4.5 The packetisation approach and functionality

4.5.1 Overview

figure 35

Transport Packet

A DTTB communication system transport bit stream can consist of either fixed length packets or variable length packets. The packetisation 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. 35.

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

figure 36

Link Header Format

The link layer is implemented using a four byte header field. Fig. 36 shows a possible link layer header with functionality assigned to each bit. Table 6 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.).

TABLE 6

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.)

field	Function/Usage
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 1) as a separate private stream with it's own PID, or 2) 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.

figure 37

Fixed-length Component of Adaptation Header

The header starts with a fixed length component that is present whenever the adaptation header is transmitted. The format is shown in Fig. 37.

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. 37 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 7.

TABLE 7

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.

figure 38

PCR and OPCR Header Format

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The PCR and OPCR fields are described in Fig. 38 and Table 8.

TABLE 8

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 milliseconds.).
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 hours.

Figure 39

transport_private_data and adaptation_field-extension Header Format

The transport_private_data and adaptation_field_extension fields are described in Fig. 39 and Table 9.

TABLE 9

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 10 is a one byte field that is present if the splicing_point_flag is set.

TABLE 10

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.

figure 40

Program Association Segment and Table Header Formats

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 1024 bytes. The program_association_table is described in Table 11. The transport decoder can extract individual table segments from the bits stream in whatever order it desires. As shown in Fig. 40, 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.

TABLE 11

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 1024. 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.

figure 41

Program Association Table Entry Format

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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. 41.

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 $x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^{8} + x^{7} + x^{5} + x^{4} + x^{2} + x + 1$.

figure 42

TS Program Map Formats

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. 42. 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.

The header format for the TS_program_map_section is shown in Fig. 42. 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.

figure 43

Elementary Stream Description

Each elementary stream description consists of a 5 byte fixed length component and a variable length elementary_stream_descriptor component as shown in Fig. 43 and described in Table 12.

TABLE 12

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*.
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.

* 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 1) 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 2) 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. 44.

figure 44

Structure of the Event Descriptor

descriptor tag	descriptor length	event id
8	8	32

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.

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. [Draft Recommendation 11-3/XI (Doc 11/37)] 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 notional 6 MHz system, or 3 bits/s/Hz for notional 7 or 8 MHz systems.

Theoretical spectral efficiencies of up to 4 bits/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/sec and the Data Frame rate is 20.66 frames/sec.

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 is the most commonly proposed MCM system.

5.3.3.1 Orthogonal Frequency Division Multiplexing (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 2048 (2k) carriers and 8192 (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 synchronisation 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 224μs 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 (7/28μs) 1/8 of the symbol period (28/112μs) and 1/4 of the symbol period (56/224μs) and 1/2 of the symbol period (112/448μs). 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 896μs and a 1/4 of symbol period guard interval results in a 224μs 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 45 shows how the FFT sampling window, which is equivalent to the symbol period can be positioned within the symbol and guard interval to minimise Inter Symbol Interference (ISI).

Figure 45

Guard Interval Utilisation

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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 maybe 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, synchronisation 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 46 provides a diagrammatic comparison of various modulation system constellations.

Figure 46

Comparison of modulation state Constellations

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Figure 47

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

a) b) c)

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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 equalise the phase and amplitude of each carrier. This can be done after the FFT using a simple equaliser. This process is known as Channel Estimation and Equalisation. Different techniques are proposed to estimate the channel and therefore equalise 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 48 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 48

Effective of frequency selective fading on carrier amplitude

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The Table below provides a typical selection of the characteristics of OFDM systems in an 8MHz channel with appropriate guard intervals. All figures are in Mbit/s.

TABLE 13

SYSTEM COMPARISON FOR SOFT DECISION DECODING (REQUIRED C/N FOR BER= 2M §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] Note - For the QAM perfect channel estimation had been used.

[2] Note - 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 synchronisation symbols has been taken [1].

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.

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 minimise 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-split 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.

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 equalisation 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 $t = (n-k)/2$ 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 LSI's 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 minimise 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 8000 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 3/4 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
[Doc. 11-3/33] 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 14 provides measured data on the peak to average power ratios of a typical COFDM signal.

TABLE 14

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 Es/No 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.

6 Planning studies and Implementation strategies

6.1 Introduction

The potential advantages of digital terrestrial television broadcasting (DTTB), in terms of service quality, lower costs and programme diversity, are summarized in § 1 of this report. In frequency planning terms it is generally agreed that, 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 - thus enabling the very considerable benefits to be gained (compared to the current analogue situation), in terms of service provision and spectrum utilization.

However it should be recognized that, in reality, the spectrum situation is far from ideal, and that 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 of course other geographic areas covered by these regional plans, where the allocated spectrum has not yet been heavily exploited, and where it will therefore 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 the concern to find 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 "domination" which 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 fully digital operations had been achieved there will be the opportunity to upgrade the DTTB services to their full potential, possibly releasing some of the allocated spectrum for reallocation to other services.

The factors that must be taken into account in the planning of DTTB service coverage areas are discussed in some detail in § 6.2

Differing perspectives on the problems to be faced in achieving a strategy for DTTB service implementation whilst retaining a migratory path to "all-digital" operations in the future are described in § 6.3.

6.2 Planning factors

6.2.1 Current analogue systems

The first television systems were developed independently in several parts of the world, and despite the substantial efforts towards standardization 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 "taboos" on the assignment of transmitters to related channels in that area had to be respected. Although consumer receiver performance has since improved significantly, these "taboos" (such as that preventing the assignment of adjacent channels to two transmitters located at the same site) have so far remained in effect.

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 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 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) 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 reorganizing 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.

6.2.2 Sharing considerations

6.2.2.1 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 spectrum has a similar spectral characteristic to Gaussian noise. The effect of CCI is therefore to raise the noise thresholds of analogue receivers which in turn reduces the picture grade (ITU 5-point 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 (continuous) and 3.0 tropospheric are the norm. For initial implementation of digital systems in the vacant channels between existing analogue channels it is possible to allow some overlap of the digital signal into the adjacent analogue channels. For DTTB power levels 20 to 23 dB below the analogue up to 0.5 MHz extra bandwidth for the digital signal can be used without significant impairment to the analogue services. This may provide an opportunity for later channel rationalisation when analogue services are phased out.

6.2.2.2 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, an increasing susceptibility as the modulation levels are increased from QPSK to higher modulation levels like 16 and 64 QAM (by approximately 7 dB and approximately 13 dB respectively in theory). However, as is shown in § 4, the increased transmission capacity of the higher modulation levels allows very sophisticated error-management schemes to be used, that more than compensate for this loss to provide an overall gain in performance.

6.2.2.3 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 § 4 can deal effectively with this type of interference to ruggedise 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 multicarrier in nature.

6.2.3 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, and indeed at many locations well beyond it.

Considering the transmission of television and sound having a particular information content, 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 which 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 which 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 which 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 which are intended to provide high quality and 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.

6.2.4 Digital system techniques

6.2.4.1 Single-carrier techniques

In the case of single-carrier systems, which are described in detail in § 4, the factors for the planning process are affected by choices made in connection with the frequency planning for the area. These choices include coverage, graceful degradation, service availability, picture quality, expected receiving conditions and effects on current analogue services. For single-carrier systems, the new digital service will be introduced as an enhancement or as a simulcasting replacement to the existing analogue TV service and will need to coexist with that service over a lengthy introduction time period. Additionally, the digital service must fit into the existing spectrum allocated to TV broadcasting.

In the present environment to coexist with the present analogue TV services, factors are influenced by the implementation choices based on population and spectrum usage density, i.e. in dense environments coverage is likely to be interference-limited, in less dense environments coverage may extend to match the protected contours of the existing analogue service.

6.2.4.2 Multicarrier techniques

As described in detail in § 4, the approach taken is to modulate the data onto a large number of carriers at closely-spaced frequencies. The symbol rate of each carrier is thus very low, giving it a narrow bandwidth. The key point is that the carriers are orthogonally spaced in frequency - that is, they are spaced such that there is no mutual interference. Benefits occurs when using this system in the presence of interference and multipath, as only some of the carriers will be affected at any one time.

Three additional features can be added to improve performance:

- The data can be interleaved in time and frequency. That is, sequential data bits are transmitted separated in time and on different carriers. The precise rule is given by an algorithm laid down in the system design. At the receiver, the effect of the loss of data from a carrier may be compensated, as bursts of errors are broken by the disinterleaving process.
- Redundancy in the form of channel coding permitting error correction can be added to

correct errors arising from carriers which suffer from interference or multipath transmission. Systems which use such coding are termed Coded OFDM (COFDM) systems.

- The symbol period of data carried on each carrier can be increased to prevent multipath causing inter-symbol interference. The additional time, known as the guard interval, not only prevents inter-symbol interference but also allows the delayed signal power to be used constructively.

The result is a system with an enhanced performance in the presence of interference and multipath.

Key benefits occur when COFDM systems are used for wide-area broadcasting (in which the same service is broadcast from many transmitters or gap-fillers). With conventional systems, interference is prevented by operating the transmitters on different frequencies. With COFDM, however, the simultaneous reception of signals from several transmitters or gap-fillers appear at the receiver to be multipath propagation with delays. COFDM systems can be designed to allow relatively long delayed echoes, and so all the transmitters or gap-fillers in a network can be operated on the same frequency. This concept is known as a single frequency network (SFN). In planning SFN systems, the power in a delayed signal can be considered to be composed of a constructive and interfering component. The relative amounts of each component depend on the signal delay.

6.2.4.3 The proposals outlined in § 6.2.7 are examples of the application of both types of the above digital techniques.

6.2.5 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 6.2.2 and 6.3.1) and the remainder of this section assumes 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 existing coverage patterns. 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 multicarrier 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.

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 multicarrier systems. If delay equalizers 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 equalizer 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.

6.2.6 Planning Factor Considerations

Protection Ratios

For the digital/analogue coexistence environment, protection ratios are dependent on a number of factors which for both services may include modulation schemes, error correction used, synchronization, picture quality and interference protection. For digital interference to the analogue service, the digital interference resembles a noise-like signal and the co-channel protection ratios fall in the range of 35 to 45 dB. For the digital service, interference may result from either an existing analogue service or a newly established digital service. For this case, the co-channel protection ratios may range from 5 dB (digital interferer) to 20 dB (analogue interferer). Additionally, in the case of the analogue interferer the protection ratio may vary depending on the picture content of the analogue signal. Other protection ratios required for planning include those for adjacent and image channel.

Receiving System

Receiver noise figure, receiver C/N ratio, antenna gain, and feeder line loss are factors used to establish the required field strength for satisfactory receiver operation (Grade 3 picture quality for analogue service in 1 to 5% of the time and available picture for digital). An example of receiving system parameters currently being considered is given in Table 15.

TABLE 15

RECEIVING SYSTEM PARAMETERS

Receiver Noise Figure	10 dB
Receiver C/N Ratio	16 dB
Receiver Bandwidth	6 MHz
Frequency	615 MHz
Antenna Gain (dipole)	10 dB
Line Loss	4 dB
Minimum Field Strength	44.5 dB

In practice, the performance of any receiving system may be improved by simple methods such as using a low noise amplifier or an antenna with higher gain.

Polarization

The use of orthogonal, horizontal and/or vertical polarization of the radiated signal may be beneficial in reducing mutual interference, particularly in densely congested areas.

Service availability

The abrupt failure characteristics of digital services requires close attention to the coverage or service availability. A location and time availability of 90% is generally assumed to be necessary.

Portable indoor reception

Portable indoor reception is not possible throughout the full coverage area derived by planning with external antennas. Indoor reception has a loss penalty in the order of 20 to 30 dB due to lower antenna gain and height and to building penetration which results in a considerable reduction in the service area. One solution to this problem is to accept the coverage where it is possible.

6.2.7 Digital proposals

Consideration of the above planning factors leads to a range of implementation strategies which are described in 6.3.1.

6.3 Digital implementation strategies

6.3.1 Service implementation in a mixed digital/analogue environment

General

Implementation strategies are under consideration around the world. A common feature of these strategies is that for most countries Digital Terrestrial Television Broadcasting (DTTB) will be implemented in the existing frequency bands of VHF or UHF television. A number of schemes under consideration involve simultaneous transmission or simulcast operation or both. To clarify simultaneous transmission involves the transmission of both the existing TV service and the new digital television service with either independent or the same programming. Simulcast operations involve the transmission of both existing and digital television services with identical programming either at the same time or within a specified time period. The implementation strategies presented in this report must be considered in the same development state as digital terrestrial television and should not be construed as the final implementation strategy chosen.

With respect of a suitable implementation strategy for digital terrestrial television two phases have to be considered - the transition period when both analogue and digital TV are present, and the period of only digital TV transmission. For the transition period three options of transmissions seem possible:

Option 1 - Simulcasting: Under this model existing terrestrial broadcasters would be given the option of broadcasting their service in parallel on a digital channel during a transition or "simulcasting" phase. Following this phase the broadcasters would cease broadcasting using the analogue standard so enabling the additional spectrum released to be used either to increase the coverage of the digital service and possibly additional broadcast services or for other services.

Option 2 - New services: Under this model any additional digital capacity would be allocated to new services. Although this would immediately increase the number and range of terrestrial channels available it would also mean that a two-tier level of services would exist so requiring dual standard television receivers and thereby reducing the possibility of an absolute transfer to a digital-only system in the foreseeable future.

Option 3 - Simulcasting and new services combined: This option combines aspects of the previous two options. Under this the simulcast strategy of option 1 would be pursued but any extra capacity identified would be allocated to additional digital services.

Since full coverage of digital TV services may not always be achievable the question arises: what is the appropriate balance between the transmitted net data rate (for the encoded video, audio and data) and the degree of ruggedness with which this data rate is transmitted? These and other system parameters are addressed in other sections of this report.

The available net data rate can then be used for broadcasting one or several television programmes with a picture quality for the individual television programme depending on the allocated data rate for it. One extreme is the transmission of only one television programme in HDTV quality. The other extreme is multi-programme transmission of television where each television programme is of VHS quality or lower. Different service providers are expected to have different priorities concerning the balance between picture quality and number of channels.

It is clear that different service providers will also have different views on the required ruggedness of the transmitted television programmes. Some would prefer a coverage very close to 100% whereas others would accept a lower coverage. A certain transmission ruggedness will lead to different service coverage depending on the receiving conditions. Some programme providers are interested in providing a service that is receivable with portable or even mobile receivers with omnidirectional antennas. Others will assume directional antennas with a certain degree of directivity and a corresponding increase in coverage.

Which of the outlined options to be implemented is a decision yet to be made. However, it should be noted that with respect to the planning and coordination of channels for digital terrestrial television, the shape of the spectrum and the modulation scheme are of importance. Decisions on future digital television implementation strategies will rest on market political and economic considerations as well as spectrum availability and technical criteria.

Use of Spectrum Overlapping to Accommodate Wider Bandwidth DTTB Signals

During the introductory phase it is likely that in most countries DTTB services will need to co-exist with current PAL/SECAM or NTSC services. Further, in countries that currently use 6 or 7 MHz channel plans there may be interest in adopting DTTB services that occupy bandwidths in excess of nominal 6 or 7 MHz channel bandwidths.

In the transition period when PAL/SECAM/NTSC and DTTB services have to co-exist in the same frequency bands it may be possible to accommodate this apparently impossible requirement by allowing a small amount of overlap of the spectrum of the DTTB and PAL/SECAM/NTSC signals..

The primary factor that limits the permissible amount of overlap between adjacent DTTB and PAL/SECAM/NTSC services is the adjacent channel interference rejection performance of domestic PAL/SECAM/NTSC television receivers. Preliminary investigations (Doc TG 11-3/-[AUS-2]) show that by operating the DTTB service at power levels in the region of 12-30 dB below adjacent channel PAL-B services it is feasible to accommodate a small overlap in spectrum of the PAL-B and DTTB signals.

The possible DTTB bandwidths that can be accommodated given either a nominal 6 or 7 MHz vacant channel are shown in Table 16 with the corresponding maximum emission mask values. The ability to successfully accommodate a greater bandwidth for a DTTB service implies an increase in the total data capacity of the digital service.

TABLE 16

BANDWIDTH CAPABILITY AND CORRESPONDING MAXIMUM RELATIVE EMISSION LEVELS OF DTTB SERVICES

DTTB bandwidth for a nominal 6MHz channel (MHz)	DTTB bandwidth for a nominal 7MHz channel (MHz)	Overlap into upper adjacent PAL-B channel (MHz)	Maximum emission mask
6.0	7.0	0	-12 dB
6.1	7.1	0.1	-12 dB
6.2	7.2	0.2	-14 dB
6.3	7.3	0.3	-17 dB
6.4	7.4	0.4	-20 dB
6.5	7.5	0.5	-23 dB

6.3.2 Modulation and the Differing Planning Environment

The differences that have arisen throughout the world in the development of digital terrestrial television systems are largely in modulation systems. They are due to a variety of factors, in part the introduction time scales envisaged and the development approaches used, but also a more fundamental factor - the different frequency planning environments which exist in different parts of the world.

There is a wide mosaic of planning environments for the introduction of digital terrestrial television. The different factors include:

- sparse or dense concentrations of population;
- national broadcasting coverage of localised coverage;
- different levels of "taboo" channel availability;
- different levels of maturity in development of analogue television;
- frequency planning criteria of countries and their neighbours;
- requirements for mobile, portable or fixed reception;
- terrain variations.

If a single modulation system were to be used throughout the world, it would have to be based on the most difficult planning environment, because this is likely to need the most sophisticated modulation system. However, the costs and time scales of doing so may outweigh the benefits, for those with less demanding planning environments.

There would certainly be value in as many areas of the world using the same modulation system, as this would lower receiver prices for the consumer. However, there may be over-riding factors, which would have a more significant impact than the economies of scale on the prices of equipment and the services available. This may mean a plurality, but limited plurality, of modulation systems are needed throughout the world. The reasons are just as valid as those for differences in modulation system between terrestrial and satellite systems.

The ITU-R needs to take account of the varying requirements and develop Recommendations for a limited and closely related family of systems. This would allow the nations of the world who have not developed the systems, and who wish to start digital terrestrial services, to select the system most appropriate for them, based on their own circumstances.

6.3.3 Status around the world

The degree of utilization of terrestrial frequency bands for analogue TV is a factor in the implementation strategies possible in different parts of the world .

In some countries the large geographical separations possible between major cities can lead to much less congested use of terrestrial frequency bands in relevant areas. This is in contrast to most countries which, by their size, or by their population concentration in certain areas, have to make extensive use of these same frequency bands.

The following gives a brief review of the current thinking on implementation strategies in the potentially mixed digital/analogue environments in different ITU Regions of the world.

Region 1

Plans for the introduction of DTTB services are under discussion in a number of countries. In particular the United Kingdom government has proposed legislation for digital terrestrial television broadcasting and plans for the introduction of services scheduled to start at the end of 1997. In many countries in ITU Region 1 the allocated terrestrial frequency bands are heavily congested with analogue transmissions especially where the national allocations are fully taken up and a large number of low-power relay transmitters are used to provide almost complete national or regional coverage. In several countries the utilization of related taboo channels could be a possibility for the implementation of DTTB services. The maximum permitted transmit power of digital services is heavily constrained by the need to avoid interference into the existing services especially those supported by low-power relays.

The limitations thus imposed upon transmission powers constrain the coverage obtainable from a single transmitter for a desired quality level and hence the number of channels of DTTB that will be economically feasible. Furthermore, in a number of countries there is interest in developing services for different types of markets. Services being considered range from those aimed previously at roof-level antennas to others primarily engineered for portable or mobile reception. A particular concern then is to define the requirements for the practical introduction of DTTB services in parallel with the existing analogue services whilst maintaining the option for up-grading such services in the long term (after the older analogue services have been phased out on the assumption that this is possible to obtain a "digital-TV-only" scenario).

An additional factor influencing DTTB strategy is that in some countries in Europe the early introduction of digital services by satellite and cable has been proposed. The introduction of such services might assist or hinder the market opportunities for certain types of DTTB service depending upon the national situation. In either case it is recognized that it will be important to harmonize standards among the different transmission media (satellite, cable and terrestrial), to ensure that the future digital television market will develop satisfactorily. This work is being progressed within the DVB project and ETSI.

Networks for digital television can be planned either in the conventional way, i.e. integrated with the analogue networks or as a single frequency network (SFN). SFNs have the potential for extremely efficient spectrum utilization in that for national coverage only one RF channel is needed for the transmission of one HDTV programme or several standard television programmes. In order to be able to start a nationwide digital television service with a SFN of course one RF channel must be completely freed for digital television. This may be more or less difficult in different countries depending on how much replanning of existing analogue services if any that is needed in order to free one RF channel. The consequence is that digital television may be introduced with integrated networks in some countries and with SFN in others. There is also the possibility that some countries will use both integrated networks and SFNs.

The limiting factor for planning of digital television is the interference to and from analogue television. The interference criterion used today allows interference on analogue television during 1% of the time (5% in some countries). One interesting idea put forward is that if this criterion could be changed to become 5% of the time everywhere, it would in many cases permit increase of the power of the digital transmitter by about 6 dB which would greatly facilitate the planning of digital television.

One proposed idea which has led to considerable interest and discussion in Europe is that the different parts in the transmitted bit stream could be unequally protected and that the more protected parts could correspond to fewer picture quality levels, e.g. SDTV, whereas the less protected levels could correspond to e.g. HDTV. With scalable coding a subset of the HDTV bit stream would correspond to SDTV quality.

Such a scheme with scalability and unequal protection allows for graceful degradation and leads to larger coverage for the lower picture quality part (SDTV) than for a system without this feature. However, this larger coverage is obtained at the expense of a reduction in coverage for the high picture quality part (HDTV) and of some reduction in its highest picture quality which requires further thorough evaluation of this scheme.

Reconfigurability refers to dynamic, static or semi-static modification of the disposition of the channel in terms of services carried. It can be combined with scalability and unequal error protection but can also be used as an alternative to those methods. Also the modulation, including the FEC, can be modified to fit robustness or other requirements. The modulation can be reconfigured to provide different spectrum efficiencies (bit/sec/Hz), with a corresponding trade-off in ruggedness. The following are examples of possible service modes:

1) one HDTV programme (the singlecast service mode) at 24 Mb/s;
2) four SDTV programmes at 4 x 6 Mb/s per SDTV programme = 24 Mb/s;
3) two SDTV programmes for very rugged transmission using 2 x 6 Mb/s = 12 Mb/s;
4) one SDTV programme for extremely rugged transmission using 6 Mb/s;
5) four LDTV programmes for extremely rugged transmission using 4 x 1.5 Mb/s = 6 Mb/s.

Many configurations could be envisaged, like the first one where HDTV is given maximum resources and any SDTV versions of the same programme, with the required degree of ruggedness, could be carried by another RF channel. The final choice of a set of configurations has to be made with respect to both non-technical and technical factors, one of them being receiver complexity and another being the coverage/bit rate trade-off.

If the reconfigurability also applies to FEC and modulation in the same system as the one that would provide maximum HDTV quality and coverage in certain RF channels, it would also be able to provide SDTV or LDTV services with a related degree of ruggedness in others.

The long term spectrum requirements for terrestrial television broadcasting in Europe have been reviewed by the CEPT in the second phase of its Detailed Spectrum investigation, covering the frequency range 29.7 to 960 MHz [2]. Noting the very significant increase in spectrum efficiency that could be realised in a transition from analogue to all-digital operation, the Investigation nevertheless recognises that the phasing out of analogue services is likely to be a very long process, especially bearing in mind their current popularity and recent improvements to service quality offered by system enhancements like NICAM sound or PAL-Plus. While the move to 'all-digital' operation is seen as an inevitable long-term trend it is recognised that any forecast of timescale for this, or for the pattern of services that might emerge, is highly speculative. One long term possibility considered is that satellite broadcasting might emerge as the primary means of providing coverage over large areas. Terrestrial broadcasting might then be primarily aimed at providing regional and local services. Furthermore terrestrial broadcasting networks may be instrumental in providing the portable and mobile outlet of the information highway. However it is recognised that, in order to compete with satellite services mooted to start in 1996, initial DTTB services will most likely be aimed at achieving a rapid build up of coverage to the existing field of roof top antennas using whatever spectrum is available in the related (taboo) channels in the existing terrestrial networks.

While there have been some frequency planning studies related to national Single Frequency Networks most 'introductory phase' planning studies have adopted a 'conventional' planning approach - that is an approach which assumes that the digital transmitters will be co-sited with (or very close to) the existing analogue transmitter, and that the frequency planning techniques are similar to those used for analogue services (albeit that the 'criteria relating to minimum field strengths and protection ratio's would be different').

An advantage of the conventional 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 polarisation can be used. This should permit viewers to re-use their existing receiving antenna and feeder system. This will be the case for some countries, while for other countries more difficult planning environments exist. In these cases other planning approaches may have to be used.

However, it may be found desirable to introduce a limited number of channel, or even site changes at some of the low power analogue stations to assist in the efficient introduction of the digital services. The use of SFN's to support low power relay stations for the digital service can also make a significant contribution to providing optimum coverage. Additionally the use of 'notches' in the digital transmitter antenna pattern so as to minimise interference in particularly sensitive directions, can allow the digital transmitter power to be increased to provide an overall benefit in coverage terms. Thus by a detailed consideration of particular interference problems, and techniques to overcome them, the digital network can be tailored to offer considerably more coverage than an initial 'broad brush' study might suggest.

'Broad brush' frequency planning studies, using somewhat simplified methods and conservative planning assumptions, have been made to get a first idea of the coverage achievable with digital techniques in a significant sample of major European cities. In general the results are encouraging and further work at a European level, covering introductory scenarios with the associated frequency planning and co-ordination issues is the responsibilty of the CEPT Project Team FM PT24.

Region 2

The United States and Canada are in the process of choosing a digital advanced television (ATV) standard to be used by terrestrial broadcasters to provide an advanced television service. In the two countries, ATV will need to fit within the 6 MHz channels of the VHF and UHF bands now used for conventional television. Hence there will be the conventional television channels and additional ATV channels which may interfere with each other in the television bands. The ATV channels will be implemented on the basis of one channel for each existing conventional television channel. The basic objective of future ATV service is to provide a service area matching as closely as possible the existing NTSC service area which it will eventually replace. Spectrum allocation and accommodation studies have demonstrated the feasibility of this approach.

Active spectrum accommodation is underway to provide for the terrestrial emission of digital ATV signals in the existing VHF and UHF allocations. The principles under consideration as a basis for ATV allotment planning are:

- Pair ATV allotments with existing NTSC allotments: An ATV channel should be identified and associated with each of the existing VHF and UHF allotments. This provides the fundamental basis for an HDTV plan and serves as a baseline for implementation.
- Comparable coverage: A desirable objective of the allotment plan (when implemented to become actual assignments) is that it be capable of providing an existing NTSC station with an ATV service area comparable to the present interference limited service area of the station with which it is paired, or the present protected service area of the Grade-B contour.
- Use existing sites: To the extent possible the assignment/allotment plan should constructed predicated on the use of existing transmitter site locations for the transmission of the ATV signal.
- Separation limits: The allotment plan is currently being developed primarily on the basis of minimum separation distances between co-channel stations, with consideration of constraints due to adjacent channels.
- Allotment model: As there are literally thousands of possible combinations of pairings of ATV allotments with NTSC allotments optimization algorithms need to be used to determine a feasible distribution of allotments.
- Taboo spectrum: The VHF/UHF spectrum not being used as a consequence of taboo (interference related) considerations in the existing NTSC VHF/UHF allotment plans may be selectively employed as necessary to provide for ATV allotments.
- ATV/NTSC interference: An objective of the ATV plan is to minimize interference from an ATV station into an NTSC or another ATV service area. A desirable goal is that the level of predicted interference be perceived as no greater than that which is currently considered as acceptable interference between existing NTSC stations, or that producing an impairment of Grade 3.
- Freeze NTSC assignment parameters: In order to know the frequency and space dimensions available for the development of the HDTV plan certain variables in the existing NTSC allotment implementation should be frozen at some fixed time. This means freezing at some point in time the existing transmitter locations. Doing so will help establish the boundary conditions within which the ATV plan may be developed.
- Vacant allotments: The spectrum presently allotted but not used may be needed to obtain full ATV accommodation for all existing on-the-air stations. The affected number of vacant allotments should be kept as small as possible being used only when an operating assignment would be prevented.
- Receiving system: The characteristics of a typical ATV receiving system should be taken into account in determining the comparable interference limited or Grade-B service area. These characteristics would include antenna parameters such as front-to-back ratio and system parameters such as tuner/decoder rejection capability.

These general principles form the basis upon which to develop a domestic allotment plan for an ATV service. The objective is to provide an ATV service with coverage areas equivalent to existing NTSC stations or allotments. Selection is accomplished through the use of optimization algorithms which search for combinations of channels in the most congested areas to find available spectrum.

During the past 6 years of development of the ATV system, significant technological advances have occurred in digital encoding/decoding and transmission. This evolution in the technology has resulted in the development of a fully digital television system with significant capabilities beyond those originally envisioned for the advanced television system. Because of these enhanced capabilities, Canada is revisiting the issues regarding Planning Factors and Implementation Strategies for the introduction of digital television in their country.

The Digital Terrestrial Television Broadcasting and Frequency Allocation Planning committees of the Joint Technical Committee on Advanced Broadcasting (JTCAB) are continuing their work and technical studies on issues impacting on planning factors and implementation strategies for the introduction of digital television. Areas of work include:

- Allotment planning principles for development of a Canadian ATV allotment plan
- Study and evaluation of multicarrier transmission modes using COFDM modulation
- Defining planning parameters re-receiving conditions, service availability and propagation reliability
- Service carriage and definition of formats for HDTV, SDTV, Sound, Data and Ancillary Services
- Harmonization of delivery media for digital television services
- Implementation guidelines.

In addition to the technical committees, the Government has established a Task Force to ensure smooth introduction of digital television in Canada. The Task Force will examine implementation issues such as types of services (HDTV, SDTV, Pay TV, Data services, Sound services, etc.) to be carried in the ATV channel, simulcasting with existing NTSC, and the transition period. Social, economic and technical issues relating to other delivery mediums and services (cable, satellite, information highway, compatibility with computer and other services, etc.) will be examined. At present, the most optimistic start-up date for digital television is mid 1998.

Region 3

Several efforts are now underway in Japan to develop a digital television broadcasting system. An important and difficult consideration is the channel allocation capability in a new digital modulation scheme. In planning the introduction of digital terrestrial television broadcasting (DTTB) to vacant channels within the current NTSC services, the reduction of protection ratios between new DTTB and current NTSC services is a major concern. As the Orthogonal Frequency Division Multiplexing scheme (OFDM) has the capability of spectrum shaping (creation of carrier holes) due to its multicarrier scheme a well designed carrier hole scheme improves protection ratios and results in an extension of digital channel availabilities.

The spectrum is almost occupied by the analogue television broadcasting in Japan. However, the channel allocation capability for digital terrestrial television broadcasting is being analyzed with computer simulations. It offers prospects of implementing terrestrial broadcasting using the OFDM scheme, which coexists with conventional analogue systems.

To investigate the efficiency of Single Frequency Network (SFN) using the existing television transmitting site, the service area that has the required field strength and the allowable multiple echoes has also been calculated by computer simulations. Field experiments using two or three transmitters will be made available to confirm the feasibility of SFN.

To compensate for the disadvantages of the single frequency relay network (SFRN), a double frequency network (DFN) is also being studied. DFN is a radiowave relay method that uses two channels alternately. The DFN does not require a large isolation between the transmitter and the receiver antennas of relay stations, and a long guard interval, which SFRN does.

BST (Band Split Transmission) is one of the candidates for the transmission scheme of the DTTB system under the constraints of a usable frequency spectrum. Programmes are transmitted by one OFDM block or more, each of which forms the basic unit of BST.

One of the features of BST is that the number of OFDM blocks can be set up depending on the total bit rate of services. Hence, BST can flexibly accept a variety of services that require a transmission capacity ranging from relatively low (e.g. a standard TV) to considerably high (e.g. studio-quality HDTV). Moreover, each OFDM block may be located at an arbitrary frequency position when contiguous spectrum cannot be secured.
In Australia the Australian Broadcasting Authority has issue a first report “Digital Terrestrial Broadcasting in Australia” which includes some key issues regarding the introduction of DTTB.

It is expected DTTB will be able to be introduced in Bands III, IV and V. Rechannelling of existing services is not considered feasible because of the difficulties in retuning modern television receivers which means the choice will need to be compatible with 7 MHz analogue channel spacing.

The initial studies indicate the current six analogue services in each area could be replicated with digital channels within the bands identified.

Australia is intending field tests to evaluate DTTB modulation options in 1996.

6.3.4 User Requirements

An important aspect of system development leading to DTTB service implementation is the derivation of a clear set of user requirements.

For example, in Region 1, the user requirements are based upon a “market -led”, approach to the implementation of DTTB services. The main elements of these requirements as defined in 1995 may be summarised as follows;

- It is planned to start services no later than end 1997. The additional cost to the viewer at the time of introduction should be less that 450 ECUs. Within two years of introduction the retail price difference between a digital TV set and the corresponding analogue set should be less that 200 ECUs.
- The system should provide maximum commonality with the satellite and cable baseline systems.
- The transmission capacity should be considered as “data containers”, that may contain different kinds of services being transmitted simultaneously.
- The system should be designed for stationary reception as well as static portable reception.
- The service should be optimised using existing transmitter sites.
- The system should permit maximum use of the flexibility in spectrum planning provided by digital broadcasting.

- The system should be designed to allow operation of single frequency networks.
- The system should be designed so that single frequency relays (gap-fillers) can already be used in the introduction phase.
- The system should provide for local and national coverage under acceptable economical and frequency management conditions.
- The system should be designed for adequate ruggedness against interference and it must minimize its own interference into existing terrestrial analogue services.
- The system should be reconfigurable in such a way that the broadcaster may trade capacity against coverage. In the first phase, a non-hierarchical system would be acceptable. In the second phase, a two-layer hierarchical system is judged to be sufficient.

6.3.5 Service implementation in an all-digital environment

Alternative distribution systems

A number of distribution systems will be used for the delivery of digital TV services to the general public. These systems such as cable, MMDS, satellite, and optical fibre networks will have their own intrinsic limitations on transmission channel performance. Ultimately, from the standpoint of the viewer, it is important that the received quality be independent of the delivery mediums.

Harmonization and interoperability

Digital technology has the potential for facilitating interoperability among various image systems. Selection of an advanced television system that incorporates attributes needed for interoperability will harmonize interchange of still and moving images from diverse sources. Digital delivery of television signal to the consumer will take place over a number of delivery media, which may differ in capacity. It is desirable that the standards take account of interoperability at the transmission level. It is also desirable that a high level of interoperability be achieved at the receiver level to ensure the maximum flexibility for the processing of differing network and computer-based services, in addition to DTTB.

In the broadcasting environment, interoperability refers to the ease with which the DTTB data stream can be transferred among delivery media such as terrestrial broadcasting, cable, satellite, public network and prerecorded media. In the receiver environment, interoperability refers to the ease with which the receiver can process the differing data streams from television, computer-based graphics, multimedia and other non broadcast sources. Interoperability considers delivery over alternative media (cable, satellite, packet networks), transcoding (with film and format conversion to other video standards), integration with computers and digital technology, interactive systems, the use of headers/descriptors and scalability.

The future benefits of video and other image technologies will be greatly enhanced if universal interchange of all kinds of image and image sequences can be implemented and managed economically. The ultimate beneficiaries are the consumers who will have at their option image information of any kind in a form chosen by them, instantly available, at an affordable price.

Rapid advances in digital semiconductors, digital communication, and digital processing algorithms will make it possible to tailor the video technology to specific applications in terms of picture quality, price, format, and performance. Such a diversity in the video marketplace will be a positive development only if it is easy to move among different formats, applications, industries, and media. The key idea is to facilitate interworking among multiple formats so that market forces can guide the developments of products and services.

Digital representation of signals is the key element in achieving interoperability for images and video. The digital nature of the signal means that all the systems that process the signal have identical material to process. The ease of storing, transporting, and processing digital data is matched by the growing speed, power, and economy of semiconductors.

Once in digital form, signals can be filtered and processed in a predictable and reproducible way so that conversions among formats can be implemented using functions selected based on mathematical theories such as sampling, interpolation and prediction.

Certain specific attributes of image-related systems contribute to interoperability and are described below:

Layered systems and scalability: Interoperability in broadcasting is enhanced by the adoption of standards based on the OSI model, in which there is a clear functional separation between layers. The packet-based approach of MPEG-2 is an excellent example of such an arrangement for video, sound and data. In the case of terrestrial broadcasting, however, consideration must be given to such standards to the possible variations in receiving environment. This may lead to differing levels of priority for data packets, effectively subdividing the channel capacity if a layered approach is employed.

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).

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.

Provision for headers and descriptors within data: An important area of agreement among advocates of interoperability and harmonization of images and video is the desirability of headers and descriptors embedded within the stream of image data. The purpose of the headers and descriptors is to identify reliably and unambiguously the form of the data. The headers could include information on how the images or images sequences were originated, processed and compressed, thus making the data stream self-identifying and offering flexibility in implementations.

Syntax: Interoperability at the receiver is greatly enhanced by the adoption of common syntax among services. For video, this must take account of colorimetry, transfer characteristics, spatial and temporal sampling structures and sample coding. Flexibility is enhanced if the syntax adopted lends itself to scalability of the image data.

Conditional access: Copyright owners, programme suppliers and service providers require highly secure broadcasted/distribution networks to allow the protection of their programmes throughout the delivery chain up to the final authorized users, thus leading to the need for highly secure end-to-end access control with flexible interfacing to authorization and confirmation systems. Interoperability is enhanced by the adoption of a common system as described in lTU-R Recommendation "Conditional-access broadcasting systems" [Doc. 11/66 period, 1990-1994].

Industries with interests in high resolution images

While the traditional entertainment television industry has been built around a single dominant format (NTSC in 60 Hz countries, PAL and SECAM in 50 Hz countries), non-entertainment industries have generated a number of formats for still and moving images. In some cases, the non-entertainment applications have used conventional television formats, although there has not always been a good match of these capabilities to requirements.

Many now seem to believe that technology has reached a point where many new standards are about to be set, and the opportunity should not be missed to harmonize the various image and video standards. One advantage of interoperability is the ability to create a common technology for use in the large consumer television market as well as other non-entertainment image and video applications.

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