will reduce data throughput in proportion to the ratio of guard interval to overall symbol duration. To avoid loss of throughput the size of the FFT used in the MCM system must be increased. This will result in a longer overall symbol duration and an increased number of more closely spaced carriers within the channel. Increasing the FFT size requires the use of processing chips (either DSPs or pipeline processors) that are faster and have greater memory capacity. However from the viewpoint of FFT requirements, implementations with up to 8 000 carriers are within the range of current technology. A more stringent requirement, however, is the receiver phase noise requirements implied by systems with a very large number of carriers. It has been reported that current consumer receiver technologies can provide satisfactory operation of systems with upto 8 000 carriers. The design of multi-carrier systems must also consider the effects of frequency selective fading. Even where guard intervals are used to overcome intersymbol interference, in-band fading can still exist which may cause severe amplitude and/or phase distortion to high order QAM signals. For example, if a very strong (0 dB) echo is present on an **uncoded** OFDM system it can increase the power of 2/3 of the OFDM carriers while decreasing the power of the remainder. However the effect of the carriers suffering a decrease in power outweighs the positive effect of those having the increase and an overall BER in the vicinity of 10^{-1} would be obtained even though system *C*/*N* was 12 dB or more. However the situation changes dramatically for a **coded** multi-carrier system. If the frequency response of the channel can be measured (for example by using a training sequence) it is possible to effectively assign a signal-to-noise ratio to each OFDM carrier. This channel state information can be communicated to the error correction system, where it can be used to dramatically improve the system performance in the presence of echoes.

The system is most easily implemented using convolutional codes and a soft decision Viterbi decoder.

As an example of the improvement possible, an uncoded system was considered to fail (Viterbi decoder suffering a BER of 10^{-4}) in the presence of a –4.5 dB echo but with the addition of rate ¾ convolutional coding ($k = 7$) with channel state estimation, the system was able to operate at an echo level of 0 dB. Research into optimum QAM-OFDM codes is ongoing. Areas of study include the appropriate code rates, and determination of appropriate interleaving factors. One of the distinct advantages of MCM over SCM with an adaptive equalizer is that MCM is less sensitive to variations in delay, as long as the multipath falls within the guard interval and the interleaver can effectively decorrelate the faded signal. Adaptive equalization performs better on short delay multipaths and is less effective on long delay multipaths. Therefore, MCM may be a better candidate for single-frequency networks (SFN).

5.5.3 Co-channel interference from analogue TV

Single carrier systems are robust to tone interference since signal power is spread over the entire spectrum.

For a single carrier system, an adaptive equalization can be used to reduce the severity of co-channel analogue television interference.

Another approach, for single carrier systems, is to use comb filtering to create notches in the spectrum at the receiver which align with the frequencies of the unwanted interfering carriers.

Multi-carrier systems can be sensitive to co-channel interference because of the very low power in each carrier. An MCM system is especially vulnerable to the non-flat spectrum of co-channel analogue TV as carriers located near the luminance, chrominance and audio carrier frequencies may suffer from strong interference.

One approach to avoiding this problem is to delete from the multi-carrier ensemble those carriers likely to suffer interference. However the disadvantage of this approach is that the data carrying capacity of the deleted carriers is lost at all points in the DTTB coverage area, even those locations where co-channel or adjacent channel interference would not have been a problem. This approach should perhaps not be rejected out-of-hand, particularly for difficult co-channel cases, as careful selection of a small number of carriers (principally around the interfering vision carrier) for deletion might produce a benefit of up-to (about) 10 dB with only a small proportion of data loss.

A second approach that avoids this disadvantage is to apply error coding to the multi-carrier system. As with the case of coding to improve the performance of the multi-carrier system in the presence of multipath, it is necessary to estimate the state of the channel – the amount of interference on each carrier. One way of achieving this is to switch the OFDM off for short periods and measure the interference power. An interleaver and channel estimator combined with a soft-decision decoding algorithm could be used to combat co-channel analogue TV interference. Using this technique on a real OFDM system it has been reported that in an extensive field trial, protection ratios of better than 0 dB were easily achievable. It is also worth noting that in locations where there is no co-channel or adjacent channel interferer, the error coding provides a residual level of error-correction capability which will improve the system's resilience against other interference.

5.5.4 Peak and average power ratio issues

Both single and multi-carrier modulations have an essentially noise-like spectrum. For single carrier modulation, the peak-to-average power ratio depends on the filter roll off. Faster roll off (which will have higher spectral efficiency) will result in a higher peak to average power ratio. It has been reported that for 99.99% of the time, the peak-to-average power ratio of a simulated 8-VSB single carrier system is 6.9 dB or less. (Lower values may be obtained if peak limiting is applied but in this case an increased level of adjacent channel energy will be produced which may well require additional transmitter filtering.) Some ATV systems may be able to take advantage of the asymmetric shape of analogue television receiver input filters to transmit more power or to implement a pilot carrier (which would improve the system ruggedness at low C/N) without increasing co-channel interference.

It is also noted that multi-carrier systems with a flat spectrum and a large number of carriers can be modelled as Gaussian distributions. Table 5.2 provides measured data on the peak to average power ratios of a typical COFDM signal.

TABLE 5.2

Peak to average ratio measurements

Peak to average ratio (%)	(dB)
99	6.5
99.5	7.0
99.9	8.2
99.99	9.5
99.999	10.3

Desired signal level : –10 dBm

If clipped to the 95% value an E_s/N_0 penalty of less than 0.25 dB applies for BERs of 10^{-3}. However the effects on adjacent channel filtering requirements need further consideration. If spectrum shaping is used in the multi-carrier system, opening holes in the spectrum, a few more dB gain may be achieved. But of course this will reduce the effective data rate of the multi-carrier system.

5.6 Coverage issues

An issue which may be of concern to system designers in implementing a modulation system for a DTTB service is the possibility of a sudden transition between "perfect service" and "no service" over a very small range of received signal variation. Furthermore this small variation might vary with time of day, propagation conditions, season of the year or other more difficult to predict factors such as aircraft or vehicle flutter or receiving antenna movement in the wind. There are a number of possible approaches to deal with this matter.

5.6.1 Hierarchical transmission

Most of the DTTB systems demonstrated so far use non-hierarchical modulation systems designed for fixed reception. They all have a sharp threshold effect in the fringe of the coverage area. From an information theory point of view, the DTTB channel differs from point-to-point communication in that the channel capacities varies with receiver location. The further away the receiver is from the transmitter, the lower the channel capacity. The design of a hierarchical system may improve service to the fringe area. For receivers closer to the transmitter, the channel capacities are not fully exploited in a non-hierarchical system. Hierarchical modulation systems are under study as one possible approach to this problem.

It is thought in some quarters that a multi-resolution coding system may be advantageous for DTTB in as much as it may be able to provide a DTTB performance which degrades gradually as received signal levels are reduced. While the aim is generally supported in principle, it has been argued that, with current source coding, a higher aggregate data rate is needed to achieve this added functionality and that this is undesirable because it increases receiver complexity and may require use of a higher spectral efficiency modulation system (which will have poorer noise performance). This topic is still being studied but here we wish to consider the topic from the viewpoint of its implications for selection of either a single or multi-carrier approach.

The main issue relates to channel data capacity.

For multi-carrier systems, a layered modulation system can be achieved by one or more of the following approaches:

– assigning groups of carriers to different coding layers so that the lower layer(s) have a greater level of error-correction than the upper layer(s);

– assigning groups of carriers to different coding layers and using more rugged modulation formats (e.g. QPSK) for carriers assigned to the lower layer(s) and less rugged codes (e.g. 64-QAM) for carriers assigned to the upper layer(s);

– multi-resolution coding where for the lower coding layers groups of states of the modulation constellation are considered as a single modulation state (for example four states of a 64-QAM multi-resolution modulator might be treated as a single state of a lower resolution 16-QAM decoder).

Other approaches to multi-layered modulation systems may also be possible.

For SCM systems using QAM, layered transmission can be achieved by using non-equally spaced constellation modulation and different channel coding.

In a single carrier VSB modulation system, layered modulation might be achieved, at some reduction in total data capacity, by transmitting a mixture of 4-VSB and 8-VSB symbols in a time division multiplex.

5.6.2 Multi-transmitter systems

A further approach is to use channel repeaters to extend or fill in the coverage in areas where the “perfect service”/“no service” transition occurs. In a DTTB system it might be possible to add repeaters without requiring the use of new transmission frequencies. This is the single frequency network (SFN) concept. In that case, the co-channel signal from the parent transmitter is processed as if it were co-channel interference. Provided the delay is within the system guard interval a seamless transition in coverage can be achieved.

Both SCM systems using adaptive equalizers and MCM systems using guard intervals could support SFNs.

In both cases, the practicality of implementing such SFNs will depend on the levels of wanted to unwanted multipath signals that the receiving equipment can cancel.

General comment

To summarize, SCM and MCM are two promising modulation techniques offering comparable performances on a Gaussian noise channel. The better peak-to-average ratio of SCM may reduce the required transmitter output back-off. Channel coding is used to reduce vulnerability to a wide range of impairments. MCM is less sensitive to variation in multipath delay (within the guard interval) and may be a better candidate for single frequency operation.

As discussed above, single-carrier and multi-carrier techniques under consideration in various countries provide a comparable performance in many areas and also some particular advantages and disadvantages. It may then be possible to use either of these techniques to create a common standard which will provide different data rates for the various bandwidths available.

CHAPTER 6

SYSTEMS OVERVIEW

6.1 The ATSC system

The ATSC system was specifically designed to permit an additional digital transmitter to be added to each existing NTSC transmitter in the United States of America, with comparable coverage and minimum disturbance to the existing NTSC service in terms of both area and population coverage. This capability is met and even exceeded.

The system is quite efficient and capable of operating under varying conditions, i. e. clear channel availability or, as implemented in the US, constrained to fit 1 600 additional channels into an already crowded spectrum, and reception with roof-top or portable antennae.

The system was also designed to be immune to multipath and to offer spectrum efficiency and ease of frequency planning.

Signals conforming to the ATSC system can travel on cables and the United States cable industry is just beginning its conversion to digital. The ATSC 16-VSB mode is suited for cable since it can double the capacity on this delivery media. The ATSC has been tested and proven to work reliably over satellite at the same or higher bit rates.

As recalled in the introduction section of this Handbook, the ATSC system was designed to permit an additional digital transmitter to be added to each existing NTSC transmitter in the United States of America with comparable coverage and minimum disturbance to the existing NTSC service in terms of both area and population coverage. Variations can be achieved in the programme formats as mentioned in the relevant section (SD or HD), and there is a great potential for data-based services utilizing the opportunistic data transmission capability of the system. The system can accommodate fixed (or possibly portable) reception with no resultant loss in payload.

6.2 The DVB-T system

The DVB-T system was essentially designed with built-in flexibility, in order to be able to adapt to all channels: it is capable of coping not only with clear channel but with interleaved planning, and even co-channel operation for the dame programme by different transmitters (single-frequency networks).

It also permits service flexibility, with the possibility of reception by roof-top antennae and also, if desired, of portable reception. Mobile reception is possible for QPSK and also for higher modulation orders, proven by extensive laboratory measurements and field trials under different channel conditions.

The system was also designed to be robust against interference from delayed signals, either echoes from terrain or buildings or signals from distant transmitters in a single frequency network, a new tool which it brings to TV service planning to improve spectrum efficiency which is necessary in the case of particularly crowded spectrum as it is the case in Europe.

The DVB-T compliant signals can also be carried over cables. However, the DVB-T specification is part of a family of specifications covering also satellite (DVB-S) and cable (DVB-C) operation. All use MPEG-2 coding for video and audio and MPEG-2 type of multiplexing. They have common features in the error protection strategy to be used. The main difference is the modulation method which is specific to the relevant bearer (satellite, cable or terrestrial). The available data capacity is also different, as higher bit rates are offered on cable and satellite. However, transferring programmes from one bearer to another is possible provided that the bit rate is available.

The DVB-T system features a number of selectable parameters, which allows it to accommodate a large range of carrier to noise ratio and channel behaviour, allowing fixed, portable, or mobile reception, with a trade-off in the usable bit rate. Table 6.1 summarizes the system possibilities. The range of parameters allows the broadcasters to select a mode appropriate to the application foreseen. For instance, a very robust mode (with correspondingly lower payload) is needed to ensure portable reception. A moderately robust mode with a higher payload could be used where the service planning uses interleaved channels. The less robust modes with the highest payloads can be used if a clear channel is available for digital TV broadcasting.

This highlights the DVB-T specific flexibility, which allows the user to tailor the system by using the most appropriate mode among the different possible modes of operation proposed.

Comprehensive discussion of the optimum use of all parameters is complex and would be lengthy. However, the following features should be kept in mind:

– the hierarchical modes when applicable split the channel in two with different (and adjustable) requirements in terms of *C*/*N*. This permits different reception conditions for the same or for different programme content;

– the code rate and the modulation scheme can be selected in order to lower down the *C*/*N* requirements to the desired form of service;

– the selection of the 2k mode instead of 8k makes mobile reception easier. However, it only permits the implementation of small single frequency networks of transmitters (SFN).

Examples of such services not using hierarchical modes are given in Table 6.1.

TABLE 6.1

Examples of DVB-T parameter use for various services

Bit rate (Mbit/s)	**Modulation**	**Code rate**	**Application**
5	QPSK	1/2	Channel featuring a high level of interference
15	16-QAM	2/3	Wide area portable reception
26	64-QAM	3/4	Maximize data rate in a clear channel

6.3 The ISDB-T system

ISDB (integrated services digital broadcasting) is a new type of broadcasting for multimedia services. It integrates systematically various kinds of digital contents, each of which may include multi-program video from LDTV to HDTV, multi-program audio, graphics, text, and so on. Most of the digital contents are, nowadays, encoded into form of MPEG-2 Transport Stream and delivered worldwide. It is highly desired to integrate the digital contents on the MPEG-TS basis.

Since the ISDB contains a variety of services, the system has to cover a wide range of requirements that may differ from one service to another. For example, a large transmission capacity is required for HDTV service, while a high service availability (or transmission reliability) is required for data services such as key delivery of conditional access, downloading of software, and so on. To integrate these signals of different service requirements, it is desired for the transmission systems to provide a series of modulation and/or error protection schemes which can be selected and combined flexibly in order to meet each requirement of services integrated.

The ISDB-T (ISDB-Terrestrial) systems has been designed to have enough flexibility to send not only television or sound programmes as digital signals but also offer multimedia services in which a variety of digital information such as video, sound, text and computer programmes will be integrated. It aims to make use of advantages provided by terrestrial radio waves so that stable reception can be provided by compact, light and inexpensive mobile receivers in addition to integrated receivers used at home by using segmented OFDM scheme.

The ISDB-T provides common elements in operation and reception between digital satellite broadcasting and communications by using MPEG-2 coding and systems in a multiplexing process. ISDB-T also provides flexible multi-program editing for different receiving conditions by hierarchical transmission in a transmission channel, which is composed of orthogonal frequency division multiplexing (OFDM)-segments in which transmission parameters can be independent of each other.

As ISDB-T system uses segmented OFDM scheme for modulation, a Transport Stream is to be re-multiplexed and arranged into data groups (data segments) prior to OFDM framing. After channel coding, data segments are formed into OFDM segments. Each segment has a bandwidth $B/14$ MHz (B means bandwidth of TV terrestrial channel: 6, 7 or 8 MHz depending the region, so one segment occupies bandwidth 6/14 MHz (~428.57 kHz), 7/14 MHz (~500 kHz) or 8/14 MHz (~571.29 kHz). The pilot signals are added to each segment and serve to transmission and multiplexing configuration control (TMCC). The TMCC carriers (added pilots) are used for the purpose of signalling parameters related to the transmission scheme, i.e. to channel coding, modulation and hierarchical condition.

Thanks to segmentation and addition of the pilot signals, each segment can have its individual error protection scheme and/or type of modulation (DQPSK, QPSK, 16-QAM or 64-QAM). Each segment can then meet requirements of service integrated, and a number of segments may be combined flexibly to integrate a wide-band service (HDTV for example).

6.3.1 Transmission bandwidths of ISDB-T

The ISDB-T signal is composed of 13 OFDM segments and has a bandwidth of $B \times 13/14$ MHz (~5.57 MHz for 6 MHz terrestrial channel, ~6.5 MHz for 7 MHz terrestrial channel, and ~7.4 MHz for 8 MHz terrestrial channel).

6.3.2 Hierarchical transmission

Terrestrial ISDB provides hierarchical transmission features. This enables part of the band to be allocated to signals for stationary reception and the rest to signals for mobile reception, which means that audio and data broadcasts for automobile and portable receivers can be performed simultaneously with television broadcasts for home use.

In ISDB-T, the transmission parameters of the modulation scheme of OFDM carriers, the coding rates of inner code, and the length of the time interleaving can be independently specified for each data segment. Hierarchical transmission of ISDB-T is achieved by transmitting OFDM segment groups having different transmission parameters in a channel. The maximum of three layers (three different segment-groups) can be transmitted in a channel at the same time.

It should be noted that partial reception is regarded as one hierarchical layer.

6.3.3 Partial reception

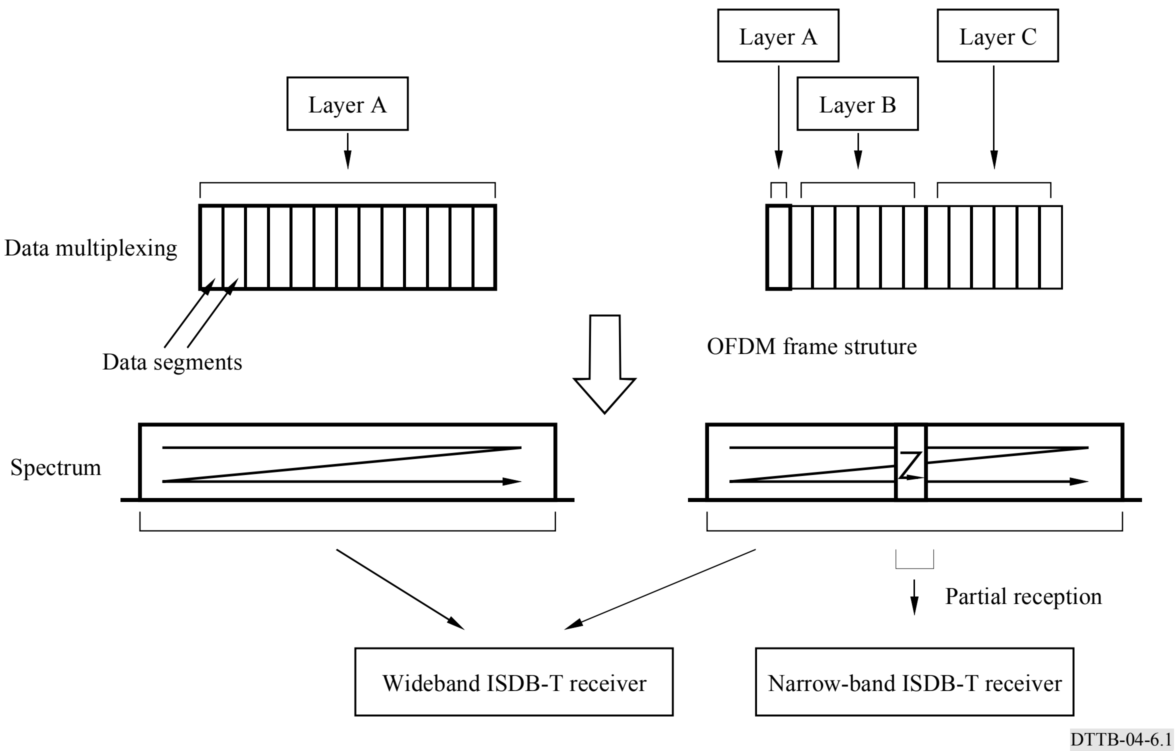
By limiting the range of frequency interleaving within a segment itself, it is possible to separate the segment independently from the remaining segments in the transmitted signal. In such a way, partial reception of services contained in a transmission channel can be obtained using a narrow-band receiver that has a bandwidth of one OFDM segment.

It should be noted that one segment is dedicated to partial reception and its position is the central segment of 13 OFDM segments.

Figure 6.1 shows an example of hierarchical transmission and partial reception.

FIGURE 6.1

Example diagram of hierarchical transmission and partial reception



6.3.4 Multiplex for Hierarchical Transmission

Multiplexing in the ISDB-T system is in accordance with ISO/IEC 13818-1 (MPEG-2 systems). For the hierarchical multiplexing in ISDB-T, in principle, a single transport stream (TS: defined in MPEG-2 systems) is transmitted in a transmission channel, whether a hierarchical transmission is in operation or not. For this reason, division and synthesis of the TS is necessary and this process is performed at both the transmission and reception sides.

It should be noted that because a signal for partial reception is part of a whole signal in a channel, part of a TS is received in partial reception.

6.3.5 Functional Block Diagram of the ISDB-T

Functional block diagram of the ISDB-T is shown in Fig. 6.2.

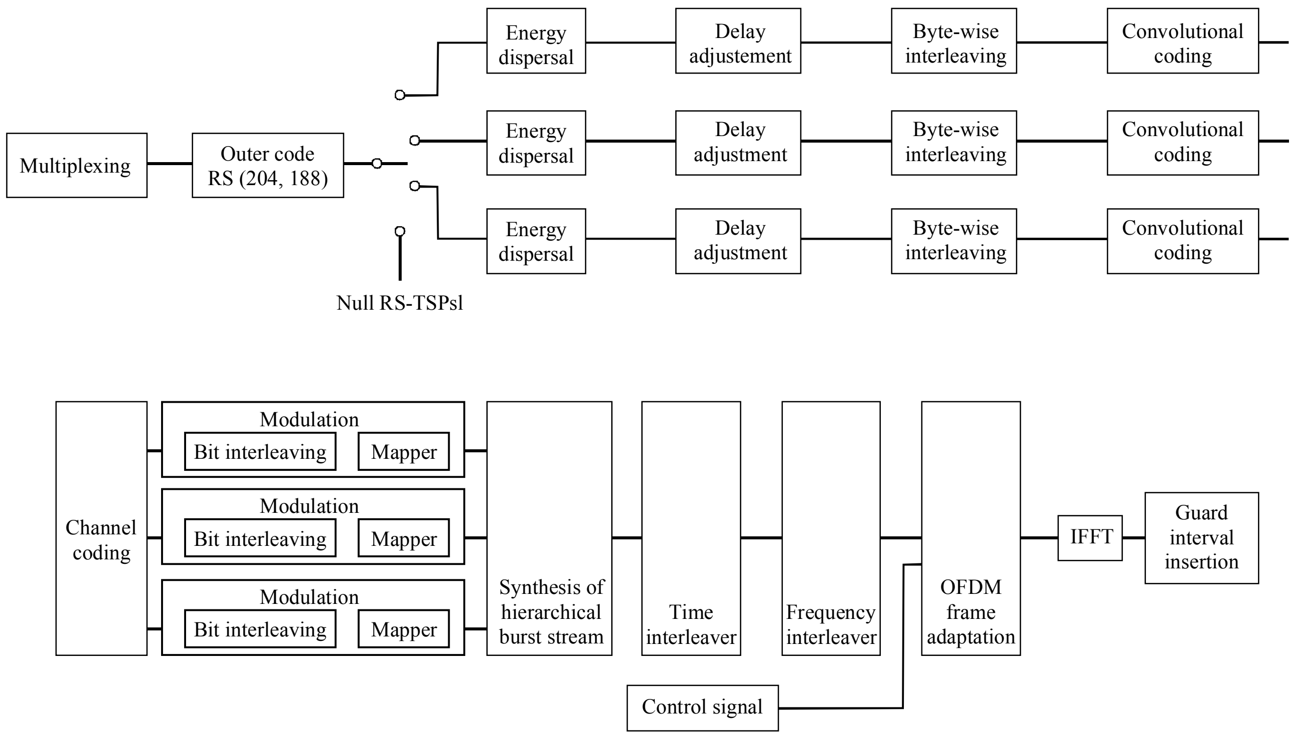


FIGURE 6.2

Functional block diagram of the ISDB-T

DTTB-06.2

6.3.6 Transmission parameters

6.3.6.1 ISDB-T for 6 MHz

TABLE 6.2

Transmission parameters for ISDB-T (6 MHz)

Mode		Mode 1	Mode 2	Mode 3
Number of segments (N_s)		13		
Bandwidth		3 000/7 (kHz) × N_s + 250/63 (kHz) = 5.575…MHz	3 000/7 (kHz) × N_s + 125/63 (kHz) = 5.573…MHz	3 000/7 (kHz) × N_s + 125/126 (kHz) = 5.572…MHz
Number of segments for differential modulation		n_d		
Number of segments for synchronous modulation		n_s ($n_s + n_d = N_s$)		
Carrier spacing		250/63 = 3.968…kHz	125/63 = 1.984…kHz	125/126 = 0.992…kHz
Number of carriers	Total	$108 \times N_s + 1 = 1\,405$	$216 \times N_s + 1 = 2\,809$	$432 \times N_s + 1 = 5\,617$
	Data	$96 \times N_s = 1\,248$	$192 \times N_s = 2\,496$	$384 \times N_s = 4\,992$
	SP(1)	$9 \times n_s$	$18 \times n_s$	$36 \times n_s$
	CP(1), (2)	$n_d + 1$	$n_d + 1$	$n_d + 1$
	TMCC(3)	$n_s + 5 \times n_d$	$2 \times n_s + 10 \times n_d$	$4 \times n_s + 20 \times n_d$
	AC1(4)	$2 \times N_s = 26$	$4 \times N_s = 52$	$8 \times N_s = 104$
	AC2(4)	$4 \times n_d$	$9 \times n_d$	$19 \times n_d$
Carrier modulation		QPSK, 16-QAM, 64-QAM, DQPSK		
Number of symbols per frame		204		
Effective symbol duration		252 μs	504 μs	1.008 μs
Guard interval		63 μs (1/4), 31.5 μs (1/8), 15.75 μs (1/16), 7.875 μs (1/32)	126 μs (1/4), 63 μs (1/8), 31.5 μs (1/16), 15.75 μs (1/32)	252 μs (1/4), 126 μs (1/8), 63 μs (1/16), 31.5 μs (1/32)
Frame duration		64.26 ms (1/4), 57.834 ms (1/8), 54.621 ms (1/16), 53.0145 ms (1/32)	128.52 ms (1/4), 115.668 ms (1/8), 109.242 ms (1/16), 106.029 ms (1/32)	257.04 ms (1/4), 231.336 ms (1/8), 218.464 ms (1/16), 212.058 ms (1/32)
Inner code		Convolutional code (1/2, 2/3, 3/4, 5/6, 7/8)		
Outer code		RS (204,188)		

(1) SP (scattered pilot), and CP (continual pilot) can be used for frequency synchronization and channel estimation.

(2) The number of CPs includes CPs on all segments and a CP for higher edge of whole bandwidth.

(3) TMCC (transmission and multiplexing configuration control) carries information on transmission parameters.

(4) AC (auxiliary channel) carries ancillary information for network operation.

TABLE 6.3

Information rates*

Carrier modu-lation	Convolu-tional code	Number of transmitting TSPs(1) (Mode 1/2/3)	Information rates (kbit/s)			
			Guard interval ratio 1/4	Guard interval ratio 1/8	Guard interval ratio 1/16	Guard interval ratio 1/32
DQPSK	1/2	156/312/624	3.651	4.056	4.295	4.425
	2/3	208/216/832	4.868	5.409	5.727	5.900
	3/4	234/468/936	5.476	6.085	6.443	6.638
QPSK	5/6	260/520/1040	6.085	6.761	7.159	7.376
	7/8	273/546/1092	6.389	7.099	7.517	7.744
16-QAM	1/2	312/624/1248	7.302	8.113	8.590	8.851
	2/3	416/832/1664	9.736	10.818	11.454	11.801
	3/4	468/936/1872	10.953	12.170	12.886	13.276
	5/6	520/1040/2080	12.170	13.522	14.318	14.752
	7/8	546/1092/2184	12.779	14.198	15.034	15.489
64-QAM	1/2	468/936/1872	10.953	12.170	12.886	13.276
	2/3	624/1248/2496	14.604	16.227	17.181	17.702
	3/4	702/1404/2808	16.430	18.255	19.329	19.915
	5/6	780/1560/3120	18.255	20.284	21.477	22.128
	7/8	819/1638/3276	19.168	21.298	22.551	23.234

* In the case of hierarchical transmission, information rate can be calculated by the combination of segment information rates.

(1) TSP: Transport Stream Packet, which contains 188 bytes and is defined in MPEG-2 Systems.

6.3.6.2 ISDB-T for 7 MHz

TABLE 6.4

Transmission parameters for ISDB-T (7 MHz)

Mode		**Mode 1**	**Mode 2**	**Mode 3**
Number of segments (N_s)		13		
Bandwidth		7 000/14 (kHz) × N_s + 500/108 (kHz) = 6.504… MHz	7 000/14 (kHz) × N_s + 500/216 (kHz) = 6.502… MHz	7 000/14 (kHz) × N_s + 500/432 (kHz) = 6.501… MHz
Number of segments for differential modulation		n_d		
Number of segments for synchronous modulation		n_s ($n_s + n_d = N_s$)		
Carrier spacing		500/108 = 4.629… kHz	500/216 = 2.3148… kHz	500/432 = 1.157… kHz
Number of carriers	Total	$108 \times N_s + 1 = 1405$	$216 \times N_s + 1 = 2809$	$432 \times N_s + 1 = 5617$
	Data	$96 \times N_s = 1248$	$192 \times N_s = 2496$	$384 \times N_s = 4992$
	SP(1)	$9 \times n_s$	$18 \times n_s$	$36 \times n_s$
	CP(1), (2)	$n_d + 1$	$n_d + 1$	$n_d + 1$
	TMCC(3)	$n_s + 5 \times n_d$	$2 \times n_s + 10 \times n_d$	$4 \times n_s + 20 \times n_d$
	AC1(4)	$2 \times N_s = 26$	$4 \times N_s = 52$	$8 \times N_s = 104$
	AC2(4)	$4 \times n_d$	$9 \times n_d$	$19 \times n_d$
Carrier modulation		QPSK, 16-QAM, 64-QAM, DQPSK		
Number of symbols per frame		204		
Effective symbol duration		216 μs	432 μs	864 μs
Guard interval		54 μs (1/4), 27 μs (1/8), 13.5 μs (1/16), 6.75 μs (1/32)	108 μs (1/4), 54 μs (1/8), 27 μs (1/16), 13.5 μs (1/32)	216 μs (1/4), 108 μs (1/8), 54 μs (1/16), 27 μs (1/32)
Frame duration		55.08 ms (1/4), 49.572 ms (1/8), 46.818 ms (1/16), 45.441 ms (1/32)	110.16 ms (1/4), 99.144 ms (1/8), 93.636 ms (1/16), 90.882 ms (1/32)	220.32 ms (1/4), 198.288 ms (1/8), 187.272 ms (1/16), 191.764 ms (1/32)
Inner code		Convolutional code (1/2, 2/3, 3/4, 5/6, 7/8)		
Outer code		RS (204,188)		

(1) SP (scattered pilot), and CP (continual pilot) can be used for frequency synchronization and channel estimation.

(2) The number of CPs includes CPs on all segments and a CP for higher edge of whole bandwidth.

(3) TMCC (transmission and multiplexing configuration control) carries information on transmission parameters.

(4) AC (auxiliary channel) carries ancillary information for network operation.

TABLE 6.5

Information rates*

Carrier modu-lation	Convolu-tional code	Number of transmitting TSPs[(1)] (Mode 1/2/3)	Information rates (kbit/s)			
			Guard interval ratio 1/4	Guard interval ratio 1/8	Guard interval ratio 1/16	Guard interval ratio 1/32
DQPSK QPSK	1/2	156/312/624	4.259	4.732	5.011	5.163
	2/3	208/216/832	5.679	6.310	6.681	6.884
	3/4	234/468/936	6.389	7.099	7.517	7.744
	5/6	260/520/1040	7.099	7.888	8.352	8.605
	7/8	273/546/1092	7.454	8.282	8.769	9.035
16-QAM	1/2	312/624/1248	8.519	9.465	10.022	10.326
	2/3	416/832/1664	11.359	12.621	13.363	13.768
	3/4	468/936/1872	12.779	14.198	15.034	15.489
	5/6	520/1040/2080	14.198	15.776	16.704	17.210
	7/8	546/1092/2184	14.908	16.565	17.539	18.071
64-QAM	1/2	468/936/1872	12.779	14.198	15.034	15.489
	2/3	624/1248/2496	17.038	18.931	20.045	20.653
	3/4	702/1404/2808	19.168	21.298	22.551	23.234
	5/6	780/1560/3120	21.298	23.664	25.057	25.816
	7/8	819/1638/3276	22.363	24.848	26.309	27.107

* In the case of hierarchical transmission, information rate can be calculated by the combination of segment information rates.

(1) TSP: Transport Stream Packet, which contains 188 bytes and is defined in MPEG-2 Systems.

6.3.6.3 ISDB-T for 8 MHz

TABLE 6.6

Transmission parameters for ISDB-T (8 MHz)

Mode		**Mode 1**	**Mode 2**	**Mode 3**
Number of segments (N_s)		13		
Bandwidth		8 000/14 (kHz) × N_s + 1000/189 (kHz) = 7.433… MHz	8 000/14 (kHz) × N_s + 500/189 (kHz) = 7.431… MHz	8 000/14 (kHz) × N_s + 250/189 (kHz) = 7.429… MHz
Number of segments for differential modulation		n_d		
Number of segments for synchronous modulation		n_s ($n_s + n_d = N_s$)		
Carrier spacing		1 000/189 = 5.291… kHz	500/189 = 2.645… kHz	250/189 = 1.322… kHz
Number of Carriers	Total	$108 \times N_s + 1 = 1\,405$	$216 \times N_s + 1 = 2\,809$	$432 \times N_s + 1 = 5\,617$
	Data	$96 \times N_s = 1\,248$	$192 \times N_s = 296$	$384 \times N_s = 4\,992$
	SP(1)	$9 \times n_s$	$18 \times n_s$	$36 \times n_s$
	CP(1), (2)	$n_d + 1$	$n_d + 1$	$n_d + 1$
	TMCC(3)	$n_s + 5 \times n_d$	$2 \times n_s + 10 \times n_d$	$4 \times n_s + 20 \times n_d$
	AC1(4)	$2 \times N_s = 26$	$4 \times N_s = 52$	$8 \times N_s = 104$
	AC2(4)	$4 \times n_d$	$9 \times n_d$	$19 \times n_d$
Carrier modulation		QPSK, 16-QAM, 64-QAM, DQPSK		
Number of symbols per frame		204		
Effective symbol duration		189 µs	378 µs	756 µs
Guard interval		47.25 µs (1/4), 23.625 µs (1/8), 11.8125 µs (1/16), 5.90625 µs (1/32)	94.5 µs (1/4), 47.25 µs (1/8), 23.625 µs (1/16), 11.8125 µs (1/32)	189 µs (1/4), 94.5 µs (1/8), 47.25 µs (1/16), 23.625 µs (1/32)
Frame duration		48.195 ms (1/4), 43.3755 ms (1/8), 40.96575 ms(1/16), 39.760875 ms(1/32)	96.39 ms (1/4), 86.751 ms (1/8), 81.9315 ms (1/16), 79.52175 ms (1/32)	192.78 ms (1/4), 173.502 ms (1/8), 163.863 ms (1/16), 159.0435 ms (1/32)
Inner code		Convolutional code (1/2, 2/3, 3/4, 5/6, 7/8)		
Outer code		RS (204,188)		

(1) SP (scattered pilot), and CP (continual pilot) can be used for frequency synchronization and channel estimation.

(2) The number of CPs includes CPs on all segments and a CP for higher edge of whole bandwidth.

(3) TMCC (transmission and multiplexing configuration control) carries information on transmission parameters.

(4) AC (auxiliary channel) carries ancillary information for network operation.

TABLE 6.7

Information rates*

Carrier modu-lation	Convolu-tional code	Number of transmitting TSPs(1) (Mode 1/2/3)	Information rates (kbit/s)			
			Guard interval ratio 1/4	Guard interval ratio 1/8	Guard interval ratio 1/16	Guard interval ratio 1/32
DQPSK QPSK	1/2	156/312/624	4.868	5.409	5.727	5.900
	2/3	208/216/832	6.490	7.212	7.636	7.867
	3/4	234/468/936	7.302	8.113	8.590	8.851
	5/6	260/520/1040	8.113	9.015	9.545	9.834
	7/8	273/546/1092	8.519	9.465	10.022	10.326
16-QAM	1/2	312/624/1248	9.736	10.818	11.454	11.801
	2/3	416/832/1664	12.981	14.424	15.272	15.735
	3/4	468/936/1872	14.604	16.227	17.181	17.702
	5/6	520/1040/2080	16.227	18.030	19.091	19.669
	7/8	546/1092/2184	17.038	18.931	20.045	20.653
64-QAM	1/2	468/936/1872	14.604	16.227	17.181	17.702
	2/3	624/1248/2496	19.472	21.636	22.909	23.603
	3/4	702/1404/2808	21.907	24.341	25.772	26.553
	5/6	780/1560/3120	24.341	27.045	28.636	29.504
	7/8	819/1638/3276	25.558	28.397	30.068	30.979

* In the case of hierarchical transmission, information rate can be calculated by the combination of segment information rates.

(1) TSP: Transport Stream Packet, which contains 188 bytes and is defined in MPEG-2 Systems.

CHAPTER 7

LIST OF ITU-R RECOMMENDATIONS RELATING TO DIGITAL TERRESTRIAL TELEVISION BROADCASTING (DTTB)

Recommendation ITU-R BT.798:	Digital terrestrial television broadcasting in the VHF/UHF bands.
Recommendation ITU-R BT.1206:	Spectrum shaping limits for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1207:	Data access methods for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1208:	Video coding for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1209:	Service multiplex methods for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1299:	The basic elements of a worldwide common family of systems for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1300:	Service multiplex, transport, and identification methods for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1301:	Data services in digital terrestrial television broadcasting.
Recommendation ITU-R BT.1306:	Error-corrections, data framing, modulation and emissions methods for digital terrestrial television broadcasting.
Recommendation ITU-R BT.1125:	Basic objectives for the planning and implementation of digital terrestrial television broadcasting.
Recommendation ITU-R BT.1368:	Planning criteria for digital terrestrial television services in the VHF/UHF television bands.

PART 2

PLANNING PART

PART 2

TABLE OF CONTENTS

Page

CHAPTER 1

INTRODUCTION

This part of the digital terrestrial television (DTTB) Handbook deals with planning considerations. It is complicated by the fact that different parts of the world use different frequency rasters and many different analogue television systems which require different treatment if digital television is to be introduced without disrupting the millions of existing viewers. A large part of the information given relates to Region 1 because most of the complications created by the use of multiple analogue television systems occur in this Region.

This part of the Handbook is intended to provide both factual information and rather general guidance, the latter being based on the experience gained in the implementation of analogue television services. Even though the primary target is to provide guidance for the introduction of digital television, many of the lessons learned when implementing analogue television remain valid and can be re-used with suitable adjustments.

CHAPTER 2

GENERAL INFORMATION AND OVERVIEW

The potential advantages of digital terrestrial television broadcasting (DTTB), in terms of service quality, lower costs and programme diversity, are summarized in the introduction of this handbook. In frequency planning terms, where new or unused frequency spectrum is available, digital television coverage from individual transmitters or from networks of transmitters can be planned to achieve the full potential of DTTB. This produces considerable benefits (compared to the current analogue situation), in terms of service provision and spectrum utilization. However, the spectrum situation is far from ideal, and there are numerous problems involved in finding the required spectrum, and in dealing with its allocation and assignment, that will have to be overcome before DTTB services can become a reality in many parts of the world.

The *allocation* of frequency spectrum to specific services on a regional or worldwide basis is subject to international treaty drawn up under the auspices of the ITU.

The *assignment* of allocated spectrum to particular uses is subject to regional treaty and cross-border negotiation, as well as to regulation on a national basis.

In Region 1, for example, the Stockholm Plan of 1961, (based upon the use of analogue television standards) has provided the framework for planning and implementing the extensive terrestrial television networks now in operation. Treaty arrangements, such as those in Region 2, are used to govern the planning and procedures of frequency assignment elsewhere. Within these regional plans there are many geographic areas where the allocated spectrum has been heavily exploited to provide the maximum number of analogue television services, each service being designed to achieve, where possible, a high population coverage. For these areas then there is little prospect that sufficient spectrum can be found for dedication to DTTB, let alone that sufficient could be found for all the DTTB services that might be required. Thus the alternative option of band sharing with the existing analogue services is being intensively studied, accepting that the DTTB transmitter power constraints that this arrangement necessarily imposes will in turn inevitably limit the performance of the DTTB system. There are other geographic areas covered by these regional plans where the allocated spectrum is not heavily exploited and where it will be feasible to consider the use of relatively high DTTB transmitter powers to achieve increased performance in terms of service quality levels or ruggedness of transmission.

It can be seen then that the constraints that apply to “frequency planning” will vary from country to country, as well as, in some cases, within national boundaries – the degree of variation being dependent on geographic/population factors as well as upon the exploitation of the national “allocations”. It is against this rather complex “frequency planning” background that strategies for the introduction and subsequent evolution of DTTB services are being considered. Central to these considerations is finding sensible “ways and means” of migrating from an initial phase of DTTB, in which limited capability DTTB services are introduced on a “sharing” basis, to a final phase of DTTB service that could allow the NTSC, PAL and SECAM services to be “phased-out”. If such a migration path can be found and followed, to the point where the “switch over” to all-digital operations had been achieved there would be the opportunity to upgrade the DTTB services to their full potential, possibly releasing some of the allocated spectrum for reallocation to other services or allowing for the introduction of new and innovative broadcasting services. Delivery of digital television and associated sound services within a single 6, 7, or 8 MHz VHF/UHF TV broadcast channel requires a number of separate technical disciplines and processes for spectrum planning which include:

– the understanding of spectrum and planning aspects of digital services including area coverage for different reception and environmental conditions; and

– the ability to provide a digital emission system in the terrestrial VHF/UHF bands allowing for possible simultaneous transmission with existing analogue television services.

2.1 Current analogue systems

The first television systems were developed independently in several parts of the world, and despite the substantial efforts towards standardisation that has been made since then, even today analogue terrestrial systems with several significantly different values of key parameters such as channel-width remain in widespread operation. Everywhere that systematic planning has been undertaken, however, it has been based on the principle that it should enable the scarce natural resource consisting of the spectrum to be exploited as fully as possible. Nevertheless, the spectrum available for terrestrial broadcasting, and the extent to which it is in fact exploited, also varies significantly from one part of the world to another. In some cases the latter is due to the high cost of operation; in others it reflects the availability of other distribution media such as cable and satellite services.

One of the most important constraints in planning for current analogue systems is the fact that the spacing between co-channel transmitters must be a significant multiple of the service radius of an individual transmitter. Furthermore, at the time when the planning criteria were established, the characteristics of consumer receivers were assumed to be such that certain other restrictions on the assignment of channels to other transmitters in the same and nearby areas had to be respected. Although consumer receiver performance has since improved significantly, these restrictions have, to a large extent, so far remained in effect. However, it must be noted that the extent to which these restrictions are regarded as mandatory varies considerably from one part of the world to another. For example, there is a commonly applied restriction which does not allow the use of two adjacent channels at the same transmitter site – this restriction may not be applied in situations where there is no other planning solution.

Furthermore, because the planning process has been designed to enable one of two main types of service to be provided, in most cases the actual configuration of transmitting stations over the territory tends to correspond to one of two characteristic types. One type is where the objective is to ensure that satisfactory reception of as many programme services as possible can be achieved virtually throughout a large territory; the other type is where the objective is to enable as many broadcasting companies as possible to compete with each other fairly in providing services within the area covered by a single high-power transmitting station located near the centre of a metropolitan area. In the former (wide-area-coverage) case, numerous low-power rebroadcast stations are also used, especially where the terrain is hilly; in the latter (local market coverage) there are few such low-power rebroadcast stations.

It should be noted that both types may coexist in the same area, because some types of programme service are inherently of mainly local interest, while others are suitable for distribution within a much larger area. Nevertheless, one such type is generally dominant in any particular case, and this has important consequences for the possibility of reorganising the usage of the spectrum, and thus for the potential introduction of digital broadcasting there. Specifically, with the wide-area coverage type there is generally much less vacant spectrum to be exploited to provide digital services.

2.2 Digital television systems

Three digital television systems have been developed for terrestrial broadcasting and details of these systems may be found in Part 1 of this Handbook. All systems use the MPEG-2 digital compression standards in the source coding and compression layer and the service multiplex and transport layer and thus have a high degree of commonality. The main difference between them is in the RF/transmission or physical layer where the type of modulation used and the RF emission mode is determined. The ATSC system, developed in North America, is a single carrier system using vestigial sideband modulation (8VSB). The DVB-T system, developed in Europe, and the BST-OFDM system, developed in Japan, are multi-carrier systems using coded orthogonal frequency division multiplex (COFDM) with QAM modulation. These modulation systems can be applied to either a single carrier modulated at a high data rate or to a large number of carriers modulated at relatively low rates – the multi-carrier approach. System parameters are scalable to permit the delivery of digital television services with a data rate of up to about 24 Mbit/s through channels with bandwidths of 6, 7 and 8 MHz.

An issue that may be of concern to system designers in implementing a modulation system for any DTTB service is the possibility of a sudden transition between "perfect service" and "no service" over a very small range of received signal variation. Furthermore this small variation might vary with time of day, propagation conditions, season of the year or other more difficult to predict factors such as aircraft or vehicle flutter or receiving antenna movement in the wind.

To summarize, single carrier and multi-carrier modulation systems are two promising digital television delivery techniques offering comparable performances in a Gaussian noise channel. The performance of both under the combined impairments of noise and co-channel analogue TV interference is also comparable. Channel coding is used to reduce vulnerability to a wide range of impairments.

2.2.1 Single carrier system

The single-carrier digital television system is designed to transmit high quality video and audio and ancillary data using the same channel bandwidth as present television systems. The system can deliver reliably about 19 Mbit/s of data throughput in a 6 MHz terrestrial broadcasting channel and higher rates in 7 and 8 MHz channels. The information bearing data is used to modulate a single carrier which occupies the entire RF channel.

In typical DTTB reception situations, multipath propagation caused by reflections or non-homogeneities in the propagation medium will cause intersymbol interference to the unprocessed received data stream. Multipath reception will also manifest itself as frequency selective fading within the channel. For practical single carrier systems an adaptive equaliser (usually a decision feedback equaliser) is used to minimise the effects of multipath distortion. For single carrier modulation, intersymbol interference, if uncorrected, will result in eye height closure and an increase in the minimum C/I at which the system can operate. For single carrier transmissions, a training mechanism is usually transmitted to assist adaptive equaliser convergence and system synchronization. An adaptive equaliser and a high-gain directional antenna can reduce the impact of co-channel digital television interference and the severity of co-channel analogue television interference. Another approach, for single carrier systems, is to use comb filtering to create notches in the spectrum at the receiver which align with the frequencies of the unwanted interfering carriers.

Single carrier systems are robust to tone interference since signal power is spread over the entire spectrum. Single carrier systems are inherently rugged against frequency selective fading because the fade will only affect a small portion of the bandwidth in which the signal energy is being received.

2.2.2 Multi-carrier system

The multi-carrier (DVB-T) system was designed originally for the 8 MHz UHF channel spacing used in Europe and has been adapted to fit 7 and 6 MHz channels. Depending on the choice of coding and modulation parameters, data rates from 20 to 30 Mbit/s can be realised to deliver high quality digital television through the broadcasting channels. Equally, lower data rates can be employed in cases where additional ruggedness is considered to be desirable.

The OFDM concept is based on spreading the data to be transmitted over a large number of carriers spread over the RF channel, each carrier being modulated at a low bit rate. In a conventional frequency division multiplex the carriers are individually filtered to ensure there is no spectral overlap. There is therefore no inter-symbol interference between carriers but the available spectrum is not used with maximum efficiency. If however, the carrier spacing is chosen so the carriers are orthogonal over the symbol period, then symbols can be recovered without interference even with a degree of spectral overlap. For maximum spectral efficiency, the carrier spacing equals the reciprocal of the symbol period.

An OFDM modulation system with concatenated error correcting coding and a guard interval is used so that the multi-carrier system can cope with short “natural” echoes due to multipath propagation, as well as with the relatively long “artificial” echoes which occur in SFNs. The system also provides good protection against high levels of co-channel interference and adjacent channel interference emanating from analogue television services. OFDM also has excellent inherent frequency spectrum shaping which will allow DVB-T to be accommodated in channels adjacent to those used for analogue television services while causing minimum interference to these services.

Mobile reception of the OFDM signal is possible given due consideration to factors including vehicle speed, carrier spacing, data rate and modulation scheme, whereas, for a service with fixed reception, high order modulation schemes and consequently high data rates could be used. OFDM signals also allow the possibility of single-frequency network (SFN) operation. This is due to OFDM’s multi-path immunity. SFN operation is possible when exactly the same signal, in time and frequency, is radiated from multiple transmitters.

Multi-carrier systems can be sensitive to co-channel interference because of the very low power in each carrier. A multi carrier system is potentially vulnerable to the non-flat spectrum of co-channel analogue TV as carriers located near the luminance, chrominance and audio carrier frequencies may suffer from strong interference. Approaches to avoid this problem are applying error coding to the multi-carrier system or deleting from the multi-carrier ensemble those carriers likely to suffer interference. An adaptation to the multi-carrier system for areas with very high spectrum congestion implements “band segmented” COFDM which carries the data in 500 kHz bands that can be located to avoid interference from analogue TV.

2.3 Reception categories

2.3.1 Fixed antenna reception

The receiving system model adopted for allotment planning should be a typical receiving installation located near the edge of the service area (i.e. weak signal conditions). Such a configuration may consist of an externally mounted antenna (fixed antenna reception), a low noise amplifier mounted at the antenna (optional), an interconnecting downlead cable and the digital television receiver. Fixed antenna reception is defined as reception where a directional receiving antenna mounted at roof level is used. In calculating the equivalent field strength required for fixed antenna reception, a receiving antenna height of 10 m above ground level is considered to be representative. In the case of fixed antenna reception it is assumed that near-optimal reception conditions (for the relevant radio frequency channels) are found when the antenna is installed. The use of the optional low noise amplifier at the antenna gives the receiving system a better noise figure and compensates for the downlead cable losses.

2.3.2 Portable reception

Portable antenna reception is defined as being reception where a portable receiver with an attached or built-in antenna is used.

– Class A (outdoor) – outdoors at no less than 1.5 m above ground level.

– Class B (ground floor indoor) – indoors at no less than 1.5 m above floor level in rooms on the ground floor and with a window in an external wall.

Portable antenna reception will, in practice, take place under a great variety of conditions (outdoor, indoor, ground floor, first floor, and upper floors). It could even be envisaged that a portable receiver is moved while being viewed.

It is to be expected that there will be significant variation of reception conditions for indoor portable reception, depending to some extent, on the floor-level at which reception is required. However, there will also be considerable variation of building penetration loss from one building to another and also considerable variation from one part of a room to another. Some estimates of the probable signal level requirements for different floor-levels are given in Chapter 5.

In both categories A and B, above, it is assumed that the portable receiver is not moved during reception and large objects near the receiver are also not moved. It is also assumed that extreme cases, such as reception in completely shielded rooms, are disregarded.

It is to be expected that portable coverage is mainly aimed at urban areas. In many countries most people living in urban areas live in apartment buildings. The second category, class B, is therefore probably the more common case of portable reception. It is to be expected that reception will be less difficult in rooms higher than the ground floor.

2.3.3 Mobile reception

Mobile reception of digital terrestrial television is becoming an attractive feature of future systems but has not been a major consideration of the planning for digital television implementation. Tests have shown that mobile reception is possible within the multi-carrier system provided parameters are optimized to overcome the difficulties of the mobile reception environment. This feature is not presently developed for the single carrier system and no testing has been done to demonstrate its feasibility, however with adaptation of the system parameters, it may be possible.

Detailed information on reception categories and required minimum median signal levels is given in Chapters 4 and 5.

No information is given in this Handbook on the minimum signal levels required by digital television receivers. The reader is advised to refer to Recommendation ITU-R BT.1368 for the latest information on this particular topic and the methodology used to derive the values given there, however, at the time of writing, it seems probable that the signal levels required for mobile reception will be similar to those required for portable outdoor reception.

2.4 Service requirements

2.4.1 Digital service possibilities

Terrestrial digital television services offer both advantages and disadvantages compared with analogue television services and in some ways these are linked. The abrupt failure characteristic of digital systems, as compared with the gradual failure typical of analogue systems, is a disadvantage as it means that more care needs to be taken to ensure that a high percentage of viewers can receive a satisfactory service. In practice, this means that coverage boundaries need to be defined for a high percentage of locations, both in terms of the minimum signal levels needed for satisfactory reception and in protection against interference. On the other hand, the full quality of the digital system is retained out to the coverage boundary.

In principle, digital systems can provide a higher quality of reception than can analogue systems for the same propagation conditions, system bandwidth and effective radiated power. However, some of this potential extra reception quality may be given up in order to provide a larger transmission capacity in a given bandwidth. This greater capacity may be used to provide higher-definition standards, more programmes, or additional features (for example, more data or sound information) with an individual programme. An alternative approach would be to trade-off both service quality and quantity in order to provide a more rugged system, for example, a service which is intended to be received on portable receivers with attached or built-in antennas.

The inherent flexibility of digital transmissions has many advantages compared to that of transmission using a "fixed-format" analogue system. However, the number of digital system configurations possible makes it difficult to provide a direct comparison between the capabilities of analogue and digital systems that are designed to occupy the same channel-width. These difficulties are compounded by the fact that some digital systems permit changes of configuration on a dynamic basis to suit broadcasters' varying needs. Nevertheless, there are some features that seem to be quite general. Digital systems:

– can provide a more flexible approach to the provision of terrestrial television services;

– can provide a greater programme capacity within a given allocation of spectrum;

– can provide for higher quality reception;

– can provide a greater degree of resistance to the impairment caused by delayed signals;

– can provide for satisfactory reception on portable receivers using attached or built-in antennas;

– can make use of somewhat lower effective radiated powers.

Even so, it is necessary to qualify some of these features. The better spectrum utilization and the lower radiated powers are the result of *C*/*N* and protection ratio values that are lower than those for analogue systems. The use of precision offset with analogue transmissions can give protection ratios comparable with those for digital systems that are intended to provide high quality. In the latter case, the saving on transmitter power may not be very high if an attempt is made to provide coverage to a very high percentage of locations. Similarly, the use of ghost-cancellation schemes can reduce the sensitivity of analogue systems to that particular type of impairment. Nonetheless, the overall balance is that the use of digital television systems offers significant advantages over their analogue equivalents.

2.4.2 Digital network possibilities

The full range of possibilities for digital television networks will only become available when it is no longer necessary for digital and analogue services to share spectrum (see Chapters 8 and 10). Assuming that digital television services have exclusive use of a given spectrum allocation, the inherent flexibility and better spectrum utilization of terrestrial digital television systems (as compared with analogue systems) makes it possible to consider a much greater range of network configurations than is available with analogue television. One obvious difference is that single frequency networks (SFNs) may become possible under some circumstances. This leads to an initial division of networks into Conventional and SFN types, although there are significant similarities and overlaps in such a division.

Conventional networks imply similar planning concepts to those used at present for analogue networks, whether these are intended to provide individual station, regional or even national coverage. It is likely that transmitter sites similar to those used at present would continue to be used in order to maintain coverage patterns comparable to those presently existing. The major differences from the existing analogue networks would be the smaller distances between co-channel transmitters and the reduced set of constraints on the channel relationships between overlapping coverages (whether the transmitters are nominally co-located or not). In practice, these apparently small differences will have major consequences because of the potentially large increase in the capacity of the available spectrum. This will lead either to a significant increase in the number of programmes available or to a reduction in the amount of spectrum allocated to television.

SFNs on a large scale imply the use of a multi-carrier digital system (such as OFDM). In addition, the basic planning concepts have major differences from those used for analogue networks. If medium or large areas require to be served with exactly the same programme material, then a complete network may have all of its transmitters on exactly the same frequency, although there are significant constraints on the timing requirements for the programme material to be transmitted. Clearly, the use of a single frequency for large area coverage of a programme leads to significant spectrum savings. In the case where multiple programmes are carried within a single channel, the savings may be even greater, although such usage implies that higher C/N and protection ratios are required and this to some extent offsets the apparent gains. In addition, it is necessary to consider carefully the symbol length and guard interval requirements if the full benefits of an SFN are to be achieved. It also has to be accepted that programme "opt-outs" for parts of the overall area are not possible.

Several variants of SFN for providing large area coverage exist, although these differ more in appearance than in reality. The primary difference lies in the spacing between transmitter sites. At one extreme would be a network based on the sites used currently for analogue services, which can be up to some 80 km apart. At the other extreme would be a dense network with transmitter spacings of only 10 or 20 km. In practice, any real network is likely to consist of some elements of both of these cases. Even a network based primarily on the existing analogue station sites would be likely to need a number of relay stations and these would have relatively small spacings. Conversely, a dense network is likely to have some "gaps" where the population density is too low to make it economically justifiable to build some stations. It cannot be assumed that SFN usage implies that large areas are to be covered. An alternative usage would be confined to urban areas in order to provide the high signal levels needed for portable reception. In this case, there could be an SFN for each urban area, with a conventional planning approach used to provide different services in separate, individual urban areas. One aspect of SFN usage may not be confined to multi-carrier systems. If delay equalisers are used with a single carrier system, then it is possible to use a single frequency for a main station and its geographically nearby relays in order to provide for coverage extensions. However, one normal requirement of a delay equaliser is that there should be a significant difference in amplitude between the main signal and any delayed component. If this is the case then there can be little or no coverage overlap between the service area of the main station and that of any of its relays, or between the coverage areas of the individual relay stations.

Detailed information on network planning is given in Chapter 6.

2.4.3 Service availability

A characteristic of digital television systems which impacts on the planning factors is the sharp degradation between the point when picture impairment is first visible and the point when the picture is unusable. With this degradation factor in the order of 1 dB, a critical review of planning criteria in terms of service availability and quality of service may be necessary in light of any objective to duplicate existing analogue coverage to the extent possible.

Service availability in location and time are factors that must be chosen to provide the required digital television service in an efficient and viable manner. The transmission and reception characteristics of a digital television system differ from an analogue system and it is believed that better location and time availability than those used for analogue service planning will be required to provide an acceptable digital television service. The provision of digital services requires close attention to the coverage or service availability and a location and time availability of 90% or even 99% is generally assumed to be necessary.

One coverage requirement for digital television was identified as that of matching the existing analogue services in the VHF/UHF TV bands. A question that needs to be resolved in the coverage planning of Digital TV is what *service availability* objectives should be established at or near the edge of the protected coverage area that would correspond to an “equivalent” availability as provided by analogue service. Whereas analogue service is planned on the basis of a specified performance for at least 50% of locations and 90% or 99% of the time, the gradual degradation characteristic of the analogue service results in a considerably higher service availability statistic at or near the edge of the coverage. If it is the goal to duplicate the analogue coverage with a digital television station, then a higher propagation margin will need to be considered due to the abrupt nature of the threshold to service outage exhibited by digital television.

Service availability can affect the level of the transmitter power required to establish the desired availability at the required coverage distance. As the service availability is increased in either location or time, the required transmitted effective radiated power increases and the separation distances required for interference protection between digital and analogue services increase.

Detailed information on this topic is given in Chapters 3 and 7.

2.5 Interference considerations

2.5.1 Digital-to-analogue interference

In considering the introduction of DTTB services on a “sharing” basis with the existing analogue services, it is necessary to define the degree of degradation to the analogue services from co-channel interference (CCI) and adjacent-channel interference (ACI) that will be acceptable. In general, the transmitted digital signal has a similar spectral characteristic to Gaussian noise. The effect of co-channel interference is therefore to raise the noise thresholds of analogue receivers which in turn reduces the picture grade (on the ITU 5-grade scale) achievable at the edge of the analogue service area. In general, the planning aim is to limit this loss of grade due to digital-to-analogue CCI, where currently grades of 4.0 (for continuous interference) and 3.0 (for tropospheric interference) are the norm.

2.5.2 Digital-to-digital interference

Given the noise-like nature of the digitally transmitted spectrum, the susceptibility of digital systems to digital CCI is almost identical to their susceptibility to thermal noise. That is, susceptibility increases as the modulation levels are increased to higher modulation levels like 16- and 64-QAM (by approximately 7 dB and approximately 13 dB respectively in theory). However, the increased transmission capacity of the higher modulation levels allows very sophisticated error-management schemes to be used, that compensate for this loss and provide an overall gain in performance.

2.5.3 Analogue-to-digital interference

The main sources of “analogue-to-digital” CCI are centred around the vision, sound and colour sub-carrier frequencies of the analogue system. While in principle this relatively high-powered “narrow-band” interference can be very damaging to the digital transmission, the sophisticated error-management schemes described in detail in Part 1 of this Handbook can deal effectively with this type of interference to provide a more rugged performance. As for the “digital-to-digital” interference case, final performance will be dependent on the choice of modulation level, the transmission capacity devoted to error-protection, as well as, to some extent, the particular characteristics of the modulation system – whether this be single or multi-carrier in nature.

Detailed information on the topic of interference calculation is given in Chapter 7.

2.6 Impact of receiving system characteristics

The receiving system parameters give rise to a number of factors that impact on allotment or assignment planning. Key factors are the receiving system noise figure, the partitioning of noise and interference at the receiver input, and the protection ratios (particularly co-channel and adjacent channel) necessary to allow interference free reception for both analogue and digital operations.

Near the edge of coverage, the receiver noise figure has a direct effect on the required field strength and hence the resulting required transmitter power. For digital television planning, a typical receiving system may consist of an antenna, an interconnecting cable and a receiver, or the same components augmented with a low noise preamplifier mounted at the antenna. For the first configuration, the noise figure of the receiver and the interconnecting cable loss impact the required field strength. In the second case with the low noise preamplifier, the noise figure of the preamplifier (in the order of 5 dB) has the major impact in determining the required field strength. The impact of the line loss and the receiver noise figure is reduced by the gain of the preamplifier. In general, the configuration with the preamplifier will require lower field strength than the receiver only configuration.

Another key factor that impacts on allotment or assignment planning is the combination of the required carrier-to-noise (C/N) at the TV receiver antenna terminal and the required co-channel carrier-to-interference (C/I). The C/N in association with the receiving antenna gain, noise figure and desired signal quality establishes the required field strength of the receiving system. The co-channel C/I in association with the receiving antenna directivity discrimination determines the required co-channel separation or protection. In the digital television case, noise and co-channel digital interference are additive as interference behaves similarly to noise. Hence, there is a minimum $C/(N + I)$ at the receiver input that needs to be met to achieve a specified threshold picture quality level, normally referred to as threshold of visibility. Once this value has been established, then for planning purposes, it is necessary to partition the threshold $C/(N + I)$ value between C/N and interference (C/I). Based on equal partitioning between noise and interference, a $C/N = C/I$ at the digital television protected contour would need to be about 3 dB higher than the value at the threshold of visibility.

Detailed information on the summation of interference and noise is given in Chapter 3.

2.7 Protection ratios

Protection ratios for various interference situations are required for the cases of:

– Digital television interfered with by digital television.

– Analogue television interfered with by digital television.

– Digital television interfered with by analogue television.

The protection ratios used in planning are usually based upon the values resulting from the measurements and tests of the digital television system. The co-channel and the adjacent channel protection ratios have the greatest impact on planning and influence system parameters and transmitting site locations. Attention should be given to the adjacent channel protection ratio to ensure that it provides protection not only for the adjacent interfering signal but also for the adjacent out-of-band spectrum of that signal (which appears as co-channel interference to the wanted signal).

No details of protection ratio values are given in this Handbook. This is because these values are likely to change considerably over the next few years as more information about the performance of practical (consumer) receivers becomes available. The reader is advised to refer to Recommendation ITU-R BT.655 and Recommendation ITU-R BT.1368 for protection ratios for analogue and digital television, respectively.

2.8 Transmission aspects

It is to be expected that broadcasters will want to use as much of their existing transmission infrastructure as possible, in particular, the transmitter sites and antenna masts represent a considerable investment which should be re-used if at all possible. In those cases where it is desired to achieve approximately the same coverage area as that of the existing analogue services, this should be possible. Even for new services, some re-use of existing facilities may be desirable, both to reduce any environmental impact and to save costs.

The peaceful co-habitation of analogue and digital television services is of major importance and this has a significant impact on the re-use of existing transmitter sites and masts, and possibly even existing antennas originally installed for analogue television.

Because of the wide variety of existing analogue television transmission installations and the probability that the variety of digital television installations will be at least as wide, it is not possible to give specific guidance in this matter. Each individual case will need to be considered on its own merits.

Detailed information on specific examples of transmission aspects are given in Chapter 9.

CHAPTER 3

PROPAGATION SIGNAL SUMMATION

3.1 Prediction of 50% location signal levels

Propagation prediction methods using information from a terrain data bank exist in a number of countries and give significant improvements in prediction accuracy when compared with simple methods such as Recommendation ITU-R P.370[1]. However, it has been found that these newer methods cannot be applied universally due to the use of empirical correction factors within each of the computer programmes which improve results for the type of terrain found in a specific country.

Tests have been carried out within the EBU to investigate the magnitude of the differences introduced in this way (by comparing predictions with measurements) and it has been found that none of the available computer programmes performs consistently better than the use of a simple method such as Recommendation ITU-R P.370. The latter is essentially statistical in nature and its curves are intended to give reasonable results for the type of terrain met in many parts of the World. Recommendation ITU-R P.370 also has the advantage of having been agreed internationally, for use at conferences, for example.

It is interesting to note that some recent experiments have indicated that Recommendation ITU-R P.370 may provide a better propagation prediction method than some of the more complex terrain databank methods as far as T-DAB signals are concerned. Because both T-DAB and the DVB-T version of terrestrial digital television are OFDM systems, it seems probable that Recommendation ITU-R P.370 (or its replacement Recommendation ITU-R P.1546) may thus provide a reasonable propagation prediction method for the case of terrestrial digital television. However, it is necessary to remember that Recommendation ITU-R P.370 is a statistical method and that it cannot predict areas of poor reception due to signal level reduction caused by obstructions on the propagation path. Indeed, some experiments have indicated that the standard deviation of the difference between a 50% location measurement and a Recommendation ITU-R P.370 prediction is around 13 dB. Such a high value indicates that it may not be very important for the accuracy of the prediction what is the exact value of location variation associated with OFDM signals – whether this value is 4 dB or 7 dB is not really very important. The latter topic is dealt with in more detail in § 3.2 and § 3.3.

Because of the very significant differences in propagation conditions for overland and over sea paths, a coastline (possibly in a simplified form) must be included in the propagation prediction calculations to permit account to be taken of these differences in the calculation of interference levels.

3.1.1 Prediction of wanted signal levels

There are no particular considerations to be taken into account when predicting *wanted* signal levels for an individual transmitter to receiver path in the case of predictions based on Recommendation ITU-R P.370. Values for 50% of the time are appropriate in this case as this time percentage is also applicable to the 99% time requirement for wanted signals. At the short distance ranges

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[1] This ITU-R Recommendation was superseded by Recommendation ITU-R P.1546 while this Handbook was being prepared for publication. The comments made in this Handbook with regard to Recommendation ITU-R P.370 may be applied equally to Recommendation ITU-R P.1546.

involved, up to about 60 km, there is negligible difference in the signal level values for 50% and 99% of the time. However, there are differences in propagation over land and sea paths and it is thus necessary to take account of the nature of any individual propagation path; that is, whether it is all-land, all-sea or a mixed land-sea path.

Where the relevant information is available, Recommendation ITU-R P.370 allows for a correction to be made using the terrain clearance angle for the path from a specific receiving location in the direction of the transmitting site.

Signal level predictions using a terrain data bank will take into account whatever information the individual model requires. As noted above, the value predicted for a given path can be expected to be dependent upon the model used.

3.1.2 Prediction of unwanted signal levels

In the course of both a planning process and a coordination process it is necessary to predict the level of interference field strength produced by one transmitting station in the service area of another. When calculating the level of interfering field strength, the 1% time curves of Recommendation ITU-R P.370 (or Recommendation ITU-R P.1546) should be used. Other methods may, however, be used if there is agreement between the countries concerned.

Ideally, the calculation should be made to points defining the coverage area of the station to be protected. However, in some circumstances, this may not be possible or necessary. Two cases can be distinguished:

a) *Prediction to points defining the service area*

Predictions of interfering field strengths would normally be made to points on the periphery of the service area of the station to be protected. It is preferable that points defining the edge of the service area are specified or calculated on 36 or 12 equally-spaced radials from the transmitter site. The terrain clearance angle correction described in Recommendation ITU-R P.370 (and Recommendation ITU-R P.1546) may be included in the calculation of interference field strengths at these points, if sufficient information concerning local terrain is available. In the case where the boundary points are specified, rather than being calculated, there is no particular requirement that they be on equally-spaced radials.

b) *Prediction to the location of the transmitting site*

In some cases it may not be possible or necessary to define the service area in the manner described in the preceding paragraph. An example of this would be where the station to be protected is a low power station with a very small coverage radius. To define the service area and calculate interference levels at many points would involve unnecessary computation. In this case, the location of the transmitting station can be taken as representative of the service area to be protected, and the prediction of interference field strength can be made to that point. However, since the terrain height of the transmitting site would not be representative of the area to be protected, terrain clearance angle corrections should not be applied.

3.2 Location statistics

Within a small area, say 100 m × 100 m, there will be a random variation of signal level with location which is due to local terrain irregularities. The statistics of this type of variation are generally characterized by a log-normal distribution for the signal levels. Recent measurements for digital signals have shown that the standard deviation will be about 5.5 dB depending, to some extent, on the environment surrounding the receiving location.

It cannot really be said that there is yet a large amount of measured data to *fully* justify any individual value of the standard deviation of location variation for digital television signals. However, the evidence which is available indicates that this standard deviation is likely to be close to 5.5 dB, at least for outdoor paths. Any values related to outdoor coverage in the remainder of this document will be based on a standard deviation of 5.5 dB. For reception indoors, the standard deviation will be larger and this subject is treated in detail in Chapter 5. The difference between 50% and 95% of locations is thus taken to be 9 dB and that between 50% and 70% of locations is taken to be 2.9 dB. It must be stressed that such a value takes no account of the inherent inaccuracies of any propagation prediction method.

In the case that the wanted signal is composed of several individual signals from different transmitters the resulting standard deviation becomes variable, depending on the individual signal strengths. As a consequence, the difference between 50% and 70 or 95% of locations becomes variable. However, it always will be smaller than that of an individual signal. This item is dealt with in more detail in § 6.3.

3.3 Calculation of coverage area for digital television

3.3.1 Necessity for complex calculation methods

The main questions when trying to build new digital television terrestrial networks are the evaluation of the service area and the population covered. These evaluations are made through the estimation of the level of the useful signal(s) and the level of the interfering signals. As indicated in § 3.3.2, because of the rapid failure of digital reception when the level of the useful signal decreases below its "minimum" value, the target for the percentage of locations nominally at any edge[2] of the service area has to be much higher for digital systems than the 50% of locations used for analogue television systems. Values ranging from 70 to 95% are usually quoted for digital television transmissions (see Chapter 4). Under such considerations, some of the simpler tools used for analogue television coverage evaluations are not completely satisfactory and it is necessary to make more complex calculations.

3.3.2 Impact of rapid failure characteristic

In the process of evaluating the coverage area of the analogue television service using usual prediction tools, the value of the field strength specified at the edge of the coverage area is a mean value. It represents the average value of all the real values of the field strength that could be measured within a small area, usually taken to be 100 m × 100 m. That means that in this small area, about half of the real values of the field strength are under this mean value and about half are above this value. For analogue television, if the value of, say, 67 dB(μV/m) is specified as the lower limit of the mean value, that indicates that smaller values of the field strength can be found inside the coverage area. But, if 67 dB(μV/m) corresponds to grade 4 for the picture quality according to the ITU scale, a lower value of field strength will give a somewhat lower quality because of the smooth degradation of analogue reception in presence of noise or in presence of interference. A reduction of about 6 dB for the C/N or C/I will lead to a loss of one grade of picture quality. Thus, at the edge of the service area, even if the real value of the wanted field strength is below the

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[2] The term "edge" is taken to mean *any* transition between a covered area and a non-covered area. These "edges" may occur at the outer boundary of a coverage area or at the boundaries of any uncovered areas which may exist inside the overall area, usually as the result of local screening on the path of the wanted signal.

specified limit value, a picture will still be received but with a lower quality. We can say that the inherent assumption for analogue television is that the "average" quality is grade 4 at the edge of the service area.

Concerning digital television, it is known that the behaviour of the receiver is completely different. When the level of signal decreases and the C/N or C/I falls below a given "minimum" value, the picture disappears completely with a further signal level reduction of less than about 1 dB. This behaviour is generally referred to as the "rapid failure characteristic of the digital system" and the limit value of the field strength is designated as the minimum field strength. If the same coverage definition as for analogue television were used for digital television, this would mean that 50% of the locations would not be served at or near the edge of the service area or in any other areas of reduced signal caused by local obstructions. This is due to the fact that there is no smooth degradation for digital receivers, the picture quality changes rapidly from grade 5 to grade 0, in effect without any intermediate levels of quality. This value of only 50% of locations receiving a picture is clearly unacceptable, higher values of the percentage of locations have to be selected in order to allow reception in a larger number of household, with a standard receiving installation.

The exact value chosen depends on the level of service quality which is aimed at, and that is why values can be different from one country to another or even from one company to another within a given country. Nevertheless, two values, 70% and 95% of the percentage of locations, have been chosen in the coverage definitions given in Chapter 4.

3.3.3 Use of *C*/*I* and *C*/*N*

The assessment of the coverage area of a wanted digital transmitter is carried out using the parameters of the chosen system and taking into account all the transmitters operating in the vicinity of the digital transmitter on the same channel or on adjacent channels. Most of these signals will interfere with the wanted digital signal; the exception is an SFN for which signals coming from nearby transmitters may make a positive contribution. It must be noted that the expression "in the vicinity" may mean "within a few hundred km".

3.3.3.1 Case of one receiving location

For one receiving location to be covered by a digital television transmission, it is known that the level of the wanted signal, expressed in dB, has to be higher than the level of noise by a certain value which is the minimum C/N ratio. This can be expressed in dB by the formula $C > \alpha + N$, α being here the minimum C/N ratio, N the noise level and C the level of the wanted signal. In the same way, to overcome the effect of an interferer, the level of the wanted signal must be higher than the level of this interferer by a certain value referred to as the protection ratio for this particular type of interferer. It can also be expressed in dB by $C > \beta + I$, β being the protection ratio (related to the minimum C/I). The sum $\beta + I$ (protection ratio + field strength of interferer) is often referred to as the nuisance field. (In practice, the receiving antenna discrimination against the interfering signal may also need to be taken into account).

Due to the different natures and bandwidths of the interferers which cause different effects on the carriers of the OFDM signal, the value of the protection ratio is very different from one type of interferer to another. Protection ratios are evaluated in laboratories with the assumption that there is only one source of interference (noise or one unwanted signal only).

In the real world, the wanted signal undergoes interference from noise and, possibly, several interferers which can be of different types. The level of the wanted signal must thus be compared with a combination of unwanted signals. It is clear that, due to the different nature of the signals, the power of the wanted signal cannot be compared directly to the power sum of the noise and the interferers.

The notation $C/(N + I)$ should thus be avoided because the term $(N + I)$ could be interpreted as an addition of the power of the noise and the power of each interferer, this would lead to a value that has no meaning. The only values that can be compared to the wanted signal are the nuisance fields $(\beta + I)$.

Due to the fact that, in this particular case of one receiving location, the levels of the signals are real values, the condition of good reception can simply be expressed as:

$$\Sigma P_C \geq P_N + \Sigma P_{(\beta + I)}$$

where:

ΣP_C: power of the wanted signals

P_N: noise power equivalent

$\Sigma P_{(\beta + I)}$: power of the nuisance fields
and all powers are expressed arithmetically.

3.3.3.2 Case of small area

In practice, it is not possible to know the real values of the field strength for each receiving location in order to apply the previous formula and to determine precisely the coverage area. The only figures that can be evaluated are the mean values of the field strengths in small areas (typically 100 m × 100 m).

The problem is then to know if one given small area is inside or outside a coverage area and for that, the probability of good reception in this areas is calculated. This probability represents the percentage of receiving locations which can receive a satisfactory signal (that is, whose power is greater than or equal to the sum of the noise and nuisance powers) within the small area. A small area is found to be inside the overall coverage area if the probability is higher than a given threshold, 70% or 95% (for the coverage definitions given in Chapter 4).

The calculation of the probability is carried out using the appropriate fixed values for the level of noise and the protection ratios of each type of interferer and for the field strengths which are random variables, with prediction of the mean level of the wanted field strength and of each unwanted signal using the prediction method of Recommendation ITU-R P.370 (or Recommendation ITU-R P.1546) or prediction models using terrain data banks.

But, because the wanted and nuisance powers are random variables which are only known through their means and standard deviations, the formula given above must not be applied only to the means of the wanted and nuisance powers. Therefore, it is necessary to refer to mathematical models for the distribution of field strength with locations and to use mathematical methods to obtain the result of the combination of several randomly distributed signals.

3.3.4 Calculation methods

The basic principle when evaluating a service area is to estimate mean value and standard deviation of wanted field strength and unwanted field strength in a large number of test locations in the assumed service area and with these values, to calculate the percentage of locations served. This could be done for different bearings originating from the transmitter location, for example every 10°, or, in some cases, with a higher density of test locations.

For analogue television, equal values of the wanted field strength and the nuisance field strength correspond to a coverage of 50% of locations. Different methods have been developed to calculate the equivalent nuisance signal level when there are several interfering signals. These methods can be found in Report ITU-R BS.945. In the case of an SFN the wanted signal may also be composed of several individual signals.

3.4 Combination of signal levels for coverage assessments

3.4.1 Introduction

One of the questions to be answered is how to combine interfering signals when there is more than one and how to take into account the effect of noise. Some of the calculation methods to deal with this question are presented below. They are all statistical methods which require computer processing and they use models of the real situation. In all the methods, except the power sum method, it is assumed that field strengths have a log normal distribution with location.

The first method is a numerical approach which is capable of providing the required accuracy but at the expense of a large amount of computer time. The remaining methods are approximations which are presented in order of growing complexity and this increasing complexity corresponds to an increasing computer processing time.

It should be noted that though there may exist some correlation between the individual signals, wanted as well as unwanted signals, none of the methods described below include the treatment of correlation in their original form. However some of them can be extended to include correlation. The effect of correlation varies with the reception situation. It can produce either an increase or a decrease of coverage depending upon the particular correlation situation.

3.4.2 The Monte-Carlo method

Apart from a deterministic (numerical integration) method, the Monte-Carlo approach is the most accurate method available to evaluate the coverage probability. With the mean value and the standard deviation of the distribution of each signal it is possible to simulate the situation for a large number of reception locations in a small area (say, 100 m × 100 m). This is done by generating one random value of the wanted field and one random value of each interferer. For each combination it is possible to check if the reception location is served or unserved by comparing the power of the useful signal with the sum of the powers of the noise and the nuisance fields.

By repeating this simulation for a large number of combinations of wanted and unwanted signals, the coverage probability for a given small area may be derived. The higher the number of combinations, the more accurate the method becomes but this can lead to very lengthy computer processing times. In addition, the process must be repeated for a large number of small areas in order to represent the overall coverage area.

3.4.3 Power sum method

This method has been used for the assessment of multiple interference at several ITU conferences. The sum of the signal levels is calculated by a non statistical summation of the individual signal powers. For the unwanted signal, the powers of the mean values of the individual nuisance fields are added to the power of the minimum field strength (representing the noise contribution). For the wanted signal in an SFN, the powers of the individual useful fields are added. A 50% location coverage is obtained if the sum of the unwanted signal levels equals the sum of the wanted signal levels.

For digital television, a margin must be added to the resulting nuisance field in order to cover more than 50% of the locations. This margin is related to the target percentage of locations. Its value is not derived by the power sum method. Usually a value derived from the standard deviation of a single signal is used.

The method gives acceptable results for a 50% locations target but shows a poor behaviour for higher percentages due to its non-statistical character. Detailed formulae are given in Annex 1 to this Chapter.

3.4.4 Simplified multiplication method

The simplified multiplication method is a statistical computation procedure which has also been used for the assessment of multiple interference, for instance at the Regional VHF/FM Broadcasting Conference (Geneva, 1984).

It gives the coverage probability in the presence of several interfering signals which are assumed to be log-normally distributed with known mean values and standard deviations. The coverage area can be determined by calculating the probability for different locations. The contour of the coverage area is made up of the set of locations where the coverage probability achieves the required value.

As the effect of noise is not taken into account in the statistical treatment, over-estimation of the coverage can be expected when the levels of the interferers are low. However, it is possible to add the effect of noise at the end of the calculation process.

This method is explained in detail in EBU Doc. Tech 3254, but it must be noted that it is not applicable to SFNs since it cannot deal with multiple useful signals.

3.4.5 Log-normal method

The log-normal method is an approximation method for the statistical computation of the sum distribution of several log-normally distributed variables. In a coverage calculation it gives the coverage probability of the small area under consideration. The method is based on the assumption that the resulting sum distributions of the wanted and unwanted fields are also log-normal. It is composed of several steps. First the distributions of the composite wanted (C) and unwanted (NF) fields are calculated. Then the corresponding distributions of C/NF and C/N are evaluated. Finally, the combination of these distributions gives the coverage probability. To some extent, the LNM is able to cope with different standard deviations of the single field distributions.

To improve the accuracy of the LNM in the high probability region (that is, a high coverage value) a correction factor can be introduced. This version of the LNM is called k-LNM.

Detailed formulas of standard LNM and k-LNM are given in Annex 2 to this Chapter. A simplified version of the standard LNM is described in Report ITU-R BS.945. (This is not to be confused with the so-called "simplified log-normal method" which is applicable only for 50% coverage calculations and therefore of no use for digital television planning).

3.4.6 The *t*-LNM method

The t-LNM method is a numerical approximation method for the statistical computation of the sum distribution of several log-normally distributed variables. Its structure is similar to that of the standard LNM and it is based on the same idea, i.e. that the sum distribution of two log-normal variables is also log-normal. However, the parameters of the sum distribution are calculated in a different way and, as a consequence, are different from those of the standard LNM.

This approach leads to a higher accuracy in the high probability region (that is, a high coverage value) compared to the standard and *k*-LNM approaches but this must be paid for with higher mathematical complexity. The *t*-LNM method is able to process different standard deviations of the single fields with few restrictions. The specific case of noise may be regarded as an interference signal with a standard deviation of 0 dB.

A description of the method is given in Annex 3 to this Chapter.

3.4.7 Schwartz and Yeh method

The Schwartz and Yeh method is an iterative method for calculation of the characteristics of the resultant of *N* interferers. It make the assumption that the combination of two log normal variables also has a log normal distribution (this is a common approximation) and it gives the formulas to calculate the resultant of two variables. For more than two signals an iterative process is applied. Its general approach is very similar to that of the *t*-LNM and the accuracy of both method is comparably high; for this reason, no further details are given here.

ANNEX 1

TO CHAPTER 3

Power sum method

The power sum method is a procedure for the approximate calculation of the mean value of a sum field. If the mean value of the (logarithmic) field strength of a single signal is denoted by $\overline{F}$ and is expressed in dB(μV/m), its power *P* (in arbitrary units) is given by:

$$P = 10^{\frac{\overline{F}}{10}}$$

For *n* individual fields the respective powers are added:

$$P_{\Sigma} = \sum_{i=1}^{n} P_i$$

and the mean value $\overline{F}_{\Sigma}$ of the (logarithmic) sum field strength is calculated as:

$$\overline{F}_{\Sigma} = 10 \log_{10}(P_{\Sigma})$$

ANNEX 2

TO CHAPTER 3

Standard LNM and *k*-LNM

The approach is based on the idea to describe the distribution of the sum of two log-normally distributed statistical variables by a new log-normal distribution, the parameters of which are determined by the prescription that the mean value and standard deviation of the new, approximate, distribution have to be identical with those of the true sum distribution:

$$M_{power}^{approx} = M_{power}^{true}, \quad S_{power}^{approx} = S_{power}^{true}$$

where M and S denote the mean value and standard deviation of the respective distributions.

Since the resulting approximate sum distribution is taken to be log-normal, it can be combined again with a third log-normal distribution, and so on, thus enabling the construction of an approximate distribution of n log-normally distributed statistical variables. This procedure can be performed analytically.

Suppose there are given:

n logarithmic fields F_i with Gaussian distribution (parameters $\overline{F_i}, \sigma_i, i = 1 \ldots n$).

The task is to find the parameters of the approximate log-normal sum distribution:

1. Transform $\overline{F_i}, \sigma_i, i = 1 \ldots n,$ from dB scale to Neper scale (this avoids nasty constants in the calculation):

$$X_{\text{Neper}} = \frac{1}{10 \log_{10}(\text{e})} * X_{\text{dB}}$$

2. Evaluate the mean values M_i and the variances S_i^2 of the n fields:

$$M_i = \text{e}^{\overline{F_i} + \frac{\sigma_i^2}{2}}, \; S_i^2 = \text{e}^{2\overline{F_i} + \sigma_i^2} * \left(\text{e}^{\sigma_i^2} - 1\right) \qquad i = 1 \ldots n$$

3. Determine the mean value M and variance S^2 of the sum field strength distribution:

$$M = \sum_{i=1}^{n} M_i, \; S^2 = \sum_{i=1}^{n} S_i^2$$

4. Determine the distribution parameters σ_Σ and $\overline{F}_\Sigma$ of the approximate log-normal sum distribution:

$$\sigma_\Sigma^2 = \log_e\left(k\frac{S^2}{M^2}+1\right), \quad \overline{F}_\Sigma = \log_e(M) - \frac{\sigma_\Sigma^2}{2} \qquad i = 1...n$$

where k is a correction factor in the range 0…1.

5. Transform $\overline{F}_\Sigma$ and σ_Σ from Neper scale to dB scale:

$$X_{\text{dB}} = 10\ \log_{10}(\text{e}) * X_{\text{Neper}}$$

The k-LNM suffers from the drawback that the correction factor k depends on the number, the powers and the variances of the involved fields. To obtain optimal results, an interpolation table would be necessary, but this is not suitable for an heuristic approach like k-LNM. Therefore, to keep the simple and analytic character of the approximation, only an average value of k can be chosen, extracted from a sample of representative field configurations. This simplicity has to be paid for with an inaccuracy which amounts to a few dB for the 1%-fractile for some, fairly typical, configurations. For the summation of fields with standard deviations between 6 and 10 dB the value $k = 0.5$ seems to represent a fair compromise. For smaller standard deviations a higher value for k should be used, e.g. $k = 0.7$. If k is set to 1.0, the k-LNM is identical with the standard LNM approach as described in Report ITU-R BS.945.

ANNEX 3

TO CHAPTER 3

t-LNM (V2)

1 Introduction

This Annex describes a method of computing the sum field from component field parameters (mean, variance) which provides a reduction of computational load compared to earlier versions of the *t*-LNM. The principal structure of computing the sum field by combining the *n*-th component field with the sum of the fields 1 to $n - 1$ by means of interpolation tables has been retained. By exploiting the properties of a suitably chosen analytical approximation of the expression for the sum of two fields it has become possible to compute the interpolation tables at run time and to replace the two trilinear interpolation steps by three bilinear interpolations, which cuts down the number of necessary operations to almost ½ of the double trilinear version *t*-LNM (V1).

2 The *t*-LNM(V2) algorithm

Let f_1 and f_2 be the (uncorrelated and normally distributed) intensity levels of the two fields to be combined. The corresponding sum field level is given by:

$$f = \log_e(e^{f_1} + e^{f_2}) \tag{1}$$

which can be written in the form

$$f = \frac{1}{2}(f_1 + f_2) + \log_e\left(e^{\frac{x}{2}} + e^{-\frac{x}{2}}\right) \tag{2}$$

where:

$$x = f_1 - f_2 \tag{3}$$

From (2) it follows that the mean value $<f>$ of the sum field level f has the form

$$\langle f\rangle = \frac{1}{2}(\langle f_1\rangle + \langle f_2\rangle) + U(\bar{x}, \sigma_x) \tag{4}$$

where $< f_1 >$ and $< f_2 >$ are the mean values of f_1 and f_2, respectively and:

$$U(\bar{x}, \sigma_x) := \langle\log_e\left(e^{\frac{x}{2}} + e^{-\frac{x}{2}}\right)\rangle \tag{5}$$

For convenience, $\bar{f}$ is used in place of $<f>$ in some of the following equations.

Clearly $U(\bar{x},\sigma_x)$ depends on the parameters of the distribution of x only; by proposition, x is normally distributed with mean $\bar{x}=\bar{f}_1-\bar{f}_2$ and variance $\sigma_x^2=\sigma_1^2+\sigma_2^2$. The variance of f can be written in the form:

$$\langle f^2\rangle-\langle f\rangle^2=\frac{1}{4}\sigma_x^2+V(\bar{x},\sigma_x)-\left[U(\bar{x},\sigma_x)\right]^2+\widetilde{W}(\bar{x},\sigma_1,\sigma_2) \tag{6}$$

where:

$$V(\bar{x},\sigma_x)=\langle\left[\log_e\left(e^{\frac{x}{2}}+e^{-\frac{x}{2}}\right)\right]^2\rangle \tag{7}$$

and

$$\widetilde{W}(\sigma_1,\sigma_2)=\langle(f_1-\bar{f}_1+f_2-\bar{f}_2)\times\log_e\left(e^{\frac{x}{2}}+e^{-\frac{x}{2}}\right)\rangle \tag{8}$$

With suitably chosen coefficients A, B, and C the term $\ln(e^{\frac{x}{2}}+e^{-\frac{x}{2}})$ can be approximated by

$$\log_e\left(e^{\frac{x}{2}}+e^{-\frac{x}{2}}\right)=\frac{1}{2}|x|+C\,e^{-A|x|-Bx^2} \tag{9}$$

Both absolute and relative approximation errors are less than 7×10^{-3} with maximum errors occurring for x in the interval $[-4,\ 4]$ when $A=0.685437037$, $B=0.08198801$ and $C=0.686850632$. When the approximation (9) is inserted into the expressions (5), (7) and (8) the mean values can be evaluated. It turns out that:

$$U(\bar{x},\sigma_x)=\bar{x}\left[\Phi\left(\frac{\bar{x}}{\sigma_x}\right)-\frac{1}{2}\right]+\frac{\sigma_x}{\sqrt{2\pi}}\,e^{-\frac{\bar{x}^2}{2\sigma_x^2}}$$

$$+\frac{C\,e^{-\frac{\bar{x}^2}{2\sigma_x^2}}}{\sqrt{1+2B\sigma_x^2}}\left[e^{\frac{K_+^2}{2}}\Phi(-K_+)+e^{\frac{K_-^2}{2}}\Phi(K_-)\right] \tag{10}$$

where:

$$K_\pm=\frac{\bar{x}/\sigma_x\pm A\sigma_x}{\sqrt{1+2B\sigma_x^2}} \tag{11}$$

and where $\Phi(y)=\frac{1}{\sqrt{2\pi}}\int_{-\infty}^{y}\mathrm{d}m\,e^{-\frac{m^2}{2}}$ is the cumulated normalized normal distribution.

V is given by:

$$V(\bar{x},\sigma_x)=\frac{1}{4}(\bar{x}^2+\sigma_x^2)+\frac{C\,\sigma_X}{1+2B\,\sigma_x^2}\,\mathrm{e}^{-\frac{\bar{x}^2}{2\sigma_x^2}}\times$$

$$\left[\sqrt{\frac{2}{\pi}}-K_+\mathrm{e}^{\frac{K_+^2}{2}}\Phi(-K_+)+K_-\mathrm{e}^{\frac{K_-^2}{2}}\Phi(K_-)\right]+\frac{C^2}{\sqrt{1+4B\sigma_x^2}}\,\mathrm{e}^{\frac{-2B\bar{x}^2+2A^2\sigma_x^2}{1+4B\sigma_x^2}}\times$$

$$\left[\mathrm{e}^{\frac{2A\bar{x}}{1+4B\sigma_x^2}}\Phi\left(-\frac{\bar{x}/\sigma_x+2A\sigma_x}{\sqrt{1+4B\sigma_x^2}}\right)+\mathrm{e}^{\frac{-2A\bar{x}}{1+4B\sigma_x^2}}\Phi\left(\frac{\bar{x}/\sigma_x-2A\sigma_x}{\sqrt{1+4B\sigma_x^2}}\right)\right] \tag{12}$$

$\widetilde{W}$ finally can be written as:

$$\widetilde{W}=(\sigma_1^2-\sigma_2^2)\,W(\bar{x},\sigma_x) \tag{13}$$

where:

$$W(\bar{x},\sigma_x)=\Phi\left(\frac{\bar{x}}{\sigma_x}\right)-\frac{1}{2}+C\,\mathrm{e}^{-\frac{\bar{x}^2}{2\sigma_x^2}}\times$$

$$\left\{\frac{1}{\sigma_x(1+2B\sigma_x^2)}\left[K_+\mathrm{e}^{\frac{K_+^2}{2}}\Phi(-K_+)+K_-\mathrm{e}^{\frac{K_-^2}{2}}\Phi(K_-)\right]\right.$$

$$\left.-\frac{\bar{x}}{\sigma_x^2\sqrt{1+2B\sigma_x^2}}\left[\mathrm{e}^{\frac{K_+^2}{2}}\Phi(-K_+)+\mathrm{e}^{\frac{K_-^2}{2}}\Phi(K_-)\right]\right\} \tag{14}$$

Once the functions U, V and W have been tabulated (which due to the many similarities of the terms appearing in equations (10), (12) and (14) consumes only a moderate amount of computing time) the combination of two fields can very simply be accomplished by first computing $\bar{x}$ and σ_x, then finding the corresponding values of the functions U, V and W by bilinear interpolation in the respective tables, and finally computing the mean sum field level by equation (4) and the variance as:

$$\langle f^2\rangle-\langle f\rangle^2=\frac{1}{4}\sigma_x^2+V(\bar{x},\sigma_x)-[U(\bar{x},\sigma_x)]^2+(\sigma_1^2-\sigma_2^2)\,W(\bar{x},\sigma_x) \tag{15}$$

CHAPTER 4

COVERAGE

4.1 Coverage definitions for fixed, portable and mobile reception

4.1.1 Introduction

It is necessary to have definitions for the coverage of a terrestrial television transmitting station or a group of such stations. Such definitions may be based primarily on technical criteria but need to be readily usable for non-technical purposes.

The above is true for analogue television transmissions as well as for digital ones. However, the case of analogue stations is relatively easy to deal with as the line defining any edge of a coverage area is rather "soft" and it is not necessary to be too precise about where the line actually lies in any given area; indeed in many cases it is not really possible to be precise.

Digital television service coverages are characterized by a very rapid transition from near perfect reception to no reception at all and it thus becomes much more critical to be able to define which areas are going to be covered and which are not. However, because of the very rapid transition of the DVB-T system, there is a cost penalty if the coverage target within a small area (say, 100 m × 100 m) is set too high. This occurs because it is necessary either to increase the transmitter powers or to provide a larger number of transmitters in order to guarantee coverage to the last few percent of the worst-served small areas.

For this reason, the coverage definition of "good" has been selected as the case where 95% of the locations within a small area are covered. Similarly, "acceptable" has been defined to be the case where 70% of the locations within a small area are covered.

The definitions do not aim to describe the area where coverage is achieved under worst case conditions. They provide a description of the area where "good" or "acceptable" coverage should be achieved under representative practical conditions.

It should be borne in mind that in a given situation it may be possible to improve reception:

– by finding a better position for the receiving antenna;

– by using a (more) directional receiving antenna with a higher gain;

– by using a low-noise antenna amplifier (in the case of fixed antenna reception).

4.1.1.1 Fixed antenna reception

Fixed antenna reception is defined as reception where a directional receiving antenna mounted at roof level is used.

It is assumed that near-optimal reception conditions (within a relatively small volume on the roof) are found when the antenna is installed.

In calculating the field strength for fixed antenna reception a receiving antenna height of 10 m above ground level is considered to be representative.

4.1.2 Portable antenna reception

Portable antenna reception is defined as:

– class A (outdoor) being reception where a portable receiver with an attached or built-in antenna is used outdoors at no less than 1.5 m above ground level;

– class B (ground floor, indoor) being reception where a portable receiver with an attached or built-in antenna is used indoors at no less than 1.5 m above floor level in rooms:
  – on the ground floor;
  – with a window in an external wall.

Portable indoor reception at the first or higher floors will be regarded as class B reception with signal level corrections applied, but indoor ground floor reception is likely to be the most common case.

In both categories A and B, above, it is assumed that:

– optimal receiving conditions will be found by moving the antenna up to 0.5 metre in any direction;

– the portable receiver is not moved during reception and large objects near the receiver are also not moved;

– extreme cases, such as reception in completely shielded rooms, are disregarded.

4.1.3 Mobile reception

Mobile reception is defined as reception using a non-directional antenna mounted at the roof level of a moving vehicle.

The dominant factor with regard to local reception effects is thought to be the fade margins in the presence of Rayleigh channels. Fade margins depend on the frequency and the velocity of the vehicle. The values of fade margins are derived from the differences between the required C/N ratio for a Gaussian channel and that for a Rayleigh channel.

The other margins can be given the same values as those for outdoor portable reception.

The specifications adopted in Japan for portable reception are shown in Table 4.1 for each frequency band.

TABLE 4.1

Mobile reception

	Band	**Band**	**Band**	**Band**
	65 MHz	200 MHz	500 MHz	800 MHz
Height gain(1)	–10 dB	–10 dB	–12 dB	–12 dB
Antenna gain	–2.2 dB	0 dB	0 dB	0 dB
Antenna directivity	0 dB	0 dB	0 dB	0 dB
Feeder loss	0 dB	0 dB	0 dB	0 dB
Fade margins	10.8 dB(2)	8.8 dB(2)	4 dB(2)	(4 dB(2))
NF	7 dB	7 dB	7 dB	7 dB

(1) For a receiving antenna height of 1.5 m above ground level (a.g.l.).

(2) The values for DQPSK $r = 1/2$.

4.1.4 Coverage area

In defining the coverage area for each reception condition a three level approach is taken.

– **Level 1: Receiving location**

  The smallest unit is a receiving location: descriptions of the reception conditions are given in § 4.1.2 to § 4.1.4.

  A receiving location is regarded as being covered if the level of the wanted signal is high enough to overcome noise and interference for a given percentage of the time. A value of 99% of time is recommended.

– **Level 2: Small area coverage**

  The second level is a "small area" (typically 100 m by 100 m).

  In this small area the percentage of covered location is indicated.

  The coverage of this small area is classified as:

  "Good", if at least 95% of receiving locations within it are covered;

  "Acceptable", if at least 70% of locations within it are covered.

– **Level 3: Coverage area**

  The coverage area of a transmitter, or a group of transmitters, is made up of the sum of the individual small areas in which a given percentage (70% or 95%) of coverage is achieved.

4.1.5 Examples of practical usage

In the case where simplified definitions of transmitter coverage are required, a phrase such as "area within which good fixed antenna reception is expected" is equivalent to:

– coverage area for a transmitter or a group of transmitters;

– at least 95% of receiving locations within every included small area are covered;

– fixed antenna reception.

In the same way "an area within which acceptable class B portable antenna reception, is expected" is equivalent to;

– coverage area for a transmitter or a group of transmitters;

– at least 70% of (indoor) ground floor receiving locations within every included small area are covered;

– portable antenna reception.

4.2 Receiving antennas

4.2.1 Fixed antenna reception

The antenna diagrams (directivity) to be used for digital television planning are given in Recommendation ITU-R BT.419.

The antenna gains (relative to half wave dipole) used in the derivation of the minimum median wanted signal levels in § 5.2.1 are:

65 MHz	200 MHz	500 MHz	800 MHz
3 dB	7 dB	10 dB	12 dB

These values are considered as realistic minimum values.

Within any frequency band, the variation of antenna gain with frequency may be taken into account by the addition of a correction term:

$$\text{Corr} = 10 \log (F_A/F_R)$$

where:

F_A: actual frequency being considered

F_R: relevant reference frequency quoted above.

4.2.1.1 Feeder losses

The feeder losses used in the derivation of the minimum median wanted signal levels in § 5.2.1 are:

65 MHz	250 MHz	500 MHz	800 MHz
1 dB	2 dB	3 dB	5 dB

4.2.2 Portable antenna reception

4.2.2.1 General

The conditions for portable reception differ from fixed reception in the:

– absence of receiving antenna gain and directivity;

– reduced feeder loss;

– generally lower reception height;

– building penetration loss in the case of indoor reception.

It has been assumed that a portable receiver and a receiver for fixed reception have the same noise figure, that is 7 dB.

From the discussion given below, it may assumed that the gain of a portable receiving antenna is –2.2 dB for VHF and 0 dB for UHF. For both frequency bands, it may be assumed that the feeder loss is 0 dB. These values were used in the derivation of the minimum median wanted signal levels given in § 5.3.1.

4.2.2.2 Elements of portable antenna reception

4.2.2.2.1 Signal level variations

4.2.2.2.1.1 General

Field strength variations can be divided into macro-scale and micro-scale variations. The macro-scale variations relate to areas with linear dimensions of 10 m to 100 m or more and are mainly caused by shadowing and multipath reflections from distant objects. The micro-scale variations relate to areas with dimensions in the order of a wavelength and are mainly caused by multipath reflections from nearby objects. As it may be assumed that for portable reception the position of the antenna can be optimized within the order of a wavelength, micro-scale variations will not be too significant for planning purposes. Another way to overcome these variations is the possibility of a receiver using antenna diversity.

Macro-scale variations of the field strength are very important for coverage assessment. In general, a high target percentage for coverage would be required to compensate for the rapid failure rate of digital television signals.

4.2.2.2.1.2 Micro-scale variations

Measurements carried out in Eindhoven in the Netherlands showed that the standard deviation of the micro-scale field strength distribution is about 3 dB. This value has been confirmed by measurements in the United Kingdom. The location variation for micro-scale variations is therefore:

Coverage target	Location variation
>95%	5 dB
>70%	1.5 dB

4.2.2.2.1.3 Macro-scale variations at outdoor locations

Recommendation ITU-R P.370 gives a standard deviation for wide band signals of 5.5 dB. This value is used here for determining the location variation at outdoor locations.

This location variation for macro-scale variations is therefore:

Coverage target	Location variation
>95%	9 dB
>70%	2.9 dB

4.2.2.2.1.4 Macro-scale variations at indoor locations

The variation factor at indoor locations is the combined result of the outdoor variation and the variation factor due to building attenuation (see § 4.2.4)

4.2.2.3 Height loss

For portable reception, the antenna height of 10 m above ground level generally used for planning purposes is not realistic and a correction factor needs to be introduced based on a receiving antenna near ground floor level. For this reason a receiving antenna height of 1.5 m above ground level (outdoor) or above floor level (indoor) has been assumed.

The propagation prediction method of Recommendation ITU-R P.370 uses a receiving height of 10 m. To correct the predicted values for a receiving height of 1.5 m above ground level a factor called "height loss" has been introduced. Measurements in the Netherlands at UHF showed a height loss of 12 dB. For VHF, Report ITU-R BS.1203 gives a value of 10 dB.

4.2.2.4 Building penetration loss

4.2.2.4.1 General

Portable television reception will take place at outdoor and indoor locations. The field strength at indoor locations will be attenuated significantly by an amount depending on the materials and the construction of the house. A large spread of building penetration losses is to be expected.

The mean building penetration loss is the difference (dB) between the mean field strength inside a building at a given height above ground level and the mean field strength outside the same building at the same height above ground level.

4.2.2.4.2 Measurements at VHF

Results of measurements carried out at VHF in the United Kingdom to investigate in-house reception of T-DAB have been reported in Report ITU-R BS.1203. The results indicate a median value of building penetration loss of 8 dB with a standard deviation of 3 dB.

4.2.2.4.3 Measurements at UHF

Measurements have been carried out in the Netherlands using a transmitted OFDM signal with a bandwidth of 8 MHz and containing 512 carriers. The measurements were made as samples with a receiver bandwidth of 12 kHz covering the channel in a series of steps.

The signal level was measured as a function of micro-scale variations at indoor and outdoor locations.

It is expected that the value $V_{10\%}$, which represents the received narrow band signal power exceeded at 10% of the locations, is most closely related to the wide band received signal level. Therefore, the values of $V_{10\%}$ for indoor, outdoor and 10 m reference measurement sites seem the most well-suited for calculation of loss and gain figures.

It appears that the median value $M(V_{10\%}(\text{outdoor})/V_{10\%}(\text{indoor}))$, which might be a good measure for building penetration loss, is in the order of 6 dB. The standard deviation is estimated to be about 6 dB.

Further measurements carried out in the Netherlands using a transmitted noise signal of 7 MHz and receiver bandwidth of 7 MHz show a median building penetration loss of about 9 dB. However these measurements were done at a limited number of locations. The number of concrete houses was relatively high. This might be the reason for the somewhat higher median value.

The influence of people walking around the receiving antenna has also been estimated. The signal level variations (10% and 90% value) ranged from +2.6 dB to –2.6 dB. These variations are relatively small and it does not seem necessary to take them into account for planning purposes.

A number of other measurements have also been carried out in the Netherlands to determine:

– influence of a wet wall;
– time variation of the received signal in a period of 11 days over a short path.

It appeared that neither of these two conditions has a significant influence on the received signal.

4.2.2.4.4 Building penetration loss values for planning purposes

Until more consistent values become available the building penetration loss for planning purposes is taken as:

Band	Median value	Standard deviation
VHF	8 dB	3 dB
UHF	7 dB	6 dB

However, the penetration loss does not become negative.

4.2.2.4.5 Location distribution indoors

The variation factor at indoor locations is the combined result of the outdoor variation and the variation factor due to building attenuation. These distributions are expected to be uncorrelated. The standard deviation of the indoor field strength distribution can therefore be calculated by taking the root of the sum of the squares of the individual standard deviations. At VHF, where the macro-scale standard deviations are 5.5 dB and 3 dB respectively, the combined value is 6.3 dB. At UHF, where the macro-scale standard deviations are 5.5 dB and 6.2 dB respectively, the combined value is 8.3 dB.

The location variation for macro-scale variations at indoor locations is therefore at VHF:

Coverage target	Location variation
>95%	10 dB
>70%	3 dB

and at UHF:

Coverage target	Location variation
>95%	14 dB
>70%	4 dB

As indicated in Chapter 3, the overall field strength prediction process must take account of both the location variation and the difference between predicted and measured values.

4.2.2.5 Portable receiving antenna properties

4.2.2.5.1 General

A roof-level antenna as used with fixed reception can be expected to have a gain of about 10 to 12 dB at UHF. For a portable receiver the antenna will most probably be of either the built-in type of very short length and in the extreme case having –20 dB gain or at best will be a set-top orientable antenna with a few dB gain (at UHF).

For planning purposes it has been assumed that the antenna of a portable receiver is omni-directional and that the gain is 0 dB for a UHF antenna and –2.2 dB for a VHF antenna. A portable receiver can be assumed to have 0 dB feeder loss. For reference, it may be noted that a roof-level antenna will be connected to a receiver by means of a feeder cable. This is likely to have a loss of 3 to 5 dB at UHF. Such values may seem high when the relatively short feeder lengths are considered, but some allowance must be included for feeder ageing effects (for example, corrosion of the copper screening).

4.2.2.5.2 Measurements of indoor antennas

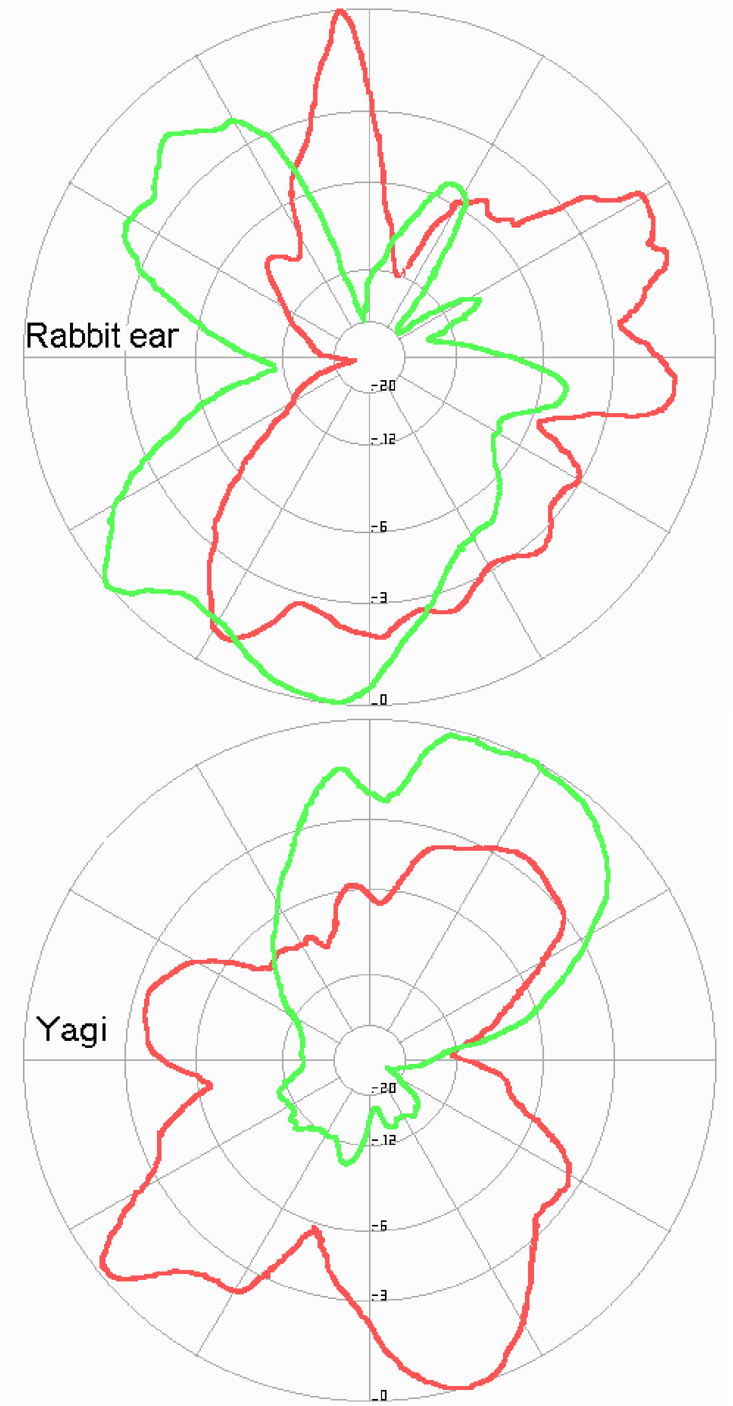
Measurements have been carried out in the Netherlands to investigate directivity of set-top antennas in practical circumstances. One "rabbit-ear" and two five element Yagi antennas of moderate quality have been selected. The results showed that gain and directivity depend very much on frequency and location.

The gain varied from about –15 to +3 dB for the Yagi antennas and from about –10 to –4 dB for the "rabbit ear" antenna.

For directivity measurements the antennas were placed in a room close to a wall to represent practical conditions. The radiation patterns changed considerably with the frequency.

In practical conditions the antenna should therefore be directed to obtain the highest signal rather than in the direction of the transmitter (assuming that this is even known).

Examples of antenna patterns for two antenna types at an indoor location close to a wall, measured in the Netherlands, are shown in Figs. 4.1 and 4.2.



FIGURES 4.1 and 4.2

Examples of indoor antenna patterns

Measurements by the BBC of two commercially available indoor antennas showed a better performance. The antennas had a gain of 5 to 6 dB throughout Band IV and V.

CHAPTER 5

MINIMUM MEDIAN WANTED SIGNAL LEVELS

5.1 General

The minimum signal levels needed to overcome noise, usually expressed as the minimum receiver input power or the corresponding minimum equivalent receiver input voltage, do not take any propagation effects into account. However, it is necessary to take account of these effects when considering television reception in a practical environment.

In § 3.3.2 and Chapter 4, it is indicated that, due to the very rapid transition from near perfect to no reception at all, it is necessary that the minimum required signal level is achieved at a high percentage of locations. These percentages have been set at 95 for "good" and 70 for "acceptable" reception. Minimum median signal levels may be derived, taking account of propagation elements, to ensure that the minimum values are achieved at the specified percentage of locations.

The minimum median signal levels are calculated for:

- three different receiving conditions:
  - fixed antenna reception;
  - portable outdoor reception;
  - portable indoor reception at ground floor;
- four frequencies representing Band I, Band III, Band IV and Band V:
  - 65 MHz;
  - 200 MHz;
  - 500 MHz;
  - 800 MHz;
- five representative C/N ratios:
  - 2 dB;
  - 8 dB;
  - 14 dB;
  - 20 dB;
  - 26 dB.

Representative C/N values are used for these examples. Results for any chosen system variant may be obtained by interpolation between relevant representative values.

All minimum median equivalent field strength values presented in this chapter are for coverage by a single transmitter only, not for single frequency networks, where there will be more than one contribution to the wanted signal. Further information on this topic is given in Chapter 6.

To calculate the minimum median power flux-density or equivalent field strength needed to ensure that the minimum values of signal level can be achieved at the required percentage of locations, the following formulas are used:

for fixed antenna reception (in Tables 5.1 to 5.4):

$$\varphi_{min} = P_{s\,min} - A_a + L_f$$

$$\varphi_{med} = \varphi_{min} + P_{mmn} + C_l$$

for portable outdoor (class A) reception (in Tables 5.5 to 5.8):

$$\varphi_{min} = P_{s\,min} - A_a$$

$$\varphi_{med} = \varphi_{min} + P_{mmn} + C_l + L_h$$

for portable indoor ground floor (class B) reception (in Tables 5.9 to 5.12):

$$\varphi_{min} = P_{s\,min} - A_a$$

$$\varphi_{med} = \varphi_{min} + P_{mmn} + C_l + L_h + L_b$$

generally:

$$E_{min} = \varphi_{min} + 120 + 10 \log (120\pi) = \varphi_{min} + 145.8$$

$$E_{med} = \varphi_{med} + 120 + 10 \log (120\pi) = \varphi_{med} + 145.8$$

C/N: RF signal to noise ratio required by the system (dB)

φ_{min}: minimum power flux-density at receiving place (dB(W/m^2))

E_{min}: equivalent minimum field strength at receiving place (dB(μV/m))

L_f: feeder loss (dB)

L_h: height loss (10 m. a.g.l. to 1.5 m. a.g.l.) (dB)

L_b: building penetration loss (dB)

P_{mmn}: allowance for man made noise (dB)

C_l: location correction factor (dB)

φ_{med}: minimum median power flux-density, planning value (dB(W/m^2))

E_{med}: minimum median equivalent field strength, planning value (dB(μV/m))

A_a: effective antenna aperture (dBm^2)

$P_{s\,min}$: minimum receiver input power (dBW).

For calculating the location correction factor C_l a log-normal distribution of the received signal is assumed. It should be noted that this standard deviation only relates to location statistics and the inherent inaccuracies of the propagation prediction method are not taken into account. The location correction factor may need to be re-assessed as more information becomes available.

The location correction factor can be calculated by the formula:

$$C_l = \mu * \sigma$$

where:

μ: distribution factor, being 0.52 for 70% and 1.64 for 95%

σ: standard deviation taken as 5.5 dB for outdoor reception.

See § 4.2.2.4.4 for σ values appropriate for indoor reception.

Planning studies for portable reception are based on a receiver able to handle signals of a broadband nature and the carrier to noise ratio requirement of a system will be moderate and may be as low as 2 dB in the case of a particularly rugged system. However, multi-channel services may need to be received by receivers having simple antennas. In practice, the possibilities for portable reception of signals with high bit rates and requiring a *C*/*N* of 20 to 26 dB will be very restricted due to the high signal level needed to overcome noise.

For these studies it has been assumed that a portable receiver and a receiver for fixed reception have same receiver noise figure, that is 7 dB.

5.2 Fixed antenna reception

Antenna diagrams and gains used for the derivation of the minimum median wanted signal levels for fixed antenna reception are given in § 4.2.

5.2.1 Minimum median power flux-density and equivalent field strength

The Tables below give the minimum median power flux-density and the equivalent minimum median field strength for 70% and 95% of location probability in Band I, III, IV and V. These values are related to the minimum power flux-density and minimum equivalent field strength at the receiving location. For Bands I and III an allowance for man-made noise has been included.

TABLE 5.1

Minimum median power flux-density and equivalent minimum median field strength in Band I for 70% and 95% location probability, fixed antenna reception

Receiving condition: Fixed antenna, Band I

Frequency	f (MHz)	65				
Minimum *C*/*N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Feeder loss	L_f (dB)	1				
Antenna gain rel. to half wave dipole	G_D (dB)	3				
Effective antenna aperture	A_a (dBm^2)	7.4				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–132.6	–126.6	–120.6	–114.6	–108.6
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	13	19	25	31	37
Allowance for man made noise	P_{mmn} (dB)	6				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–123.7	–117.7	–111.7	–105.7	–99.7
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	22	28	34	40	46

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–117.6	–111.6	–105.6	–99.6	–93.6
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	28	34	40	46	52

For 7 MHz channels, 0.6 dB is to be subtracted from the input signal power, the power flux-density and field strength values given in the Table above.

TABLE 5.2

Minimum median power flux-density and equivalent minimum median field strength in Band III for 70% and 95% location probability, fixed antenna reception

Receiving condition: Fixed antenna, Band III

Frequency	f (MHz)	200				
Minimum *C/N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Feeder loss	L_f (dB)	2				
Antenna gain rel. to half wave dipole	G_D (dB)	7				
Effective antenna aperture	A_a (dBm^2)	1.7				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–125.9	–119.9	–113.9	–107.9	–101.9
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	20	26	32	38	44
Allowance for man made noise	P_{mmn} (dB)	1				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–122	–116	–110	–104	–98
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	24	30	36	42	48

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–115.9	–109.9	–103.9	–97.9	–91.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	30	36	42	48	54

For 7 MHz channels, 0.6 dB is to be subtracted from the input signal power, the power flux-density and field strength values given in the Table above.

TABLE 5.3

Minimum median power flux-density and equivalent minimum median field strength in Band IV for 70% and 95% location probability, fixed antenna reception

Receiving condition: Fixed antenna, Band IV

Frequency	f (MHz)	500				
Minimum C/N required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Feeder loss	L_f (dB)	3				
Antenna gain rel. to half wave dipole	G_D (dB)	10				
Effective antenna aperture	A_a (dBm^2)	–3.3				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–119.9	–113.9	–107.9	–101.9	–95.9
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	26	32	38	44	50
Allowance for man made noise	P_{mmn} (dB)	0				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–117	–111	–105	–99	–93
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	29	35	41	47	53

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–110.9	–104.9	–98.9	–92.9	–86.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	35	41	47	53	59

TABLE 5.4

Minimum median power flux-density and equivalent minimum median field strength in Band V for 70% and 95% location probability, fixed antenna reception

Receiving condition: Fixed antenna, Band V

Frequency	f (MHz)	800				
Minimum *C*/*N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Feeder loss	L_f (dB)	5				
Antenna gain rel. to half wave dipole	G_D (dB)	12				
Effective antenna aperture	A_a (dBm^2)	–5.4				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–115.8	–109.8	–103.8	–97.8	–91.8
Min equivalent field strength at receiving place	E_{min} (dB(μV/m))	30	36	42	48	54
Allowance for man made noise	P_{mmn} (dB)	0				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–112.9	–106.9	–100.9	–94.9	–88.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	33	39	45	51	57

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–106.8	–100.8	–94.8	–88.8	–82.8
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	39	45	51	57	63

5.3 Portable antenna reception

Consideration of portable receiving antenna gain and feeder losses is given in § 4.2.2.

5.3.1 Minimum median power flux-density and equivalent field strength

The Tables below give the minimum median power flux-density and the minimum median equivalent field strength for location probabilities of 70 and 95% in Band I, III, IV and V.

TABLE 5.5

Minimum median power flux-density and equivalent minimum median field strength in Band I for 70% and 95% location probability, portable outdoor reception

Receiving condition: Portable outdoor (Class A), Band I

Frequency	f (MHz)	65				
Minimum *C/N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	–2.2				
Effective antenna aperture	A_a (dBm^2)	2.2				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–128.4	–122.4	–116.4	–110.4	–104.4
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	17	23	29	35	41
Allowance for man made noise	P_{mmn} (dB)	6				
Height loss	L_h (dB)	10				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–109.5	–103.5	–97.5	–91.5	–85.5
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	36	42	48	54	60

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–103.4	–97.4	–91.4	–85.4	–79.4
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	42	48	54	60	66

For 7 MHz channels, 0.6 dB is to be subtracted from the input signal power, the power flux-density and field strength values given in the Table above.

TABLE 5.6

Minimum median power flux-density and equivalent minimum median field strength in Band III for 70% and 95% location probability, portable outdoor reception

Receiving condition: Portable outdoor (Class A), Band III

Frequency	f (MHz)	200				
Minimum C/N required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	–2.2				
Effective antenna aperture	A_a (dBm^2)	–7.5				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–118.7	–112.7	–106.7	–100.7	–94.7
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	27	33	39	45	51
Allowance for man made noise	P_{mmn} (dB)	1				
Height loss	L_h (dB)	10				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–104.8	–98.8	–92.8	–86.8	–80.8
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	41	47	53	59	65

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–98.7	–92.7	–86.7	–80.7	–74.7
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	47	53	59	65	71

For 7 MHz channels, 0.6 dB is to be subtracted from the input signal power, the power flux-density and field strength values given in the Table above.

TABLE 5.7

Minimum median power flux-density and equivalent minimum median field strength in Band IV for 70% and 95% location probability, portable outdoor reception

Receiving condition: **Portable outdoor (Class A), Band IV**

Frequency	f (MHz)	500				
Minimum C/N required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	0				
Effective antenna aperture	A_a (dBm^2)	–13.3				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–112.9	–106.9	–100.9	–94.9	–88.9
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	33	39	45	51	57
Allowance for man made noise	P_{mmn} (dB)	0				
Height loss	L_h (dB)	12				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–98	–92	–86	–80	–74
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	48	54	60	66	72

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–91.9	–85.9	–79.9	–73.9	–67.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	54	60	66	72	78

TABLE 5.8

Minimum median power flux-density and equivalent minimum median field strength in Band V for 70% and 95% location probability, portable outdoor reception

Receiving condition: Portable outdoor (Class A), Band V

Frequency	f (MHz)	800				
Minimum *C/N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	0				
Effective antenna aperture	A_a (dBm^2)	–17.4				
Min power flux-density at receiving place	φ_{min} (dB(W/m^2))	–108.8	–102.8	–96.8	–90.8	–84.8
Min equivalent field strength at receiving place	E_{min} (dB(μV/m))	37	43	49	55	61
Allowance for man made noise	P_{mmn} (dB)	0				
Height loss	L_h (dB)	12				

Location probability: 70%

Location correction factor	C_l (dB)	2.9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–93.9	–87.9	–81.9	–75.9	–69.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	52	58	64	70	76

Location probability: 95%

Location correction factor	C_l (dB)	9				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–87.8	–81.8	–75.8	–69.8	–63.8
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	58	64	70	76	82

TABLE 5.9

Minimum median power flux-density and equivalent minimum median field strength in Band I for 70% and 95% location probability, portable indoor reception at ground floor

Receiving condition: Portable indoor ground floor (Class B), Band I

Frequency	f (MHz)	65				
Minimum *C*/*N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	–2.2				
Effective antenna aperture	A_a (dBm^2)	2.2				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–128.4	–122.4	–116.4	–110.4	–104.4
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	17	23	29	35	41
Allowance for man made noise	P_{mmn} (dB)	6				
Height loss	L_h (dB)	10				
Building penetration losss	L_b (dB)	8				

Location probability: 70%

Location correction factor	C_l (dB)	3				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–101.4	–95.4	–89.4	–83.4	–77.4
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	44	50	56	62	68

Location probability: 95%

Location correction factor	C_l (dB)	10				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–94.4	–88.4	–82.4	–76.4	–70.4
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	51	57	63	69	75

NOTE – Minimum median equivalent field strength values at 10 m a.g.l. for 50% of time and 50% of locations are expected to be:

– 5 dB lower than the values shown if reception is required in rooms at the first floor;

– 10 dB lower than the values shown if reception is required in rooms higher than the first floor.

For 7 MHz channels, 0.6 dB is to be subtracted from the input signal power, the power flux-density and field strength values given in the Table above.

TABLE 5.10

Minimum median power flux-density and equivalent minimum median field strength in Band III for 70% and 95% location probability, portable indoor reception at ground floor

Receiving condition: Portable indoor ground floor (Class B), Band III

Frequency	f (MHz)	200				
Minimum C/N required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	–2.2				
Effective antenna aperture	A_a (dBm^2)	–7.5				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–118.7	–112.7	–106.7	–100.7	–94.7
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	27	33	39	45	51
Allowance for man made noise	P_{mmn} (dB)	1				
Height loss	L_h (dB)	10				
Building penetration losss	L_b (dB)	8				

Location probability: 70%

Location correction factor	C_l (dB)	3				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–96.7	–90.7	–84.7	–78.7	–72.7
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	49	55	61	67	73

Location probability: 95%

Location correction factor	C_l (dB)	10				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–89.7	–83.7	–77.7	–71.7	–65.7
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	56	62	68	74	80

NOTE – Minimum median equivalent field strength values at 10 m a.g.l. for 50% of time and 50% of locations are expected to be:

– 5 dB lower than the values shown if reception is required in rooms at the first floor;

– 10 dB lower than the values shown if reception is required in rooms higher than the first floor.

For 7 MHz channels, 0.6 dB is to be subtracted from the input signal power, the power flux-density and field strength values given in the Table above.

TABLE 5.11

Minimum median power flux-density and equivalent minimum median field strength in Band IV for 70% and 95% location probability, portable indoor reception at ground floor

Receiving condition: Portable indoor ground floor (Class B), Band IV

Frequency	f (MHz)	500				
Minimum *C/N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\ min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\ min}$ (dB(μV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	0				
Effective antenna aperture	A_a (dBm^2)	–13.3				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–112.9	–106.9	–100.9	–94.9	–88.9
Min. equivalent field strength at receiving place	E_{min} (dB(μV/m))	33	39	45	51	57
Allowance for man made noise	P_{mmn} (dB)	0				
Height loss	L_h (dB)	12				
Building penetration losss	L_b (dB)	7				

Location probability: 70%

Location correction factor	C_l (dB)	4				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–89.9	–83.9	–77.9	–71.9	–65.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	56	62	68	74	80

Location probability: 95%

Location correction factor	C_l (dB)	14				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–79.9	–73.9	–67.9	–61.9	–55.9
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(μV/m))	66	72	78	84	90

NOTE – Minimum median equivalent field strength values at 10 m a.g.l. for 50% of time and 50% of locations are expected to be:

– 6 dB lower than the values shown if reception is required in rooms at the first floor;

– 12 dB lower than the values shown if reception is required in rooms higher than the first floor.

TABLE 5.12

Minimum median power flux-density and equivalent minimum median field strength in Band V for 70% and 95% location probability, portable indoor reception at ground floor

Receiving condition: Portable indoor ground floor (Class B), Band V

Frequency	f (MHz)	800				
Minimum *C/N* required by system	**(dB)**	**2**	**8**	**14**	**20**	**26**
Min. receiver signal input power	$P_{s\,min}$ (dBW)	–126.2	–120.2	–114.2	–108.2	–102.2
Min. equivalent receiver input voltage, 75 Ω	$U_{s\,min}$ (dB(µV))	13	19	25	31	37
Antenna gain rel. to half wave dipole	G_D (dB)	0				
Effective antenna aperture	A_a (dBm^2)	–17.4				
Min. power flux-density at receiving place	φ_{min} (dB(W/m^2))	–108.8	–102.8	–96.8	–90.8	–84.8
Min. equivalent field strength at receiving place	E_{min} (dB(µV/m))	37	43	49	55	61
Allowance for man made noise	P_{mmn} (dB)	0				
Height loss	L_h (dB)	12				
Building penetration losss	L_b (dB)	7				

Location probability: 70%

Location correction factor	C_l (dB)	4				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–85.8	–79.8	–73.8	–67.8	–61.8
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(µV/m))	60	66	72	78	84

Location probability: 95%

Location correction factor	C_l (dB)	14				
Minimum median power flux-density at 10 m a.g.l. for 50% of time and 50% of locations	φ_{med} (dB(W/m^2))	–75.8	–69.8	–63.8	–57.8	–51.8
Minimum median equivalent field strength at 10 m a.g.l. for 50% of time and 50% of locations	E_{med} (dB(µV/m))	70	76	82	88	94

NOTE – Minimum median equivalent field strength values at 10 m a.g.l. for 50% of time and 50% of locations are expected to be:

– 6 dB lower than the values shown if reception is required in rooms at the first floor;

– 12 dB lower than the values shown if reception is required in rooms higher than the first floor.

5.4 DTTB receiving environment

The planning of broadcasting services has traditionally embraced the concepts of three distinct reception zones – urban, suburban and rural – with there being different requirements for signal level and quality in the different zones. In line with the methods to measure radio noise as outlined in Recommendation ITU-R SM.1753 and the environmental categories for man-made noise as outlined in Recommendation ITU-R P.372, the following table presents the interpretation and translation of the environmental categories used in the context of noise measurements into the three common location categories used in the planning of broadcasting services.

TABLE 5.13

Interpretation of environmental categories in broadcasting context

<table>
<tr><th>SG 1 SM.1753-2</th><th>SG 3 P.372-10</th><th>SG 6 Broadcasting Usage</th></tr>
<tr><td>Remote rural</td><td>Quiet rural</td><td rowspan="2">Rural</td></tr>
<tr><td>Rural</td><td>Rural</td></tr>
<tr><td>Residential</td><td>Residential</td><td>Suburban</td></tr>
<tr><td>Urban</td><td rowspan="5">City</td><td rowspan="5">Urban</td></tr>
<tr><td>City</td></tr>
<tr><td>Industrial area</td></tr>
<tr><td>Railway</td></tr>
<tr><td>Road</td></tr>
</table>

CHAPTER 6

NETWORK PLANNING

6.1 Introduction

Throughout much of the World analogue television is highly developed, and most countries achieve more than 99% population coverage on at least two or three national networks. In parallel, a large number of local networks with lower coverage are operated. The need to achieve a large percentage of coverage leads to the use of very many television transmitters. The radiated power of these covers a wide range: from about 1 MW effective radiated power (e.r.p.) for major stations serving large areas, to less than 1 W e.r.p. for small stations intended to serve perhaps a few hundred people.

Analogue TV systems (PAL, SECAM) are very sensitive to interference from other analogue TV signals, and require high co-channel protection ratios (of the order of 30 dB to 45 dB, depending on the value of frequency off-set). Furthermore, adjacent channels cannot generally be used from the same transmitting location.

In addition, analogue TV systems cannot operate in a single frequency network (SFN) configuration, where neighbouring transmitters cover overlapping service areas with the same programme, on the same RF channel. Therefore the existing analogue services are planned in multi-frequency network (MFN) configurations, covering adjacent service areas with different RF channels. The same RF channel is re-used only in regions separated by a large distance, to avoid harmful co-channel interference.

Television coverage is therefore characterized by an intensive exploitation of VHF/UHF channels, with large areas where a given channel cannot be re-used because of the high protection ratios required by analogue systems. The total available RF channels (at maximum 10 channels at VHF and 48 at UHF, at least in Region 1) allow for only up to about 2 VHF programmes and 3 to 5 UHF programmes per coverage area, if high protection from interference is required. Higher spectrum exploitation could be obtained by using precision offset techniques.

It should be noted that in some countries a more intensive use of the spectrum is achieved, but in this case it is commonly found that many programmes show a very poor technical quality, due to interference or noise, especially in the less densely populated areas.

Analogue MFNs are usually based on relatively few high power transmitters, located where possible on hills or mountains. They are fed by cable or by radio links or sometimes by satellite or optical fibres. In addition to these, to follow precisely the terrain orography in the presence of hills or mountains or other obstacles, or to improve reception in highly populated regions, a very large number of lower power transmitters has been put in operation. They are usually fed by the signals broadcast by the higher power transmitters or, sometimes, by radio links.

In conclusion, the current analogue TV networks make use of a large percentage of the available VHF/UHF spectrum, and operate in MFN configurations with medium to high transmitter density. In each service area, a large number of RF channels cannot be re-used for high power analogue services, because of potential interference. Since digital systems can be significantly less sensitive to noise and interference, this spectrum could be used to introduce digital TV services, capable of operating at reduced e.r.p. levels. (However, care must still be taken to ensure that these digital services do not cause interference to existing analogue services).

The digital TV systems may offer improved RF performance over the analogue systems, in terms of spectrum efficiency and power requirements. First of all, digital systems allow multi-programming: in a single 8 MHz channel, 2 to 4 standard definition programmes (SDTV), at about 6 Mbit/s each, can be transmitted in time division multiplex. The total capacity (from 12 to 24 Mbit/s) can also be allocated to higher quality TV standards, such as enhanced definition TV (EDTV, requiring about 10 to12 Mbit/s per programme) or HDTV (requiring about 24 Mbit/s per programme). Of course, the higher capacity systems also have higher minimum C/N ratio requirements.

Digital systems can be significantly less sensitive to noise and interference, especially when the system spectrum efficiency is not too high and sophisticated modulation and error correction techniques are adopted. This can offer the possibility to operate at low e.r.p. levels (depending on the modulation) thereby reducing interference to existing analogue services.

Nevertheless, it should be taken into account that the best modulation and error correction systems show a very steep failure characteristic; a digital system can operate under severe reception condition without decoding errors, but an increase of 1 to 2 dB of the noise or interference level can suddenly interrupt the service. Therefore wide margins must be kept in the planning procedures to allow for service availability at a high percentage of locations and a high percentage of time.

Digital modulation and channel coding systems can achieve different trade-offs between spectrum efficiency and ruggedness against noise and distortion. For example, for fixed reception, a suitable spectrum efficiency can be of the order of 4 bit/s/Hz, (i.e., a useful bit-rate of about 24 Mbit/s in an 8 MHz channel), while for static portable reception a more suitable value may be 1 to 2 bit/s/Hz.

The introduction of digital terrestrial television in the immediate future has a main constraint which is the need to protect the existing analogue services. In addition, good digital service coverage is needed to provide an attractive development basis.

In many countries, due to the intensive spectrum exploitation, there is no possibility of access to previously coordinated but unused television networks or individual station assignments at least at relatively high power. In these countries, the use of new channel assignments is almost essential in order to introduce any new digital services.

6.2 Multi-frequency networks

The advantage of the multi-frequency planning approach is that a large part of the existing analogue network infrastructure may be re-used. This has obvious cost-saving implications for the broadcaster but should also provide benefits for the viewer. The latter will arise in any case where it is found possible to use channels for the digital transmissions from a particular site which are close to the channels used for the analogue transmissions from the same site, especially if the same polarization can be used. This should permit viewers to re-use their existing receiving antenna and feeder system. Some form of signal splitter or switch may be needed to permit separate feeds to the analogue and digital receivers although this could be avoided if the digital receiver provides loop-through facilities.

During the transition period of co-existence of analogue and digital services, and especially at the first introduction of digital services, it may be important not to place unnecessary difficulties in front of potential viewers and thus avoiding the necessity of a new receiving antenna system can be regarded as desirable.

Another aspect of multi-frequency planning is that it makes an inherent assumption that the existing analogue services, which currently serve more than 98% of the population in many countries, will remain in use for many years and that relatively few changes to the analogue stations will occur in that time. In particular, there are likely to be no generally-applied channel or site changes within the analogue networks.

However, it may be found desirable to introduce a limited number of channel changes, or even site changes, at some of the lower power analogue stations where this can be shown to have a significant impact on the implementation opportunities for digital stations and services.

In most countries there are few (or even no) opportunities for the introduction of new analogue stations with a significant population coverage. Opportunities exist for the introduction of new digital stations because of their greater immunity to interference and the ability of digital receivers to make use of lower input signal levels, given a suitable digital television system. Even so, these opportunities are limited by the need to protect existing analogue viewers from additional interference.

6.2.1 Conventional planning of MFNs

The term "conventional planning" is used to describe the situation where the network for a digital service has a similar configuration to that for an analogue service, at least for the higher power stations. This means that digital stations would use much the same transmitter sites as analogue stations and would have comparable transmitting antenna heights, although the e.r.p.s would be lower.

The primary reasons for the lower e.r.p. values are:

– the lower minimum field strength requirement;

– the need to protect existing analogue viewers.

It seems likely that in many cases the digital services would use channels close to those of the analogue services, for example the adjacent channels. Generalisations with regard to choice of polarization are not possible, but the use of the same polarization for the digital and analogue services would at least mean that existing domestic receiving antennas could be used for the digital services without change. Because the services come from the same site and because the digital service e.r.p. would be lower than that of the analogue service (for the reasons given earlier), there would be little risk of causing adjacent channel interference to the existing analogue service viewers. If this type of interference were to exist, it would be present for 100% of the time and therefore must be avoided. It must be noted that many of the technical aspects related to the use of adjacent channel transmissions from the same site still need to be investigated.

The coverage areas for the digital services are likely to be reduced in size compared with those for the analogue services, the amount of reduction being dependent on the C/N value required. Nonetheless, it is to be expected that significant population coverages can be obtained, provided that some degradation of the analogue service due to co-channel interference can be accepted.

It seems unlikely that channels could be found which would permit the duplication of existing analogue services by digital services at all higher power analogue transmitter sites in all countries.

6.3 Single frequency networks

6.3.1 General

In a single frequency network (SFN) all transmitters of a network use the same channel. They possess a common coverage area and cannot be operated independently. MFN and SFN concepts are based, in principle, on the same network topology, i.e. main transmitters with auxiliary gap fillers, if necessary.

OFDM, the modulation technique which enables the reception (and constructive summation) of more than one useful RF signal, is described in Part 1 of this Handbook.

6.3.2 Spectrum utilization efficiency

Spectrum utilization efficiency is regarded as a major advantage of the SFN concept as compared to the MFN approach. Spectrum utilization efficiency is an important feature in a situation where spectrum is scarce, for example, in the introductory phase of digital television, when most of the TV spectrum is still occupied by analogue services, as well as in the long term, when a large number of programmes has to be provided to make terrestrial digital television attractive for the consumer.

Current analogue services are operated as MFNs. Within the UHF band, using 40 of any available channels, 2 to 4 well-protected full-coverage analogue programmes can be achieved (depending upon the geographic situation of an individual country). Digital systems will be more efficient than this. Using MFNs, it can be expected that 3 to 6 full coverage networks could be implemented; with 4 programmes per channel, this would amount to 12 to 24 programmes. Using SFNs, it can be expected that the number of full-coverage networks (and the number of programmes delivered) is two to three times higher. If the target for coverage were changed to be the more densely populated areas only, the number of channels available could theoretically be around 40. All of these figures are based on theoretical considerations and the effect of practical considerations has to be checked case by case, for example, taking account of services in neighbouring countries.

6.3.3 Echo delay in SFNs

Terrestrial television broadcasting in VHF/UHF bands is characterized by attenuation and multipath propagation, due to the presence of obstacles and reflections in the propagation environment. Therefore the signal at the receiver is characterized by the presence of a main signal component, and of many echoes, with variable amplitude and delay (Ricean channel). In the case of portable reception, the principal signal can be absent (Rayleigh channel). The delay of these "natural echoes" is usually limited to 20 to 30 μs, corresponding to difference of propagation path of about 6 to 9 km.

The presence of SFN transmitters and gap-fillers produces a significantly more critical multipath propagation environment, introducing "artificial echoes" of high amplitude and long delay. These artificial echoes are superposed on the natural echoes. The range of delay times of the artificial echoes is proportional to the transmitter distance, and it is determined by the transmitter network geometry. For example, assuming that in a large SFN with transmitter distance $D = 100$ km the range of delay times is 330 μs, for a dense SFN with $D = 10$ km it would be only 33 μs.

6.3.4 Network gain

In an SFN many receiving locations can be covered by more than one transmitter, thus introducing a certain level of redundancy in the signal sources and improving the service availability, especially when portable reception is required. Particularly in portable reception, the field strength from a single transmitter shows statistical variations due to the presence of obstacles on the propagation path. This field strength variation can be reduced by the presence of several transmitters, located in different directions, since when one source is shadowed, others may be easily receivable. This is known as "network gain" (see also Annex 1 to Chapter 6). Numerical examples are given in § 6.4.2.

As a result of network gain, SFNs can be operated at lower power for the main transmitters and the field strength distribution is more homogeneous as compared to MFNs. The impact of these features for fixed reception may not be very prominent but portable reception with its non-favourable receiving sites and non-elaborate receiving antennas will benefit from these features to a large amount. The SFN approach seems to be the most reasonable way to provide satisfactory coverage for larger areas when portable reception is envisaged.

6.3.5 Planning of SFNs

Since the MFN and SFN approaches are based on the same network transmitter topology, SFNs can use, in principle, the network structure of the existing MFN analogue networks. In general, it can be expected that fewer gap fillers are needed with SFNs because of their more uniform field strength distribution.

The introduction of SFN-based DVB-T services is faced with the major problem that most (or all, in some countries) of the TV spectrum is occupied by analogue services which use a MFN structure. Even if some unused assignments exist (in a given country) which could be used for digital television, this is only of limited use for the introduction of an SFN-based large area service since a network can only operate in the SFN mode under the condition that its channel is cleared for the entire service area. If there are still analogue services using this channel – and this is probable as long as there is any national or regional analogue service in operation – the affected analogue transmitters would have to be shifted in frequency. Among these transmitters there will be main stations with a considerable amount of population coverage. It is questionable whether it makes sense, from the accompanying large cost efforts for broadcasters and consumers, to re-arrange an analogue service which will later be phased out. However, there may be suitable channel configurations that make this transformation practicable. In particular, for smaller area networks comprising only two or three high power transmitters the SFN approach may be applicable and attractive.

In some countries the possibility exists that in the UHF band one or more channels will be released for the implementation of digital services on a nation-wide scale. These channels are either not yet allocated to TV broadcasting, or they are already allocated but not used by TV services. These countries are offered a good chance to implement an SFN-based digital service on a national or regional scale, which potentially represents the introduction of an attractive long term scenario from the very beginning. In general, the use of these channels may not be possible on an entire nation-wide basis because of neighbouring countries which probably use these channels for analogue TV or other services.

6.3.6 Types of SFNs

SFNs can be implemented in different ways. Annex 2 to Chapter 6 gives definitions of the various types of SFN being considered.

6.3.6.1 Large area SFNs

Large area SFNs are built from at least two up to several dozens of high power transmitters together with associated medium and low power transmitters. They form the best way of exploiting the high spectrum efficiency inherent in the SFN approach.

In the case that a group of new frequencies is allocated to the new digital services, a straightforward approach is the introduction of some national SFNs, and smaller SFNs to cover the regional programming requirements. This scenario could equally apply to the long term situation for digital television, when the analogue services will have been phased-out.

On the other hand, in a country with fully developed analogue networks and few unused but accessible assignments it is unlikely that large-scale SFNs could be implemented. One possibility which might be explored would be to implement general channel changes at existing analogue stations. However, it seems unlikely that this could be undertaken in practice because of the widespread disruption of existing reception for the country concerned and for its neighbours.

6.3.6.2 Mini SFNs

In a mini SFN an existing main station and many (perhaps all) of its auxiliary low power stations would share the same channel. This is an attractive concept in terms of channel economy and homogeneity of the field strength distribution but there are still a number of technical matters to be examined.

Among these, is the fact that there are likely to be viewers of the existing analogue transmissions from the main station who are situated close to the relay station sites. Such viewers are likely to experience interference from the digital transmissions from the relay station if these use channels adjacent to those of the analogue services. In addition, in the case of fixed reception, the receiving antennas used by the viewers of the analogue services from the relay stations may not be suitable for reception of the new digital services because of channel differences. On the other hand, for portable reception the concept of mini SFNs provides an attractive means for increasing the coverage of digital television.

6.4 Multiple signal effects

In general, the reception of digital services is faced with a multi-signal environment, multiple interference as well as multiple wanted signals in the case of SFNs. To assess the wanted and unwanted resultant field strengths the individual signals have to be combined. Since signal strengths are described by statistical quantities they have to be combined statistically.

Basically, this is true for both location and time statistics. However, it is usual to treat them in different ways. Time statistics are taken account of by using tabulated field strength propagation curves for the appropriate time percentages. Location statistics are dealt with by using field strength distributions.

General aspects of time and location statistics and mathematical methods to perform statistical summation are described in Chapter 3. The impact of signal summation effects on planning methods and parameters is discussed in the present section.

6.4.1 Single signals and propagation margins

Location statistics of an individual (logarithmic) field strength originating from one transmitter is described by means of a normal distribution which is characterized by two parameters, mean value and standard deviation. Accordingly, the power of the signal is then distributed log-normally.

Coverage probability targets play a key role as planning parameters for a digital system is discussed (see Chapter 4). These target figures are related to the field strength distribution parameters. 50% coverage probability is determined by the mean value of the distribution, for the calculation of higher (and also lower) coverage probabilities both mean value and standard deviation of the signal distribution are needed.

In the case of a single signal, where the distribution parameters are known a priori, propagation margins to cater for higher coverage probabilities, as described in Chapter 3, are easily calculated, e.g., the propagation margin for a 95% coverage probability is given by 1.64 σ where σ denotes the standard deviation. This is how the minimum median field strengths for planning are determined in Chapter 5. The same applies to propagation margins for protection ratios when one wanted and one unwanted field are involved.

6.4.2 Multiple signals and network gain

When a multi-signal situation is encountered, the parameters of the resulting sum signal distribution are no longer known a priori. Mean value and, especially, standard deviation strongly depend on the particular signal configuration having to be determined by means of statistical procedures. As a consequence, minimum field strengths and propagation margins to be used in coverage calculations no longer have fixed values, they rather become variables depending on the number, strength and spread of the individual single fields. However, two general trends can be identified. Firstly, the mean value of the combined sum signal is larger than the arithmetic sum of the mean values of the individual signals and, secondly, the standard deviation of the combined sum signal is smaller than that of the individual signals, both facts creating the effect of network gain (in the case of wanted signals).

The following examples may illustrate the significance of field strength summation effects. A coverage probability target of 95% is assumed in the examples and the standard deviation of the single fields is chosen to be 5.5 dB. The examples are calculated for receiving systems with a non-directional antenna pattern thus taking maximum benefit from the space diversity introduced by the active echoes in an SFN. For receiving systems with directional antennas, e.g., fixed roof top reception, the effect of network gain is reduced, since active echoes are attenuated by the space selectivity of the antenna.

Maximum statistical network gain is achieved if the contributing fields are of equal strength. In the case of, e.g., 3 single signals it amounts to 5.1 dB. This means, it would allow for an overall power reduction in an SFN by a factor of 3 as compared to a single transmitter coverage.

Not all locations covered by an SFN benefit to such an amount from network gain. As a second example, an edge location of a typical, closed 7-transmitter hexagon SFN is chosen. Though situated on the fringe of the coverage area it still experiences a network gain of about 4 dB, reducing the minimum field strength for planning by that amount.

Similarly, propagation margins for protection ratios are reduced by signal summation effects. Again, as an example, an edge location in a 7-transmitter hexagon SFN is chosen, now interfered with by a second, identical SFN situated at the re-use distance, i.e. fairly close. Here field summation effects for both wanted and unwanted signals lead to a reduction of the necessary propagation margin of about 4.5 dB, indicating the ability of the wanted SFN to cope with a roughly three times higher interference without losing the coverage probability target.

The examples show that signal summation effects in SFNs may impact the coverage of a digital service to a significant amount.

It has already been stated that signal summation effects increase the mean value and lower the standard deviation of the resulting sum signal distribution as compared to the outcome of the standard treatment. This is an important finding, since it gives the possibility to fix the results of the standard treatment as an upper bound for initial planning estimates. Allowing for some additional implementation margin, they form an appropriate basis for planning when detailed information about the transmitter characteristics of a network is not available, e.g., when setting up an allotment plan.

On the other hand, detailed planning, e.g., installing an assignment plan or implementing a real transmitter network, has to take account of signal summation effects. Propagation margins for minimum field strengths and protection ratios then no longer form suitable planning parameters. They have to be replaced by the more basic coverage probability targets. Their relation to the sum distribution parameters of the wanted and unwanted sum fields is described in Chapter 3.

6.4.3 Multiple interference and self interference

Time statistics for interfering fields are taken account of by basing calculations on 1% time propagation curves, whereas wanted field calculations are based on 50% (or 99%) time propagation curves. Statistical signal summation effects for interfering fields with respect to location statistics are effective, in principle, in the same manner as described for wanted fields in the previous section. However, their impact on coverage calculations is not that important because of the asymmetrical characteristics of the sum field distribution. Therefore, it is often justified to treat multiple interference with simpler statistical procedures.

In considering SFNs it should be recognized that not all the transmitters in a network will contribute to the wanted signal. Depending on the system and network parameters, such as guard interval and distances between transmitters, some signals may become interferers. This effect is called the self interference of the SFN. It is of larger importance for the DVB-T system with its higher protection demand than for T-DAB, and has to be met by careful network design.

With respect to signal summation, self interference fields are treated as “normal” unwanted signals. 1% time propagation curves are used, and they are added to the other source of possible interference from outside the SFN.

However, a problem arises with the treatment of the interfering and the contributing signal parts. Usually, unwanted field strengths are calculated on the basis of 1% time propagation curves and wanted fields on the basis of 50% time propagation curves. If both parts emerge from the same field the question is whether to adopt the 1% time or the 50% time propagation curves as the basis of the calculation. A possible treatment could be to base it on 50% time propagation curves as long as the major part of the signal shows a contributing behaviour, otherwise to base it on 1% time propagation curves.

In addition, a similar problem is encountered with location statistics. Usually, wanted and unwanted signals are treated as statistically independent. In the case of an interfering and a contributing signal part emerging from the same field this is obviously not true. The impact of this “self correlation” effect on coverage calculations has not yet been evaluated and needs further investigation.

6.4.4 Correlation

Spatial correlation between RF signals have been reported to be of non-vanishing significance for the evaluation of the coverage of broadcasting services. Nevertheless, no generally agreed assessment of correlation has yet been established.

Basically, correlation is not a signal summation effect, it may also occur in the presence of only one wanted and one unwanted field. In this case, correlation increases coverage for a given configuration of wanted and unwanted field strengths.

In a multi-signal situation, the opposite effect can be observed. Correlation between wanted signals reduces the network gain of a transmitter network and correlation between unwanted signals increase their interference potential, both effects lowering coverage for a given configuration of wanted and unwanted signal strengths.

In view of the uncertain general assessment of correlation and the different effects they produce with respect to coverage, it seems to be justified to neglect them in planning calculations, at least at the present time.

ANNEX 1

TO CHAPTER 6

Characterization of theoretical SFNs

Studies have been undertaken to characterize the impact of the digital TV system performance (guard interval, protection ratios) on the service availability in the service area, versus the dimension of a SFN (large or dense). The analysis methodology has been defined and a number of results have been obtained for DVB-T networks.

In a single frequency network, all transmitters broadcast exactly on the same RF channel. The service areas of these transmitters are overlapping, and the transmitted signals are fully synchronized.

Compared with a conventional multiple frequency network (MFN), an SFN allows significant improvements in spectrum utilization, but it imposes heavy constraints in the design of the transmission system. In fact the useful signal is interfered with by the artificial echoes produced by the other transmitters, characterized by large amplitudes and long delays. The echo delays depend on the difference of the propagation path lengths, and can be of the order of some tens to some hundreds of microseconds, depending on the transmitter distance (e.g., a path difference of 10 km corresponds to a delay of about 33 μs).

These SFN echoes are superposed on the echoes generated by the obstacles (mountains, hills, buildings) often present in the propagation environment (multipath echoes). In general the delays associated with the multipath echoes are shorter than 20-30 μs. This Annex refers only to the artificial SFN echoes, while the natural multipath echoes are neglected.

The performance of a digital television system in an SFN heavily depends on the echo delay and amplitude characteristics. Only OFDM systems are considered here, with powerful channel coding schemes suitable to operate under very severe multipath propagation conditions, such as those produced in SFNs. These systems can process the echoes (natural or artificial) in such a way that, up to a certain amount of delay spread, all the fields contribute constructively to the wanted signal. This offers the possibility of establishing SFNs.

A uniform semi-infinite lattice network is used to study the theoretical performance of a SFN, usually based on hexagonal coverage areas.

The propagation model described in Recommendation ITU-R P.370 is usually used to evaluate the field strength produced by each transmitter in the network in a given receiving point. Mean values for the field-strength location distributions are taken for the land 50% location/50% time-curves for the useful components of the SFN signal. For interfering signals, it is more usual to use the 50% location/1% time-curves, which correspond to greater impairment. Calculations are carried out with a location variation standard deviation of 5.5 dB for individual signals.

For DVB-T services, both fixed reception (with roof-top directive antennas) and portable reception are of interest. Recommendation ITU-R P.370 propagation curves give field strengths which are valid for reception with an antenna situated 10 m above ground level (a.g.l.). This model is suitable for DVB-T fixed reception with roof-top antennas. For portable DVB-T reception, it is necessary to subtract some 10-20 dB from the field strength value predicted for 10 m a.g.l.

A location coverage of 95% is generally demanded for DVB-T services, at least for fixed reception and roof-top antennas. It is well known that in this high-probability domain, the statistical network gain of a SFN provides a significant part of the overall coverage. Particularly for portable reception in shadowed areas, the space diversity of the signal sources provides a reduction in the field-strength variations and improvement in the coverage. It is therefore necessary to treat the statistical aspects as thoroughly as possible. Non-statistical approaches lead to severe under-estimation of the coverage and give an incorrect impression of the validity of SFN concept.

The statistical summation of the field-strengths is performed by means of a "Monte Carlo" technique. For every location, the signal components from the various transmitters are randomly generated with the suitable statistical distributions, and according to the delay and the system guard interval, an equivalent aggregate carrier to noise C/N and interference ratio C/I is obtained. For a first investigation of the SFN performance, a simplifying assumption of "interference limited network" (i.e. no noise) can be done. In the case of roof-top directive antenna, it is assumed that it is pointed in the direction of the strongest transmitter (although this may not always be appropriate).

The combination of C/N and C/I is compared with the system threshold (relevant to a severe multipath environment, such as a Rayleigh channel), to determine whether this particular receiving point is served or not served. To achieve statistically significant results, this procedure is repeated thousands of times for each "small area" or "pixel", and then repeated on a regular grid over the complete service area.

This method can provide the coverage probability for the service area and overall aggregate values of percentage of served locations. An important indicator of the network performance is the percentage of locations served in the worst pixel. Alternatively, the performance can be quantified by means of the percentage of pixels for which a given coverage target is achieved.

This analysis can allow the optimisation of the system parameters (guard interval, threshold C/N and C/I) given a theoretical network configuration (transmitter distance, antenna height), or alternatively, permits a choice of network parameters given a digital modulation/coding system.

ANNEX 2

TO CHAPTER 6

Definitions related to transmitting stations and single frequency networks for digital television services

Transmitting stations for digital services

High power station:

A station with an e.r.p. greater than 10 kW and an effective antenna height usually greater than 150 m.

Medium power station:

A station with an e.r.p. in the range 100 W to 10 kW (inclusive) and an effective antenna height usually in the range 75 to 150 m.

Low power station:

A station with an e.r.p. less than 100 W and an effective antenna height usually less than 75 m.

Single frequency networks

Large area SFN:

An SFN which contains more than one high power station together with any associated medium and low power stations, usually with a composite coverage greater than about 10 000 km^2.

Mini SFN:

One high power station together with at least one (and probably several) associated medium or low power stations.

National SFN:

An SFN covering a whole country.

Regional or Local SFN:

An SFN covering part of a country.

CHAPTER 7

PLANNING METHODS

7.1 Introduction

Terrestrial digital television services can be planned using assignments and/or allotments. The methods for both these approaches are given in this section and they can be used in preparation for and during, an international planning conference. The planning of an individual station (or group of stations), intended to provide coverage to a specified area, is also covered.

7.1.1 Assignment planning for terrestrial digital television

In the past, terrestrial television planning in Europe has been by way of assignment conferences. In assignment planning, a significant amount of individual station planning is needed to prepare for a planning conference. Stockholm '52 and Stockholm '61 were two such conferences related to terrestrial television and European broadcasters have gained much experience in assignment planning, particularly since the planning methods and criteria of the '61 conference are still applied to analogue television planning.

Assignment planning for terrestrial digital television is appropriate where:

– a broadcaster wishes to use an existing transmission infrastructure for environmental and economic reasons;

– there is a need to share spectrum with existing analogue television transmissions in the same country;

– where digital assignments are envisaged.

At the completion of the assignment plan, the locations and characteristics of the transmitters in the planning area are known, and the transmitters can be brought into service without further coordination.

7.1.2 Allotment planning for terrestrial digital television

The alternative possibility of obtaining allotments at a conference has received considerable attention in recent years, partly because of the opportunities offered by SFNs. Allotment planning for SFNs is probably best carried out where spectrum is available or can be made available in large region or throughout a country. Allotments may also be applicable for MFN planning where a country has no plans to use specific transmitter sites and wishes to retain some flexibility for late future.

In allotment planning, a channel is "given" to an administration to provide coverage over all or part of its territory, but as there are no agreed definitions of words such as "national" or "regional", care is needed in their application. At the allotment planning stage, in general nothing is known of the actual location of the transmitter sites, nor of the specific transmission characteristics to be used. The only parameters available are a definition of the area to be covered and the channel to be used. Thus in order to carry out the planning exercises it is necessary to define some reasonably realistic reference transmission conditions so that any necessary compatibility calculations can be made.

To implement networks within an allotment it is necessary to convert the allotments into individual transmitter assignments.

7.1.3 Planning constraints for coordination

The constraints on coordination are the same for allotment and assignment planning. These are:

– compatibility with existing analogue television services;

– compatibility with other services (Chapter 8);

– mutual protection of digital television allotments or assignments.

7.2 Digital television planning in the United States of America

The planning of digital terrestrial television broadcasting services is subject to many considerations. Planning can be accomplished using assignments and/or allotments. Geographical variables such as border and terrain must be taken into account. There are several parts of the world where television broadcasting planning is subject only to the requirements of individual administrations with special consideration given to bilateral planning in border areas. Taking into account these and other factors, various methods have been developed and used to optimize frequency assignment and allotment planning.

To handle this task, sophisticated operations research methodology has led to development of computer software used to optimize the allotment of channels for DTTB in the United States. Such software incorporates methodologies for calculating expected service areas and the quantification of intra-service interference effects. This software can be used under an analogue-to-digital service replication approach.

Based on these needs, a computer model was developed to optimize and balance various policy objectives. This model was used to generate a DTTB allotment plan which also takes into account existing analogue television assignments. The software incorporates an operations research optimisation methodology known as “simulated annealing”. This methodology employs a system of penalties that attach to conditions that fall short of specified objectives. To achieve an optimum condition, the simulated annealing method seeks to minimise the sum of these penalties, or “costs”. Its use permits the definition and quantification of costs based on various conditions.

The computer model permits the rapid computation and analysis of service area coverage provided by analogue and digital television systems (NTSC and ATSC, respectively, in the United States), both on an overall cumulative basis and for individual stations. The plan could be modified to consider other analogue and digital television systems. The actual service area of an individual analogue station is defined as the area within the station’s predicted service contour, reduced by any interference; and is computed based upon the actual transmitter location, power, and antenna height. The actual service area of a digital station is defined as the area contained within the station’s noise-limited contour, reduced by the interference within that contour. Digital station coverage calculations assume locations and antenna heights identical to those of the companion analogue station with a radiated power generally sufficient to achieve noise-limited coverage equal to or less than the companion analogue station's coverage.

There may be instances where the allotment of channels in specific local situations can best be resolved on a case-by-case basis. The software, therefore, is able to merge specific local designs into complete tables and, where necessary, make changes in other allotments to preserve a balance of the specified policy considerations. This capability allows the incorporation of allotment/pairing agreements that broadcasters may reach in any negotiated settlements.

In the United States, cessation of analogue television broadcasting is anticipated in the near-term. Issues related to the protection of future digital service areas and the licensing of new services have been considered. For example, trade-offs can be considered between the radiated power of new digital facilities and the extent to which those facilities will be protected from interference after the shutdown of analogue television services. One significant advantage of digital broadcasting is the fact that assignment or allotment plans can be developed that more efficiently use spectrum that was previously unavailable for use due to the need to protect the so-called taboo channels associated with analogue television. When an analogue service is discontinued, it becomes possible to reassign spectrum to other services and to provide decisional trade-offs in the construction of digital facilities. In the United States, a “core” spectrum was defined, consisting of channels 2-51, each channel being 6 MHz wide. A process was developed to recover the spectrum in channels 52-69 for other purposes.

The allotment plan accommodates all eligible existing broadcasters, replicates existing service areas, and ensures effective and efficient spectrum management. The plan was designed to facilitate the short-term recovery of 60 MHz of spectrum at channels 60-69 and longer-term recovery of an additional 78 MHz of spectrum at the end of the transition period. Policy development efforts are ongoing in the United States and decisions on use of that spectrum (channels 60-69) are in progress. The United States has allocated 24 MHz for public safety uses, and is currently considering the disposition of some, or all, of the remaining 36 MHz.

The United States frequency plan (The U.S. DTV Table of Allotments) provides channels for digital television operations to all eligible broadcasters. Those eligible for a digital television channel in the plan include: parties licensed to operate a full service television broadcast station and those holding a construction permit for such a station. These eligibility criteria follow the initial eligibility set forth in national law. This approach was intended to promote an orderly transition to digital television by ensuring that all eligible full service broadcasters are able to provide digital services.

While all eligible broadcasters have been provided with a channel that will allow them to provide a digital television service to an area that is generally comparable to any existing analogue television service area, service replication is no longer required. If individual broadcasters elect to do so, they may construct digital television stations that do not replicate the existing analogue service coverage, but if they choose to do so, they will not be protected beyond the defined edge of the new digital service area. As an initial stage, broadcasters were allotted digital television channels that will replicate the service areas of their existing stations after the end of the transition period, that is to say, after the analogue television stations have ceased to operate. Over 50% of all broadcasters received a digital television channel that provides 100% replication during the transition period, and over 93% of all broadcasters received a channel that provides a service area replication of at least 95% during the transition period.

7.3 Digital television planning in Europe

It is likely that terrestrial digital television planning in Europe in the reasonably near future will have to be based on a mixture of assignment and allotment planning. The information given in § 7.4 to 7.9 is based upon ideas developed in Europe essentially for assignment planning, but it could also be applied to other parts of the world with little change.

7.4 Elements of planning

7.4.1 Planning criteria

The planning criteria consist of the following elements:

– Protection ratios (Chapter 2).

– Percentage of time for which protection is required (Chapter 4).

– Percentages of locations for which protection is required (Chapter 4).

– Signal levels and *C*/*N* values (Chapter 5).

The range of *C*/*N* values under discussion for different digital systems and their variants is very large and the differences between some of the *C*/*N* values are smaller than the inherent accuracy of the propagation prediction methods available (including the assumptions necessary in the case of portable reception). For planning the introduction of digital television it is usually necessary to restrict the interim planning studies to a representative sub-set of *C*/*N* values.

7.4.2 Propagation prediction methods

The signal level prediction method is Recommendation ITU-R P.370 for individual transmitters and the statistical approach given in § 3.4 for SFNs.

7.4.3 Combination of multiple signals

The methods to combine the wanted and unwanted signals are given in § 3.4.

7.4.4 Databases for planning

Terrestrial digital television broadcasting will be primarily accommodated in the same bands as analogue television. Extensive compatibility calculations will be needed in planning studies and subsequent coordination to facilitate this accommodation. These calculations require databases containing:

– analogue transmitter stations;

– digital transmitter station assignments;

– allotment plans containing, for example, areas to be covered;

– details of other services.

7.5 Procedures for the protection of analogue television services

It is necessary to ensure that the existing and planned analogue television services continue to be protected. This applies to both assignment and allotment planning for digital television. In either case, before a channel is chosen for a digital television service, it is necessary to establish the size of the coverage area for each coordinated analogue station. This may be done by the calculation process given in § 7.7 or by specifying boundary test points in special cases. Examples of the latter would be where a calculated coverage area would cross a national boundary or in mountainous terrain where propagation predictions based on Recommendation ITU-R P.370 cannot be expected to give accurate results.

7.6 Definitions of test points

Two categories of test points are needed. One category represents the coverage area for a given station, or SFN, while the other represents country boundaries.

All test points are defined by their geographical coordinates.

7.6.1 Test points representing coverage areas

The transmitter site will normally be inside the contour described by the test points, however, in special cases the transmitter can be located outside of this area.

For small stations, i.e. stations with a coverage area whose width is less than, say, 5 km, only one test point, located at the transmitter site, may be sufficient. However, up to 36 test points can be defined if necessary. If only one test point is given, no receiving antenna directivity should be assumed.

For stations with a coverage area whose width is 5 km or more up to 36 test points are used. These test points may be located on radials at 10° intervals.

If the contour of the coverage area crosses a country boundary, the test points in this area are located at the crossing points between a radial and the boundary unless otherwise agreed by the concerned administrations.

7.6.2 Test points at the country boundary

A larger number of test points can be used to represent the boundary of a country.

The location of test points at a boundary should be agreed between the countries sharing this boundary and be used as boundary test points by all other countries, where necessary.

The set of test points representing the boundary of a country should be a complete individual set, as should a set representing a coverage area.

7.6.3 Availability of test point locations

The location of test points, i.e. their geographical coordinates, should be commonly available to all relevant administrations in order to facilitate calculations of interference into other countries or coverage areas of stations in other countries.

7.7 Calculation of the location for test points representing coverage areas

To calculate the coverage area of a television station on a given channel, two elements are necessary:

– the parameters particular to an individual transmitting station (coordinates, height of the antenna, radiated power, etc.) which are used to calculate the wanted signal;

– the system parameters such as the protection ratios, which are used to calculate the individual nuisance field strengths and the usable field strength, and the minimum median field strength.

These calculations should take into account:

– interference from analogue television assignments;

– interference from digital television assignments.

The individual nuisance field strength, E_n, is the field strength of an unwanted signal to which has been added the relevant protection ratio and receiving antenna discrimination. It is calculated as follows:

$$E_n = E + PR + \mathrm{A}$$

where:

E: field strength of the unwanted signal. The appropriate time percentage according to the wanted signal is to be chosen (see Note 1)

PR: appropriate protection ratio (see Note 1)

A: receiving antenna discrimination (taking into account polarization discrimination), ($A \leq 0$);

and all quantities are expressed in dB or dB(µV/m).

NOTE 1 – In the case of a wanted digital service the 1% time 50% location field strength of the unwanted service is to be chosen. In the case of a wanted analogue service the larger of the 1% time 50% location field strength of the unwanted signal together with the protection ratio for tropospheric interference and the 50% time 50% location field strength of the unwanted signal together with the protection ratio for continuous interference is to be chosen.

The usable field strength is the minimum value of the field strength necessary to permit a desired reception quality, under specified receiving conditions, in the presence of natural and man-made noise and interference. The usable field strength is calculated by combining the individual nuisance fields and the minimum median field strength taking account of the effect of the location variation of the wanted and interfering signals. The combination may be done by means of the power sum method or by one of the statistical methods described in detail in Chapter 3.

The test points representing a coverage area can thus be determined in three stages:

Stage 1 Calculation of noise limited coverage area

Using Recommendation ITU-R P.370, the locations of the noise-limited test points are found, which represent the area that could be served if there were no interference. This area may be approximated on the basis of up to 36 radials, using the e.r.p. and the effective antenna height. For each radial, that location is determined where the field strength of the wanted transmitter equals the minimum median field strength.

Stage 2 Identification of interferers

The impact of co-channel, adjacent channel and image channel interference from other transmitters is calculated for each wanted station and each noise-limited test point from Stage 1. First, the sub-set of possible interferers is established. This consists of the stations which can produce a nuisance field which is no more than 15 dB below the minimum median field strength at any of the noise-limited test points from Stage 1. (The value of 15 dB is used to ensure that all signals are included which could cause an increase in the usable field strength of more than 0.3 dB. Any other valued agreed between the relevant administrations could also be used.)

Stage 3 Calculation of the test points for the interference limited coverage

The individual nuisance field strength En caused by each of the interfering stations in this sub-set of interferers is calculated at each of the noise limited test points from Stage 1 (see Fig. 7.1). The usable field strength is calculated for each of these test points.

In the case of no interferers the usable field strength at a test point is equal to the minimum median field strength, no further calculation is required, and the coverage radius is that of Stage 1 above (see also Fig. 7.1).

If the usable field strength at a test point is greater than the minimum median field strength plus the combined location correction, it is then necessary to find the new coverage radius on this bearing at which the field strength from the wanted station equals the usable field strength.

Because, in general, the coverage radius thus calculated will not equal the radius previously calculated for the same bearing and thus the nuisance field strengths will change, the process of the previous paragraph is repeated to obtain a close approximation to the required coverage radius on each of the bearings.

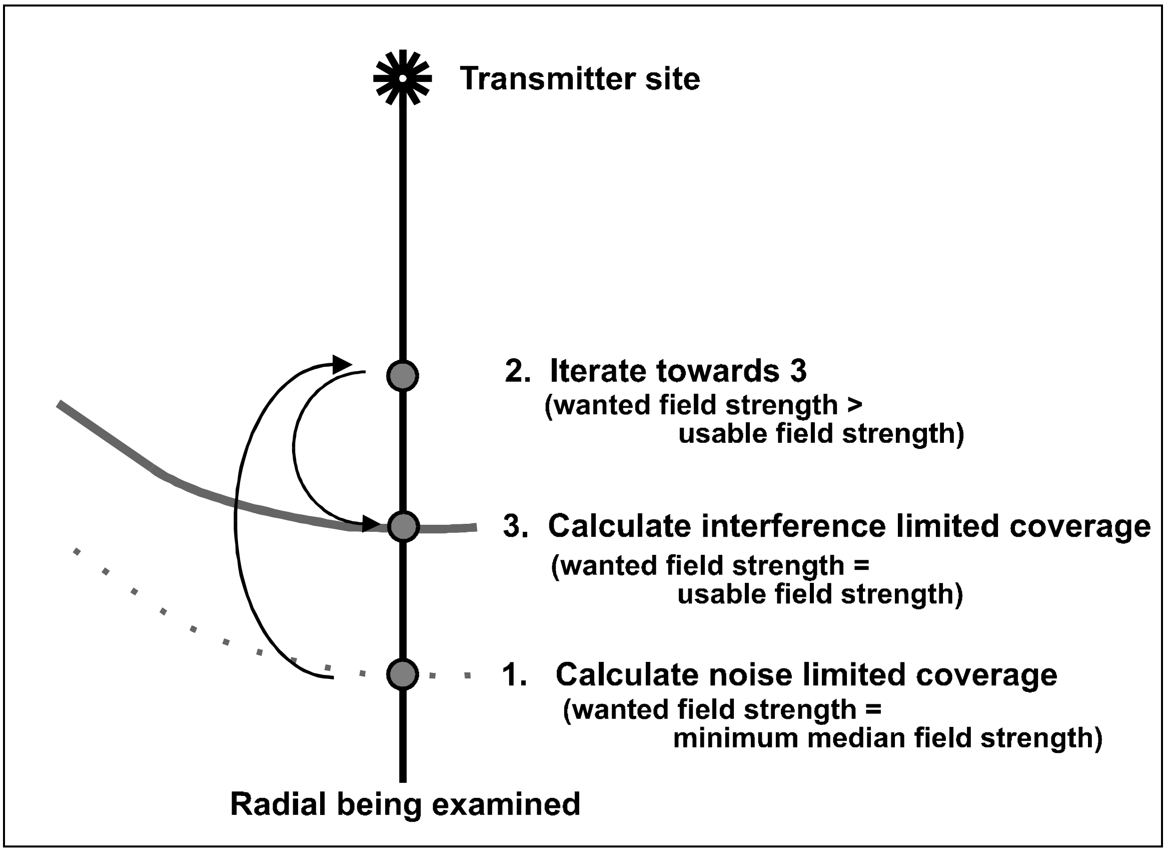


FIGURE 7.1

Illustration of the calculation of location of test points for the interference limited coverage

It must be noted that a given station will normally have different coverage areas on different channels and that this can be important when considering the relative coverage of digital and analogue services.

7.8 Method for combination of signals (power sum method)

The power sum method is a procedure in which individual field strengths are combined such that the power of the resultant field strength equals the sum of the powers of the individual field strengths. If the (logarithmic) field strength of a single signal is denoted by E_i and is expressed in dB(µV/m), the combined field strength E_Σ is given by:

$$E_\Sigma = 10 \times \log_{10}\left(\sum_{i=1}^{n} 10^{\frac{E_i}{10}}\right)$$

where n is the number of individual field strengths.

7.9 Planning methods for digital television assignments

This section describes a method for finding **potentially** available channels for individual new digital television transmitters in the case of MFN planning. The basic idea is to determine the amount of power which may be radiated in each channel without causing excessive interference (that is beyond a pre-determined increase of say, 0.3 dB) to existing analogue services or an equivalent increase for digital services. The results obtained by using the process described in § 7.9.1 and § 7.9.2 may be used to identify which channels could provide useful digital coverage from any given transmitter site.

The actual channel or channels to be used at that site, rather than at any other, nearby, site would need to be selected during a coordination process, which would also need to take account of the potential mutual interference between digital television transmitters which could use the same channel.

An alternative, but related, approach would be to identify the channels and the station characteristics for a group of digital assignments arranged in a mini SFN. The basic concept remains the same as that described above but the actual processing is significantly more complex.

Of course, the station characteristics determined as in § 7.9.1 and agreed in a coordination process could be used as the basis for the implementation of a mini SFN, provided that the latter did not have a larger interference potential (at any of the test points for other stations agreed during the coordination process).

7.9.1 Establishment of the characteristics of a digital television station

For a given transmitting site, the following characteristics have to be established for a digital television station:

- channel(s);
- polarization;
- transmitting antenna effective height above mean terrain;
- maximum e.r.p.;
- horizontal radiation pattern.

Information on the effective antenna effective height and radiation pattern is required at 10° intervals, starting at true north.

Many iterations may be required in the choice of the characteristics which will give maximum coverage of the digital service, while at the same time not causing unacceptable levels of interference to existing analogue television stations and other services.

For a particular choice (or trial choice) of channel and polarization, the maximum permissible radiated power of the digital service must be determined.

For each analogue television station which may suffer interference from the DVB-T station with the proposed characteristics, the following steps should be carried out:

Step 1: at each point representing the coverage area of the analogue station, calculate the power sum of the nuisance field strengths of other existing analogue stations. These values have already been established in the course of the procedures described in § 7.7;

Step 2: at the same points, calculate the combined nuisance field of the proposed DVB-T station together with the existing analogue television stations;

Step 3: compare the combined nuisance field strengths calculated as above. If there is an increase of no more than an agreed value, say 0.3 dB, then the e.r.p. of the digital station is regarded as acceptable (for the purpose of establishing a list of potentially available channels). Otherwise, the e.r.p. is adjusted so that an acceptable value of interference is achieved.

In this way the maximum permissible radiated power of the digital television station in all directions can be determined.

7.9.2 Establishment of the size of digital television coverage areas

When the characteristics of the digital station have been established, its coverage can be calculated for a specified digital television system variant requiring a certain *C*/*N*. This calculation should take into account:

– interference from analogue television stations to the potential digital station coverage area;

– interference from other digital television services;

– interference from other services.

In order to do this, the signal summation methods described in Chapter 3 should be used. As a result of this process, for each small area, the percentage of locations served is obtained. The predicted digital service area can also be calculated using the method given in § 7.7.

7.9.3 Establishment of the characteristics of a group of digital television stations in a mini SFN

The basic concept employed would be similar to that of § 7.9.1 and the details given there are not repeated. It is necessary that the set of transmitter sites to be regarded as a mini SFN be identified to the computerised process. The additional complexity, compared with the process given in § 7.9.1, is that multiple combinations of power levels radiated from each of the transmitters in the mini SFN would need to be considered. It would be necessary for a choice to be made among the many possibilities, on each possible channel, by considering the coverage which could be achieved by the given combination under study. This would involve an interactive process in which intermediate results produced by a computer process were studied to determine which combination of digital power levels is regarded as optimum for each channel.

7.10 Protection of digital television services

Because of the rapid failure of digital reception when the level of the useful signal decreases below its "minimum" value, the target for the percentage of locations nominally covered at any edge – where edge means any transition between a covered area and a non-covered area – of the coverage area has to be much higher for digital systems than the value used for analogue television systems. For coordination, the reference conditions, including the percentage of location value, would need to be agreed between the relevant administrations.

Since the reception conditions of the **actual** implementation of a digital television service may differ from the **reference** reception conditions agreed, the test points representing a digital assignment do not necessarily lie on the boundary of the actual coverage area of that digital assignment. The test points may lie inside or outside the actual coverage area of the digital assignment.

CHAPTER 8

INTERACTION WITH OTHER SERVICES

8.1 General

Broadcasting does not have exclusive access to the frequency bands allocated to the broadcasting service. A number of sharing situations exist and these vary from one country to another, both in terms of the "other service" involved and its status in Radio Regulatory terms. Indeed, in some cases, the status may be in the process of change, for example from "permitted" to "primary". Neither the status nor the right to protection are of direct concern here. However, it is clear that methods for calculating any potential interference either from or to the broadcasting service are required. This calculation process is complicated by the fact that it may be necessary to consider either assignments or allotments as the basis for digital television planning. This is also a difficulty which must be dealt with in the planning of television services (in other words, the interaction between television services) with no consideration of interference to or from other services.

8.2 Other service stations

Regardless of whether planning is done on the basis of digital television assignments or allotments, it is essential to have a clear definition of the other service requirements, in terms of their susceptibility to interference and their protection needs and also in terms of their potential to cause interference. In the case of a receive-only service, such as radio astronomy, the potential for it to cause interference to television may be regarded as zero.

8.2.1 Protection needs of other services

In addition to obvious elements:

– centre frequency;
– signal level to be protected;
– protection ratio as a function of frequency separation between other service and digital television centre frequencies;
– percentage time for which protection is required;
– other service receiving antenna orientation and discrimination (if relevant),

it is also necessary to determine the area or the locations for which the protection is required. The latter may conveniently be done by specifying a set of test point locations (as longitude, latitude and height above ground level) which represent either:

– the boundary of the area within which protection is required; or,
– the actual locations at which a receiving installation is, or may be, installed.

In order to avoid some ambiguities which have created difficulties in the past, special care needs to be taken when obtaining information about other service receiving antenna characteristics:

– in the case of mobile reception, it is assumed that there is neither directivity nor polarization discrimination; and,
– in the case of fixed reception, it is necessary to specify the orientation of the receiving antenna, as well as its co- and cross-polar discrimination as a function of relative bearing.

8.3 Technical elements of other services needed for compatibility calculations

The parameters which are needed for compatibility calculations are for transmitting and/or receiving terminals:

– modulation;
– frequency;
– bandwidth;
– maximum radiated power;
– azimuthal radiation pattern;
– polarization;
– polarization discrimination;
– site coordinates and height information (longitude, latitude and height above ground level, or sea level, as appropriate);
– protection ratio as a function of frequency separation;
– minimum signal level to be protected for a given installation;
– time percentage to be protected;
– coverage area defined by calculation test points (up to 36).

8.4 Calculation of the protection of other services

A calculation should be made for each of the test points used in the definition of the other service. This calculation should take into account:

– the protection ratio for the frequency difference between the other service and the digital television service;
– the signal level from the interfering assignment;
– other service receiving antenna discrimination (polarization and directivity), where relevant.

From the above information, the nuisance field strength (at each of the test-points) may be calculated for the other service.

The nuisance field strength, E_n, is defined as:

$$E_n = E_i + PR + A$$

where, expressed in dB:

E_i: field strength value of DVB-T assignment

PR: relevant protection ratio

A: relevant receiving antenna discrimination ($A \leq 0$).

During any necessary coordination discussions, the nuisance field strength (at each of the test points) may be compared with the minimum signal level to be protected for the other service.

The calculation of the interfering signal level is dependent upon the other service being considered. Recommendation ITU-R P.1546 (or former Recommendation ITU-R P.370), for individual transmitters, or a statistical method, for SFNs, may be used for terrestrial other services, taking into account the relevant percentage of time. However, free-space calculations will be needed for aeronautical (or satellite) services if a line-of-sight condition between other service receiver and interfering transmitter exists.

8.5 Calculation of the protection of digital television

A calculation should be made for each of the test points used in the definition of a digital television coverage area. This calculation should take into account:

– the protection ratio for the frequency difference between the other service and the digital television service;

– the signal level from the other service transmitter;

– the digital television service receiving antenna discrimination (in the case of fixed antenna reception).

From the above information, the nuisance field strength (at each of the boundary test-points) may be calculated for the digital television service.

The nuisance field strength, E_n, is defined as:

$$E_n = E_i + PR + A$$

where, expressed in dB:

E_i: field strength value of the other service assignment

PR: relevant protection ratio

A: relevant receiving antenna discrimination ($A \leq 0$).

During any necessary coordination discussions, the nuisance field strength (at each of the boundary test points) may be compared with the minimum signal level of the digital television service, to which it is necessary to add the effect of the combined location variation of the wanted and interfering signals.

CHAPTER 9

TRANSMISSION ASPECTS

9.1 Transmitting antennas

9.1.1 Introduction

The implementation of a network for terrestrial television is of course dependent on the provision of suitable transmitting antennas radiating from suitable locations. In general, the most appropriate aperture on existing structures is already used by the service for which the mast was built. On the supposition that new structures for most stations will be prohibitively expensive, the re-use of existing antennas is of primary interest. If this is not possible, other alternatives will need to be considered.

The purpose of this section is to consider the re-use of existing antennas and the options available for mounting digital television antennas on structures already used for analogue television.

9.1.2 Description of existing television transmitting antennas

In Europe, a typical television network consists of high-power main stations using horizontal polarization and low to medium power relay stations using horizontal or possibly vertical polarization.

The transmitting antennas are often mounted on a cantilever spine on top of the mast or tower. Mounting on a spine, rather than directly on the structure ensures that the arrays of radiating elements are as close as possible to one another in the horizontal plane. The closer the radiating phase centres, the more uniform (and controllable) the horizontal radiation pattern.

In most main stations and, in some countries, also many relay stations, the entire antenna system is enclosed in a fibreglass cylinder. This cylinder provides weather protection for the antenna and in many cases also forms part of the mechanical support structure for the antenna.

UHF antennas are generally designed for a specific set of channels spread over the entire UHF band or grouped in sub-bands, for example, Band IV, Lower Band V or Upper Band V. Typically, the antennas only show a satisfactory impedance match for analogue television at the channels they are designed for and in the close vicinity thereof. This is normally also the case even where antenna systems are equipped with wide band panels.

9.1.3 Options for digital television antennas

9.1.3.1 Share antenna with analogue television

This is possible if:

– the digital television and analogue television transmissions are co-polarized;

– the existing antenna will operate satisfactorily at the frequencies being proposed for digital television;

– the radiation pattern of the existing antenna satisfies any restrictions in the radiated power of the digital television which are needed to avoid interference into other services;

– the antenna system is capable of handling the total power of all services to be transmitted.

If these conditions are satisfied, then only additional combining equipment is required.

The performance of the existing antenna is worth some consideration. If the digital television channel is close to one of the analogue channels, the radiation pattern will almost certainly be similar to that for analogue television. The impedance match should also be similar. For other channels the radiation pattern may not be very different but the impedance match will in many cases be unacceptable for analogue television. However, digital television may not be so sensitive in this respect. Even if the reflected power is problematic, it may be the case that it can be diverted into a load. Alternatively, the internal feed system of the antenna could be re-engineered.

If digital television and analogue television are cross-polarised it may be possible to engineer new antennas shared by both analogue and digital television, which would have individual input ports to produce either vertical polarization or horizontal polarization for either of the services. The isolation between the ports may be such that special measures to improve isolation are not necessary. Of course, separate main feeders would be required for each service.

9.1.3.2 Share analogue television aperture

This option is for a separate digital television antenna to be constructed in the same aperture as that used for the analogue television and will mainly be of interest when the digital television will be radiated with a different linear polarization than that of the analogue television service. Whether this is a feasible option will depend on the design of the existing antenna. For certain types of antennas (such as a batwing) it will probably not be feasible. Where it is possible, the coupling between antennas (and supporting metalwork) must be low enough to avoid mutual interference to the radiation patterns of each service.

A significant factor concerning this option is the available space in which to engineer any new antenna. As already stated, many main and relay stations use fibreglass cylinders for weather shielding. It is unlikely that the relay stations in this category will have sufficient space for any new metalwork inside the fibreglass cylinder. The main stations may be slightly less difficult, but this could still impose a severe logistical constraint on an already difficult design.

Another way out could, in some cases, be to remove half of the existing antenna, used for analogue television, and replace it with a new antenna for digital television. This will imply a loss in antenna gain of about 3 dB for the analogue television service.

Construction of a new antenna for digital television on the outside of the glass-fibre cylinder is considered unrealistic for structural reasons. Furthermore such a spacing between the radiating elements and the structure axis is not conducive to providing satisfactory radiation patterns.

9.1.3.3 Completely separate digital television antenna occupying its own aperture

For sites where acceptable aperture on an existing antenna is available, this may be the preferred option. In MFNs, the radiation pattern of the new antenna(s) can be designed to match any restrictions in radiated power which are necessary to prevent interference into existing analogue television services. If the available aperture is relatively low on the structure, the required coverage may not be realized. If the structure tapers, the lower the aperture the greater the face width. The phase centres of the radiating elements become further apart and as a consequence dips in the horizontal radiation pattern become deeper or the antenna becomes very complex and thereby very expensive. Nevertheless, less-than-ideal coverage must be balanced against complex and expensive engineering solutions.

Other practical considerations must also be taken into account. There must be room for new feeders to be connected to the antennas and the structure must be strong enough to carry the windload of the new antenna as well as the new feeder cables.

9.1.3.4 Further considerations

If digital television is to match analogue television coverage, similar antenna heights will be necessary. This being the case, it is a reasonable assumption that at the majority of transmitting sites, no space will be readily available. It may be that some reconfiguration of existing antennas will provide the required aperture, although it is unlikely that this will be usual.

If digital television and analogue television are to be co-polarized, it may be most effective to look into the possibility of sharing existing antennas. In some cases, the existing antennas may need to be re-engineered to accommodate the new channel(s). This may not necessarily be prohibitively expensive.

If digital television and analogue television are not co-polarized, serious thought may be given to the following options:

– **Sharing aperture**

  Computer modelling can give an indication of possible configurations and interactions, but as the situation is so complex, practical models and measurements will also be necessary. Such development work is costly, time consuming and requires a wide range of expertise. Bearing in mind the earlier comments regarding available space inside fibreglass cylinders, it must be accepted that the outcome of such a study may not be promising;

– **Dual polarized antenna**

  Antenna panels suitable for dual polarization (separate input connectors) are available on the market, normally used for elliptic (circular) polarization. The use of a dual polarized antenna for the radiation of both analogue digital television will in fact imply a complete renewal of all parts of the existing antenna. The gain in each of the planes in a dual polarized antenna will generally not differ from that of a single polarized antenna by more than 1 dB. The radiation pattern for analogue television can be maintained while the pattern for digital television can be designed to have either the same or a different shape;

– **Reduction of the existing aperture**

  If a reduction of the existing antenna to half of its original size, causing a 3 dB reduction of the antenna gain is acceptable, a new antenna for digital television having approximately the same gain as the remaining part of the existing antenna can be accommodated within the aperture available. The horizontal radiation pattern may be the same as that of the analogue service(s) or it may be different in order to allow for any required restrictions.

It must be noted that all three options also imply the installation of an extra set of feeders.

9.1.3.5 Conclusions

If digital television and analogue television services are co-polarized, it is possible that they may be able to share transmitting antennas. As far as cost and complication is concerned, this is the preferred outcome.

If digital television and analogue television services are not co-polarized and no space is available for a new separate digital television antenna, significant re-engineering on the existing antenna will be necessary.

For sites with suitable free aperture the preferred option may be to construct an exclusive digital television antenna.

9.2 Suppression of unwanted emissions

Considerable detail is given in this section about the spectrum masks used in Europe and the way in which they were derived. Such masks are specific to a given planning environment and would need to be derived in accordance with the requirements of that environment. However, the same approach as is described here is likely to be of value in any environment except where special constraints are to be applied.

In the starting phase of terrestrial digital television, channels will have to be found mainly between those already in use for analogue television. In some cases it will be necessary to use channels adjacent to existing analogue television channels. To avoid interference into the analogue television services it is considered important to limit the out-of-channel emissions from digital television transmitters as much as possible. This leads to a need for defined spectrum masks for digital television transmitters.

The modulation scheme to be used for digital television will be quite complex, for example OFDM 64-QAM. Such a modulation scheme will demand a very high degree of linearity in the transmitter power amplifier in order to avoid intermodulation products.

The "natural" sidebands of the OFDM spectrum can (and must) be cut off in a suitable filter at IF in the modulator. The sidebands will, however, re-appear at RF due to intermodulation products, between the individual carriers, occurring in the amplifier chain of the transmitter. In order to achieve a reasonable (although still rather low) efficiency of the transmitter, extensive linearity pre-correction must be applied. Very non-linear amplifying components like klystrons are not expected to be readily usable for digital television.

The prevailing types of intermodulation products falling in or near the digital television channel are the third and the fifth order products. The third order products will fall in the range:

Channel centre frequency +/–1.5 (OFDM signal bandwidth)

and the fifth order intermodulation products will fall in the range:

Channel centre frequency +/–2.5 (OFDM signal bandwidth)

Intermodulation products falling inside the channel will act as interference from a co-channel (non-SFN) digital television transmitter and cause an increased bit error rate. The maximum acceptable level of intermodulation inside the channel can thus be estimated to be approximately equal to the minimum required C/N for the digital system in question. If this maximum level is reached no margin for noise or interference is left.

Intermodulation products falling outside of the channel could cause a noise-like co-channel interference to existing analogue television services operating on one or more channels adjacent or near to the digital television channel. The protection ratio needed for the analogue television service will be near to 40 dB, depending on the analogue system used. If the analogue television signal is radiated from the same station (or antenna) sufficient attenuation of the intermodulation products from the digital television transmitter is fairly easily specified. If the analogue television signal is not radiated from the same site as the digital television signal but still covering the same area or a part of it, the necessary attenuation of digital television intermodulation products can be quite difficult to achieve. In both cases a suitable spectrum mask filter is needed.

Where analogue and digital television transmitters using adjacent channels are co-sited and serving a common area also the out-of-band emissions from the analogue transmitter must be considered.

Due to non-linearity, mainly in the power amplifier(s), the suppressed part of the lower (vestigial) sideband tends to re-appear. This can affect a DVB-T signal transmitted in the lower adjacent channel. In analogue television transmitters using a common power amplifier for vision and sound also an image to the sound carrier(s) is found below the vision carrier.

Above the sound carrier(s) in the analogue channel harmonic distortion products from the video signal components appear causing an extension ranging into the upper adjacent channel.

In order to deal with adjacent channel compatibility defined spectrum masks for analogue television are needed.

9.2.1 Asymmetrical spectrum masks for DVB-T

It is generally expected that digital television transmitters to a large extend will be co-sited with existing analogue television transmitters and, as far as possible, will use the same polarization. On this basis spectrum masks for digital television transmitters, covering interference into various analogue television systems, can be derived on the basis of known protection ratios for the individual parts of the analogue signal.

Digital television transmitters are expected only to operate in the frequency bands envisaged for television. In most cases only protection of analogue television in adjacent channels has to be considered, exceptions being channels such as 5, 21, 60 or 69, where other services demanding high protection operate at frequencies just outside of the television channel. However, even in such cases only one side of the spectrum mask needs to show the shape of the mask for critical cases while the other side can have an out-of-band attenuation satisfactory for analogue television.

The protection ratios used for the adjacent channels have been taken from the EBU publication BPN 003 “Technical Bases for T-DAB Services Network Planning and Compatibility with existing Broadcasting Services” assuming that there is no influence on the value of the protection ratio whether the digital signal (OFDM) is a T-DAB signal or a DVB-T signal.

The examples presented have all been based on the following assumptions:

– the digital and the analogue transmitters are co-sited;

– no polarization discrimination;

– no offset is used on either of the transmitters except System L sound transmitters, where a positive offset of 50 kHz has been taken into account;

– the e.r.p. of the analogue transmitter (peak-sync) and the digital transmitter (total power) are the same.

Proportional corrections must be applied if:

– the radiated powers of the analogue and the digital television transmitters are not equal;

– the analogue and the digital television signals are not radiated with the same polarization and if polarization can be assumed.

The protection ratios used for analogue television are based on protection ratios taken from Recommendation ITU-R BT.655 and recalculated to impairment grade 4.5. From these protection ratios the maximum permissible relative power in a 4 kHz bandwidth is calculated for a set of representative frequencies in the analogue channel.

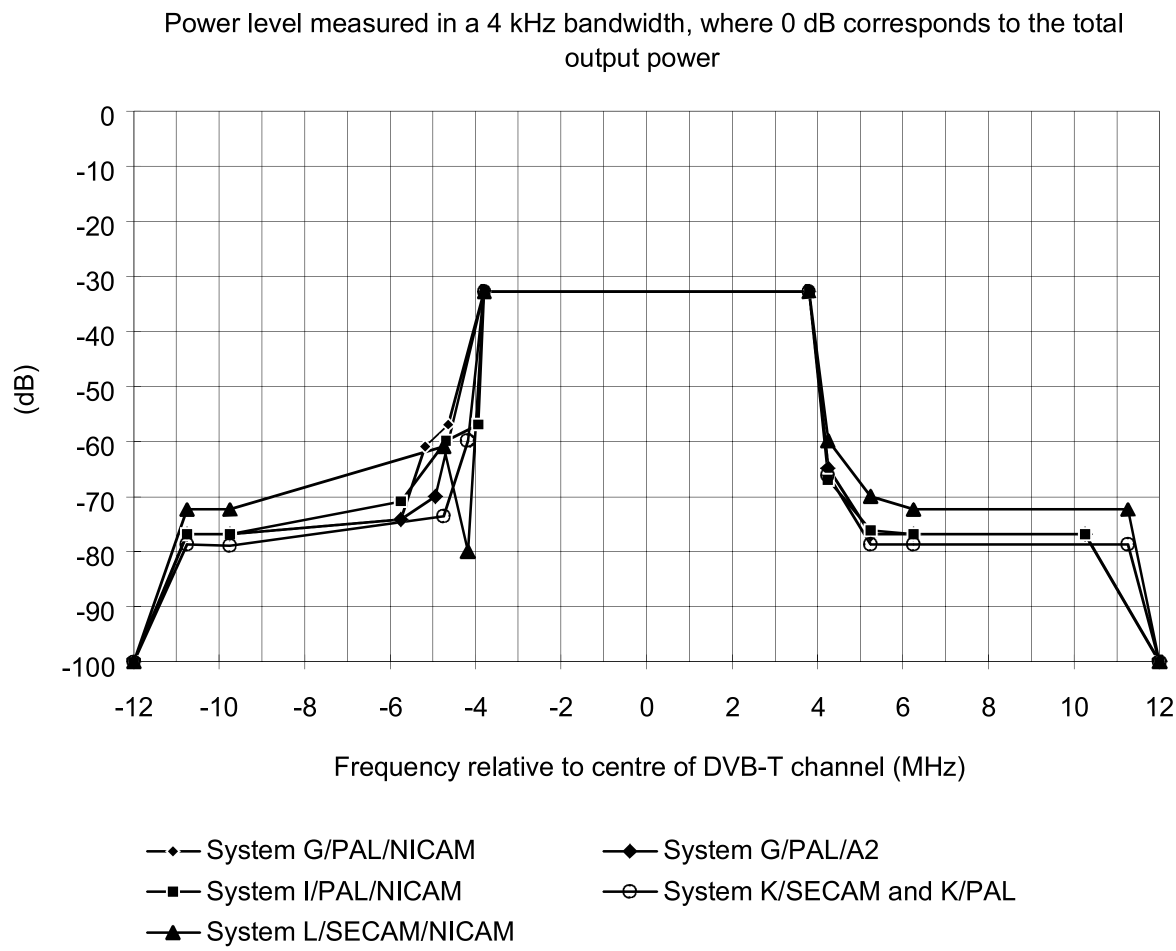
Information on the sound systems used is given in Recommendation ITU-R BS.707. The two channel FM system described there is given the designation “A2” in the remainder of this section.

The relative level in a 4 kHz bandwidth at the lower end of the lower adjacent channel and at the upper end of the upper adjacent channel has been chosen to –100 dB.

For details about the derivation of the values see Annex 1 to Chapter 9.

Two sets of spectrum masks for 8 MHz channels are given in Figs. 9.1 and 9.2 respectively and for 7 MHz channels in Fig. 9.3 and 9.4. The sets shown in Fig. 9.1 and Fig. 9.3 are based directly on the protection ratios derived in Annex 1 to Chapter 9 for the lower adjacent channel. In the upper adjacent channel the sound carrier demands less protection than the vision carriers. This would lead to spectrum masks having less attenuation further away from the DVB-T channel than just outside of it. For this reason the protection ratios for the vision carrier are repeated at frequencies corresponding to the upper end of the video sideband in the upper adjacent channel. This will, however, lead to an over-protection of about 5 dB at these frequencies.

These masks are considered to cover the minimum protection needed for co-sited analogue and digital television transmitters having equal radiated powers.

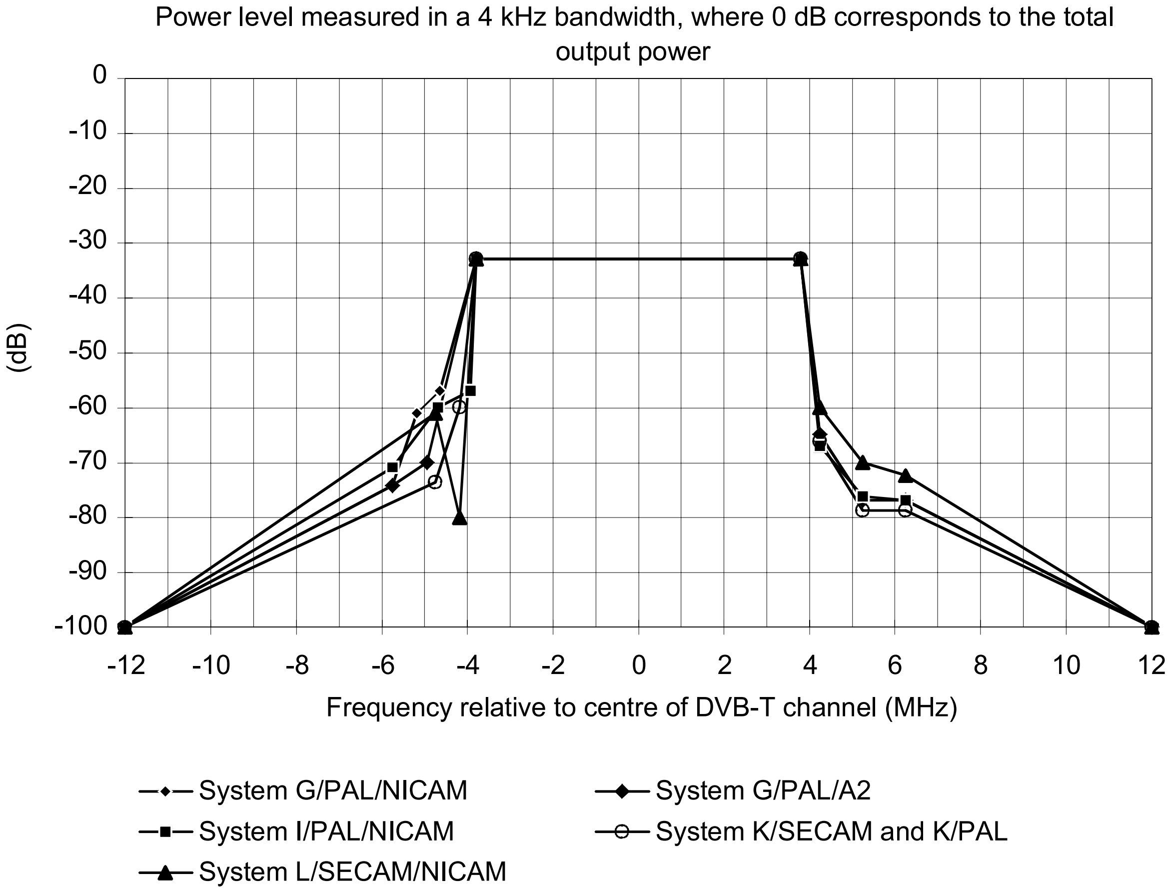


Breakpoints										
	G/PAL/NICAM		G/PAL/A2		I/PAL/NICAM		K/SECAM K/PAL		L/SECAM/NICAM	
See Notes to Fig. 9.4	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)
1	−12	−100	−12	−100	−12	−100	−12	−100	−12	−100
2	−10.75	−76.9	−10.75	−76.9	−10.75	−76.9	−10.75	−78.7	−10.75	−72.4
3	−9.75	−76.9	−9.75	−76.9	−9.75	−76.9	−9.75	−78.7	−9.75	−72.4
4	−5.75	−74.2	−5.75	−74.2	−5.75	−70.9	−4.75	−73.6	−4.75	−60.9
5	−5.185	−60.9	−5.185	N/A	−4.685	−59.9	−4.185	−59.9	−4.185	−79.9
6	N/A	N/A	−4.94	−69.9	N/A	N/A	N/A	N/A	N/A	N/A
7	−4.65	−56.9	N/A	N/A	−3.925	−56.9	N/A	N/A	N/A	N/A
8	−3.8	−32.8	−3.8	−32.8	−3.8	−32.8	−3.8	−32.8	−3.8	−32.8
9	+3.8	−32.8	+3.8	−32.8	+3.8	−32.8	+3.8	−32.8	+3.8	−32.8
10	+4.25	−64.9	+4.25	−64.9	+4.25	−66.9	+4.25	−66.1	+4.25	−59.9
11	+5.25	−76.9	+5.25	−76.9	+5.25	−76.2	+5.25	−78.7	+5.25	−76.269.9
12	+6.25	−76.9	+6.25	−76.9	+6.25	−76.9	+6.25	−78.7	+6.25	−72.4
13	+10.25	−76.9	+10.25	−76.9	+10.25	−76.9	+11.25	−78.7	+11.25	−72.4
14	+12	−100	+12	−100	+12	−100	+12	−100	+12	−100

FIGURE 9.1

Spectrum masks for a digital terrestrial television transmitter operating on a channel adjacent to a co-sited analogue television transmitter, 8 MHz

DTTB-09-1

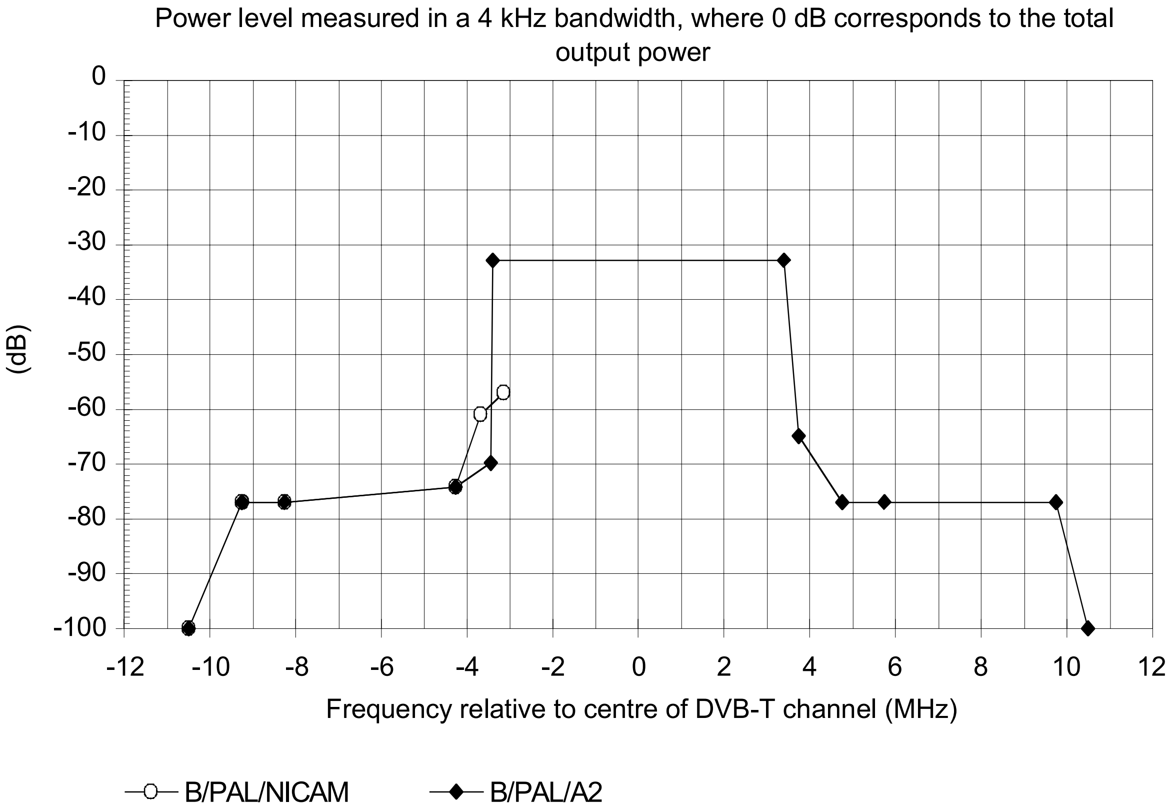


Breakpoints										
See Notes to Fig. 9.4	G/PAL/NICAM		G/PAL/A2		I/PAL/NICAM		K/SECAM K/PAL		L/SECAM/NICAM	
	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)
1	–12	–100	–12	–100	–12	–100	–12	–100	–12	–100
4	–5.75	–74.2	–5.75	–74.2	–5.75	–70.9	–4.75	–73.6	–4.75	–60.9
5	–5.185	–60.9	–5.185	N/A	–4.685	–59.9	–4.185	–59.9	–4.185	–79.9
6	N/A	N/A	–4.94	–69.9	N/A	N/A	N/A	N/A	N/A	N/A
7	–4.65	–56.9	N/A	N/A	–3.925	–56.9	N/A	N/A	N/A	N/A
8	–3.8	–32.8	–3.8	–32.8	–3.8	–32.8	–3.8	–32.8	–3.8	–32.8
9	+3.8	–32.8	+3.8	–32.8	+3.8	–32.8	+3.8	–32.8	+3.8	–32.8
10	+4.25	–64.9	+4.25	–64.9	+4.25	–66.9	+4.25	–66.1	+4.25	–59.9
11	+5.25	–76.9	+5.25	–76.9	+5.25	–76.2	+5.25	–78.7	+5.25	–69.9
12	+6.25	–76.9	+6.25	–76.9	+6.25	–76.9	+6.25	–78.7	+6.25	–72.4
13	+10.25	–76.9	+10.25	–76.9	+10.25	–76.9	+11.25	–78.7	+11.25	–72.4
14	+12	–100	+12	–100	+12	–100	+12	–100	+12	–100

FIGURE 9.2

Spectrum masks for a digital terrestrial television transmitter operating on a channel adjacent to a co-sited analogue television transmitter, 8 MHz

DTTB-09-2



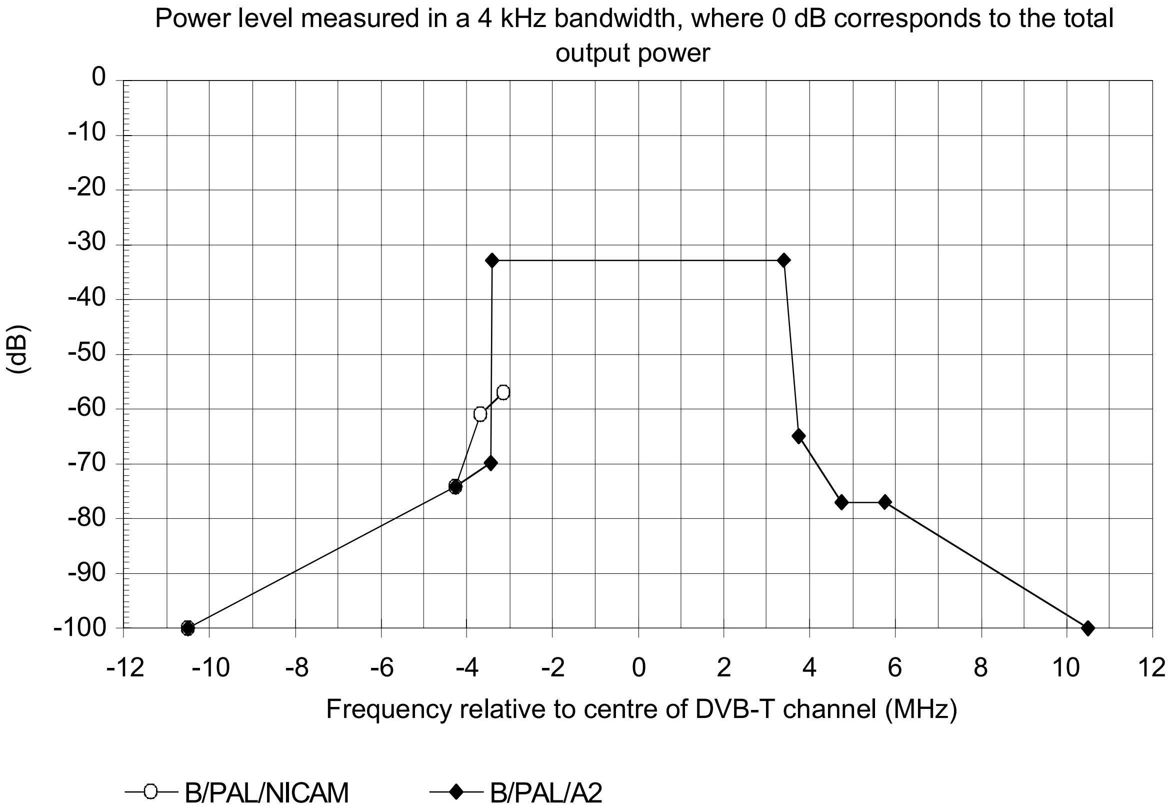
Breakpoints				
See Notes to Fig. 9.4	B/PAL/NICAM		B/PAL/A2	
	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)
1	–10.5	–100	–10.5	–100
2	–9.25	–76.9	–9.25	–76.9
3	–8.25	–76.9	–8.25	–76.9
4	–4.25	–74.2	–4.25	–74.2
5	–3.685	–60.9	–3.685	N/A
6	N/A	N/A	–3.44	–69.9
7	–3.15*	–56.9	N/A	N/A
8	–3.35	–32.8	–3.4	–32.8
9	+3.35	–32.8	+3.4	–32.8
10	+3.75	–64.9	+3.75	–64.9
11	+4.75	–76.9	+4.75	–76.9
12	+5.75	–76.9	+5.75	–76.9
13	+9.75	–76.9	+9.75	–76.9
14	+10.5	–100	+10.5	–100

* The NICAM signal overlaps the DVB-T signal if relative offset is less than 200 kHz.

FIGURE 9.3

Spectrum masks for a digital terrestrial television transmitter operating on a channel adjacent to a co-sited analogue System B television transmitter, 7 MHz

DTTB-09-3



Breakpoints				
See Notes below	B/PAL/NICAM		B/PAL/A2	
	Relative frequency (MHz)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)
1	–10.5	–100	–10.5	–100
4	–4.25	–74.2	–4.25	–74.2
5	–3.685	–60.9	–3.685	N/A
6	N/A	N/A	–3.44	–69.9
7	–3.15*	–56.9	N/A	N/A
8	–3.35	–32.8	–3.4	–32.8
9	+3.35	–32.8	+3.4	–32.8
10	+3.75	–64.9	+3.75	–64.9
11	+4.75	–76.9	+4.75	–76.9
12	+5.75	–76.9	+5.75	–76.9
14	+10.5	–100	+10.5	–100

* The NICAM signal overlaps the DVB-T signal if relative offset is less than 200 kHz.

FIGURE 9.4

Spectum masks for a digital terrestrial television transmitter operating on a channel adjacent to a co-sited analogue System B television transmitter, 7 MHz

DTTB-09-4

Notes to the Tables of breakpoints in Fig. 9.1, 9.2, 9.3 and 9.4:

For details about the determination of breakpoints and attenuation, see Annex 1 to Chapter 9:

1 Lower end of lower adjacent channel

2 Vision carrier in lower adjacent channel

3 Vision carrier + 1 MHz in lower adjacent channel

4 Upper end of video sideband in lower adjacent channel

5 Upper end of the RF bandwidth of the first sound carrier in lower adjacent channel

6 Upper end of the RF bandwidth of the A2 second sound carrier in lower adjacent channel

7 Upper end of the RF bandwidth of the NICAM signal in the lower adjacent channel

8 Lower end of the RF bandwidth of the DVB-T signal

9 Upper end of the RF bandwidth of the DVB-T signal

10 Lower video sideband (vision carrier – 1 MHz) in upper adjacent channel

11 Vision carrier in upper adjacent channel

12 Vision carrier + 1 MHz in upper adjacent channel

13 Upper end of video sideband in upper adjacent channel

14 Upper end of upper adjacent channel

Additional Notes:

In the Tables in Fig. 9.1, 9.2, 9.3 and 9.4 some cells are marked with "N/A". This means that this part of the analogue television signal does not exist or has no influence on the shape of the spectrum mask.

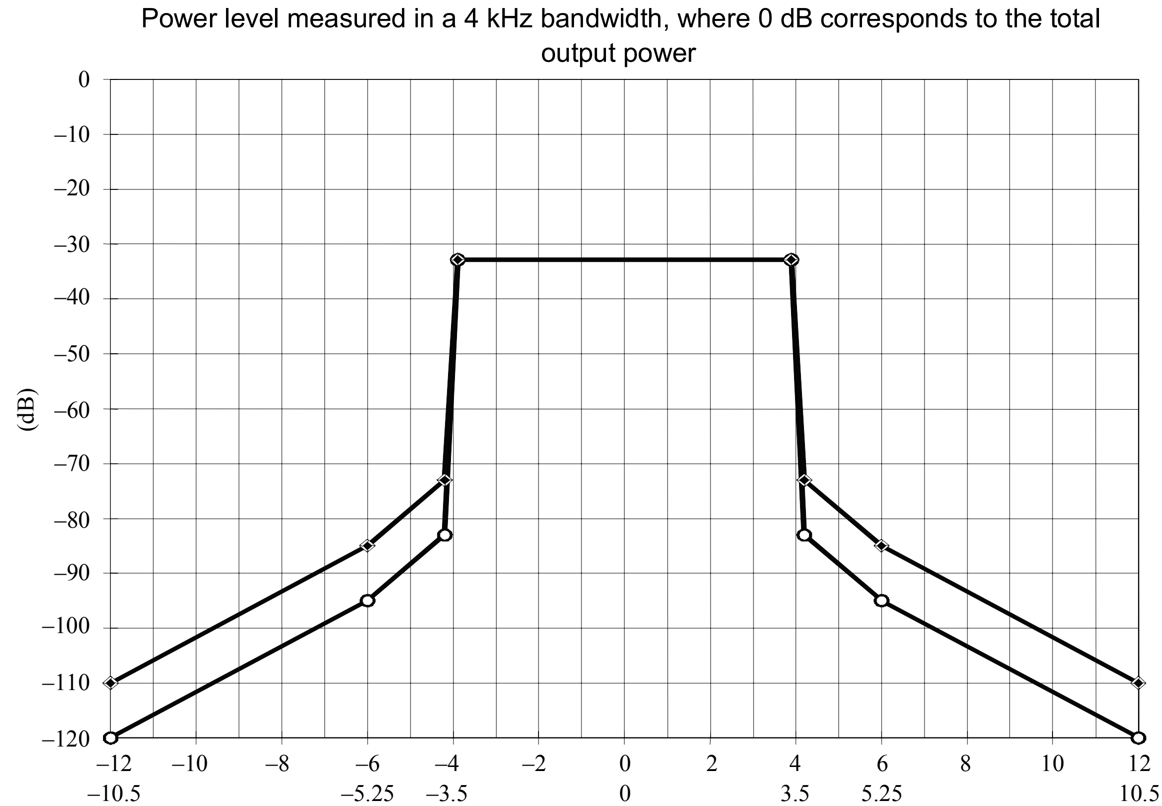
As it can be assumed that some degree of general selectivity will be introduced with a spectrum mask filter it is believed that in general, straight lines can be drawn from the breakpoints representing the upper end of the video sideband in the lower adjacent channel to the end point at the lower end of the lower adjacent channel. In the same way straight lines are drawn from the breakpoints representing the vision carriers in the upper adjacent channel to the end point at the upper end of the upper adjacent channel. Spectrum masks corresponding to those shown in Fig. 9.1 but based on the assumption above are shown in Fig. 9.2 for 8 MHz channels and in Fig. 9.4 for 7 MHz channels.

9.2.2 Symmetrical spectrum mask for DVB-T in 7 MHz and 8 MHz channels

For digital television transmitters using the channels adjacent to other services (low power or receive only) this spectrum mask may not give enough attenuation on the side of the digital television channel falling in the frequency band where the other service operates.

In such cases, special spectrum masks have to be defined, based on the characteristics of the other service and the distance between the digital television transmitter and the service area (or receiving installation) of the other service. It must, however, be borne in mind that spectrum mask filters showing a higher attenuation close to the digital television channel will be very expensive and imply a higher insertion loss.

Two symmetrical spectrum masks are shown in Fig. 9.5. The mask having a shoulder attenuation of 40 dB is intended for non-critical cases and the mask with a shoulder attenuation of 50 dB is intended for sensitive cases.



Frequency relative to centre of DVB-T channel (MHz)
Upper scale: 8 MHz channel
Lower scale: 7 MHz channel
Upper curve: non-critical cases
Lower curve: sensitive cases

Breakpoints					
8 MHz channels			7 MHz channels		
	Non-critical cases	Sensitive cases		Non-critical cases	Sensitive cases
Relative frequency (MHz)	Relative level (dB)	Relative level (dB)	Relative frequency (MHz)	Relative level (dB)	Relative level (dB)
–12	–110	–120	–10.5	–110	–120
–6	–85	–9	–5.25	–85	–95
–4.2	–73	–83	–3.7	–73	–83
–3.9	–32.8	–32.8	–3.35	–32.8	–32.8
+3.9	–32.8	–32.8	+3.35	–32.8	–32.8
+4.2	–73	–83	+3.7	–73	–83
+6	–85	–95	+5.25	–85	–95
+12	–110	–120	+10.5	–110	–120

FIGURE 9.5

Symmetrical spectrum masks non-critical and for sensitive cases

DTTB-09-5

The mask for non-critical cases should also be used for measurements of protection ratios for analogue television interfered with by DVB-T.

The shape of the masks have been based on:

– the natural spectrum of a 7.6 MHz OFDM signal (for 8 MHz channels) and a 6.7 MHz OFDM signal (for 7 MHz channels);

– the amplitude response of an IF SAW-filter;

– the power amplifier of the transmitter produces intermodulation outside of the channel at a level limited by the amount of intermodulation acceptable inside the channel;

– the mask for sensitive cases also include the amplitude response of a six-cavity bandpass filter at the output of the transmitter.

9.3 Analogue television

Based on information from the Radio Regulations (Article 3 and Appendix 3) and Recommendation ITU-R BT.470 spectrum masks have been established for a number of analogue television systems in use.

For transmitters operating in the frequency range 30 MHz to 235 MHz the unwanted power measured at the output terminal of the transmitter shall be attenuated at least 60 dB relative to the mean output power and shall not exceed 1 mW. The maximum limit is thus proportional for transmitters with output powers up to 1 kW and fixed for higher output powers.

For transmitters operating in the frequency range 235 MHz to 960 MHz the unwanted power measured at the output terminal of the transmitter shall be attenuated at least 60 dB relative to the mean output power and shall not exceed 20 mW. The maximum limit is thus proportional for transmitters with output powers up to 20 kW and fixed for higher output powers.

The mean power of a television transmitter depends considerably on the picture content. For transmitters using negative modulation the highest mean power is achieved at “Black with sync” and no pedestal where the mean power of the vision signal is 2.5 dB lower than the peak sync power. This leads to an out-of-band attenuation of 62.5 dB relative to peak sync power for transmitters within the “proportional” power range.

For transmitters using positive modulation the highest mean power occurs with an all-white picture where the mean power of the vision signal is 1.2 dB lower than the nominal transmitter output power.

If intermodulation products between the vision carrier and the sound carrier(s) are considered it is assumed that the sum of the vision power and the sound power(s) shall be used as reference.

When typical antenna gains and feeder losses for VHF transmitters are taken into consideration then transmitter output powers up to 1 kW corresponds to radiated powers of up to 10 kW and the fixed limit of 1 mW corresponds to an e.r.p. of 10 mW. For UHF the values become 400 kW and

400 mW respectively. It is assumed that most antenna systems will show the same or nearly the same gain in the adjacent channels as in the channel used. It is also assumed that diplexers inserted into the feeder line do not contribute to the attenuation of unwanted emissions in the adjacent channels.

At this state only analogue UHF transmitters are considered. In the original ST61 Plan totally 4479 UHF stations are listed. These stations can be subdivided into the following categories:

2 041 stations	e.r.p. ≤ 400 kW
818 stations	400 kW < e.r.p. ≤ 500 kW
1 589 STATIONS	500 kW < e.r.p. ≤ 1 mW
31 stations	e.r.p. > 1 mW (maximum = 2 MW)

The 400 kW limit thus only applies to less than the half of the stations. For stations with an e.r.p. > 400 kW the out-of-band attenuation shall be increased accordingly (for 2 mW by additional 7 dB).

9.3.1 Reference bandwidth for analogue television spectrum masks

Generally it is considered desirable to use a low reference bandwidth in order to show the real spectrum of the signal in question, on the other hand it is necessary to use a bandwidth wide enough to make it possible to measure the RF spectrum realistically.

For DVB (and DAB) spectrum masks are based on the power measured in a 4 kHz bandwidth.

In analogue television three different types of modulation are used (ignoring the colour subcarrier): AM, FM and QPSK. The signal components have different bandwidths and use different "quiescent" modulation, e.g. the vision carrier is always modulated with at least a sync-signal and the NICAM subcarrier is always occupying a constant bandwidth whereas FM- or AM- sound carriers are not modulated when no sound signal is present.

Tests have shown that power spectrum of the vision carrier and its sideband(s) can be measured correctly with a spectrum analyser using "max. hold" and resolution bandwidths down to 50 kHz. At 10 kHz resolution bandwidth the level is shown about 0.2 dB too low and at 3 kHz resolution bandwidth the level error is about 1 dB. In all cases the video bandwidth of the spectrum analyser was 100 kHz. A resolution bandwidth of 300 kHz and a video bandwidth also of 300 kHz were used as reference. When measuring the contour of the video sideband(s) with narrow resolution bandwidths a very slow sweep rate is mandatory, 10 s is recommended for video frequencies swept from 100 kHz to 6 MHz.

The power spectrum of FM sound carriers can only be measured correctly (at full deviation) if the resolution bandwidth of the spectrum analyser is at least equal to the highest modulation frequency, i.e. 15 kHz, otherwise the result will depend on the modulation frequency and the deviation. The video bandwidth of the spectrum analyser should be somewhat higher, e.g. 30 kHz.

AM sound carriers can be measured correctly with even very low resolution bandwidths as long as the modulation frequency is kept constant or swept very slowly.

For QPSK carriers like NICAM the measured level depends only on the resolution bandwidth of the spectrum analyser and scaling can be done from or to any relevant bandwidth.

As a result of the above described differences between the individual components of an analogue television signal a reference bandwidth of 50 kHz is used for spectrum masks for analogue television.

TABLE 9.1

Breakpoints for spectrum masks for analogue television systems. based on 50 kHz bandwidth

Description of breakpoint	Frequency relative to vision carrier in analogue channel (MHz)	Frequency relative to centre of channel (MHz)	Analogue tv-system G/PAL (mono), V/S ratio = 10 dB	Analogue tv-system G/PAL/ NICAM V/S ratio = 13 dB(1)	Analogue tv-system G/PAL/A2 V/S ratio = 13 dB(1)	Analogue tv-system I/PAL/ NICAM V/S ratio = 10 dB(1)	Analogue tv-system K/ SECAM V/S ratio = 10 dB	Analogue tv-system L/ SECAM V/S ratio = 10 dB
Lower end of lower adjacent 8 MHz channel	–9.25	–12	–62.5	–62.5	–62.5	–62.5	–62.5	–61.2
Image of colour subcarrier system G and I. Lower limit of colour subcarrier. Image system K	–4.43	–7.18	–46	–46	–46	–46.7	–46	N/A
Image of colour subcarrier system L	–4.3	–7.05	N/A	N/A	N/A	N/A	N/A	[–13] –30 = –43
Upper limit of colour subcarrier image. System K	–4.23	–6.98	N/A	N/A	N/A	N/A	–46	N/A
Attenuation of lower video sideband. System G and I	–3	–5.75	–36	–36	–36	–36.7	N/A	N/A
Attenuation of lower sideband system L	–2.7	–5.45	N/A	N/A	N/A	N/A	N/A	[–13] –15 = –28
Lower end of channel	–1.25	–4	–36	–36	–36	–16.7	–36	[–13]
Lower corner of vestigial sideband. System G and K	–0.75	–3.5	–16	–16	–16	N/A	–16	N/A
Lower end of sync. signal spectrum	–0.13	–2.88	–16	–16	–16	–16.7	–16	[–13]
Vision carrier (for System L at an all 100% white picture)	0	–2.75	0	0	0	0	0	0
Upper end of sync. signal spectrum	0.13	–2.62	–16	–16	–16	–16.7	–16	[–13]
Upper end of video sideband. System G	5	2.25	–16	–16	–16	N/A	N/A	N/A
Gap between video sideband and 1st sound carrier. System G	5.25	2.5	–20	–20	–20	N/A	N/A	N/A
Lower corner of 1st sound carrier. System G	5.435	2.685	–10	–13	–13	N/A	N/A	N/A
Upper end of video sideband. System I	5.5	2.75	N/A	N/A	N/A	–16.7	N/A	N/A
Upper corner of 1st sound carrier. System G	5.565	2.815	–10	–13	–13	N/A	N/A	N/A
Lower corner of NICAM signal. System G/NICAM	5.6	2.85	N/A	–20	N/A	N/A	N/A	N/A

TABLE 9.1 (*end*)

Description of breakpoint	**Frequency relative to vision carrier in analogue channel (MHz)**	**Frequency relative to centre of channel (MHz)**	**Analogue tv-system G/PAL (mono), V/S ratio = 10 dB**	**Analogue tv-system G/PAL/ NICAM V/S ratio = 13 dB(1)**	**Analogue tv-system G/PAL/A2 V/S ratio = 13 dB(1)**	**Analogue tv-system I/PAL/ NICAM V/S ratio = 10 dB(1)**	**Analogue tv-system K/ SECAM V/S ratio = 10 dB**	**Analogue tv-system L/ SECAM V/S ratio = 10 dB**
Lower corner of 2nd sound carrier. System G/A2	5.675	2.925	N/A	N/A	–20	N/A	N/A	N/A
Gap between video sideband and 1st sound carrier. System I	5.75	3	N/A	N/A	N/A	–20	N/A	N/A
Upper end of spectrum used by System G/mono	5.8	3.05	–62.5	N/A	N/A	N/A	N/A	N/A
Upper corner of 2nd sound carrier. System G/A2	5.805	3.055	N/A	N/A	–20	N/A	N/A	N/A
Lower corner of 1st sound carrier. System I	5.9346	3.1846	N/A	N/A	N/A	–10	N/A	N/A
Upper end of spectrum used by System G/A2	5.97	3.22	N/A	N/A	–62.5	N/A	N/A	N/A
Upper end of video sideband. System K and L	6	3.25	N/A	N/A	N/A	N/A	–16	[–13]
Upper corner of 1st sound carrier. System I	6.0646	3.3146	N/A	N/A	N/A	–10	N/A	N/A
Upper corner of NICAM signal. System G/NICAM	6.1	3.35	N/A	–20	N/A	N/A	N/A	N/A
Gap between video sideband and 1st sound carrier. System K and L	6.25	3.5	N/A	N/A	N/A	N/A	–20	–20
Upper end of spectrum used by System G/NICAM	6.28	3.53	N/A	–62.5	N/A	N/A	N/A	N/A
Lower corner of NICAM signal. System I/NICAM	6.302	3.552	N/A	N/A	N/A	–25	N/A	N/A
Lower corner of 1st sound carrier System K and L	6.435	3.685	N/A	N/A	N/A	N/A	–10	–10
Centre of NICAM signal. System I/NICAM	6.552	3.802	N/A	N/A	N/A	–20	N/A	N/A
Upper corner of 1st sound carrier System K and L	6.565	3.815	N/A	N/A	N/A	N/A	–10	–10
Upper end of 8 MHz channel	6.75	4	N/A	N/A	N/A	N/A	–54	–54
Upper end of spectrum used by System K and L(2)	6.8	4.05	N/A	N/A	N/A	N/A	–62.5	–61.2
Upper corner of NICAM signal. System I/NICAM	6.802	4.052	N/A	N/A	N/A	–25	N/A	N/A
Upper end of spectrum used by System I/NICAM	6.94	4.19	N/A	N/A	N/A	–62.5	N/A	N/A
Upper end of upper adjacent 8 MHz channel	14.75	12	–62.5	–62.5	–62.5	–62.5	–62.5	–61.2

(1) The Vision to NICAM ratio and the Vision to second sound carrier ratio are both 20 dB.
(2) Because of 50 kHz reference bandwidth used.

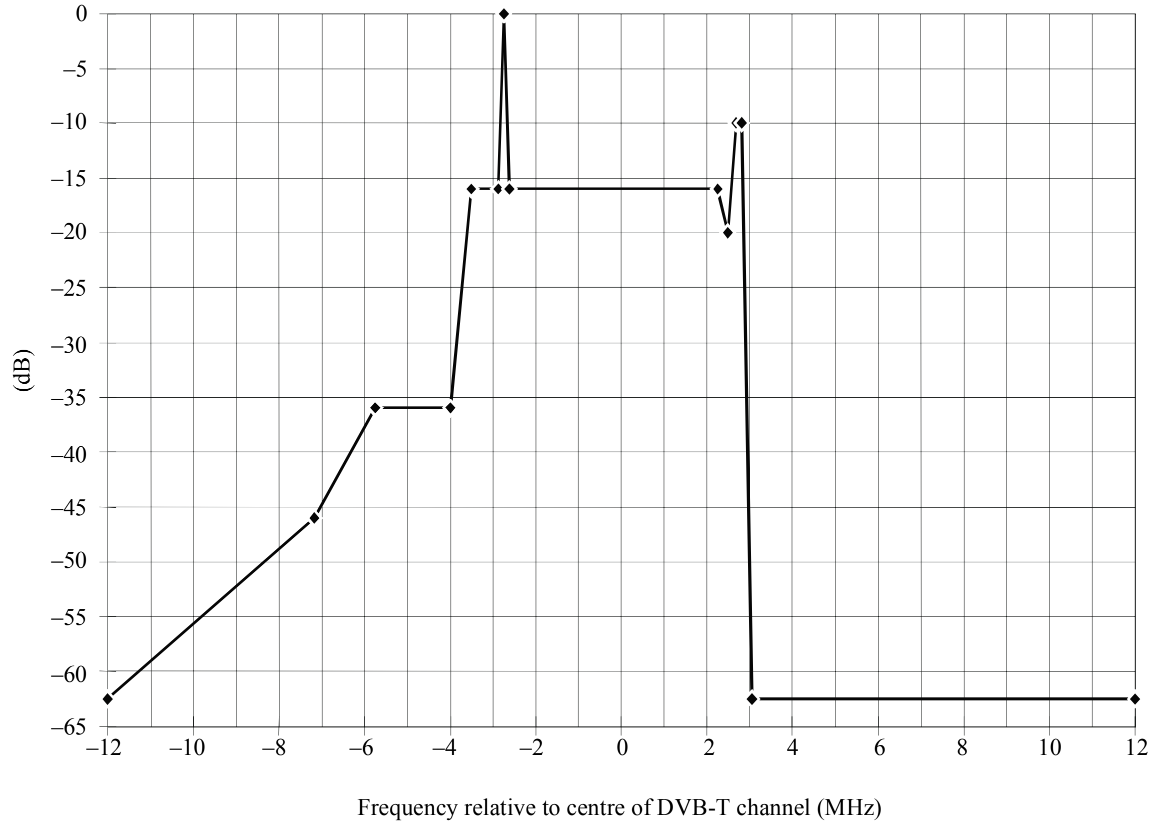


FIGURE 9.6

Analogue system G/PAL/mono. V/S ratio = 10 dB

DTTB-09-6

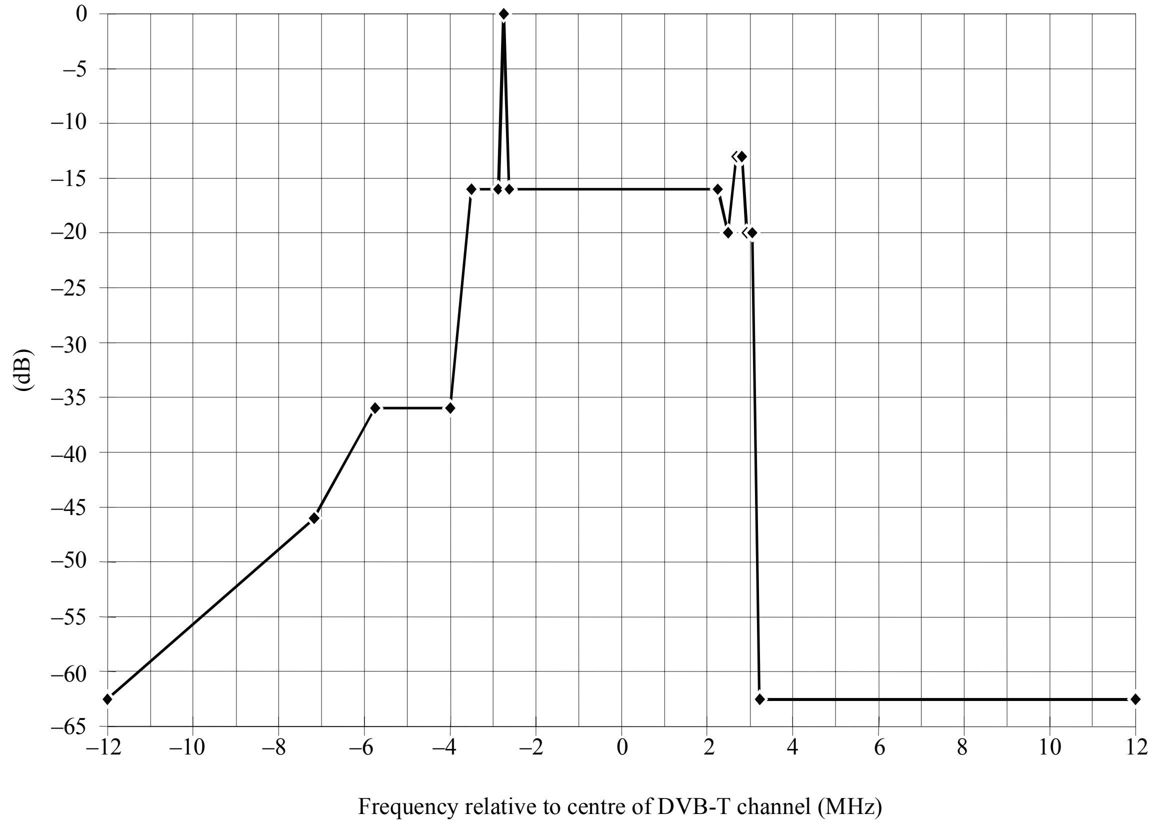


FIGURE 9.7

Analogue system G/PAL/A2. V/S/s ratio = 13 dB/20 dB

DTTB-09-7

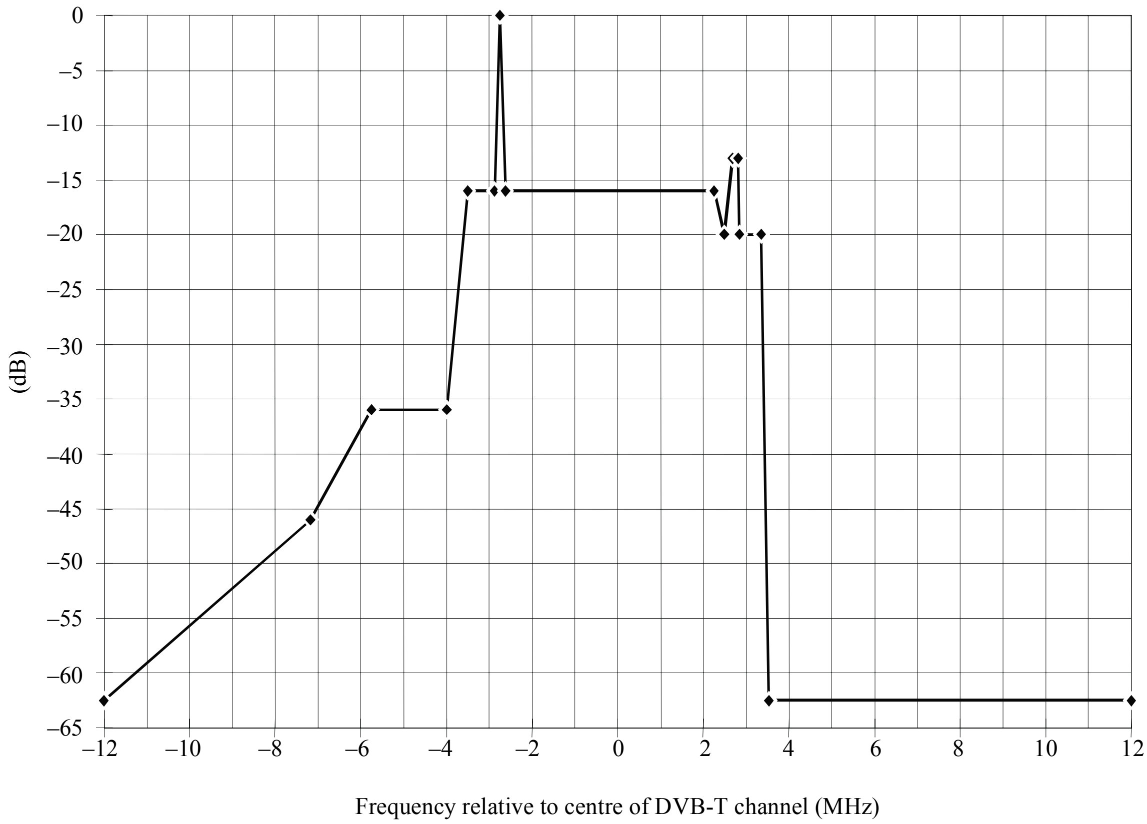


FIGURE 9.8

Analogue system G/PAL/NICAM. V/S/N ratio = 13 dB/20 dB

DTTB-09-8

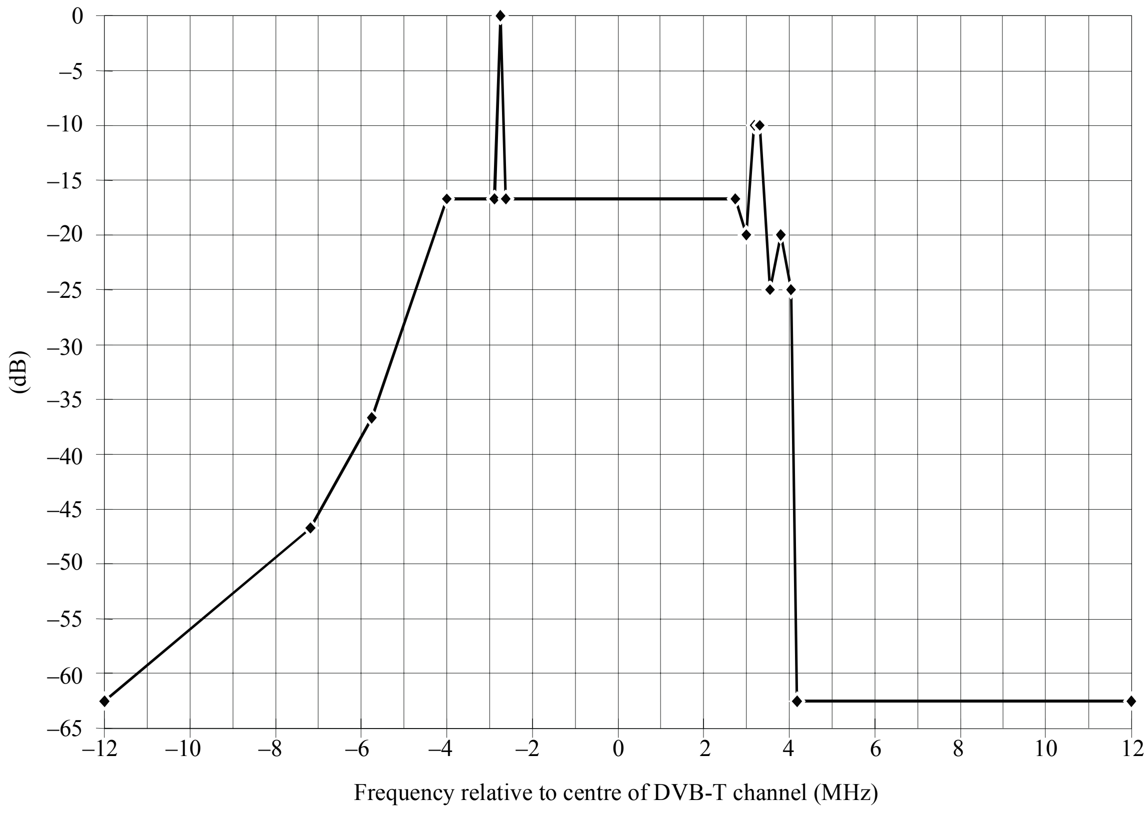


FIGURE 9.9

Analogue system I/NICAM. V/S/N = 10 dB/20 dB

DTTB-09-9

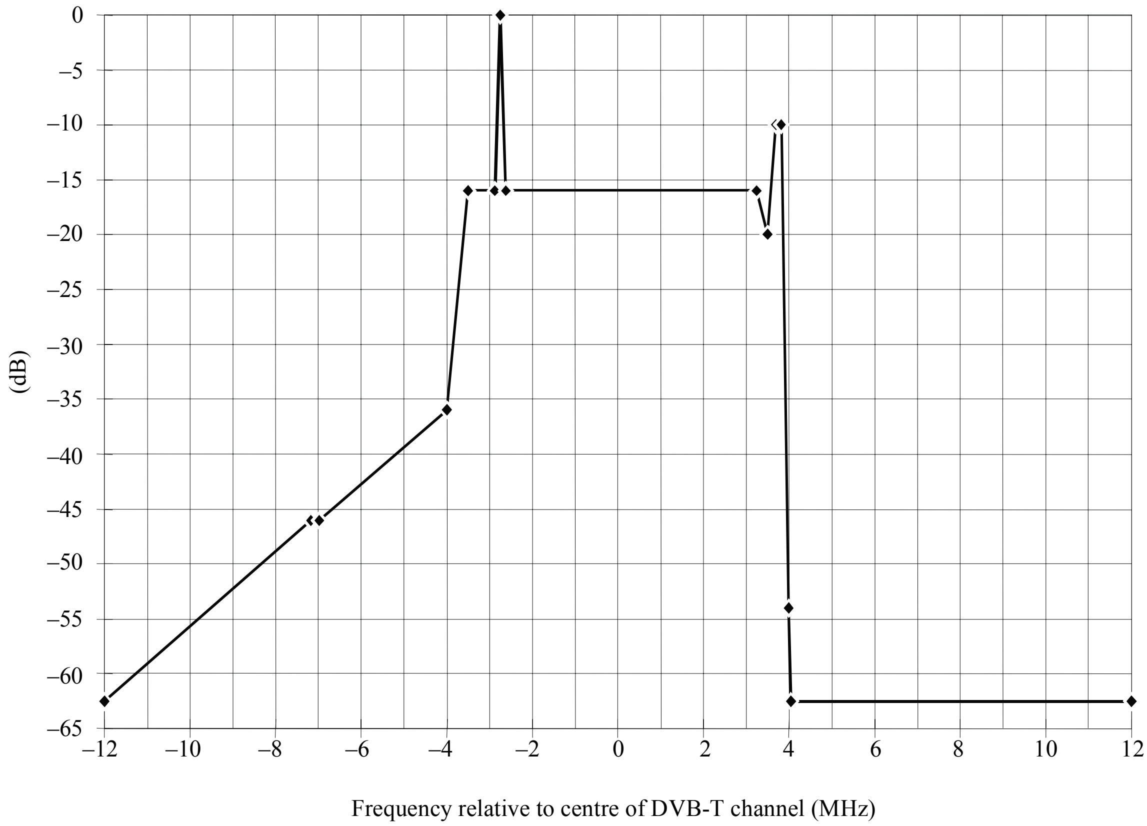


FIGURE 9.10

Analogue system K/SECAM. V/S ratio = 10 dB

DTTB-09-10

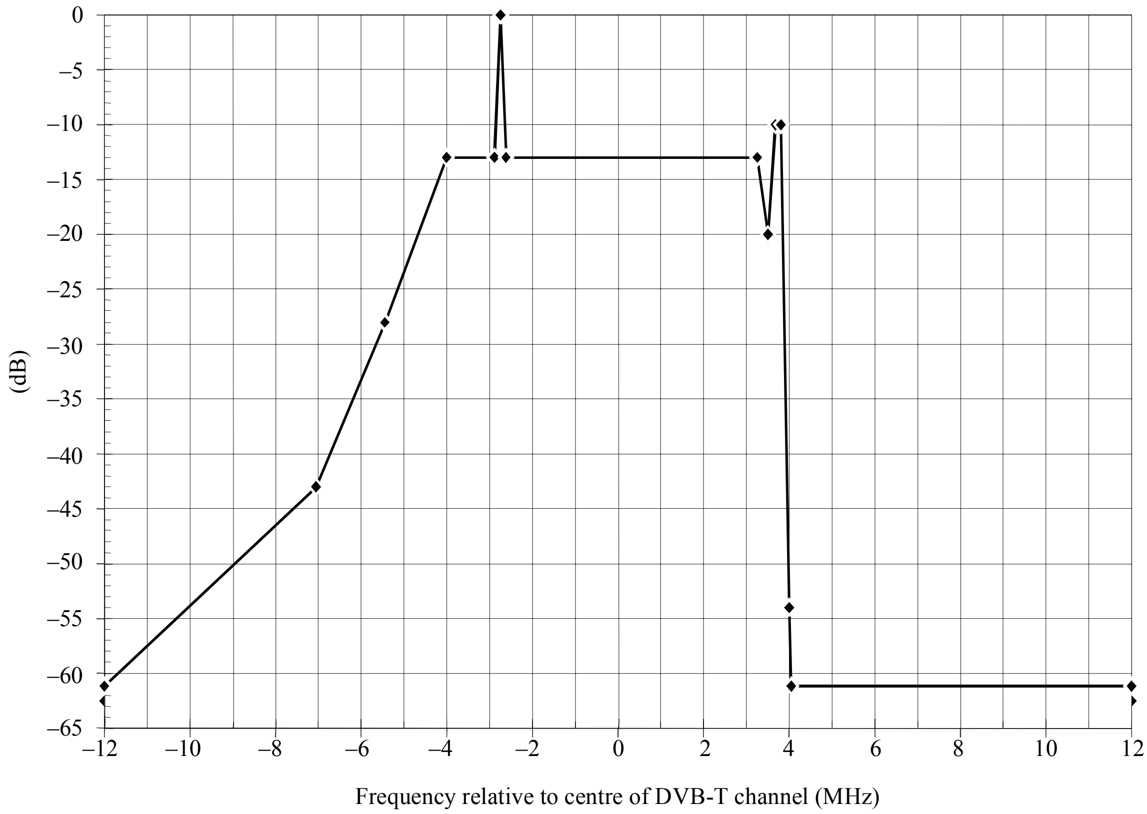


FIGURE 9.11

Analogue system L/SECAM. V/S ratio = 10 dB

DTTB-09-11

9.4 Measured transmitter power spectra

To illustrate the performance of typical high power transmitters the power spectrum was measured on three UHF transmitters. Two of these being identical 40 kW pulsed klystron transmitters but operating on different channels (31 and 53) while the third transmitter is a 10 kW tetrode transmitter operating on channel 53. All three transmitters are less than 10 years old.

The residual carrier level was set to 11%.

The spectra with FM-sound and NICAM are shown in Figs. 9.12. 9.13 and 9.14 respectively.

The "extra sideband" appearing above the NICAM subcarrier has been identified as the second harmonic of the sine-wave contained in the video signal. It is seen that the suppression of this unwanted signal is significantly different for the two types of transmitters tested. It is also seen that the suppression of the (re-inserted) lower sideband differs between the two (identical) klystron transmitters.

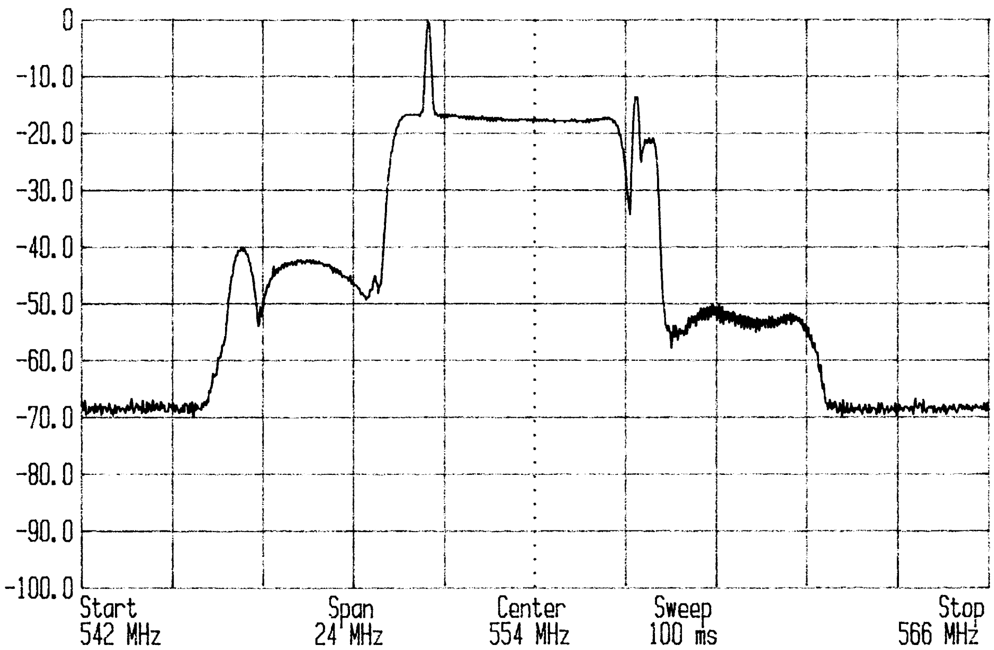


FIGURE 9.12

Spectrum for a system G 40 kW pulsed klystron transmitter with FM-sound and NICAM carriers

DTTB-09-12

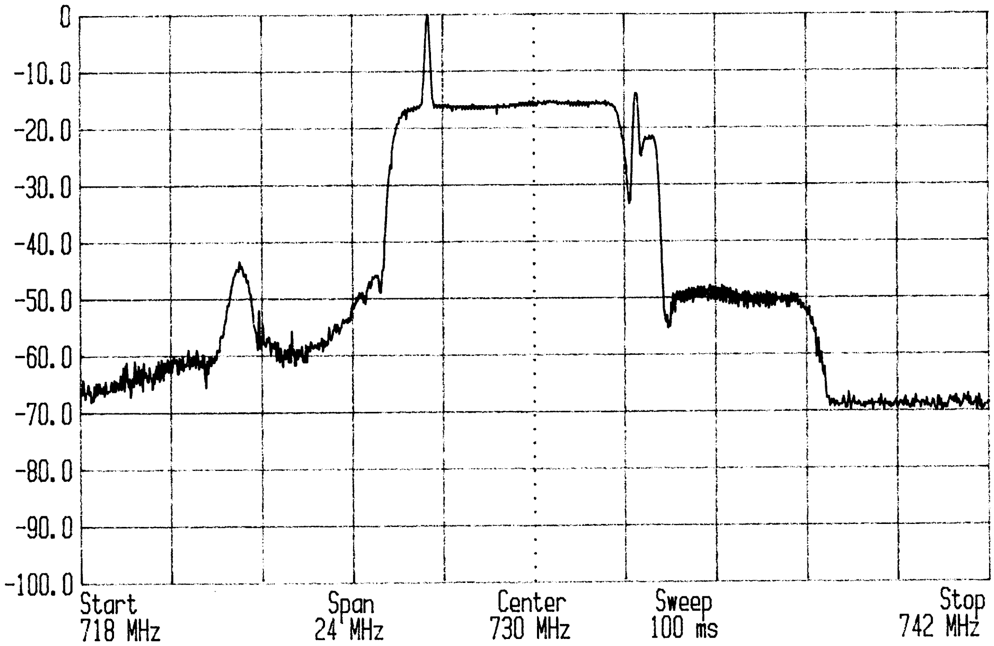


FIGURE 9.13

Spectrum for a system G 40 kW pulsed klystron transmitter with FM-sound and NICAM carriers

DTTB-09-13

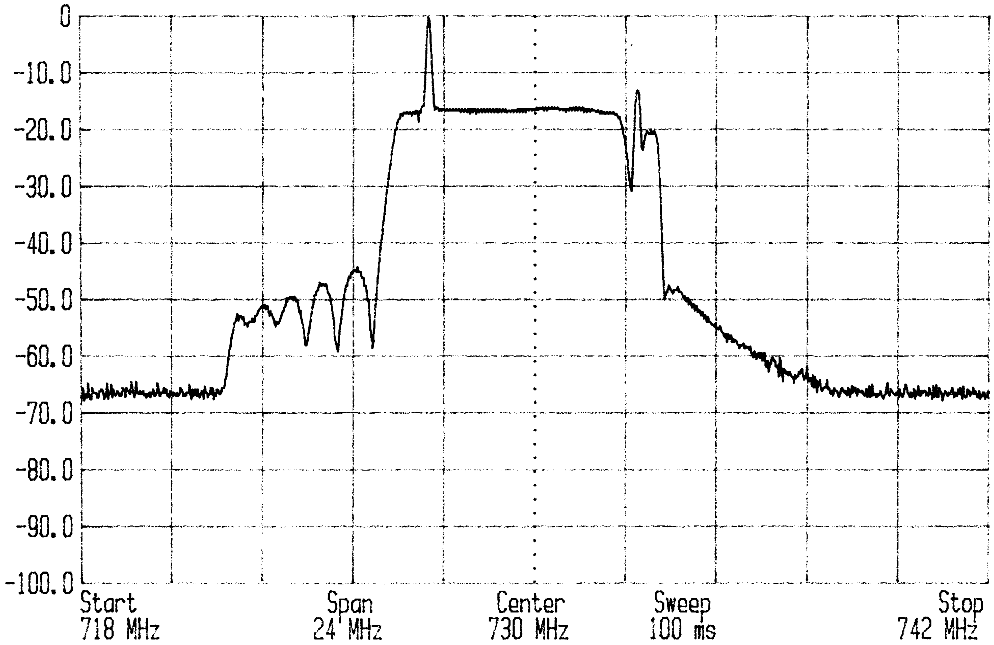


FIGURE 9.14

Spectrum for a system G 10 kW tetrode transmitter with FM-sound and NICAM carriers

DTTB-09-14

ANNEX 1

TO CHAPTER 9

Derivation of protection ratio values used for the asymmetrical DVB-T spectrum masks

8 MHz channels

G/PAL/NICAM interfered with by DVB-T

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –10.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –9.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –5.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 48.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(48.3 + 25.9) dB = –74.2 dB

Analogue mono FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.5 MHz above the vision carrier

Upper end of band: (1.25 + 5.5 + (0.130/2)) MHz = 6.815 MHz.

corresponding to –5.185 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 35 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(35 + 25.9) dB = –60.9 dB

NICAM subcarrier in lower adjacent channel:

Bandwidth: 500 kHz

Centre frequency of subcarrier: 5.85 MHz above the vision carrier

Upper end of NICAM signal: (1.25 + 5.85 + (0.5/2)) MHz = 7.35 MHz.

corresponding to –4.65 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 31 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(31 + 25.9) dB = –56.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +4.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 39 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(39 + 25.9) dB = –64.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +6.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +10.25 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –76.9 dB

G/PAL/A2 interfered with by DVB-T

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –10.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –9.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –5.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 48.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(48.3 + 25.9) dB = –74.2 dB

Analogue mono FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.5 MHz above the vision carrier

Upper end of band: (1.25 + 5.5 + (0.130/2)) MHz = 6.815 MHz.

corresponding to –5.185 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 35 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(35 + 25.9) dB = –60.9 dB

As the necessary protection ratio is lower than that for the second sound carrier and the centre frequency is further away from the DVB-T channel this value is ignored.

Second analogue FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.742 MHz above the vision carrier

Upper end of band: (1.25 + 5.742 + (0.13/2)) MHz = 7.06 MHz.

corresponding to –4.94 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 44 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(44 + 25.9) dB = –69.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +4.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 39 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(39 + 25.9) dB = –64.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +6.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +10.25 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –76.9 dB

G/PAL (Vision/sound ratio = 10 dB) interfered with by DVB-T

For information only, not included in the curves in Figs. 9.1 and 9.2

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –10.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –9.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –5.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 48.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(48.3 + 25.9) dB = –74.2 dB

Lower adjacent mono FM-sound carrier:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.5 MHz above the vision carrier

Upper end of band: (1.25 + 5.5 + (0.130/2)) MHz = 6.815 MHz.

corresponding to –5.185 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 34 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(34 + 25.9) dB = –59.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +4.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 39 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(39 + 25.9) dB = –64.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +6.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +10.25 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –76.9 dB

I/PAL/NICAM interfered with by DVB-T

Vision carrier in lower adjacent channel:

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 50.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(50.3 + 25.9) dB = –76.2 dB

As the necessary protection ratio is lower than that for the vision carrier and the centre frequency is further away from the DVB-T channel. the value for the maximum relative level is replaced by that for vision carrier + 1 MHz: –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –9.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –5.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 45 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(45 + 25.9) dB = –70.9 dB

Analogue mono FM-sound carrier in lower adjacent channel: (–10 dB)

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 6.0 MHz above the vision carrier

Upper end of band: (1.25 + 6.0 + (0.130/2)) MHz = 7.315 MHz.

corresponding to –4.685 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 34 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(34 + 25.9) dB = –59.9 dB

NICAM subcarrier in lower adjacent channel: (–20 dB)

Bandwidth: 550 kHz (–10 dB). used to determine the upper limit and
364 kHz (–3 dB). used to determine the 4 kHz correction factor

Centre frequency of subcarrier: 6.55 MHz above the vision carrier

Upper end of NICAM signal: (1.25 + 6.55 + (0.55/2)) MHz = 8.075 MHz.

corresponding to –3.925 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 31 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(31 + 25.9) dB = –56.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +4.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 41 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(41 + 25.9) dB = –66.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 50.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(50.3 + 25.9) dB = –76.2 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +6.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +10.25 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier + 1 MHz: –76.9 dB

K/SECAM. K/PAL. D/SECAM and D/PAL

(Vision/sound ratio = 10 dB) interfered with by DVB-T

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –10.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 52.8 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(52.8 + 25.9) dB = –78.7 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –9.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 52.8 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(52.8 + 25.9) dB = –78.7 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 6 MHz used to determine the 4 kHz correction factor

5 MHz is the significant point on the protection ratio curve

Upper sideband frequency: (1.25 + 6) MHz = 7.25 MHz.

corresponding to –4.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 47.7 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(47.7 + 25.9) dB = –73.6 dB

Analogue mono FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 6.5 MHz above the vision carrier

Upper end of band: (1.25 + 6.5 + (0.130/2)) MHz = 7.815 MHz.

corresponding to –4.185 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 34 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(34 + 25.9) dB = –59.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +4.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 40.2 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(40.2 + 25.9) dB = –66.1 dB

Vision carrier in upper adjacent channel:

Bandwidth: 6 MHz

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 52.8 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(52.8 + 25.9) dB = –78.7 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +6.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 52.8 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(52.8 + 25.9) dB = –78.7 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 6) MHz = 7.25 MHz.

corresponding to +11.25 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –78.7 dB

L/SECAM/NICAM interfered with by DVB-T

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –10.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 44 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(44 + 25.9) dB = –69.9 dB

As the necessary protection ratio is lower than that for the vision carrier and the centre frequency is further away from the DVB-T channel. the value for the maximum relative level is replaced by that for the vision carrier + 1 MHz: –72.4 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz.

corresponding to –9.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 46.5 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(46.5 + 25.9) dB = –72.4 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 6 MHz

Upper sideband frequency: (1.25 + 6) MHz = 7.25 MHz.

corresponding to –4.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 35 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(35 + 25.9) dB = –60.9 dB

Analogue mono AM-sound carrier in lower adjacent channel:

Bandwidth: 30 kHz

Centre frequency of subcarrier: 6.5 MHz above the vision carrier

Allowance for positive sound carrier offset: 50 kHz

Upper end of band: (1.25 + 6.5 + 0.05 + (0.030/2)) MHz = 7.815 MHz.

corresponding to –4.185 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 54 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(54 + 25.9) dB = –79.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +4.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 34 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(34 + 25.9) dB = –59.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +5.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 44 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(44 + 25.9) dB = –69.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz.

corresponding to +6.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 46.5 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(46.5 + 25.9) dB = –72.4 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 6) MHz = 7.25 MHz.

corresponding to +11.25 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier + 1 MHz: –72.4 dB

7 MHz channels

B/PAL/NICAM interfered with by DVB-T

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –9.25 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –8.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –4.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 48.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(48.3 + 25.9) dB = –74.2 dB

Analogue mono FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.5 MHz above the vision carrier

Upper end of band: (1.25 + 5.5 + (0.130/2)) MHz = 6.815 MHz.

corresponding to –3.685 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 35 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(35 + 25.9) dB = –60.9 dB

NICAM subcarrier in lower adjacent channel:

Bandwidth: 500 kHz

Centre frequency of subcarrier: 5.85 MHz above the vision carrier

Upper end of NICAM signal: (1.25 + 5.85 + (0.5/2)) MHz = 7.35 MHz.

corresponding to –3.15 MHz relative to centre of the DVB-T channel.

NOTE – This frequency is inside the DVB-T bandwidth (+/–3.33 MHz).

The values given below are thus only relevant if the analogue System B and the DVB-T transmitters are offset from each other by more than 200 kHz.

Protection ratio for Grade 4.5: 31 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(31 + 25.9) dB = –56.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +3.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 39 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(39 + 25.9) dB = –64.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +4.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +5.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +9.75 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –76.9 dB

B/PAL/A2 interfered with by DVB-T

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –9.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –8.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –4.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 48.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(48.3 + 25.9) dB = –74.2 dB

Analogue mono FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.5 MHz above the vision carrier

Upper end of band: (1.25 + 5.5 + (0.130/2)) MHz = 6.815 MHz.

corresponding to –3.685 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 35 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(35 + 25.9) dB = –60.9 dB

As the necessary protection ratio is lower than that for the second sound carrier and the centre frequency is further away from the DVB-T channel. this value is ignored.

Second analogue FM-sound carrier in lower adjacent channel:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.742 MHz above the vision carrier

Upper end of band: (1.25 + 5.742 + (0.13/2)) MHz = 7.06 MHz.

corresponding to –3.44 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 44 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(44 + 25.9) dB = –69.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +3.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 39 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(39 + 25.9) dB = –64.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +4.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +5.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +9.75 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –76.9 dB

B/PAL (Vision/sound ratio = 10 dB) interfered with by DVB-T

For information only, not included in the curves in Figs. 9.3 and 9.4

Vision carrier in lower adjacent channel

Frequency: 1.25 MHz.

corresponding to –9.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in lower adjacent channel:

Frequency: 2.25 MHz

corresponding to –8.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in lower adjacent channel:

Bandwidth: 5 MHz

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to –4.25 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 48.3 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(48.3 + 25.9) dB = –74.2 dB

Lower adjacent mono FM-sound carrier:

Bandwidth: (2 * ($\Delta f + f_{mod.\ max}$)) = 130 kHz

Centre frequency of subcarrier: 5.5 MHz above the vision carrier

Upper end of band: (1.25 + 5.5 + (0.130/2)) MHz = 6.815 MHz.

corresponding to –3.685 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 34 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(34 + 25.9) dB = –59.9 dB

Lower video sideband in upper adjacent channel:

Frequency: (1.25 – 1) MHz = 0.25 MHz

corresponding to +3.75 MHz relative to centre of the DVB-T channel

Protection ratio for Grade 4.5: 39 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(39 + 25.9) dB = –64.9 dB

Vision carrier in upper adjacent channel:

Frequency: 1.25 MHz.

corresponding to +4.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Vision carrier + 1 MHz in upper adjacent channel:

Frequency: 2.25 MHz

corresponding to +5.75 MHz relative to centre of the DVB-T channel.

Protection ratio for Grade 4.5: 51 dB

Correction for 4 kHz bandwidth: 10 * log(1 540/4) = 25.9 dB

Maximum relative level in 4 kHz: –(51 + 25.9) dB = –76.9 dB

Upper end of video sideband in upper adjacent channel:

Upper sideband frequency: (1.25 + 5) MHz = 6.25 MHz.

corresponding to +9.75 MHz relative to centre of the DVB-T channel.

Taken equal to the value for the vision carrier: –76.9 dB

CHAPTER 10

IMPLEMENTATION STRATEGIES

10.1 Introduction

The introduction of digital terrestrial television can be seen from a short-term or a long-term point of view. Objectives, constraints and possibilities differ in both cases, leading to several possible introduction scenarios, some in line with the short-term objectives and other better suited to the long-term. In addition, adequate transition methods from the short-term to the long-term scenarios have to be found. The three scenarios, short-term, transition phase and long-term are introduced below. Further details are given in § 10.4.

Spectrum usage differs in various countries. In spite of of these differences, it seems that there are some common threads in the ways in which terrestrial digital television may be introduced, how it may develop and what its longer term uses may be. However, it has to be accepted that consideration of the long-term situation is rather speculative as there are so many variables involved. This section attempts to provide a general overview rather than concentrating on the detail in individual countries. In this overview, a distinction is made between short to medium term scenarios (indicated as S1, S2 below and long-term scenarios (indicated as L1, L2 below)).

10.2 Implementation scenarios

10.2.1 Short-term scenarios

Short-term scenarios deal with the introduction of digital terrestrial television within the next few years. In this phase, digital television has to be accommodated in frequency bands, which are already extensively used by existing analogue broadcasts. Thus to fit into the current analogue spectrum, the new service has to cope with the major constraints of:

– being forced to adopt the existing channel structure, and

– protecting the existing analogue services.

In addition, it is necessary to reach the maximum achievable coverage for the digital service, at minimum expense for the viewer interested in the reception of such services and to provide an attractive development basis for the new technology.

For these reasons it seems reasonable to classify the various short-term scenarios according to the guideline of the different spectrum constraints that an introductory strategy encounters.

Current studies have been confined to the UHF band, because VHF Band III has various channel rasters in different countries and, in addition, might not be exclusively available for television broadcasting in future. The opinion of some television receiver manufacturers is that digital receivers for multiple channel bandwidths may be somewhat more expensive.

10.2.2 Long-term scenarios

Long-term scenarios deal with the final implementation of digital television. At this stage, if the new technology has a successful market penetration, analogue services will have been phased out and digital transmission will be the only means of television broadcasting.

The scenarios can be classified according to the different objectives, which are aimed at with digital services (e.g. service area, service mode, implementation, maintenance costs, etc.). The classification is done according to the size of intended digital service area (national or local) and variants of each class with respect to service modes network modes and implementation costs are described.

The chosen subdivision along coverage areas must not be seen as exclusive, rather it is complementary since, in general, countries will implement networks of at least two kinds, and possibly all three, in a layered structure.

In the long-term future a mixed analogue/digital environment is not envisaged, because it does not make the most efficient use of the spectrum. On the other hand, if terrestrial digital technology does not encounter the favour of the market, terrestrial broadcasting either remains analogue or is phased out. There would then be no portable reception and the coverage of local/regional programmes would be limited.

10.2.2.1 Transition phase

Short-term and long-term scenarios differ in their objectives and, as a consequence, in their technical implementation features. Therefore, a third phase considering the transition of short into long-term implementations has to be analysed. It describes various ways to extend digital services in a mixed environment, where constraints on digital transmission should decrease and networks should be modified as appropriate, with frequency changes and site modifications. By decreasing constraints, implementation possibilities of digital terrestrial television will increase.

In addition, to facilitate the transition for the viewer, the receiving equipment should have some facilities such as automatic tuning of receiver, wide band receiving antenna systems, and so on. Until now only few studies have been carried out on this subject.

10.3 Frequency management

10.3.1 Spectrum requirements

Terrestrial digital television services started in 1998 in North America and in Europe.

Partly because of the complex arrangement existing in Bands I and III, the studies concerning the possible introduction of DVB-T in Europe have mainly concentrated on the possibilities offered by the UHF band. On these channels, the new digital signals have to share the available spectrum with the analogue programmes.

It is possible that, in the long-term, a uniform channel bandwidth, together with a uniform alignment of channels for terrestrial digital television, could be adopted in the VHF band throughout Europe, although this is by no means certain.

10.3.2 The DSI phase II investigation

Within Europe, the CEPT has carried out its second detailed spectrum investigation (DSI), which covers the frequency range 29.7 to 960 MHz (therefore including the bands used for terrestrial television broadcasting). The final aim of the DSI is to establish a European Common Frequency Allocation Table (ECA) for all the CEPT countries. Such a process may lead to changes in spectrum allocation that could be implemented by about 2008.

The results of the DSI which affect the television situation in the VHF/UHF bands are given in Table 10.1.

TABLE 10.1

Final DSI 2 proposals

Frequency band (MHz)	Proposals
47 to 68	Proposal for television to be stopped
174 to 216	Shared with mobile services; possible reallocation after transition period
470 to 862	New channels mainly for digital television; after transition period digital television only

For some time, there has been strong pressure from mobile services for access to the broadcasting band below 900 MHz. Although recent projections made by the European Radiocommunication Office indicate a decreasing usage of frequencies below 900 MHz by the mobile services, the pressure for access remains.

On the other hand, many administrations and organisations are making alternative proposals to the CEPT regarding future use of this part of the spectrum. In particular, the EBU has proposed that the frequency ranges 174 to 216 MHz and 470 to 862 MHz should be available for terrestrial television on an exclusive basis, to permit development of the new market opportunities.

10.3.3 Short-term period

The introduction phase of DVB-T will require the maximum spectrum availability, because of the current heavy usage of the VHF/UHF bands by analogue television services and the need for DVB-T to share with these services.

Thus, in the short-term phase, all of the spectrum dedicated to television in Bands III, IV and V will be needed for the successful introduction of DVB-T, including those channels which cannot be used for broadcasting at present.

10.3.4 Long-term period

Digital technology provides enhanced spectrum efficiency. Thus, by today’s criteria, in the long-term, less spectrum might be necessary for the existing terrestrial broadcasts, and the present need of new channels expressed by European broadcasters could also be satisfied. In this hypothesis, part of the spectrum could be allocated to other services.

However, the following factors may lead to the complete utilization of most, if not all of the spectrum currently available for television:

– it is likely that new market opportunities, based on additional services, will be developed thanks to the added flexibility which digital television can offer. Such market possibilities are restricted today by the limited capacity and versatility available within the existing analogue systems;

– the desirable future implementation of HDTV, when wide flat-screen displays become available on the market, at a reasonable price, will substantially increase the spectrum occupied by each programme;

– with the introduction of digital broadcasting services – which are more robust than the current analogue ones, a wider use of portable receivers is to be expected. To satisfy all the foreseen requirements for portable or mobile reception, a rugged system variant may be necessary and, as a consequence, more spectrum would be required for each programme.

In the future, all these possibilities might lead to the complete utilization of all the spectrum currently available for television, considering Europe as a whole. However, it is very difficult, if not impossible, to estimate today the possible long-term future trend, although it is already clear that there is a significant demand for new broadcasting services.

10.3.5 Transition period

Because there are several hundred million television receivers in use in Europe, and because the minimum lifetime of a modern television set is at least seven years, it is essential that the transition period is sufficiently long to avoid any disruption of service.

For these reasons it is perhaps not unreasonable to assume that the transition period will last some 15 years from the introduction date of digital services. As the latter date will not be the same in all European countries and as it is unlikely to occur in any country, it is easy to see that the transition period will last for such a long time that any prediction about the post-transition era must be considered somewhat speculative.

Therefore, it is quite unrealistic to envisage a rapid phasing out of the existing analogue services. It is possible that, around the year 2015, DVB-T might be nearing the end of the implementation phase in some countries, while in others the penetration of digital television might only be partially completed.

10.3.6 Some speculative considerations

If the digital terrestrial television is successfully introduced and replaces analogue television, it is possible to give some consideration to band usage.

10.3.6.1 Band I

Measurements have shown that man made noise levels in Band I (47 to 68 MHz) are considerably higher than in other television bands (Bands III, IV and V). Moreover Band I suffers from sporadic E propagation which may cause abrupt failure for digital systems at low time percentages. Consequently, Band I is deemed to be less suitable for DVB-T than the other television Bands and can be considered for reallocation.

In a number of countries, analogue television transmissions in Band I (particularly in the lowest channels) could be closed in the short-term. Nevertheless, a too early closure of such services could lead to their transition to the UHF band, lessening the possibilities for the introduction of DVB-T.

It is therefore recognized that, for operational reasons, there may be the need to continue the operation of analogue television in Band I, in particular when the replacement of the transmitters in UHF is either not possible, or limits the introduction of DVB-T.

10.3.6.2 Bands III, IV, V

To facilitate the introduction of terrestrial digital television, all of the spectrum in Bands III, IV and V, which is normally available for terrestrial analogue television, will be required for DVB-T services for at least 20 years. Indeed, these bands will continue to be the primary medium for many broadcasting services and will also form a key part in the strategies for the transition from analogue to digital.

In addition, taking into account the increasing pressures in the spectrum during the transition period (while analogue services are being complemented and then replaced by digital services), it could be very highly desirable for broadcasting to have access to some spectrum not currently available in some countries.

Concerning the frequency sharing between analogue and digital services, a CEPT indication proposed that any channel that may become available in the short and medium term should not be used for extending the analogue networks. This requirement is quite a delicate issue. Very distinct situations exist in the different European countries and different digital start up philosophies are predictable. Therefore, a limitation on the development of analogue television may be too severe for the time being. In a general way, analogue television should still be allowed to be introduced.

In the long-term, the market potential for terrestrial transmission, will probably develop new opportunities and services, that only terrestrial transmission has the ability to provide. Therefore, despite the enhanced spectrum efficiency, the traditional bands assigned to the television terrestrial transmission could continue to be required for these services. On the basis of future periodical reviews of the situation, the appropriate decisions have to be taken.

10.3.7 Can spectrum be released for use by other services?

Summarising the aforementioned ideas, it is very difficult and may be premature to predict today either the spectrum requirements for digital television in the long-term, or the time scale for its penetration.

As already said, it is possible that, by today's criteria, less spectrum will be needed for DVB-T in the future, but is very speculative to estimate the required amount at this stage. During the transition period, which could last well until 2015, the number of terrestrial digital television services, their nature and several other factors will become clearer as they evolve.

Therefore, it is not possible, at the moment, to assess if any spectrum might be transferred to other services. Further reviews of spectrum usage and requirements will be needed periodically in future years to establish what spectrum allocation should be undertaken.

10.3.8 Conclusions

DVB-T was introduced in the UHF bands by some countries in Europe in 1998. To facilitate early start-up, short-term strategies have been adopted by these countries. The spectrum has to be shared between the terrestrial analogue and the new terrestrial digital television services.

As the number of receivers for DVB-T increases, it will be possible to withdraw the analogue services by a phased closing down of analogue transposers and transmitter stations. This is the transition phase between the shared use of the spectrum by the analogue and digital services and the long-term aim of an all-digital television scenario. During this phase, channels will be released by the closure of analogue transmitters and these channels can be used to increase terrestrial digital coverage.

Finally, in a long-term scenario, the overall coverage will be achieved by DVB-T networks in Bands III, IV and V. The analogue transmissions will have been phased out and the released spectrum used by digital services. The completion of the conversion in any particular country could take a period of about 10 to 20 years from the start of implementation.

It is highly probable that an ITU Conference will be needed in the middle of the current decade in order to produce a Plan, at least for the European Broadcasting Area, to replace the Stockholm '61 Plan.

10.4 Some possible implementation scenarios

10.4.1 Short-term period

The main constraint for the introduction of digital terrestrial television in the immediate future (for example in the next five years) is the protection of existing analogue services.

Countries in Europe can be considered in two broad categories:

Category 1

Those countries which have access to assignments which are currently unused for television stations, or even networks, at relatively high effective radiated power which have been fully coordinated. Such channels are referred to below as "free channels". Countries which can obtain access to, for example, channels above 60, may also be considered to be in Category 1.

Category 2

Those countries which do not have access to unused (relatively high power) television station assignments.

This separation is convenient because different implementation strategies are possible in the two cases. However, even countries in Category 1 will probably not have access to sufficient spectrum to satisfy all of their requirements and will thus come into Category 2 for some requirements.

Three types of introductory scenarios, denoted in the paper by S1 to S3, can be foreseen. They refer to the various kinds of spectrum usage introductory digital scenarios have to cope with.

– S1: use of existing or planned assignments;

– S2: re-use of existing (used) channel assignments;

– S3: use of "free channels".

The chosen separation into the various scenarios is not exclusive. Countries may take up some or all approaches simultaneously, depending on the spectrum situation they are faced with.

10.4.1.1 Short-term scenario 1 (S1): Use of existing or planned (but unused) channel assignments

The first scenario is possible for those countries belonging to Category 1.

In this case, large coverage areas are achievable, since there are relatively few restrictions on the radiated power. Therefore, it can provide an excellent starting point for the introduction of DVB-T and may make the core of a future digital MFN or represent, in the case of an entire network, the backbone of an MFN-based long-term scenario.

In general, it can be assumed that the coverage of plan assignments will be similar for digital and analogue services. However, caution has to be taken in those cases where planning of analogue service has been performed with precision off-set, which significantly reduces the restrictions on the analogue transmitter, or where a non-rugged digital service with the requirement for a high protection is intended. These circumstances might reduce the digital coverage.

It is essential that the relevant protection ratios and e.r.p. values for the digital and analogue services are considered on a case-by-case basis to ensure that an existing assignment can be used to achieve the digital service required.

The concept of mini SFNs provides an adequate means to cope with such kind of restrictions. Indeed, the interference potential of a mini SFNs configuration is significantly lower compared to a single transmitter solution.

The concept of mini SFNs can also be used to improve coverage, in particular in the case of portable reception.

Implementation costs of scenario S1 will be relatively low for the broadcasters, since in almost all cases transmitter installations already exist, if a conventional single transmitter solution is chosen.

Where the digital channel is close to the analogue one, there is a particular cost advantage for the viewers, because they can use their existing receiving antenna system. This cost advantage for the consumer may be a decisive aspect for the assessment of a digital scenario in the introductory phase when the supply of digital services is not yet completed and, therefore, has not yet reached its full attractiveness.

Use of the dense network approach substantially increases implementation costs for both broadcasters and consumers, since additional transmitters have to be installed as well as new receiving antennas.

Coordination effort is likely to be negligible since already coordinated channels are used.

10.4.1.2 Short-term scenario 2 (S2): Re-use of existing channel assignments

Scenario (S2) applies to Category 2 countries without free assignments and to Category 1 countries having already used their free assignments for their first digital networks, looking for alternative channels.

Channels that are only of very limited use for analogue services may be available for DVB-T, because of its higher robustness and smaller interference potential. Therefore, even an highly saturated UHF spectrum might offer some resources for the introduction of digital television. Of course, this situation is not favourable for the realization of large scale SFNs.

The implementation of re-used channel assignments necessitates coordination with neighbouring countries. However, the approach fits into the existing spectrum usage.

For fixed reception it is possible to consider two variants (described in the Chapter):

– **scenario S2a** – using existing transmitter sites and, whenever possible, channels near the existing analogue ones for the start of digital transmission;
– **scenario S2b** – adding new broadcasting sites to existing analogue ones.

For portable reception this distinction is not relevant.

The feasibility of scenario S2 strongly depends on the density of the current analogue services and differs significantly from country to country. Once started, it may serve as a basis for a long-term digital service in MFN mode. Until now, scenario S2 is the most investigated introductory approach in Europe.

The main features of Scenario 2 are:

a) *Protection of existing or planned analogue services*

Any new station, digital or analogue, causes some increase of interference to existing viewers and thus causes a coverage reduction. The constraints on the power of digital stations will be set by considerations of how much additional interference to analogue viewers is tolerable and for what percentage of the time. The size of the digital coverage areas will be determined by a combination of factors:

– the radiated power of the digital transmitter;

– the amount of interference from analogue or other digital transmitters;

– the required *C*/*N* ratio for the digital service.

The required protection of analogue services will lead to e.r.p. restrictions, therefore to limitations of the achievable coverage areas for digital television.

b) *Fixed roof top reception and limited portable reception*

Most studies have shown that, at least during the transition period when analogue and digital services will have to coexist, the coverages achievable for portable reception are likely to be rather limited. However, useful portable coverage could be achieved if a transmitter is close to a population centre.

The implementation scenario where existing transmitter sites are used, may therefore take reception with fixed roof top antenna as a basis. Portable reception is subject to variable conditions compared to fixed roof top reception and depends on receiving height, building penetration loss and local signal variations. Depending on the situation (indoor or outdoor, high or moderate location probability, bit-rate requirement, network configuration), portable reception may be possible to a large proportion of the population.

It must be noted that in countries with high cable and satellite penetration, portable reception is seen as the primary target for future terrestrial services.

c) *Possible change of frequency for low power analogue stations*

It is clear that it will not be possible, in general, to change the channels used by the higher power analogue transmissions because of the widespread disturbance to analogue reception which could be caused. However, some of the e.r.p. restrictions on digital stations may be caused by the need to protect low power (and low coverage) analogue stations. In such cases, analogue channel changes may be feasible and this could make a significant improvement to the digital coverage achievable. In this context, it has to be remembered that there are more than 30 000 operating stations with less than 10 W e.r.p. in Europe in addition to the 30 000 stations above 10 W.

10.4.1.3 Scenario S2a: use of existing transmitter sites

Most homes already have a domestic receiving antenna, which is both frequency selective and oriented with a particular direction and polarization. In order to maximise the commercial attraction of digital transmissions, it is desirable that they should be easily receivable, for example by the existing receiving antenna systems. Indeed, the added cost of installing a new antenna represents a significant disadvantage for most viewers, especially if there are no additional programmes to be received (simulcast of analogue service).

Therefore a good starting solution is the use for the digital programmes of the same sites already transmitting the analogue ones. In addition, the new channels should be close to those used for existing analogue services and the same polarization should be used.

Because the services come from the same site and because the digital power would be lower than that of the analogue service (for protection purpose), there would be little risk of causing adjacent channel interference to the existing analogue viewers.

It should be noted that if a country wishes to prepare a long-term future with SFNs, the choice of adjacent channels to the analogue services may lead to transition problems between the short and long term scenarios.

10.4.1.4 Scenario S2b: adding new transmitter sites

Scenario S2b is a variation of scenario S2a. It is mainly based on the same assumptions for most areas, but with the addition of new low power stations in those areas where the protection of existing analogue services prevents adequate digital coverage from existing sites.

In such configurations, the receiving antennas used by the viewers of the analogue services are unlikely to be suitable for reception of the digital services from the new relay stations, because of the different channel and direction.

On the other hand, they have the advantage of reducing the impact of interference from the digital network and thus the coverage areas may be increased in size. However, there is the risk of causing interference to existing analogue viewers in some marginal parts of the analogue coverage area. Indeed, there are likely to be viewers of the analogue transmissions from the main station who are situated close to the new digital relay station sites. Such viewers may experience interference from the digital transmissions, if these use channels adjacent to those of the analogue services, due to the high level digital signals in areas where the analogue signal is relatively weaker.

Concerning implementation costs, obviously, such networks are likely to be more expensive to install than conventional ones, because of the need for additional transmitter sites.

10.4.1.5 Short-term scenario 3 (S3): Use of "free channels" on a national or regional scale

In some countries there exists the possibility that in the UHF band one or more channels will be released for the implementation of digital services on a nation-wide basis. These channels are either not currently allocated to television broadcasting, or they are allocated but not used by television services.

In some European countries, the UHF channels 61 to 69 are used by the military or by fixed services. There is a prospect, encouraged by CEPT considerations, that some or all of these channels can be made available for digital television broadcasting.

10.4.1.6 SFN implementation

This kind of situation offers a unique chance to implement an SFN-based digital service on a national or regional scale. It potentially represents the introduction of an attractive long-term scenario from the very beginning.

In general, the use of these channels will not be possible on an entire nation-wide basis because of neighbouring countries, which probably use these channels for analogue or other services. Coordination will be difficult in these cases, as long as the neighbours operate these channels on an MFN basis.

As compared to single transmitter or MFN introductory strategies, portable reception finds larger coverage because of the higher homogeneity of the field strength within an SFN. Additional increase of transmitter density, by means of low power stations, would offer the option for a national or regional coverage for portable reception from the very beginning. This would imply further, new transmitter sites which increase the implementation costs.

10.4.1.7 Conventional planning

If the existing conditions do not permit the implementation of SFNs (e.g. when neighbours have already access to these channels, and the realization on a single frequency basis becomes hard to coordinate) or, simply, if the SFN mode is not seen as the long-term scenario, the free channels can be utilized in the conventional mode.

10.4.1.8 Implementation of SFNs

Some studies advocate extensive use of single frequency network techniques at an early stage in order to prepare for a long-term future. SFNs offer the major advantage of requiring few channels to cover large areas. Indeed, in theory, only a single channel may be needed for a complete national network. Equally, regional SFNs may be possible in many countries preferring such coverage. In addition, SFN technique improves portable reception.

However, care should be taken not to put too much emphasis on this technique. Indeed, the heavy congestion in the UHF band with analogue systems will, in many instances, prevent or lessen the possibilities of using the SFN technique – for instance the use of channels near to the analogue ones. In some situations these considerations may lead to an easier introduction of digital television using conventional planning.

10.4.2 Long-term period

Terrestrial digital television introduction requires not only good perspectives in the near future, but needs also to be examined from the long-term strategy point of view (15 to 20 years from start).

In this phase it is also possible to classify the potential scenarios into three categories. These include two different maximisation strategies:

– scenario L1: maximizes the amount of coverage;

– scenario L2: maximizes the number of programmes in limited areas;

– scenario L3: no digital terrestrial transmissions (put in only to provide further thought).

Compromises between these extremes are not considered here.

10.4.2.1 Long-term scenario 1 (L1): Maximising size of coverage areas

Scenario L1 is based on single frequency networks.

10.4.2.1.1 Wide coverages

As already stated, SFNs offer the major advantage of requiring few channels to cover large areas. Indeed, in theory, only a single channel may be needed for a complete national network. MFN planning would require several channels.

With an SFN a national coverage can be achieved on the same channel, subject to any limitations caused by self-interference. As neighbouring countries will claim an equitable share of the spectrum, not all channels can be used for national coverage. Experience in planning SFNs for T-DAB has shown that, at least in Western Europe, 6 or 7 channels are needed to provide regional/national coverage in all countries. Of course, channels from neighbouring countries can be re-used for more local services at a certain distance from the border.

For band IV/V, channels 21 to 60, there could be 8 channels available in each country for national coverage. In a digital mode with 4 programmes per channel this means 32 programmes

10.4.2.1.2 Portable reception

As an alternative to large number of programmes, one can prefer portable reception over the wide area, the capacity available being used to achieve robustness instead of a large number of programmes.

10.4.2.1.3 Analogue networks closure

In most cases, finding a clear channel all over a wide area means the closure of existing analogue stations working on this channel. Therefore, scenario L1 is a straight forward scenario, but it implies the end of analogue transmissions at a given time and needs a strong will and a good management of the transition period.

10.4.2.2 Long-term scenario 2 (L2): Maximising number of programmes in limited areas

Scenario L2 applies when wide coverages, as in scenario L1, are not the main objective. In this case more possibilities become available. Indeed, it may be reasonable to ask if full national coverage is a requirement for terrestrial digital television in the presence of alternative delivery media such as cable and satellite.

Therefore, the aim would be to concentrate on coverage of urban areas which may limit the necessary investment costs. Elsewhere, people would receive the programme from satellite or by any other means, although it has to be remembered that few cable systems extend into really rural areas.

The aim could then be to maximize the number of programmes available, especially for portable reception.

SFNs with wide areas (Scenario L1) require sharing of the available channels between neighbouring countries, thus dividing the total possibilities in the available frequency range, as indicated above.

In scenario L2, where services are concentrated over limited areas, possible interference between co-channel service areas are of less importance. All channels can be used in these limited areas.

For the band IV/V channels there could be some 40 channels available for local coverage. In a digital mode with 4 programmes per channel this means 160 programmes. However, the high bit rate, implied by having four programmes per channel, implies that there would be an increase in the size of the areas near country boundaries where channels would need to be shared between countries and this means that fewer programmes would then be available in such areas.

10.4.2.3 Scenario long-term 3 (L3): No terrestrial broadcasting

In this scenario, DVB-T services would not start and the existing analogue services will eventually be phased out. Digital programmes would only be delivered by satellite or cable. It now seems that this is an unlikely scenario.

10.4.3 Transition period

During the transition period, analogue and digital services have to co-exist. While constraints on digital transmission should decrease, networks should be modified to extend digital service. The main uncertainty associated to this phase is the duration of the transition period.

Any spare capacity which can be found during the transition period will mainly be used for digital transmission. In those countries in which the number of available channels is insufficient, the coverage attainable by digital services will be limited. This constraint increases the problem of terminating the transition period.

This section outlines some ways to decrease the constraints on digital transmission. It should be noted, however, that an increase of interference to analogue stations may make the conversion of these stations to digital more difficult in the future.

10.4.3.1 Reduction of analogue service protection

Generally, it can be assumed that the transition period will be characterized by decreasing importance of analogue coverage as the penetration with digital receivers grows. This gives the possibility to reduce gradually the protection of some or all analogue transmitters, allowing for an increase of digital coverage by higher transmitter powers and/or implementation of new digital transmitters (although in some countries they may be legal constraints which would prevent this reduction of coverage). It is a means to expand MFN-based digital services on a national, regional or local scale.

The protection of the existing analogue transmission is controlled through three major parameters:

– level of impairment of the analogue service due to the digital transmission: the level of interference is measured through the increase of the "usable field strength". For example, an increase of 3 dB corresponds to an impairment of half a grade on the ITU-R quality scale;

– percentage of time during which analogue transmission is protected: in principle, 99% of the time;

– percentage of locations where the analogue reception is protected in its service area: in principle, 50% (at the edge of the service area).

By modifying progressively the above parameters, the implementation possibilities of digital terrestrial television will increase. This could be measured in percentage increase of the covered population.

10.4.3.2 Reduction of the number of protected analogue transmitters

Closure of analogue transmitters or change of their frequencies are important possibilities for the improvement of digital coverage, by allowing either for a larger coverage of already existing digital transmitters, or for the installation of new digital transmitters at the sites of the closed analogue transmitters.

It may therefore be important to reduce the number of analogue transmitters to be protected during the transition stage. However, not all analogue transmitters are likely to be removable without substitution, implying additional implementation costs for analogue installations which are to be phased out in the near future.

Another variant of this scenario arises if a country decides to move a national (or regional, if the region is large enough) analogue service from terrestrial to satellite service. In particular, this case may occur if the country intends to provide this service from satellite as a digital one. The free spectrum may then serve for the extension or implementation of one or more regional MFN-based digital services.

Implementation costs for the latter scenario are likely to be low, since transmission infrastructures and receiving antenna systems already exist.

Also coordination with neighbouring countries should be relatively easy – the new digital transmitters being in the place of already coordinated analogue transmitters.

10.4.3.3 Completing SFNs

In a similar evolutionary way, the transformation of a short-term SFN into a long-term SFN can be accomplished. In this case, the installation of new transmitters may be necessary to increase the density of the network, allowing for a higher degree of portable reception.

This scenario does not imply much additional coordination effort since the channel is already coordinated for the whole service area but it would be necessary to ensure that the additional transmitters do not significantly increase interference levels.

Implementation costs are the usual ones required for the installation of SFNs.

The transition from a former short-term national SFN to a long-term regional SFN seems to pose no problems, as after the introductory phase, more channels are available for digital within the envisaged coverage areas.

10.4.3.4 Transformation of MFNs into SFNs

A difficult transformation process is encountered if a country introduces its digital services on an MFN basis but aims at the SFN mode in the long-term.

A network can only operate in the SFN mode under the condition that its channel is cleared for the entire service area. If there are still analogue services using this channel (and this is probable as long as there is any analogue service in operation) the affected analogue transmitters would have to be shifted in frequency.

Among these transmitters, there will certainly be main stations realising a considerable amount of population coverage. It is questionable whether it makes sense (because of the large cost efforts for broadcasters and consumers) to re-arrange an analogue service which will be soon phased out. However, there may be suitable channel configurations that make this transformation practicable.

If the envisaged SFN channels are thoroughly used by digital transmitters, on an MFN basis, the transition seems to be less problematic. Transmitters would have to be changed from their MFN channels to the SFN channel. This will result in some costs with respect to transmitter installations since, in general, MFN and SFN channels will not be situated close together, and possibly re-arrangements of the receiving digital antenna installations would be necessary.

With respect to national/regional SFN coverage, this scenario makes coordination with all neighbouring countries inevitable. Since SFN and MFN approaches are not compatible, at least on a large scale, neighbouring countries would have to agree on which to use. Therefore, a large coordination effort is expected to be required for this scenario and a lot of good will would be necessary for it to succeed. In practice, a Regional Planning Conference would be needed. The preparation for this would take some time and would involve agreements with neighbouring countries in other Regions.

21453

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