

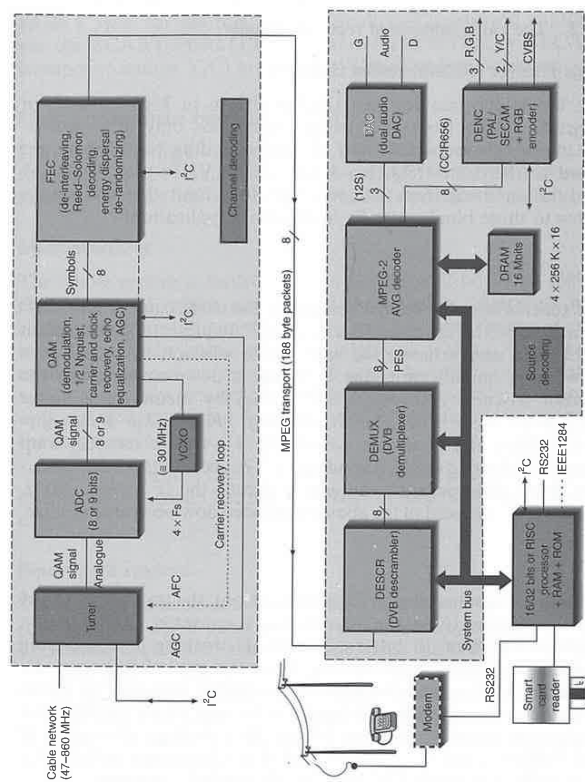


Fig. 8.3 Block diagram of a DVB cable receiver (1995/96 generation) (the dotted lines represent functions which are sometimes combined)

Composition of the integrated receiver decoder (IRD) 119

half-Nyquist filtering, echo equalization of the I and Q signals, and reformatting/demapping into an appropriate form for the FEC circuit (generally 8 bits parallel). It also plays a part in the clock and carrier recovery loops mentioned before, as well as generating the AGC for control of the IF and RF amplifiers at the front end.

FEC

The FEC block performs de-interleaving, Reed-Solomon decoding and energy dispersal de-randomizing; as is the case for satellite, the output data are the 188-byte transport packets in parallel form (8-bit data, clock and control signals).

The three blocks – ADC, QAM and FEC – now form a single integrated circuit (single chip cable channel decoder).

Other functions

The processor, conditional access, descrambling, demultiplexing, MPEG-2 audio/video decoding and all other 'secondary' functions (OSD, interfaces, modem, etc.) are, in principle, identical to those described above for the satellite for the same level of functionality.

Interactive receiver (with cable return channel)

In principle, the interactive receiver differs from the basic receiver only by its 'network interface' part (NIM), because of the need of an *upstream* for the return channel.

This needs, in addition to the existing circuitry of the *downstream* channel of the receiver described in the previous paragraph, specific functional blocks necessary for the emission of data to the network head-end station as well as software to control them.

In order that all receivers have the opportunity to send data to the head station, the frequency band allocated to the upstream (generally located between 5 and 65 MHz) is divided into channels on which time slots are dynamically allocated to receivers by the head station.

Management of the communication with the head station requires a sophisticated control software known as **MAC** (media access control), which requires significantly more important resources (processing power, FLASH and RAM memory size) than a receiver without interactivity.

Three main standards exist today for bidirectional communication over cable:

- The **DOCSIS** standard, originating from the USA, is originally mainly intended for cable modems allowing high speed access to the Internet, but can also be used in an interactive digital TV set-top box. The upstream uses QPSK or 16-QAM modulation in the low VHF band (5 to 42 MHz). Interactive data use the native Internet Protocol (IP) with variable length frames.

The downstream uses 6 MHz channels with 64- or 256-QAM and ITU-J83 Annex B FEC.

It can in principle be placed anywhere in the UHF/VHF band (70 to 860 MHz).

The original standard (DOCSIS 1.0) has been upgraded to DOCSIS 1.1 in order to ensure quality of service (QoS) required by voice over IP telephony.

- **Euro-DOCSIS** is a variant adapted to the European 8 MHz channels and frequency plan. The downstream complies with DVB-C specs (ETS 300 429/ITU-J83 Annex A FEC).

The upstream uses QPSK or 16-QAM modulation in the low VHF band extended to 65 MHz, which increases substantially the return channel capacity over DOCSIS 1.x.

The block diagram of a DOCSIS or Euro-DOCSIS cable modem can be seen in Fig. 8.4.

DOCSIS and Euro-DOCSIS can also be used in set-top boxes, but require in this case a second downstream channel (one for broadcast services and one for data).

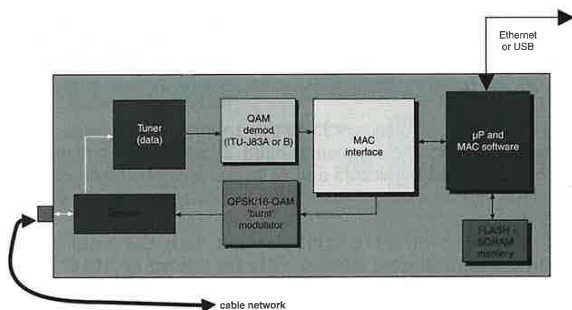


Fig. 8.4 Block diagram of a DOCSIS/Euro-DOCSIS cable modem

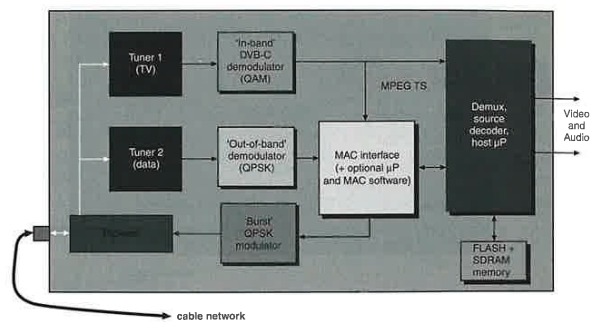


Fig. 8.5 Block diagram of an interactive DVB/DAVIC cable receiver

- The **DVB/DAVIC** standard originates from Europe (specification ETS 300 800) and is mainly intended to bring interactivity to digital cable TV receiver-decoders. It adds to the conventional DVB-C downstream channel used for TV (so-called 'in-band' channel) a second downstream at lower bit-rate (so-called 'out-of-band' (OOB)) for data.

The OOB channel is using QPSK modulation in 1 or 2 MHz wide channels which can carry bit-rates of 1.5 or 3.0 Mb/s, and is situated in the mid-VHF range (70 to 130 MHz), but it can also use UHF channels (300 to 862 MHz), depending on the constraints of the network.

The upstream channel uses differential QPSK (DQPSK) modulation in the European low VHF range (5 to 65 MHz) with channel widths of 200 kHz, 1 MHz or 2 MHz.

Interactive data use the Internet Protocol (IP) encapsulated in ATM cells.

The block diagram of a DVB/DAVIC cable receiver can be seen in Fig. 8.5.

8.2.3 The digital terrestrial TV receiver (DVB-T)

The block diagram (Fig. 8.6, overleaf) differs from its cable (without return channel) and satellite counterparts only by its front-end (RF/IF, demodulation and channel decoding parts). These parts have to be suited to the UHF and VHF frequency bands already used by analogue television, and to the COFDM modulation prescribed

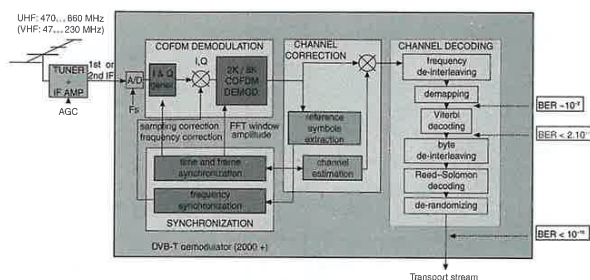


Fig. 8.6 Block diagram of a DVB-T front-end (source: Philips Semiconductors)

by the DVB-T standard. As it appears on Fig. 8.6, the DVB-T front-end is much more complex than those of cable or satellite due to the COFDM modulation used.

We will limit ourselves to the description of the blocks.

Tuner

The tuner and IF part are functionally similar to the cable parts, but they have to satisfy more stringent requirements regarding the phase noise of the tuner's PLL and also the selectivity (because of the coexistence with adjacent analogue channels, which can be more powerful than the desired channel). The tuner selects the desired channel in the VHF/UHF bands (47 to 230 and 470 to 860 MHz), transposes it into an intermediate frequency F_{IF} centred on 36.15 MHz and realizes the required selectivity by means of two SAW filters (only one is required by some very recent COFDM demodulators); the AGC amplified IF signal is either transposed at a frequency F_S in the order of 4.5 MHz by means of a 'supradyné' mixer-oscillator ($F_{osc} = F_{IF} + F_S \cong 40.65$ MHz), or directly applied to a recent COFDM demodulator (since 2001 products only) which accepts IF signals at 36.15 MHz at their input, thus avoiding an additional conversion stage.

ADC

The COFDM signal, transposed or not, is applied to an analogue-to-digital converter with a resolution of 9 or 10 bits, and digitized

with a sampling frequency in the order 20 MHz (undersampling when it is the IF signal at 36.15 MHz).

COFDM demodulation

This block is the main element of the demodulation process. I and Q signals are reconstituted from the digitized IF signal, and the OFDM signal is demodulated by means of a fast Fourier transform (FFT) on 2K or 8K points, depending on the mode (number of carriers) of the received signal.

Channel correction

This block performs the estimation of the transmission channel and its correction, and participates in the time and frequency synchronization.

Synchronization

From the information it gets from the channel correction block, this block performs time and frequency correction of the COFDM demodulation.

Channel decoding

This block performs frequency de-interleaving and demapping of the COFDM symbols. It is followed by a similar error correction to a satellite receiver (Viterbi decoding, Forney de-interleaving, RS decoding and energy dispersal removal).

The transport stream data at the output (188 bytes transport packets) are generally delivered in parallel format (8 bits data + control signals, one of which signals non-correctible packets).

Recent circuits generally deliver the transport stream in a serial format.

All the functional blocks contained in the large grey rectangle in Fig. 8.6 are now included in a single integrated circuit.

The processor, conditional access, descrambling, demultiplexing, MPEG-2 audio/video decoding and all 'secondary' functions (OSD, interfaces, modem, etc.) are, in principle, identical to those described above for the satellite for the same level of functionality.

9 Middleware and interoperability aspects

The size of investments required for setting up and running digital television transmissions as well as the bewildering increase in transmission rights of attractive 'contents' (sports, recent movies) are such that the vast majority of digital TV programmes are not free-to-air in countries where they have a significant penetration (USA, United Kingdom, France, Italy, Spain).

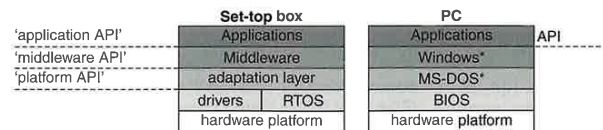
Even if the DVB consortium, like the GSM in the field of mobile telephony, played the role of a powerful unifier in Europe, it could not impose a common conditional access system nor a unique user interface specification.

We have seen in Chapter 5 that there were numerous conditional access systems using the DVB-CSA scrambling algorithm. These systems differ from one another due to the protocols used to manage and transmit access rights and keys, mainly by software embedded in the set-top box and the subscriber's smart card.

Besides the conditional access system, the most important part of the software for ensuring complete interoperability between different operators of digital TV 'bouquets' (beyond the simple passive reception of TV programmes compliant with the DVB standard) is what is generally referred to as '**middleware**' or sometimes '**interactivity engine**' or, more seldomly, just **API** (application programming interface).

In fact, the term 'middleware' is rather vague and does not mean anything more than that it is situated somewhere between the hardware and (application) software.

The term *API stricto sensu* points only at the interface between an application and the software layer immediately below it. It is essentially made of a predefined set of function calls of the middleware.



* Windows includes the DOS functionality since Windows 95.

Fig. 9.1 Comparison of the software layers of a set-top box and a PC

The middleware can be functionally compared to a high-level operating system with graphical user interface (GUI) such as Windows, which is very different from a low level real time operating system (RTOS, such as pSOS, vxWorks, Nucleus to cite just a few), on which the middleware is based.

One of the crucial functions of most middlewares is to make the application independent of the hardware platform on which it is run, assuming it has sufficient resources (processing power and device functionality).

In order that the same hardware platform can relatively easily support many middlewares, which not only differ from one another by their 'upper' API (to the applications), but also by their interface to the lower layers of the software (which we will name 'middleware API'), the set-top box manufacturers most often interface the middleware to their own API (which we will name 'platform API') through an adaptation layer.

Figure 9.1 illustrates the different software layers of a set-top box using a middleware in comparison to the layers of a personal computer (PC).

It is the middleware that defines the 'look and feel' of the user interface, the aspect and the possibilities of the Electronic Program Guide (EPG), the interactivity level of the applications which are offered to the user (pay-per-view programmes, games, specific services requiring a return channel to the service provider, special functions such as hard disk recording, etc.).

All middlewares generally offer two levels of interactivity:

- A local or 'off-line' interactivity of the 'carousel' type, which means that the user can only access data which are broadcast cyclically, in a manner similar to the Teletext information of analogue TV transmissions (this is particularly the case with the Electronic Program Guide).
- An 'on-line' interactivity, where the user is connected to a server by means of a 'return channel' (telephone modem or cable

Table 9.1 Some popular middlewares used for digital TV

Middleware	Type	Origin	Operators
Betanova	V	Betaresearch (D)	Premiere World, Deutsche Telekom
Liberate	P	Liberate (USA)	CWC, Telewest
MediaHighway	V/P	Canal+ Tech nologies (F)	Canal+, Canal Satellite, ITV Digital
Microsoft TV	P	Microsoft (USA)	TV Cabo, cable operators US
OpenTV	P	OpenTV (USA/F)	BSkyB, TPS, Via Digital, Quiero

return). The user can thus access remote data, either in a manner comparable to Videotex (e.g. Prestel or Minitel), the Internet, within the limits allowed by both the service provider and or from a terminal connected to a TV screen.

All middlewares allow downloading of data and software into the FLASH memory of the set-top box by means of the broadcast channel, which allows updating and upgrading (bug corrections, new functionalities) as well as downloading of applications.

Table 9.1 lists some proprietary middlewares in use today in the world of digital television.

Middlewares are classified as 'proprietary' (P) when they belong to a company which is not linked to a pay TV operator, or as 'vertical' (V) when a specific middleware has been developed by an operator or an affiliate company for its own use. They are noted P/V if they are both vertical and proposed to other service providers.

However, independently of the middleware core, pay TV service providers (whose terminals are most of the time rented or subsidized) can personalize their own version of the middleware or restrain it in order that it cannot execute non-approved applications (for instance coming from a competitor), even if they are compliant to the middleware's API.

Likewise, rented receivers are sometimes deliberately limited in order not to be able to receive transmissions (even free-to-air) from competitors who did not conclude an agreement with the service provider.

If this situation can be accepted in the context of pure pay TV (even if it is leading the customer to rent or buy one specific terminal for each service provider if he/she wants to access all the services), it is at least not desirable in view of 'digital TV for everybody'. In many countries, this should happen with digital terrestrial TV when it replaces analogue TV at the end of the decade and include a large proportion of free-to-air transmissions.

This is why alternative solutions are being developed to cover the needs of any service provider of pay or free-to-air TV, with different levels of functionality.

These solutions are said to be 'horizontal' as apposed to, 'vertical', integrated solutions of pay TV service providers (which range from the delivery of programmes to specific set-top boxes).

Before the availability of boxes with such a standard middleware, it is possible to find on the market DVB compliant receivers that do not use any middleware, commonly called 'zapper boxes'.

These receivers have a proprietary control software, built directly on the RTOS and drivers of the hardware platform; most of the time they do not have a modem, because its use would imply that it could run the application of a service provider requiring its middleware.

So this proprietary software does not incorporate real interactive (on-line) functionalities, but most of the time it can be updated by downloading from satellite or from a PC connected to its serial port.

This software ensures the basic functions required for reception of TV programmes, often with a 'mini-EPG' limited to the information on the current programme and the next one 'now-next' function, when this information is transmitted according to the DVB-SI standard.

In addition to the free-to-air (FTA) transmissions some receivers are able to receive encrypted pay TV transmissions either by means of an embedded conditional access (Viaccess is one of the most common) or by means of an appropriate DVB-CI module. In both cases, of course, the appropriate smart card containing valid rights for the programmes to be received is required.

We will first have a brief look at three of the most common proprietary middlewares in use in Europe, then we will examine two open and standardized solutions.

9.1 Main proprietary middlewares used in Europe

9.1.1 MediaHighway: the middleware of the Canal+ 'galaxy'

MediaHighway was developed in 1993 by the R&D department of Canal+ (which subsequently became Canal+ Technologies) initially for the launch of the first French digital TV service in April 1996: Canal Satellite Numérique (CSN).

MediaHighway is used by all the national variations of Canal Satellite which were launched shortly afterwards (Spain, Italy, Poland, etc.). More recently it was proposed to service providers not belonging to the Canal+ group such as OnDigital (UK) in 1998, which later became ITV Digital, and some other satellite and cable operators in Europe, Asia and the USA.

Canal+ Technologies requires a tough certification process, which includes an important suite of 'unitary tests' before allowing the marketing of products with the MediaHighway label.

Many versions of MediaHighway exist, which correspond to a **DLI** (device layer interface) number. The DLI is in fact the interface of the middleware to the lower layers of the software. The hardware platform together with its operating system and its drivers must comply with the DLI specification, generally via an adaptation layer. The DLI defines the functionalities supported by the middleware and ensures an independence (or abstraction) from the hardware platform and RTOS used.

MediaHighway supports near video-on-demand (NVoD) applications, impulse pay-per-view (IPPV) and allows downloading of data or applications. MediaHighway is most of the time used with MediaGuard, the 'in-house' conditional access system, but can in principle be used with any embedded CA system; it also supports the DVB-CI interface standard for detachable conditional access modules.

DLI versions currently in use:

- **3.xx versions:** used by most satellite service providers of the Canal+ group, because of their early starting date. They have been enhanced by satellite downloads.
- **4.xx versions:** OnDigital (now ITV Digital) is the first commercial service with the 4.1 version which in addition supports the MHEG5 API which is mandatory by law in UK for digital terrestrial services. Other variants of the 4.xx DLI now include an Internet browser and management of a hard disk drive for recording TV programmes and other data.

Recent versions of MediaHighway are based on 'MediaHighway virtual machine' which can execute applications developed under many different standard languages (Java, etc., HTML, MHEG-5, etc.) by loading the appropriate interpreter.

This feature has allowed MediaHighway to be one of the first proprietary middlewares to be able to support applications compliant with the new DVB-MHP open standard, which is based on the Java language of Sun Microsystems.

(Website: www.canalplus-technologies.com)

9.1.2 OpenTV: the most widely used middleware!

OpenTV is a middleware proposed by a company (now independent) bearing the same name. It was originally a joint venture between Thomson MultiMedia and Sun Microsystems (Sun Interactive Alliance formed in 1994).

The first digital TV service provider to use OpenTV was the French satellite bouquet TPS in 1996. OpenTV is now used by more than 30 digital TV service providers worldwide and is installed in more than 10 million set-top boxes produced by approximately 30 suppliers. Consequently, OpenTV has to support many different embedded conditional access systems as well as the DVB-CI interface standard for detachable conditional access modules.

OpenTV supports near video-on-demand (NVoD) applications, impulse pay-per-view (IPPV) and allows downloading of data or applications.

Many OpenTV versions exist. Their numbers reflect their date of introduction:

- **OpenTV 1.0.x** is the version still used by the 'pioneers', mainly French cable and satellite service providers. It has no longer been proposed to new customers since early 2000. It has nevertheless been enhanced by many downloads over time.
- **OpenTV 1.2** is a specific derivative for the British satellite service provider BSkyB.
- **OpenTV EN** is the second generation of OpenTV.
- **OpenTV EN2** is the third generation, inheriting from both OpenTV EN and OpenTV 1.2.

OpenTV EN2 is based on two libraries:

- Basic functions are included in the 'core' library, which is the heart of the middleware and is compatible with many real-time operating systems (RTOS).
- Optional functions are located in the 'extensions' library, which allows service providers to personalize the middleware and extend its functionality by downloading.

OpenTV EN2 executes OpenTV applications written in 'O code' (conceptually comparable to the 'byte code' of Java) by means of a virtual machine (interpreter).

This is why the architecture allows a relatively easy porting of MHP by means of adjunction of a second virtual machine (Java),

which will enable the middleware to be compliant with both OpenTV and MHP applications.

OpenTV functionalities extend now to the management of a hard disk and Internet browsing, notably thanks to the acquisition in 2000 of the company Spyglass.

(Website: www.opentv.com)

9.1.3 Betanova: a very 'vertical' middleware

Developed by Betaresearch, the technology branch of the big German media group Kirch, Betanova, in contrast to most other middlewares, does not pretend to be independent of the hardware platform or the RTOS upon which it runs. It is intimately linked to the 'd-box', the set-top box developed for the satellite pay TV bouquet Premiere World. A cable variant of this platform has been adopted by Deutsche Telekom for its cable networks.

Two generations of Betanova exist:

- **Betanova 1.xx:** this is the 'historical' version, which runs on more than one million d-boxes. It supports near video-on-demand (NVOD) applications, impulse pay-per-view (IPPV), and allows downloading of additional applications to the Electronic Program Guide (EPG). On the satellite d-box, Betanova also supports the DiSEqC antenna control system.
- **Betanova 2.xx:** this new version supports all the functionalities of the 1.xx version. It is written in Java, which simplifies adaption and accelerates the writing of new applications. In addition, this new architecture is more flexible and foresees the support of the API of DVB-MHP (also based on Java), but it is still dedicated to the d-box. This new version of Betanova is also intended to support access to the Internet as well as home banking and home shopping applications.

Mainly due to the strict interdependence between Betanova and the d-box, both have found themselves limited to the German market, on which their monopoly and lack of openness have been bitterly criticized by their competitors.

However, things in favour of MHP and open platforms on the German market, among others, are changing due to the end of the *de facto* monopoly of Deutsche Telekom on the cable networks.

(Website: www.betaresearch.de)

9.2 The open European 'middlewares'

9.2.1 MHEG-5: the first standardized middleware for digital TV

The MHEG (Multimedia and Hypermedia Expert Group) is a descriptive language of 'multimedia presentations', comparable to HTML (HyperText Markup Language) for hypertext pages. It is based on an object-oriented multimedia exchange format independent of the hardware and software (OS) platform on which it is executed. It is a declarative, as opposed to a procedural, language (such as Java).

The MHEG-5 version, standardized under the ISO/IEC 13522-5 reference, is a subset of MHEG-1 particularly dedicated to digital TV receivers (set-top box or integrated digital TV receiver, iDTV) with limited processing power and memory size. It is practically the only MHEG version in real volume usage.

In the MHEG vocabulary, a multimedia presentation is an MHEG application or an MHEG scene. An MHEG application is a grouping of MHEG scenes.

An MHEG scene is made up of three main groups of objects or 'ingredients':

- the 'interactive objects' such as buttons, cursor or other similar objects;
- the 'link objects' (links) which define processes triggered by user actions on 'interactive objects';
- the 'presentation objects' (objects) controlled by the above-mentioned 'link objects'.

A certain number of other objects exist, which regroup many presentation objects. When an MHEG application is started, it generally expects an action from the user. If the user presses a button, the application generates the event 'button selected'. A link can be defined on this (interactive) object, for example in order to start a video sequence (presentation object). Objects themselves can in turn generate events which can trigger links.

Relatively complex applications (video games, catalogue sales, etc.) can be realized in this way.

An application is normally made up of scenes and some common objects and links. A scene is only made up of objects and links.

Other versions of MHEG exist:

- MHEG-1 to 4: (for the record) the 'ancestors' of MHEG-5, little used.

- MHEG-5: the most important, used by the digital TV services in the UK.
- MHEG-6: an extension of MHEG-5 by a virtual machine allowing the inclusion of applications based on a procedural language.
- MHEG-7: defines test and conformance procedures of MHEG-5 applications.
- MHEG-8: a project aimed at combining XML and MHEG.

The small amount of resources required to run MHEG-5 makes it a good candidate for the basic digital terrestrial receivers in countries starting transmissions.

9.2.2 DVB-MHP: the 'universal' middleware for digital TV of the future?

From 1997 onwards, since the standardization of coding of digital television signals was achieved, the DVB consortium decided to tackle the problem of software and hardware interoperability between digital TV services compliant with the DVB standard for broadcasting their signals.

In addition to the initial assumption of interoperability, the main objectives of the development of **MHP** (Multimedia Home Platform) have been evolutivity (with the possibility of extending the functionality by downloading), backwards compatibility, modularity and stability.

The Internet has also been taken into account, as well as the integration to a local home network made up of various terminals (PC, telephones, domestic appliances, etc.).

One of the prerequisites was that this new standard would be based on 'open' standards in order to guarantee a non-discriminatory access to anybody desiring to use it.

Another concern was to use an efficient format in terms of bandwidth requirements but still ensuring a complete separation between data and applications.

A total neutrality versus the different optional conditional access systems, embedded or based on the DVB-CI interface, was another prerequisite.

MHP defines a generic software interface (API) between the interactive applications coming from different service providers and the terminals on which they should be run, independently of their hardware and software implementation. The performance level of these terminals can vary greatly.

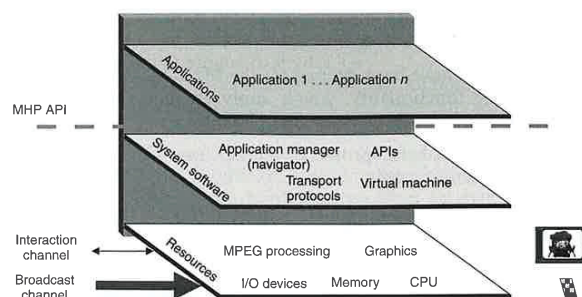


Fig. 9.2 The three levels of the MHP architecture (© DVB 2000)

MHP can support various applications (in increasing order of interactivity):

- Electronic Program Guide (EPG);
- information services comparable to a 'super teletext';
- applications linked to the current TV programme (betting on matches, games, etc.);
- electronic commerce (e-commerce) with secure banking transactions.

The MHP architecture has three levels: resources, system software and applications (Fig. 9.2).

- Resources include all the essential parts of the set-top box: MPEG decoding, input/output devices, host processor, graphics subsystem, etc.
- System software enables presentation of an abstract view of the platform to the applications. It includes a navigator (or *application manager*) which takes control of the platform and applications which are run on it. The software kernel of MHP is called DVB-J (DVB-Java), based on the *Java Virtual Machine* defined by Sun Microsystems.
- MHP applications access the platform through the MHP API. A number of software interfaces required to make use of specific resources of the hardware platform are implemented by means of extensions. The task of any practical implementation of MHP

is to ensure correspondence between the API and the hardware resources of the platform.

Three 'profiles' have been defined for the MHP platform, by order of increasing functionality, which imply an increasing need for processing power and hardware complexity:

- enhanced broadcast profile – makes use of unidirectional broadcast services only;
- interactive broadcast profile – adds support for interactive services requiring only a low speed return channel (V22 modem for instance) via the telephone network;
- Internet access profile – adds the supplementary functionalities brought by Internet access within the limits of TV screen display. It requires a higher speed connection (e.g. V90 modem).

Other profiles or elements of profiles will be added in the future, for instance for the support of hard disk recording.

Figure 9.3 illustrates the functionalities and the ascending compatibility between the three different profiles of MHP.

MHP **version 1.0** includes the first two profiles, and is now standardized by ETSI under the reference TS 101 812.

MHP **version 1.1** is currently being finalized. It will cover the 'Internet access' profile and will thus add the HTML functionality. It will be specified in the ETSI document TS 102 812.

At the time of writing, DVB-MHP seems to be the only open standard which can deliver at the same time a unified user interface

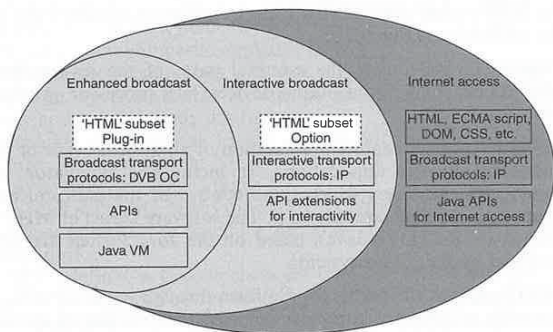


Fig. 9.3 Ascending functionality of the three profiles of MHP (© DVB 2000)

with different levels of functionality and a satisfactory evolutivity. These features can be seen as prerequisite conditions to a full switch-over from analogue to digital TV.

This is why MHP has obtained the support of many actors in the world of European television and beyond, especially in view of the generalization of digital terrestrial TV.

However, MHP will probably not be mandatory in the terminals, because of a significant cost increase which is criticized by its detractors, due to the hardware resources (higher processing power and memory size) required by the Java virtual machine.

(Website: www.dvb-mhp.org)

10 Evolution: state of the art and perspectives

Digital consumer applications (camcorders, MP3 players, DVD players/recorders and many other applications) which started in the mid-1990s by the first satellite TV transmissions in the USA and France, are now part of the consumer electronics landscape at the beginning of the third millennium.

Digital cable television, and even digital terrestrial television in some countries, has now taken a significant share of the market. However, many evolutions in receiver functionalities as well as services proposed to the users are to be expected in the near future.

Given that we now have more than five years of experience since the beginning of digital television transmissions, it is probably less risky than at the time of the first edition of this book to make some anticipation of the evolution of services and receivers. This is what we will try to do in the following paragraphs, limiting our horizon to the next three to four years.

10.1 Digital terrestrial television

The European digital television system (standardized under the reference ETS 300 744) makes use of the **OFDM** modulation (also named **COFDM**) already used for the DAB digital radio system, in two variants (2K and 8K) which have been described in Chapter 7.

The United Kingdom was the first European country to start (end 1998) a digital terrestrial television service using the DVB-T standard (mainly based on pay TV transmissions of OnDigital, recently renamed ITV Digital).

Due to this early start, DVB-T demodulators were not available for 8K COFDM at that time, so the 2K mode of the DVB-T system was chosen; it is likely that the UK will be the only European country to use this DVB-T version (at least for services destined for fixed reception).

Sweden (service provider Senda) and Spain (service provider Quiero) have followed suit respectively at the end of 1999 and mid-2000, and use the 8K mode of DVB-T. This enabled Spain to develop a single frequency network (SFN) on the whole territory, using the four highest channels of the UHF band (66 to 69) which were hitherto unused by analogue television.

At the time of writing, the commercial success of these first transmissions is mixed, probably due to the fact that these almost entirely pay TV terrestrial bouquets can hardly compete with pre-existing satellite and cable offers from the point of view of the number of programmes (the diffusion capacity of digital terrestrial TV is in the order of ten times less than satellite or cable as long as analogue television transmissions continue).

This is why digital terrestrial television needs an important share of free-to-air transmissions to be successful, in order to reach the two-thirds to three-quarters of the population not yet prepared to pay a subscription to receive more television programmes.

France plans to commercially launch its digital terrestrial TV network in 2003 with six multiplexes covering around 50% of the population. This should enable the transmission of 30 to 36 programmes (five to six TV programmes per multiplex) out of which approximately half should be 'free-to-air'. However, the coverage will be limited to 85% of the population as long as analogue television stays in service.

It is hoped that this relatively high number of free-to-air programmes, as well as the conversion of Canal+ analogue subscribers, will enable a relatively quick ramp-up.

Some countries like Italy aim at stopping analogue TV transmissions as early as 2006. However, the complete replacement of today's analogue television transmissions promises to be a long process (in the order of 10 years), unless there is a real political willingness to accelerate the movement, for example by delivering free of charge or at very low cost a basic 'digital adapter' or 'converter' to all homes that pay their TV tax.

The DVB-T system has already been selected by a number of non-European countries (such as Australia and Singapore), but for countries which have not yet made a choice, it is in competition with a new Japanese system (**ISDB-T**) which is also based on COFDM modulation with additions aiming at making it even more robust

(especially for mobile reception), however at the cost of an important increase in complexity and thus price (see Appendix D).

Digital terrestrial TV transmissions in the USA started in 1998, in high definition format based on the ATSC system (see Appendix D), with the initial objective of stopping analogue NTSC transmissions in 2008. However, its development has been very slow until now, mainly due to questionable technical and political choices (disappointing performances of the 8-VSB modulation chosen, high cost of high definition capable displays, absence of defined associated services), which makes the switch-off of analogue transmissions in 2008 unlikely.

Generalized integration of digital television decoders in TV receivers will only happen when technical choices (standard middleware, a cheap and secure way of adding conditional access) as well as political choices (simulcasting, network deployment, subsidizing of receivers) are very clear. As a first step, these receivers will have to be a hybrid (analogue and digital), even if the '100% simulcast' approach is followed, since the geographical coverage will be incomplete for a few years.

10.2 Evolution of the set-top box

10.2.1 Integration at (quasi) constant functionality

The block diagram of Fig. 8.2 corresponds to the functional partitioning of the first generation (1996) of a European receiver/decoder (IRD). At that time, each block corresponded to a major integrated circuit. Most of these circuits were realized in CMOS technology with a 0.5μ feature. Integration has progressed rapidly, and for the first time (1997–1999), one could observe a grouping of functions in circuits with a 0.35μ feature, at (almost) constant functionality, following the scheme below:

- Integration of external interfaces (RS232, IEEE1284, etc.) and demultiplexer/descrambler with the main processor (often a RISC), with a maximum clock frequency in the order of 50 MHz.
- Integration of the video encoder with the MPEG audio/video decoder, to which more sophisticated graphics capabilities have been added (8 bits per pixel or more, sometimes a 2D accelerator).
- Reduction of the required (S)DRAM chips from four times $256K \times 16$ to a one $1M \times 16$ chip common to the MPEG decoder and to the graphics processor.

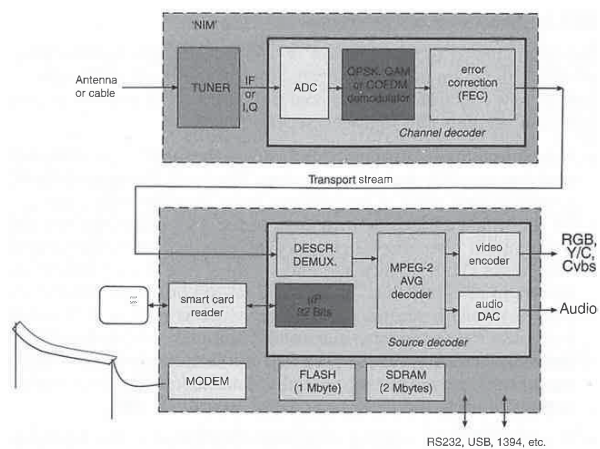


Fig. 10.1 Block diagram of a typical set-top box for pay TV (from 2000 onwards)

- Integration in a single 'channel decoder' chip (first for satellite then for cable and later for terrestrial) of demodulation, error correction and input ADCs.
- For the satellite tuner part, progressive adoption of the so-called 'zero IF' or ZIF technology (also known as direct conversion) with a view to finally integrating it on the main board.

Functional integration continues inexorably, thanks to the availability of ever smaller geometries (0.25 then 0.18μ), which made possible, at the dawn of 2000, the integration of the processor (a 32-bits RISC with a clock frequency up to 100 MHz or more) and its peripherals with the source decoder, video encoder and sometimes the audio DAC.

A 32-bit unified memory architecture (UMA) for the processor, MPEG audio/video decoding and graphics functions reduce for most common applications the memory requirements to a unique SDRAM chip of 8 Mbytes ($2M \times 32$).

Figure 10.1 illustrates the block diagram of such a receiver, commercially available in 2002.

10.2.2 Functional enhancements

Simultaneous to the integration of almost constant functionality, of which the main goal is cost reduction of the basic IRD, adjunction of new functions which increase its attractiveness is becoming common on mid- and high-end set-top boxes.

- A function missing from set-top boxes is the facility to record one programme while watching another. This is all the more difficult to understand since it has always been possible with analogue VCRs, which all include their own tuner. This deficiency and the concomitant phenomenal capacity increase and price decrease of hard disks (HDD) since the end of the 1990s have allowed set makers to integrate them in high-end set-top boxes. In order to be able to record a second programme which is part of a different multiplex (transponder), a second front-end or NIM (tuner+channel decoder) is required, as well as a demultiplexer in order to record only the interesting part of the multiplex. This functionality is generally referred to as 'watch and record' (W&R).
- Another, more innovating, functionality allowed by the hard disk (which has the advantage over a tape recorder of random access capability and simultaneous read and write) is what is generally called 'time shift recording' (TSR). This allows a programme to be stopped, for example to answer an incoming phone call, and resumed at the place where it was interrupted. From the moment the user stops watching, the hard disk records the programme, the user can then resume watching from where he left while the hard disk continues recording in real time. The user watches from this moment with a time shift which is equal to the duration of the interruption. Figure 10.2 shows the block diagram of a satellite IRD

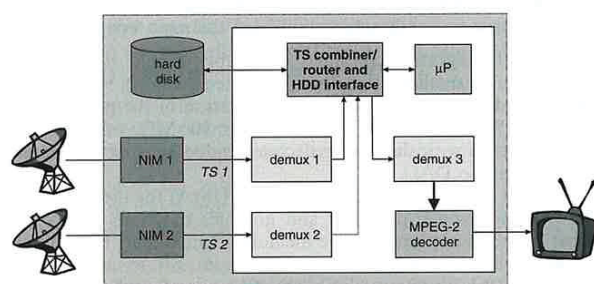


Fig. 10.2 Principle block diagram of a satellite receiver with HDD and dual front-end

with a dual front-end which allows a programme to be watched, if required in 'time shift' mode, and another one to be recorded not necessarily belonging to the same multiplex. This requires of course two front-ends, but also three demultiplexers as well as a device (which we have named TS combiner) able to reconstruct a partial transport stream, since the information will most of the time have to be recorded in the form of a scrambled transport stream for copyright reasons.

- Another important advantage of the hard disk compared to the VCR is that the original quality of picture and sound is fully maintained, since the recorded bitstream does not undergo any bit-rate reduction before being written to the disk.
- Interfaces of the set-top box to the external world evolve, even if the analogue PAL/SECAM output to an external recorder has not yet been replaced as expected by a high speed IEEE1394 serial bus (see Appendix E), mainly due the very relative 'success' of the digital D-VHS recorder. This situation is due both to the new possibilities brought by the hard disks discussed above and the opposition of the owners of rights (mainly of motion pictures) to allow digital recording of their works on a removable support. For the pure 'data' interfaces, the current standards (RS232 serial and/or IEEE1284 parallel) tend to be progressively replaced by the USB interface, following with some delay the trend in the PC world.
- For decoders connected to a cable network, a high speed return channel using the cable network instead of the telephone line becomes progressively standard, but it generally requires the upgrade of the network to make it bidirectional. The main standards used are DOCSIS in the USA and DVB-DAVIC or EURODOCSIS in Europe. The set-top box can thus become an attractive way of accessing the 'data superhighways', enabling real interactivity, high speed access to the Internet on the TV set as well as the possibility of using the set-top box as a high speed modem by a PC connected, for example, by a USB or Ethernet link.

10.3 New architectures

There are two main conflicting 'philosophies' which are explained below about the long-term future of domestic digital TV receiving systems. It is likely that both solutions will have a share of the

market, but will leave an important place to the 'classical' set-top box for many years.

10.3.1 The integrated digital television receiver (iDTV)

Currently, the major players in analogue television, as well as TV set makers, defend the integration of the digital decoder (essentially terrestrial) into the TV set—a first step to a hybrid receiver (analogue and digital, see block diagram Fig. 10.3)—as long as analogue terrestrial broadcasts continue. They see this as a way to phase in the eventual move to a digital-only receiver.

This type of receiver is often referred to as the integrated digital television receiver (iDTV).

This relatively high priced solution had until recently encountered little success (mainly in the United Kingdom) due to its lack of competitiveness because of the 100% subsidization of set-top boxes by pay TV operators (OnDigital for terrestrial TV as well as BSkyB for satellite) and also maybe because of a certain fear of obsolescence of the integrated decoder during the useful lifetime of a TV set.

Actually, the features of a digital TV decoder evolve almost at the same speed as those of a PC (cycles in the order of 12 to 18 months) in comparison to a classical TV set, which appears not to evolve functionally any more and has an average lifetime in the order of ten years.

This reason, added to the new possibilities of interconnection through a home network (wired or wireless), tends to give weight to

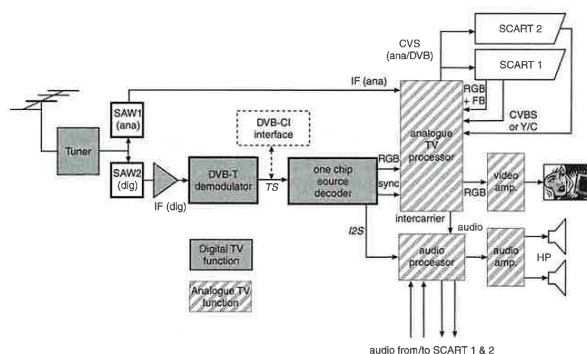


Fig. 10.3 Example of a block diagram of a hybrid TV set

the 'home gateway' concept as 'the' access point to all external networks which pass the user premises.

10.3.2 The 'home gateway' or the set-top box at the centre of the home multimedia network

In order to allow equal access to all external information sources to all the terminals of a home with full interactivity, it is tempting to propose a 'client/server' type of architecture as illustrated by Fig. 10.4.

In this concept, a set-top box (server) with an important processing power and equipped with high capacity hard disk(s) ensures interface with all external networks (cable, satellite, terrestrial, ADSL and/or plain old telephone). It is directly connected to the main TV set or screen in the home and to simpler terminals (*thin clients*) by means of an internal high speed network either wired (e.g. an extension of IEEE1394 as proposed by the HAVI committee) or wireless in the 2.4 or 5 GHz frequency band (e.g. one of the versions of IEEE802.11 already used in the PC world).

One of the advantages of this solution is its evolutivity which can be ensured mainly by updating or upgrading the server, since the clients are 'dumb' terminals merely able to send requests to the server and to display the requested audiovisual services, whatever their origin (external networks or local disks) and their nature (audio/video, graphics, Internet pages).

However, this solution faces many obstacles, even if we assume the problems linked to the standardization of the communication

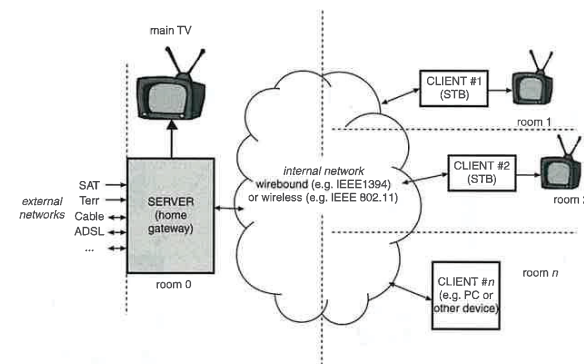


Fig. 10.4 Architecture example of a home multimedia network

network are resolved: for example, compatibility with the many middlewares and conditional access systems of pay TV operators, wiring constraints for the wired solution and bandwidth constraints for the wireless solution, as well as complete unavailability of all terminals in the case of a problem at the server.

In a less ambitious approach, perhaps more realistic in the short term, a **Bluetooth** radio link (also at 2.4 GHz but at a lower bit-rate than the various IEEE802.x standards – max. 1 Mb/s) is also proposed by some companies to link the set-top box to some peripherals. Originally developed by Ericsson to connect wirelessly and at low cost a mobile phone to its peripherals, Bluetooth is now supported by a consortium of participants belonging to all branches of the electronics industry.

A possible application is the connection to the set-top box of light terminals equipped with an LCD screen; this allows, for example, for the operation of the interactive functionalities of the set-top box without disturbing the display of the current programme on the screen of the TV set.

The same link allows a telephone (fixed or mobile) equipped with a Bluetooth interface to connect interactively and without additional wires, the set-top box to the external world.

Figure 10.5 illustrates this type of application.

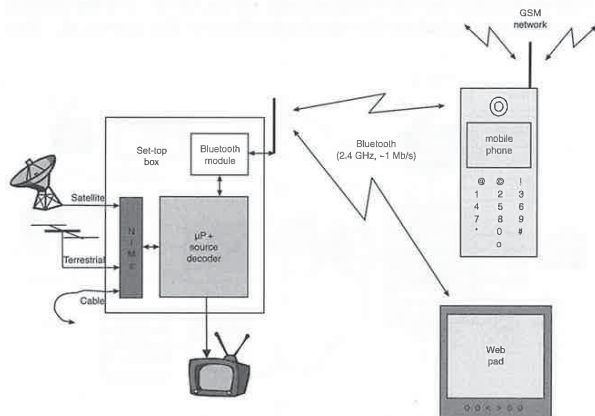


Fig. 10.5 Application example of Bluetooth around a set-top box

In conclusion, the objective of this book has been achieved if at this point readers have acquired a (hopefully clear enough) global view of the new techniques used in digital television systems, which perhaps will give them the desire to investigate this subject further.

Appendix A: error detection and correction in digital transmissions

From the very first digital data transmissions, the need to detect and, as far as possible, to correct errors introduced by the transmission link has been recognized. Various more or less complex solutions are used, depending on the characteristics of the link, but they invariably consist of adding a calculated redundancy to the original message.

A1.1 An error detecting code: the parity bit

The simplest means of detecting errors is the parity bit; it is still employed in short distance serial transmissions (for instance RS232), where it usually applies to words of 6–8 bits. On the transmission side, all the bits of the word to be transmitted are summed, and a supplementary ‘parity’ bit is added at the end of the word; this supplementary bit is equal to 0 if the sum of the bits is even, and 1 if the sum is odd (even parity), or the reverse (odd parity). On the reception side, the receiver does the same calculation and compares its result with the transmitted parity bit:

- if there is no error (or an even number of errors), the bits are identical;
- if there is one error (or an odd number of errors), the bits are different.

This system allows the detection, without being able to identify or correct it, of a 1-bit error in a word. The parity is only useful

when the probability of more than one error per word is very low, as even numbers of errors are undetected, and when the communication link is bidirectional: in this case, the receiver which detects an error can ask the transmitter to retransmit the corrupted message. This principle of error correction is called *feedback error correction*.

A1.2 Block error correction codes

The block error correction codes apply to finite length words composed of k symbols (bits for instance) to which they add a calculated redundancy which increases the word's length to n symbols (bits in our example), so that $n > k$. The coding algorithm therefore adds $n - k$ bits (called control or parity bits) to the end of the original word, and the redundancy comes from the fact that only 2^k combinations out of the 2^n possible from the resulting code are used. The ratio k/n of the code (obviously < 1) is called the *yield* of the code.

In order to be able to distinguish between them in the case of an error, all elements of the code should be as ‘distant’ as possible from each other. The *Hamming distance* between two elements of a code is defined as the number of different bits situated in the same position. The *minimum* Hamming distance d of the code is defined as the number of different bits between any two elements of the code; one can demonstrate that the number, t , of correctable errors in a word is equal to $t = (d - 1)/2$.

The whole mechanism of the coding consists, therefore, of finding, for given n and d values (which define the correction capacity of the code), the coding algorithm giving the greatest k , and thus the best possible yield. This principle of error correction, which does not require a bidirectional link, is called *forward error correction* (FEC), by contrast to the preceding example (parity bit).

We will illustrate this principle by two simple codes (repetition and Hamming codes), in order to understand the philosophy of the more sophisticated codes used in digital video broadcasting (Reed–Solomon and convolutional codes) which we will not detail since they are based on rather complex mathematical concepts. The interested reader will, however, find some books entirely dedicated to these subjects in the bibliography.

A1.2.1 Repetition

One of the simplest means of correcting error transmissions is to repeat the bits of the message and to use a majority logic (*vote*) on the reception side to decide whether the received bit is a '1' or a '0' in the case of an error. For example, with a repetition factor of 3 ($k = 1, n = 3$), the coding of a bit would be:

Useful bit	Transmitted code
0	000
1	111

In the above example, the code is made up of two words of 3 bits, all of which are different; the minimum Hamming distance is therefore $d = 3$, and the correction capacity of the code is $t = (3 - 1)/2 = 1$ erroneous bit per word.

On the reception side, if we assume that the transmission channel does not introduce more than one error per word, a majority logic will be able to correct it by deciding, according to the table below, where the corrupted, albeit correctable, messages (one erroneous bit out of three) are in the shaded area:

Sent	000				111			
Received	000	001	010	100	011	101	110	111
Decoded	0	0	0	0	1	1	1	1

This type of error correction algorithm has a very low yield, as it triples the amount of information to be transmitted ($k/n = 0.33$), but it can correct high error rates (up to one bit out of three).

A1.2.2 The Hamming code (7, 4, d=3)

This code applies to 4-bit words ($k = 4, 2^4 = 16$ code elements) to which it adds 3 bits of redundancy thus $n = 7$; therefore the yield is $k/n = 4/7 = 0.57$.

It is used for, among other things, the coding of critical parts of teletext magazines, and it works as follows. Let u be the original 4-bit words made up of the bits u_1, u_2, u_3, u_4 . Let c be the resulting 7-bit code composed of bits $u_1, u_2, u_3, u_4, v_1, v_2, v_3$, obtained by means of the generating matrix G below:

u_1	1	0	0	0	0	1	1
u_2	0	1	0	0	1	0	1
u_3	0	0	1	0	1	1	0
u_4	0	0	0	1	1	1	1
	u_1	u_2	u_3	u_4	v_1	v_2	v_3

The parity bits v_1, v_2, v_3 (situated in the shaded area of the matrix) obey the following relations:

$$v_1 = 0 + u_2 + u_3 + u_4$$

$$v_2 = u_1 + 0 + u_3 + u_4$$

$$v_3 = u_1 + u_2 + 0 + u_4$$

On the receiving side, the decoder receives the message y composed of bits $u'_1, u'_2, u'_3, u'_4, v'_1, v'_2, v'_3$, and does the same calculation as the encoder for the bits u'_1, u'_2, u'_3, u'_4 , which yields the bits w_1, w_2, w_3 :

$$w_1 = 0 + u'_2 + u'_3 + u'_4$$

$$w_2 = u'_1 + 0 + u'_3 + u'_4$$

$$w_3 = u'_1 + u'_2 + 0 + u'_4$$

The decoder calculates the *syndrome* $S = s_1, s_2, s_3 = v'_1 - w_1, v'_2 - w_2, v'_3 - w_3$. The syndrome's calculation can be represented by the parity control matrix H below:

0	1	1	1	1	0	0	s_1
1	0	1	1	0	1	0	s_2
1	1	0	1	0	0	1	s_3
u'_1	u'_2	u'_3	u'_4	v'_1	v'_2	v'_3	

If we assume that there is a maximum error of 1 bit per received word, the results will be as follows:

- if there is no error, the syndrome will be $S = 0, 0, 0$;
- if the bit u'_4 is erroneous, the 3 bits v'_1, v'_2, v'_3 will be wrong, and the syndrome will be $S = 1, 1, 1$;
- if 1 of the 3 bits u'_1, u'_2, u'_3 received is erroneous, two of the three relations used to calculate w_1, w_2, w_3 will be wrong (as each bit is

used once in two of the three equations); the syndrome S will therefore have 2 bits at '1' and 1 bit at '0', the position of which will indicate which of bits u'_1, u'_2, u'_3 is in error;

- if the error affects a parity bit, its position will be indicated by the presence of only one '1' in the syndrome.

From this, it can be seen that this code allows the detection and correction of one error in the received word ($t = 1$) which corresponds to a minimum Hamming distance of $d = 3$.

A1.2.3 Reed-Solomon coding

This is also a block error correction code, the symbols of which are not bits but *finite field elements* (very often bytes); the mathematical theory of these codes can be found in specialized books, some of which can be found in the bibliography. We will therefore not present a theoretical description of the Reed-Solomon code, which would require long mathematical developments, but merely mention that it belongs to the cyclic class of codes and that it is a particular case of the so-called BCH codes (from their authors Bose, Ray-Chauduri, Hocquenghem).

Like the two previous codes seen before, it is characterized by the three parameters (n, k, t) which define the size of the blocks on which it acts and the number of errors that it can correct:

- n is the size (in symbols) of the block after coding;
- k is the size (in symbols) of the original block;
- t is the number of correctible symbols.

This code is well adapted to the correction of burst errors introduced by a transmission channel, which is why it has been chosen as the *outer coding* algorithm for all the variants of the DVB digital television standard, with a symbol size of 1 byte.

In the DVB case, the size of the original block is the transport packet ($k = 188$ bytes); the specified Reed-Solomon coding increases the block size by 16 bytes ($n = 204$ bytes), and the minimum Hamming distance is $d = 17$ bytes, which allows the correction of up to $(d - 1)/2 = 8$ erroneous bytes per block. This code is denoted RS (204, 188, $t = 8$), and its yield is $k/n = 188/204 = 0.92$. It is a shortened version of the RS(255, 239, $t = 8$) obtained at encoding by adding 51 null bytes before the 188 byte packets in order to form 239 byte blocks applied at the input of an RS(255, 239, $t = 8$) encoder. This coding adds 16 parity bytes to

the end of the original blocks, thus forming new protected 255 byte blocks. After encoding, the first 51 null bytes added before coding are discarded, which gives the new protected transport packets of 204 bytes.

Decoding of the Reed-Solomon code uses a fast Fourier transform (FFT) to calculate the syndromes and a Euclidean algorithm to find the error evaluation and localization polynomials. Error words are calculated by using the *Forney formula* and are corrected within the limit of a maximum of 8 bytes per block.

A1.3 Convolutional coding

We will not develop the theory of these codes either, which is quite complex and requires high level mathematical explanations, and will again refer the interested reader to some of the references in the bibliography.

This coding acts on blocks of indefinite length: in practice it will be a *continuous bitstream* of arbitrary length. It is sometimes improperly called 'Viterbi coding', from the name of the author of the decoding algorithm generally used. It is intended to correct random errors, most often as a complement to a block code.

The convolution coding transforms the input stream into n output streams (two in the case of Fig. A.1), thus adding redundancy to it. There can be as many branches in parallel as one wants, but the most common case has two. The incoming stream is applied to the input of a shift register which has intermediate outputs (taps) after each stage; the stream advances one stage in the shift register after each new incoming bit. The incoming stream is also applied to the input of the first modulo 2 adding machine of each branch, which receives the output of the first intermediate tap of the shift register at its other input. The output of this adding machine is applied to one of the inputs of the next adding machine, which

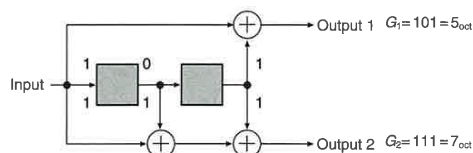


Fig. A.1 An example of convolutional coding producing two bitstreams ($R_c = 1/2, K = 3, d_{free} = 5$)

receives the output of the next stage of the shift register at its other input, and so on. Some adding machines are omitted and are assigned a coefficient of '0', while those that are not omitted are given the coefficient '1'. Each branch is then described by the binary or octal number formed by the sequence of '0's and '1's. Figure A.1 illustrates the process for a simple case with two outputs X and Y and three taps.

The code is characterized by the following parameters:

- R_c , code rate: ratio between the input and output bit-rates (here $1/2$)
- K , constraint length: number of usable taps (here 3)
- G_1 , generator sum for X (here 101_{bin})
- G_2 , generator sum for Y (here 111_{bin})

The generator sums G_1 and G_2 are obtained by assigning, from the left to the right, a '1' to the taps actually used and a '0' to the unused ones. The binary number obtained is generally represented in octal form (base 8). Hence, in the above example, the standard description is:

$$R_c = 1/2, K = 3, G_1 = 5_{\text{oct}}, G_2 = 7_{\text{oct}}$$

Another parameter, called the *free distance* (d_{free}), describes the correction capacity of the code: the higher the value of d_{free} , the more efficient is the correction (in the above example, $d_{\text{free}} = 5$).

The most standard method used to decode convolutional codes is based on the Viterbi algorithm, so named because of its inventor (1969). An explanation of the Viterbi algorithm would also be rather complex, and so we will simply indicate that its principle consists of finding the maximum likelihood route on a trellis representing the different possible states of the coder in order to find the most probable original message from the received stream.

Appendix B: spectral efficiency of cable and satellite transmissions with DVB parameters

The bandwidth occupied by an AM modulated signal is twice the bandwidth of the modulating signal, due to the two lateral bands generated by the modulation process. In the case of a digital signal of symbol period T which is Nyquist filtered with a roll-off α , the bandwidth will be:

$$BW = (1 + \alpha) \times 1/T$$

From this formula, we can now calculate the capacity of the channel and the spectral efficiency for various hypothetical cases:

Cable

With a roll-off $\alpha = 0.15$, a European channel of bandwidth $BW = 8$ MHz can accept a maximum symbol rate $F_s = 1/T = BW/(1 + \alpha) = 8/1.15 \cong 7$ MHz. A commonly used value is 6.875 MHz.

The usable order of QAM will depend on the network quality which determines E_b/N_o . In the worst case, 16-QAM (4 bits/symbol) will have to be used, resulting in a *brutto* bit-rate of:

$$R_b = 6.875 \times 4 = 27.5 \text{ Mb/s (without Reed-Solomon coding)}$$

Hence the brutto spectral efficiency will be $27.5/8 = 3.437$ b/s per Hz. The *useful* bit-rate R_u is the product of R_b and the Reed-Solomon factor r_{RS} (188/204), and hence:

$$R_u = R_b \times r_{RS} = 27.5 \times 188/204 \cong 25.34 \text{ Mb/s}$$

The useful spectral efficiency is therefore $25.34/8 = 3.16 \text{ b/s per Hz}$.

In the best case, 64-QAM (6 bits/symbol) will be possible, resulting in a brutto bit-rate of:

$$R_b = 6.875 \times 6 = 41.25 \text{ Mb/s (without Reed-Solomon coding)}$$

Hence, the brutto spectral efficiency will be $41.25/8 = 5.156 \text{ b/s per Hz}$. The useful bit-rate R_u is calculated as before,

$$R_u = R_b \times r_{RS} = 41.25 \times 188/204 \cong 38.0 \text{ Mb/s}$$

The useful spectral efficiency is therefore $38.0/8 = 4.75 \text{ b/s per Hz}$.

Satellite

Theoretically with a roll-off $\alpha = 0.35$, a channel of $BW = 33 \text{ MHz}$ supports a maximum symbol rate $F_s = 1/T = BW/(1 + \alpha) = 33/1.35 = 24.4 \text{ MHz}$.

With QPSK modulation (2 bits/symbol), the brutto bit-rate is therefore

$$R_b = 2 \times F_s = 48.88 \text{ Mb/s} \\ \text{(without Reed-Solomon and inner codings)}$$

In practice, this capacity can be slightly higher (a ratio BW/F_s of 1.20 between bandwidth and symbol rates results in only a marginal degradation). The maximum symbol rate thus becomes $33/1.20 = 27.5 \text{ MHz}$; the brutto bit-rate $R_b = 55 \text{ Mb/s}$ and the brutto spectral efficiency is $55/33 = 1.66 \text{ b/s per Hz}$.

The useful bit-rate depends on the code rate r_c chosen for the inner coding; it is equal to the product of the brutto bit-rate R_b , r_c and r_{RS} (Reed-Solomon factor = $188/204$). In the worst case (minimal E_b/N_o) the code rate $r_c = 1/2$ will have to be used to ensure a satisfactory service ($\text{BER} < 10^{-10}$); in this case the useful bit-rate is:

$$R_u = R_b \times r_c \times r_{RS} = 55 \times 1/2 \times 188/204 = 25.34 \text{ Mb/s}$$

The useful spectral efficiency is therefore $25.34/33 = 0.76 \text{ b/s per Hz}$.

In the best case (maximal E_b/N_o), the same service quality can be obtained with $r_c = 7/8$; the useful bit-rate becomes:

$$R_u = R_b \times r_c \times r_{RS} = 55 \times 7/8 \times 188/204 = 44.35 \text{ Mb/s}$$

The useful spectral efficiency is therefore $44.35/33 = 1.34 \text{ b/s per Hz}$.

In fact, most digital satellite TV transmissions in use today on ASTRA and EUTELSAT (33 MHz channels) use this maximum symbol rate of 27.5 MHz and a code rate $r_c = 3/4$. This corresponds to a useful bit-rate of $27.5 \times 2 \times 3/4 \times 188/204 \cong 38.0 \text{ Mb/s}$, which allows transparent retransmission on a cable network with 8 MHz channels by using 64-QAM at a signal rate of 6.875 MHz (best case example in our calculation for cable above). In this case, the spectral efficiency of the satellite channel is $38.0/33 = 1.15 \text{ b/s per Hz}$.

Appendix C: reception of the digital TV channels of ASTRA and EUTELSAT

The new digital satellites operate mainly in the so-called DBS band (11.7–12.5 GHz); this band has until now been used in circular polarization by high power satellites such as TDF 1/2, Hispasat, etc. At 19.2°E, ASTRA 1E and 1F provide 40 channels in the DBS band. These channels are 33 MHz wide and situated on a grid with a 19.5 MHz step. ASTRA 1G added 16 channels of 26 MHz on a grid of 14.75 MHz in the so-called 'Telecom' band (12.5 to 12.75 GHz).

Positioned on 28.2°E, the new ASTRA 2 family of satellites (used among others by BSkyB digital services) use the same channels.

The tables opposite give the frequency allocation of the digital channels of ASTRA. EUTELSAT's 'Hot Birds' 2, 3 and 4 at 13E use the same frequency bands, but with different channels (33 MHz channels on a uniform grid of 19.18 MHz from 11.700 to 12.750 GHz).

A dozen (as yet unspecified) channels between 12.50 and 12.75 GHz (FSS band currently used by telecommunication satellites such as EUTELSAT, Kopernikus, Telecom) will be used later by ASTRA 1G and HOT BIRD 4.

In order to receive these new channels, ASTRA has specified a LNC (Low Noise Converter) or LNB covering the frequency range from 10.70 to 12.75 GHz in two bands, now called 'universal' LNB. The two bands are obtained by switching the local oscillator

frequency by means of a 22 kHz signal superimposed on the LNC supply voltage; polarization is controlled, as is usual, by the value of the supply voltage (14 V = vertical, 18 V = horizontal):

11.7–12.1 GHz (ASTRA 1E)				12.1–12.5 GHz (ASTRA 1F)			
Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)
65 H	11 719.50	75 H	11 914.50	85 H	12 109.50	95 H	12 304.50
66 V	11 739.00	76 V	11 934.00	86 V	12 129.00	96 V	12 324.00
67 H	11 758.50	77 H	11 953.50	87 H	12 148.50	97 H	12 343.50
68 V	11 778.00	78 V	11 973.00	88 V	12 168.00	98 V	12 363.00
69 H	11 797.50	79 H	11 992.50	89 H	12 187.50	99 H	12 382.50
70 V	11 817.00	80 V	12 012.00	90 V	12 207.00	100 V	12 402.00
71 H	11 836.50	81 H	12 031.50	91 H	12 226.50	101 H	12 421.50
72 V	11 856.00	82 V	12 051.00	92 V	12 246.00	102 V	12 441.00
73 H	11 875.50	83 H	12 070.50	93 H	12 265.50	103 H	12 460.50
74 V	11 895.00	84 V	12 090.00	94 V	12 285.00	104 V	12 480.00

12.5–12.75 GHz (ASTRA 1G)							
Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)
105 H	12 515.25	109 H	12 574.25	113 H	12 639.25	117 H	12 692.25
106 V	12 522.00	110 V	12 581.00	114 V	12 640.00	118 V	12 699.00
107 H	12 544.75	111 H	12 603.75	115 H	12 662.75	119 H	12 721.75
108 V	12 551.50	112 V	12 610.50	116 V	12 669.50	120 V	12 728.50

11.7–12.1 GHz (HOT BIRD 2)				12.1–12.5 GHz (HOT BIRD 3)			
Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)
50 V	11 727.48	60 V	11 919.28	70 V	12 111.08	80 V	12 302.88
51 H	11 746.66	61 H	11 938.46	71 H	12 130.26	81 H	12 322.06
52 V	11 765.84	62 V	11 957.64	72 V	12 149.44	82 V	12 341.24
53 H	11 785.02	63 H	11 976.82	73 H	12 168.62	83 H	12 360.42
54 V	11 804.20	64 V	11 996.00	74 V	12 187.80	84 V	12 379.60
55 H	11 823.38	65 H	12 015.18	75 H	12 206.98	85 H	12 398.78
56 V	11 842.56	66 V	12 034.36	76 V	12 226.16	86 V	12 417.96
57 H	11 861.74	67 H	12 053.54	77 H	12 245.34	87 H	12 437.14
58 V	11 880.92	68 V	12 072.72	78 V	12 264.52	88 V	12 456.32
59 H	11 900.10	69 H	12 091.90	79 H	12 283.70	89 H	12 475.50

12.5–12.75 GHz (Hot Bird 4)							
Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)	Channel/ polarization	Frequency (MHz)
90 V	12 519.84	93 H	12 577.38	96 V	12 634.92	99 H	12 692.46
91 H	12 539.02	94 V	12 596.56	97 H	12 654.10	100 V	12 713.28
92 V	12 558.20	95 H	12 615.74	98 V	12 673.28	101 H	12 730.82

- The low band (10.70–11.70 GHz) is, for the time being, almost exclusively used by analogue TV channels; it is converted into the 950–1950 MHz range by means of a 9.75 GHz local oscillator (22 kHz signal not active).
- The high band (11.70–12.75 GHz) is dedicated to the new digital TV broadcasts; it is converted into the 1100–2150 MHz range by means of a 10.60 GHz local oscillator (22 kHz signal activated). 10.60 GHz is left out of the low band in order to avoid any interference with analogue channels.

It should be noted that a good reception of digital TV signals requires an LNC with a local oscillator having better phase noise characteristics than has been required until now for analogue reception.

In order to allow more complex antenna switching functions, e.g. to receive two or more satellites, each transmitting in two bands, EUTELSAT has proposed, in cooperation with the receiver and antenna industries, an antenna control protocol via the coaxial cable:

- This proposal called **DiSEqC** (for Digital Satellite Equipment Control) is based on a modulation of the 22 kHz signal by digital messages. This will require a microcontroller in LNCs and switching devices for decoding and answering these messages as well as executing the corresponding commands.
- It is currently being examined for standardization by the ETSI and the IEC.
- If generalized, this system should allow an almost 'plug and play' connection of any receiver to any antenna system, all devices being identified by a unique address, and their functionality being readable by the receiver.

Appendix D: the main other digital TV systems

Even if most countries in the world (in particular countries using PAL or SECAM systems for analogue TV) have adopted the DVB system, the USA, Japan and some other 'NTSC countries' use it either partly or not at all. Other systems are also used or proposed, and the three main ones are briefly described below.

D1.1 The DSS system (satellite, USA)

DSS (Digital Satellite System), launched in 1994, is historically the first 'direct to home' digital satellite TV broadcasting system and the most widely used in the USA.

DSS started with the DirecTV® network, which was shortly after complemented on the same orbital position with the USBB® bouquet. These two networks, now merged, offer together more than 200 TV channels which can be received from anywhere in the USA with the same set-top box and a small 18" (45 cm) dish. Other satellites are also positioned on 101° and 119° west.

Contrary to DVB which is an open system and an international standard, DSS® is the proprietary system of Hughes Electronics Corporation, the detailed specifications of which are not available to the general public. Furthermore, as it was specified before the finalization of the MPEG-2 specifications, it uses an audio/video coding scheme sometimes called 'MPEG-1.5' (somewhere in between MPEG-1 and MPEG-2, as the name suggests), an AC-3 audio coding (also known as Dolby® digital) and a specific multiplexing.

Table D.1 summarizes differences between the DVB-S and DSS systems.

The main competitor of DirecTV®, the DISH® Network from Echostar, uses the DVB system in the same band (DBS) and with the same channel width.

Table D.1

Parameter	Recall DVB-S	DSS	Remarks (DSS)
Video coding	MPEG-2 MP@ML	MPEG-1.5	Format up to 720 × 480
Audio coding	MPEG-1 level 2	AC3	Under licence of Dolby®
Multiplexing	MPEG transport	Proprietary	Statistical MPX allowed
Transport packet	188 bytes	130 bytes	Before channel coding
Scrambling	DVB-CSA	Proprietary	CA: Videoguard (NDS)
Energy dispersal	$G_{(15)} = 1 + X^{14} + X^{15}$	None	
External coding	RS(204, 188, $T = 8$)	RS(204, 188, $T = 8$)	Sync byte not RS protected
Internal coding	Convolution	Trellis	
Puncturing rates	1/2, 2/3, 3/4, 5/6, 7/8	2/3 or 6/7	Depending on transponder power
Modulation	QPSK	QPSK	
Roll-off factor	35%	20%	
Channel width	Up to 54 MHz	27 MHz	Only one channel width
Symbol rate	From 1 to 45 Ms/s	20 Ms/s	
Allocated Ku band	10.7 to 12.75 GHz	12.2 to 12.7 GHz	So-called DBS band

Table D.2 gives the frequency of the 32 DBS channels which use circular left (L) or right (R) polarization.

Table D.2

DBS band (USA) 12.200–12.700 GHz							
Channel/ polar.	Frequency (MHz)	Channel/ polar.	Frequency (MHz)	Channel/ polar.	Frequency (MHz)	Channel/ polar.	Frequency (MHz)
1 L	12224	9 L	12341	17 L	12457	25 L	12574
2 R	12239	10 R	12355	18 R	12472	26 R	12588
3 L	12253	11 L	12370	19 L	12486	27 L	12603
4 R	12268	12 R	12384	20 R	12501	28 R	12618
5 L	12282	13 L	12399	21 L	12516	29 L	12632
6 R	12297	14 R	12414	22 R	12530	30 R	12647
7 L	12311	15 L	12428	23 L	12545	31 L	12661
8 R	12326	16 R	12443	24 R	12559	32 R	12676

D2.1 The ATSC system (terrestrial, USA)

The ATSC (Advanced Television System Committee) system is the result of work started in the mid-1980s which had as their original objective the specification of a high definition television system for

terrestrial diffusion, in order to compete with the Japanese MUSE and the European HD-MAC systems.

Many industrial interest groups were put in competition, out of which came three proposed digital solutions. In May 1993, they agreed to cooperate under the name of 'Grand Alliance' in order to combine their individual proposals into a unique system, of which specifications were published in 1995 (Document ATSC A/53).

This system is based on the MPEG-2 standard for the video coding (MP@HL), and can transmit pictures at high definition (max. 1920 × 1080, 16/9), enhanced definition (1280 × 720, 16/9) and standard definition (720 × 480, 4/3 or 16/9).

A 640 × 480, 4/3 format has also been foreseen in view of computer graphics compatibility (VGA). In all, no less than 18 different display formats (interlaced or progressive) have to be supported.

Audio compression uses the AC-3 process of the Dolby® laboratories (Document ATSC A/52). Multiplexing complies with the MPEG-2 system specification (ISO/IEC 13818-1), and 188-byte transport packets are thus identical to those used in the DVB system. Regarding channel coding and modulation used for terrestrial transmission, a process known as **8-VSB** (vestigial side band with 8 states) has been retained.

On the transmission side, after removal of the sync byte, energy dispersal followed by an internal coding RS(207, 187, $T = 10$) are applied to the 187 remaining bytes. The resulting 207-byte packets follow a convolutional interleaving followed by a trellis coding with a puncturing of 2/3.

A mapping forms 'segments' of 828 symbols coded on 3 bits (8 levels) which are grouped in frames of 312 segments. Each frame starts by a synchronization segment formed with the synchronization bytes of the transport packets and each segment starts with a synchronization symbol. The structure obtained in this way resembles an analogue television frame with a duration of 24.2 ms and is made of 313 segments (comparable to a TV line) of 77.3 μs.

The frames are grouped by two in a similar manner to an analogue interlaced picture. This signal modulates a unique carrier (amplitude modulation with suppressed carrier), which can take eight discrete values during the useful duration of the segments. The signal is then Nyquist filtered (raised cosine with $\alpha = 0.1152$) in order to limit the upper sideband to 5.69 MHz and the lower sideband to 0.31 MHz. A pilot carrier with reduced amplitude is reinserted in place of the suppressed carrier.

Table D.3

Parameter	recall DVB-T	ATSC	Remarks (ATSC)
Video coding	MPEG-2 MP@ML	MPEG-2 MP@HL	18 formats, SDTV to HDTV
Audio coding	MPEG-1 level 2	AC3	Licence of Dolby® labs
Multiplexing	MPEG transport	MPEG-2 transport	Same as DVB
Transport packet	188 bytes	188 bytes	Before channel coding
Scrambling	DVB-CSA	Not specified	NRSS detachable CA module
Energy dispersal	$G_{(15)} = 1 + X^{14} + X^{15}$	$G_{(16)} = X^{16} + X^{13} + X^{12} + X^7 + X^6 + X^3 + X + 1$	
External coding	RS(204, 188, $T = 8$)	RS(207, 187, $T = 10$)	Sync byte not RS protected
Internal coding	Convolutional	Trellis	
Puncturing rates	1/2, 2/3, 3/4, 5/6, 7/8	2/3	Fixed value
Modulation	COFDM 2K/8K	8-VSB	16-VSB mode for cable
Roll-off factor	NA	0.1152	Attenuation band : 0.31 MHz
Channel width	6, 7 or 8 MHz	6 MHz	No 7 or 8 MHz variant
Useful bit-rate	From 5 to 31 Mb/s	19.28 Mb/s	Fixed rate

One can see that this process has nothing to do with the COFDM used in the DVB-T standard, but it does not have its flexibility (all the parameters are fixed). More importantly, it does not have the same performance, especially in the presence of multiple echos.

These unsatisfactory performances, added to the complexity and cost of the complete system due to the numerous display formats supported, including HD, mean that three years after its commercial launch in 1998, the ATSC system did not really take off (some tens of thousand sets in 2001).

Table D.3 summarizes differences between the DVB-T and ATSC systems.

D3.1 The ISDB-T system (terrestrial, Japan)

In 1999 Japan proposed another system for digital terrestrial television in view of standardization by the ITU. This system, named **ISDB-T** (Integrated Services Digital Broadcasting for Terrestrial transmission), is like DVB-T, based on COFDM modulation, and can be seen as a kind of extension of it. The input format (MPEG-2 transport stream) and the channel coding are identical to those of

DVB-T. The philosophy behind this system is on the one hand to increase flexibility in allocating services by dividing the channel into segments capable of transmitting different services able to support different receiving conditions, and on the other hand to have the possibility of further increasing the signal's robustness for mobile reception by adding a time interleaving function.

The TV channel (which can have a 6, 7 or 8 MHz width) is divided in 14 'segments', 13 of which (numbered 0 to 12) are effectively used, the 14th corresponding to a guard space between adjacent channels.

Three COFDM modes are foreseen which are characterized by the number of carriers per segment:

108 carriers/segment (corresponding to a 2K mode in a single segment)

216 carriers/segment (corresponding to a 4K mode in a single segment)

432 carriers/segment (corresponding to a 8K mode in a single segment)

The segments are associated in groups (maximum three groups per channel), which have the same purpose as the hierarchical modes of DVB-T. The minimum size of a group is one segment.

In order to ensure different receiving conditions, each segment group can use different internal coding parameters (puncturing rate) and modulation orders (dQPSK, QPSK, 16-QAM, 64-QAM) as well as different time interleaving depths.

The system thus allows partial, narrow-band reception (single segment, which has to be the central segment), for example for mobile audio reception.

Figure D.1 represents the 13 segments and their position in the channel, where three groups can be seen, of which segment 0 in the middle of the channel is intended for narrow-band reception.

The only significant improvement of ISDB-T over DVB-T is the robustness if a relatively long time interleaving is used, which allows it to withstand relatively long reception 'holes' or disturbances. However, this feature is balanced by a considerable cost increase, mainly due to the RAM capacity required by time interleaving.

At the time of writing, ISDB-T is still not in commercial use in Japan and no consumer circuit is available for this system, which



Fig. D.1

Table D.4

Parameter	Recall DVB-T	ISDB-T	Remark
Multiplexing	MPEG-2 transport	MPEG-2 transport	Same as DVB-T
Transport packet	188 bytes	188 bytes	Same as DVB-T
Energy dispersal	$G_{(15)} = 1 + X^{14} + X^{15}$	$G_{(15)} = 1 + X^{14} + X^{15}$	Same as DVB-T
External coding	RS(204, 188, $T = 8$)	RS(204, 188, $T = 8$)	Same as DVB-T
Internal coding	Convolutional	Convolutional	Same as DVB-T
Puncturing rates	1/2, 2/3, 3/4, 5/6, 7/8	1/2, 2/3, 3/4, 5/6, 7/8	Same as DVB-T
Modulation	COFDM 2K/8K	Segmented COFDM	108, 216, 432 carriers/seg.
Modulation orders	4, 16 or 64	4d, 4, 16 or 64	dQPSK supported
Segment number	1	13 useful	segment width = channel/14
Hierarchical modes	By constellation	By segmentation	3 groups max.
Time interleaving	No	Yes	Improvement of mobile rec.
Guard interval	1/4, 1/8, 1/16, 1/32	1/4, 1/8, 1/16, 1/32	Same as DVB-T
Channel width	6, 7 or 8 MHz	6, 7 or 8 MHz	Same as DVB-T
Useful bitrate (8 MHz)	From 5 to 32 Mb/s	From 5 to 31 Mb/s	Slightly less than DVB-T

does not prevent its promoters from strong lobbying actions that put it in direct competition with DVB-T in countries which have as yet not chosen a system for digital terrestrial TV.

Table D.4 shows the comparison of the main parameters of DVB-T and ISDB-T.

Appendix E: the IEEE1394 high speed serial AV interconnection bus

Interconnection of new digital audiovisual consumer equipment such as an IRD, a VCR, a camcorder and a multimedia computer will soon require a high speed link between the different pieces of equipment. As the equipment will be operating in a consumer environment, it has to be designed to be low cost and as easy to use as possible ('plug and play' with 'hot plugging' capability).

The recently standardized IEEE1394 (1995) high speed serial bus fulfils these requirements. It is based on Apple computer's 'Fire-Wire' bus and allows communication at up to 400 Mb/s (an upgrade to more than 1 Gb/s is under study). Three standard speeds are defined (100, 200 and 400 Mb/s), but most current implementations only cover the first two speeds.

The IEEE1394 bus architecture enables the construction of a kind of network, made up of multiple (up to 63) participants called **nodes**. Each of the nodes has three basic functional components:

- a physical layer interface (PHY) which carries out the physical interfacing to the cable, bus arbitration and an active repeater function;
- a link layer controller (LINK) which assembles/disassembles data packets and handles handshaking and acknowledgement;
- a host controller (MCU) which deals with the higher levels of the bus protocols.

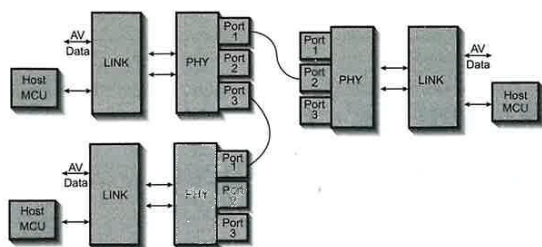


Fig. E.1 Structure of an IEEE1394 network (example with three nodes)

Figure E.1 shows an example of a three node IEEE1394 network where the three above-mentioned components can be seen.

In the first implementation of IEEE1394 chipsets, the blocks denoted 'LINK' and 'PHY' generally correspond to two different integrated circuits. The IEEE1394 specification defines two kinds of data transfer which can coexist on the bus:

- **Isochronous** transfers are reserved for time-critical data, such as audio and video streams which need to be 'real time', as delays in audio or video frames cannot be tolerated. These types of transfer are allocated a guaranteed bandwidth and use time division multiplexing.
- **Asynchronous** transfers are reserved for non-time-critical data, which are used, for instance, for data interchange or storage and employ a more conventional process of handshaking and acknowledgement.

The proportion of bus bandwidth reserved for isochronous data packets is defined during the initialization phase by electing as the 'root node' a node which includes an isochronous resource manager (IRM). The IRM is then responsible for allocating isochronous resources for a whole session, based on $125\mu\text{s}$ cycles on the bus. However, this is not sufficient to guarantee the very low jitter required to transmit, for example, an MPEG-2 transport stream over the bus (a variation in transport delay of less than 500 ns is required in this case).

In the case of a DVB set-top box, the AV data interface of the link circuit in Fig. E.1 would be connected to the transport stream path (i.e. between the output of the channel decoding part and the input of the source decoding part, before or after descrambling). This would allow, for example, the non-descrambled transport

stream to be sent to a digital VCR, and in this case playback would require suitable access rights for viewing.

In order to enable the receiver to compensate for the delays introduced by the bus, an AV layer has been defined and standardized in the IEC1883 specification. It adds a 32-bit header (a quadlet in IEEE1394 terminology) to the 188-byte MPEG-2 packets which then become 192 bytes long, the header quadlet containing a time stamp which allows resynchronization at the receiving end. In addition, if a 192-byte packet (= 48 quadlets) cannot be transmitted in one $125\mu\text{s}$ cycle, the AV layer allows it to be split into two, four or eight smaller equal parts (or blocks) which then can fit within one cycle.

Electrically, the bus is made up of two signals (DATA and STROBE), which are both transported in differential mode at low impedance on a shielded pair of conductors. An optional third pair is used for power distribution over the bus. Each branch of the network between two nodes can be up to 4.5 m long and is terminated at both ends by two 56Ω resistors connected to a common DC reference. A low cost six pin connector (similar to those used on Nintendo Game Boy[®] consoles) and a specific cable have been standardized (see Fig. E.2). A version of the bus with only two pairs (without the power pair) has also been standardized.

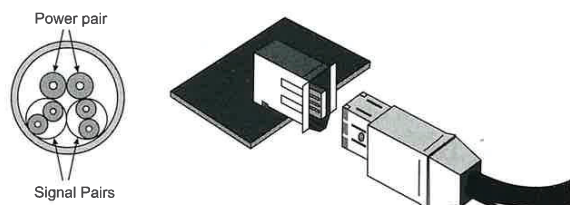


Fig. E.2 Cable and connector for IEEE1394 cabling (six pin version with power)

Appendix F: the DiSEqC bus for antenna system control

Back in 1995, EUTELSAT proposed, in cooperation with the satellite receiver and antenna industries, the introduction of a 'control bus' using the coaxial cable of a satellite antenna called **DiSEqC™** (Digital Satellite Equipment Control).

This solution is compatible with the most common switching system in use today in recent individual satellite installations (14/18 V for the polarization and 22 kHz for the band). It is royalty free and the **DiSEqC™** label may be used, provided the EUTELSAT specifications are fulfilled.

F1.1 The DiSEqC levels

DiSEqC has three implementation levels: 'ToneBurst', DiSEqC 1.0 and DiSEqC 2.0 and above.

- The 'ToneBurst' is a short-term, simple extension to the 22 kHz tone intended to allow switching of two antennas (or LNBs fitted on the same dish, e.g. for Astra 19.2°E and Eutelsat 13°E).
- The first 'true' DiSEqC level (DiSEqC 1.0) is a unidirectional, relatively simple implementation allowing more sophisticated controls, sufficient for most consumer installations.
- Upper levels (DiSEqC 2.0 and above) use a bidirectional communication between the master (set-top box) and slaves, with a view to longer-term 'plug and play' installation.

Table F.1 Minimum functionality of receiver for the various DiSEqC levels

Level	Communication	Minimal receiver functionality
ToneBurst	Unidirectional	14/18 V + 0/22 kHz + A/B switch control
DiSEqC 1.0	Unidirectional	Idem ToneBurst + control of 4 committed switches
DiSEqC 2.0	Bidirectional	Idem 1.0 + reading of slave data (LO frequency, etc.)
DiSEqC 2.1	Bidirectional	Idem 2.0 + control of 4 uncommitted switches
DiSEqC 2.n	Bidirectional	Increasing functionality (unspecified) between 2.1 and 3.0
DiSEqC 3.0	Bidirectional	Idem 2.n + configuration of receiver by the bus ('plug and play')

Table F.2 Implementation of various DiSEqC levels in slaves

Level	Communication	Implementation in peripheral slave devices
ToneBurst	Unidirectional	Simple analogue circuitry
DiSEqC 1.0	Unidirectional	Specific microcontroller (<i>1.0 level not recommended in slaves</i>)
DiSEqC 2.0	Bidirectional	Specific microcontroller
DiSEqC 3.0	Bidirectional	Idem 2.0 + 'intelligent interface'

For each DiSEqC level, a minimal functionality has been defined by EUTELSAT for receivers (Table F.1) and peripherals (Table F.2).

F2.1 DiSEqC basic principles

DiSEqC is an extension to the 22 kHz tone control in use today (mostly for bandswitching of Universal LNBs) which simplifies its implementation in existing receivers.

It is a system with a single master (the satellite receiver) and many slaves (peripheral devices such as switches, LNBs, etc.); the master is the only one to have the capability of starting a communication on the bus, slaves can only react to a command from the master.

DiSEqC philosophy has some similarities with the popular I²C bus from Philips. DiSEqC messages are obtained by PWK (pulse width keying) modulating the duty cycle of the 22 kHz with two values: '0' corresponds to two-thirds duty cycle and '1' to one-third duty cycle (Fig. F.1).

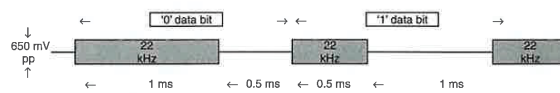


Fig. F.1 Definition of the DiSEqC bit

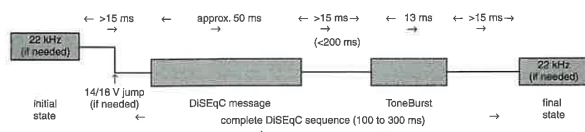


Fig. F.2 DiSEqC sequence compatible with Universal LNB and ToneBurst

The bit period is 33 cycles of 22 kHz ($1.5 \text{ ms} \pm 20\%$ taking into account the 22 kHz tolerance). Each DiSEqC message must be followed by at least 6 ms 'silence'.

The 22 kHz level superimposed on the LNB supply voltage is nominally 650 mV pp. Above level 2.0, the master should have a source impedance of 15Ω at 22 kHz in order to be able to receive the response from the slaves.

DiSEqC is specified in a way which is compatible with existing hardware:

- In order to be compatible with existing peripherals (14/18 V, 0/22 kHz, ToneBurst) new receivers must first apply the appropriate voltage for polarization control and the 22 kHz tone if required, then after a short 'silence' the true DiSEqC message followed by the ToneBurst (see Fig. F.2).
- In order that new DiSEqC peripherals be at least partly controllable by existing (non-DiSEqC) receivers, they will initially have to respond to 'traditional' 14/18 V and 22 kHz controls.

F2.1.1 DiSEqC 'ToneBurst'

The ToneBurst consists of either a plain 22 kHz burst of 12.5 ms (satellite A selection) or of a sequence of nine '1' (one-third duty cycle) for satellite B selection (see Fig. F.3).

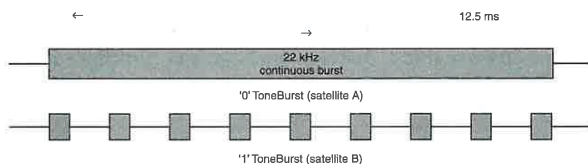


Fig. F.3 Definition of the ToneBurst

F2.1.2 DiSEqC 1.0 and above formats

DiSEqC messages are made of one or more bytes (MSB first) each followed by an odd parity bit in order to be able to detect transmission errors.

- Commands from *master* consist of a minimum of 3 bytes (framing, address, control), followed by a certain number of data bytes for some commands. The general format is illustrated below:

FRAMING	P	ADDRESS	P	CONTROL	P	DATA	P	DATA	P
---------	---	---------	---	---------	---	------	---	------	---

In order to keep the duration of a message within acceptable limits, a maximum of 1 byte of data is used with level 1.0, and a maximum of 3 for levels 2.0 and above.

- The response from a *slave* is at the master's discretion, and occurs only if they are both at level 2.0 or above. It is made of a framing byte (used as acknowledge) optionally followed by one or more data bytes carrying the message content. Its format is shown below:

FRAMING	P	DATA	P	DATA	P
---------	---	------	---	------	---

Slave response must occur within 150 ms after the master's command end. The error recovery method is not defined by the specification; the most usual strategy consists of repeating the message, but is entirely at the master's software discretion.

F3.1 Different fields of the DiSEqC message

F3.1.1 Framing byte

The framing byte is the first byte of a master's command or a slave's response.

- The 4 MSBs are used for synchronization (*run-in*) and recognition (*framing*) of a DiSEqC message. They are currently fixed at '1110', but the two last bits might be affected later.
- The 5th bit is reserved (currently '0'); it might be used later.

Table F.3

Hex.	Binary	Meaning
E0	1110 0000	Master command, no response expected, 1st transmission
E1	1110 0001	Master command, no response expected, 2nd transmission
E2	1110 0010	Master command, response expected, 1st transmission
E3	1110 0011	Master command, response expected, repetition
E4	1110 0100	Slave response, repetition unnecessary, no error
E5	1110 0101	Slave response, repetition unnecessary, command not supported
E6	1110 0110	Slave response, repetition required, parity error
E7	1110 0111	Slave response, repetition required, command not recognized

- The 6th bit indicates the origin of the message: '0' if it comes from the master, '1' if it comes from a slave.
- The 7th bit is set to '1' if a response is expected, '0' otherwise. In the case of a slave response, it can be set to '1' to ask repetition of the message in the case of a transmission error.
- The 8th bit is set to '1' by the master if it is a retransmission. In the case of a slave response, it indicates the reason of non-execution of the command: '0' in the case of a parity error, asking for retransmission, '1' if the command is not supported by the slave.

The meaning of framing bytes is described in Table F.3; only the first two lines apply to level 1.0.

F3.1.2 Address byte

The second byte sent by the master indicates the slave to which the message is destined; it is subdivided into two nibbles. The MS nibble indicates the family to which the slave belongs (LNB, switch, positioner, etc.); the LS nibble divides the family into subgroups.

For each nibble the value '0000' indicates a value for all families (MS nibble) or all subgroups of a family (LS nibble).

The address hex. 00 is recognized by all peripherals.

Table F.4 indicates addresses currently specified (addresses above hex. 20 are only used by levels 2.0 and above).

F3.1.3 control (or command) byte

This byte allows up to 256 basic commands, optionally controlled by one or more data bytes. DiSEqC specification (version 4.0 of

Table F.4

Hex. address.	Binary	Category
00	0000 0000	All families, all subgroups (general call)
10	0001 0000	Switch (all types, including LNB or SMATV)
11	0001 0001	LNB
12	0001 0010	LNB with loop-through
14	0001 0100	Switch
15	0001 0101	Switch with loop-through
18	0001 1000	SMATV
20	0010 0000	Polarizer (all types)
21	0010 0001	Linear polarization control (full skew)
22	0010 0010	Fine polarization adjustment
30	0011 0000	Positioner (all types)
40	0100 0000	Installation support (all types)
41	0100 0001	Signal intensity adjustment support
60	0110 0000	Reserved for address reallocation
70	0111 0000	Slave intelligent interface for proprietary multimaster bus
Fx	1111 xxxx	Manufacturer extensions

Table F.5

Hex.	Priority	Bytes	Name	Function	Response
03	M	3	Power On Peripheral supply		—
38	M	4	Write N0	Write to port 0 (committed switches)	—
39	R	4	Write N1	Write to port 1 (uncommitted switches)	—

22/03/96) defines around 50 commands, divided into three priority categories: mandatory (M), recommended (R) and suggested (S).

For the 1.0 level, only the first two commands of Table F.5 are mandatory (M), the third is recommended (R):

These commands determine the state of the output ports of a specific microcontroller defined by EUTELSAT by means of the byte which follows them.

F3.1.4 Data bytes

Commands 38 and 39 must be followed by a byte, of which each bit at '1' of the MS nibble (clear) clears individually one of the four outputs of one of the two ports of the slave microcontroller, the LS 'nibble' (set) setting them individually in the same way

CLEAR	SET
$b_7b_6b_5b_4 = 1111 = F_h$	$b_3b_2b_1b_0 = xxxx = x_h$

Table F.6

Bit	b_3		b_2		b_1		b_0	
Function	OPTION (1)		SATELLITE (2)		POLARIZATION		BAND	
State	1	0	1	0	1	0	1	0
Result	β	α	B	A	Horiz.	Vert.	High	Low

(1) Example: α = satellites group ASTRA/EUTELSAT, β = satellites group TELECOM 2A/2B.

(2) Example: A = Astra, B = Eutelsat (group α), A = Telecom 2A, B = Telecom 2B (group β).

A DiSEqC 1.0 message intended to control an output port of the slave microcontroller will follow the general form below:

FRAMING (E0 or E1) ADDRESS (10 to 18) COMMAND (38 or 39) DATA (Fx)

For command 38h, three of the four outputs of the 'committed' port are dedicated to a predefined function (the fourth one being usable freely) according to Table F.6.

Command 39h functionality depends on the use of the 'uncommitted' switches by the manufacturer.

F3.1.5 Example of a DiSEqC 1.0 message

In the case where the OPTION bit is used as defined above:

- If one wants to receive a channel on Astra in the LOW band and VERTICAL, the data byte will be 1111 0000 (hex. F0).
- For Telecom 2B in HIGH band and VERTICAL polarization it will be 1111 1111 (hex. FF).

In the first case, if it is the first transmission of the message with the generic address 10h (all switches and LNBs), the message will be:

FRAMING = E0 ADDRESS = 10 COMMAND = 38 DATA = F0

In the second case, if it is a repetition, the first byte becomes E1:

FRAMING = E1 ADDRESS = 10 COMMAND = 38 DATA = FF

We will not detail further the DiSEqC system, for which the interested reader can consult the reference documents, freely available on the EUTELSAT website (www.eutelsat.org).

Appendix G: the common interface (DVB-CI)

In order to allow access to pay TV transmissions scrambled according to the DVB standard with a non-service provider specific terminal, the DVB consortium has defined a standardized interface known as **DVB-CI** (DVB common interface) which allows adjunction of detachable conditional access modules with a PC-Card (ex PCMCIA) form factor, of which DVB-CI is an extension.

This interface is defined in the European standard EN 50221 and the ETSI extension TS 101699 and a few additional documents (implementation guides).

The DVB-CI interface can in principle also be used for other purposes, for example to increase the functionality of the set-top box by means of FLASH memory containing software extensions, to add a NIM for another transmission medium, or even add some modules from the PC world (modems or others) if the set-top box software can exploit them.

Conditional access, however, is by far the main use, which is why we will describe briefly its principle of operation.

Figure G.1 represents the block diagram of a set-top box equipped with a DVB-CI conditional access module, where it can be seen that the module receives the scrambled transport stream at its input, and, assuming that a subscriber's smart card with valid access rights is inserted in it, delivers it 'in the clear' to the source decoder.

The module is, in addition, connected to the main processor by means of a conventional address, data and control bus to exchange control and OSD information.

Figure G.2 represents a more detailed view of the DVB-CI module and its communication with the set-top box and the subscriber's smart card.

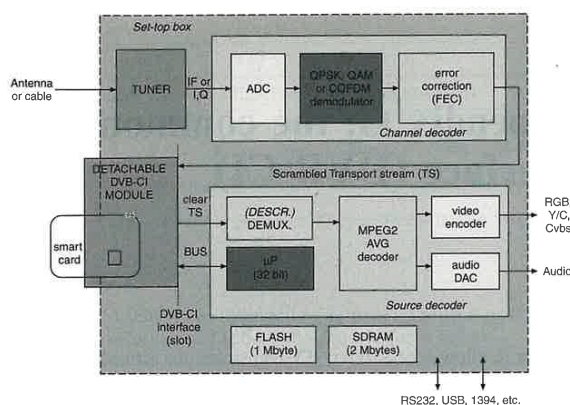


Fig. G.1 Block diagram of a receiver with a DVB-CI interface and CA module

The CA module includes mainly a DVB-CSA descrambler (or possibly another one) and a microcontroller, on which the conditional access software is executed. The microcontroller is connected to a smart card reader via the ISO 7816 interface and to the host processor of the set-top box via the DVB-CI interface. Operation of the CA module is similar to that of an embedded CA (see Chapter 5).

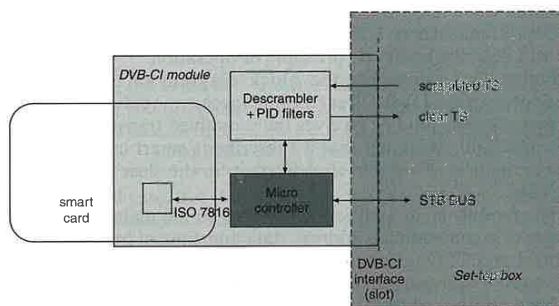


Fig. G.2 Simplified block diagram of a DVB-CI CA module

The microcontroller receives, after selection by a PID filtering, the encrypted control words transmitted in the ECM messages and returns them to the descrambler after decryption by means of the keys contained in the smart card, on the condition that it has the required access rights which are sent by the EMM messages.

This process allows the module to return a transport stream partly or fully descrambled (depending on the level – program or transport – of the scrambling at the transmission side).

More than one DVB-CI (most often two) may be present in a set-top box. In this case, the transport stream interfaces are connected in cascade (daisy chain).

The set-top box's internal DVB-CI interface circuitry contains switches which enable a direct path for the transport stream when no module is inserted in a slot, or when the transmission is in the clear or scrambled with a system not supported by the module.

In addition to the physical interface, the DVB-CI specification also defines a communication protocol between the module and the host processor, in order that (in principle) all modules are compatible with all DVB-CI compliant boxes, which is unfortunately not always the case.

The internal software of the DVB-CI module is of course dependent on the conditional access supported, whereas the hardware implementation of the module can be common to many systems based on the DVB-CSA scrambling algorithm.

Modules are available today for most conditional access systems used in Europe, and the presence of at least one DVB-CI slot is mandatory for terrestrial digital TV receivers in most countries where these transmissions are available.

Appendix H: example chipset for DVB decoders

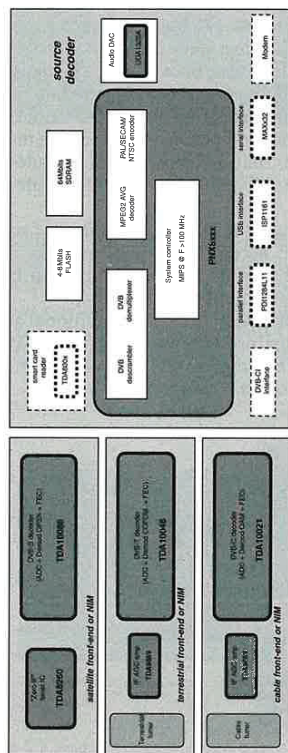


Fig. H.1 Example chipset for DVB decoders (2002)
(source: Philips Semiconductors)

Glossary of abbreviations, words and expressions

Abbreviations

- AC-3** – multichannel digital audio system developed by the US company Dolby®.
- ADC** – analogue-to-digital converter (device converting an analogue voltage into a binary number).
- AF** – adaptation field (data field used to adapt the PES to transport packet length).
- API** – application programming interface (programming interface of middleware for its compatible applications; sometimes used to designate the middleware itself).
- ASK** – amplitude shift keying (digital amplitude modulation with two states).
- ATSC** – Advanced Television System Committee (US digital terrestrial TV system supporting HDTV and using 8-VSB modulation).
- AU** – access unit (MPEG coded representation of a presentation unit: picture or sound frame).
- B** – bidirectional (picture): MPEG picture coded from the preceding and the following picture.
- BAT** – bouquet association table (optional table of DVB-SI).
- BER** – bit error rate (ratio of the number of erroneous bits to the total number of bits transmitted).
- BPSK** – bi phase shift keying (two-state phase modulation, carrying one bit per symbol).

CA – conditional access (system allowing to limit the access to pay TV broadcasts).

CAM – conditional access message (specific messages for conditional access: ECM and EMM).

CAT – conditional access table (MPEG-2 table indicating the PID of conditional access packets).

CAZAC/M – constant amplitude zero auto correlation (reference symbol in the DVB-T proposal).

CCIR – Comité Consultatif International de Radiodiffusion (now IUT-R).

CCIR-601 (ITU-R 601) – recommendation for digitization of video signals ($F_s = 13.5$ MHz, YUV signals in 4:2:2 format).

CCIR-656 (ITU-R 656) – recommendation for interfacing CCIR-601 signals (most common variant: 8 bits parallel multiplexed YUV format).

CCITT – Comité Consultatif International du Télégraphe et du Téléphone (now ITU-T).

CIF – common intermediate format (360×288 @ 30 Hz).

CNR – carrier-to-noise ratio (or C/N) ratio (in dB) between the received power of the carrier and the noise power in the channel bandwidth.

COFDM – coded orthogonal frequency division multiplex (see OFDM).

CSA – common scrambling algorithm (scrambling algorithm specified by DVB).

CVBS – composite video baseband signal (colour composite video, e.g. NTSC, PAL or SECAM).

D1 – professional digital video recording format in component form (CCIR-656).

D2 – professional digital video recording format in composite form (NTSC, PAL or SECAM).

D2-MAC – duobinary multiplex analogue components (hybrid standard used on satellite and cable).

DAB – digital audio broadcasting (new European digital audio broadcasting standard).

DAC – digital-to-analogue converter (device converting a binary value into an analogue voltage or current).

DAVIC – digital audio visual council, a non-profit making association based in Geneva with members from all industries involved in digital technologies applied to audio and video, from content producer to service provider to hardware manufacturer. One of its goals is to define and specify open interfaces allowing maximal interoperability across countries, applications and services.

DBS – direct broadcast satellite (satellite in the 11.7–12.5 GHz band reserved for TV broadcast).

DC – direct current (coefficient of the null frequency in the DCT).

DCT – discrete cosine transform (temporal to frequency transform used in JPEG and MPEG).

DiSeq[®] – digital satellite equipment control (control protocol allowing a bidirectional digital communication between the set-top box and the antenna for complex switching functions by modulating the 22 KHz tone with digital messages). Two versions (1.0 and 2.0) currently exist.

DOCSIS – data over cable service interface specifications (main standard of US cable modems).

DPCM – differential pulse code modulation (coding of a value by its difference to the previous one).

DRAM – dynamic random access memory (read/write memory requiring a periodic refresh of the information, the most widespread due to its low cost).

DSM – digital storage medium (mass storage devices such as hard disk, tape or CD).

DSP – digital signal processor (processor specialized in processing of digitized analogue signals).

DSS – Digital Satellite System (main direct-to-home digital satellite TV system in the USA).

DTS – decoding time stamp (indicator of the decoding time of an MPEG access unit).

DVB – digital video broadcasting: European digital TV standard with three variants, DVB-C (cable), DVB-S (satellite), DVB-T (terrestrial).

DVB-CI – common interface (DVB interface for conditional access modules in PCMCIA form).

DVB-SI – system information (group of tables specified by DVB, additional to MPEG-2 PSI).

DVD – digital versatile disk (new unified format of laser disk with 4.7–19 Gbyte capacity).

E_b/N_o – ratio between the average bit energy E_b and noise density N_o (related to C/N).

EBU – European Broadcasting Union (French: Union Européenne de Radiodiffusion, UER). Organization grouping the main European broadcasters which, among other things, works on new broadcasting standards (ex: DAB, DVB) which are then approved by the ETSI.

ECM – entitlement control message (first type of conditional access message of the DVB standard).

- EIT** – event information table (optional table of DVB-SI indicating a new event).
- ELG** – European Launching Group (group at the origin of the DVB project in 1991).
- EMM** – entitlement management message (second type of conditional access message of the DVB standard).
- EPG** – electronic programme guide (graphical user interface for easier access to DVB programmes).
- ES** – elementary stream (output stream of an MPEG audio or video encoder).
- ETSI** – European Telecommunications Standards Institute (organization issuing the European standards in the field of telecommunications, ETS).
- FCC** – Federal Communications Commission (regulatory authority for telecommunications in the USA).
- FEC** – forward error correction (addition of redundancy to a digital signal before transmission, allowing errors to be corrected at the receiving end; synonym: channel coding).
- FFT** – fast Fourier transform (digital Fourier transform on sampled signals).
- FIFO** – first in, first out (type of memory often used as a buffer).
- FSK** – frequency shift keying (digital frequency modulation with two states).
- FSS** – fixed service satellite (satellite in the 10.7–11.7 GHz or 12.5–12.75 GHz bands originally reserved for telecommunications).
- FTA** – free to air (or free-to-view; designates free access TV programmes).
- GOP** – group of pictures (MPEG video layer: succession of pictures starting by an I picture).
- H261** – compression standard used for ISDN videotelephony (bitrate of $p \times 64$ kb/s).
- HD-MAC** – high definition MAC (high definition extension to the D2-MAC standard with 1250 lines).
- HDTV** – High Definition TeleVision (TV system with more than 1000 scanning lines).
- HTML** – HyperText Markup Language. The main coding standard used for creating web pages. Tags (codes) are used, for example, to specify text styles and colour, display images, include links to other web pages and/or web sites etc.
- I** – in-phase; for QAM, designates the signal modulating the carrier following the 0° axis.
- I** – intra (picture); MPEG picture coded without reference to other pictures.

- I²C** – inter-integrated circuits (serial interconnection bus between ICs developed by Philips).
- I²S** – inter-integrated sound (serial link between digital sound ICs developed by Philips).
- iDTV** – integrated digital TV (hybrid TV set able to receive analogue and digital transmissions).
- IDTV** – Improved Definition TeleVision (TV system with a resolution between standard TV and HDTV).
- IEC** – International Electrotechnical Commission (international organization for standardization in the field of electrotechnics, electricity and electronics).
- IEEE 802.11b** – wireless LAN standard at 2.4 GHz and 11 Mb/s.
- IEEE1284** – bidirectional high speed parallel interface (enhanced Centronics interface).
- IEEE1394** – high speed serial interface (up to 400 Mb/s) which is the likely future standard for consumer digital A/V links (already used in some digital video recorders and camcorders).
- IRD** – integrated receiver decoder; popular synonym: set-top box.
- ISDB-T** – Integrated Services Digital Broadcasting-Terrestrial (Japanese digital terrestrial TV and multimedia system also based on COFDM modulation).
- ISI** – inter-symbol interference (interference between successive symbols in a digital transmission).
- ISO** – International Standards Organization (international standardization organization within the UNO).
- ITU** – International Telecommunications Union (world regulation organization for telecommunications, previously CCITT).
- JPEG** – Joint Photographic Experts Group (standard for fixed pictures compression).
- LNC** – low noise converter (down-converter situated at the focus of a satellite antenna). Synonym = LNB (low noise block).
- MAC** – media access control (designates hardware and software parts of a network adapter or of an interactive cable receiver, in charge of management of network access and bandwidth sharing between users).
- MHEG-5** – Multimedia and Hypermedia Expert Group version 5 (standard middleware based on the MHEG descriptive language requiring only limited resources; used in the UK for digital terrestrial TV).
- MHP** – multimedia home platform (standard, open middleware proposed by the DVB consortium for interoperability of all digital TV transmissions in Europe).

- MP3** – MPEG-1 level 3 (audio compression format approximately two times more efficient than level 2; MP3 is the *de facto* standard for exchange of music files on the Internet).
- MP@ML** – main profile at main level (main video format of the DVB standard).
- MPEG** – Motion Pictures Experts Group (group which developed the MPEG-1 and MPEG-2 standards, currently working on MPEG-4 for very low bit-rate compression).
- MUSE** – Japanese high definition television system (analogue system with digital assistance).
- MUSICAM** – masking universal sub-band integrated coding and multiplexing (coding process of MPEG-1 audio, layer 2 used by DAB and DVB).
- NICAM** – near instantaneous companded audio multiplexing (digital sound system for analogue TV using a QPSK modulated carrier at 5.85 or 6.55 MHz).
- NIT** – network information table (optional table of DVB-SI).
- NTSC** – National Television Standard Committee (colour TV system used in USA and most 60 Hz countries).
- OFDM** – orthogonal frequency division multiplex (digital modulation system based on a high number of carriers used for DAB and proposed for terrestrial DVB-T).
- OOB** – out of band (additional medium bit-rate downstream channel used for user interaction and signalling data in the DVB-DAVIC interactive cable standard).
- P** – predictive (picture) MPEG picture coded with reference to the preceding I or P picture.
- PAL** – phase alternating line (colour TV system used in most European and 50 Hz countries).
- PAT** – program allocation table (DVB table indicating the PID of the components of a programme).
- PCM** – pulse code modulation (result of the digitization of an analogue signal).
- PCMCIA** – (now renamed PC-Card) – Personal Computer Memory Card Association (designates the format used for PC extension modules and proposed by DVB for the detachable conditional access modules using the DVB-CI common interface).
- PCR** – program clock reference (information sent at regular intervals in MPEG-2 to synchronize the decoder's clock to the clock of the programme being decoded).
- PES** – packetized elementary stream (MPEG elementary stream after packetization).
- PID** – packet identifier (PES identification number in the DVB standard).

- PMT** – program map table (DVB table indicating the PID of the PAT of all programmes in a transport multiplex).
- PRBS** – pseudo-random binary sequence (used for signal scrambling).
- PSI** – program specific information (MPEG-2 mandatory tables: CAT, PAT, PMT).
- PTS** – presentation time stamp (information indicating the presentation time of a decoded image or sound).
- PU** – presentation unit (decoded picture or audio frame in MPEG).
- Q** – quadrature; for QAM, designates the signal modulating the carrier following the 90° axis.
- QAM** – quadrature amplitude modulation (modulation of two orthogonal derivatives of a carrier by two signals).
- QCIF** – quarter common intermediate format (180 × 144 @ 15 Hz used for videotelephony).
- QEF** – quasi error-free (designates a channel with a BER < 10⁻¹⁰).
- QPSK** – quadrature phase shift keying (phase modulation with four states, equivalent to 4-QAM).
- RISC** – reduced instruction set computer.
- RLC** – run length coding (data compression method exploiting repetitions).
- RS(204, 188, 8)** – abbreviated notation of the Reed-Solomon coding used by DVB.
- RS232** – standardized asynchronous serial communication interface (relatively slow).
- RST** – running status table (optional table of DVB-SI informing on the current transmission).
- SCR** – system clock reference (information sent at regular intervals in MPEG-1 to synchronize the decoder's clock to the system clock).
- SDRAM** – synchronous dynamic random access memory (a new kind of high speed DRAM (16 bits organized) used with recent MPEG decoders).
- SDT** – service description table (optional table of DVB-SI).
- SECAM** – séquentiel couleur à mémoire (colour TV system mainly used in France and in eastern European countries).
- SFN** – single frequency network (terrestrial TV transmitter network using the same frequency on all its coverage area; it uses the 8K COFDM mode of the DVB-T standard and a relatively long guard interval).
- SIF** – source intermediate format (360 × 288 @ 25 Hz or 360 × 240 @ 30 Hz; basis for MPEG-1).
- ST** – stuffing table (optional table of DVB-SI).
- STD** – system target decoder (hypothetical reference decoder used in MPEG standards).

- TDT** – time and date table (optional table of DVB-SI).
TPS – transmission parameter signalling (modulation and channel coding parameters transmitted on pilot carriers of the OFDM multiplex in the DVB-T proposal).
TSR – time shift recording (facility to watch a complete live TV transmission in spite of interruptions thanks to the simultaneous record and playback capability of a hard disk).
USB – universal serial bus (high speed serial bus – up to 11 Mb/s for version 1.1 and 480 Mb/s for version 2 – replacing progressively the RS232 and IEEE1284 in PCs and now digital TV STBs).
VBS – video baseband signal (monochrome composite signal).
VLC – variable length coding (data compression method consisting of coding frequent elements with fewer bits than infrequent ones).
VLIW – very long instruction word (new type of processor using parallelism).
VSF – vestigial sideband (AM with one of the two sidebands truncated); used by all analogue TV standards and proposed for the terrestrial broadcasting of the digital 'Grand Alliance' US HDTV system (8 or 16-VSB).
WSS – wide screen signalling (signalling information on line 23 of the PAL+ signal, also used on standard PAL or SECAM, to indicate the format and other characteristics of the transmission).
ZIF – see zero IF.

Words and expressions

- aliasing** – disturbance caused by spectrum mixing when sampling a signal with a bandwidth exceeding half of the sampling frequency (during an A-to-D conversion, for instance).
asynchronous – in IEEE1394 terminology, designates the mode specified for the transport of non-time-critical data (used, for example, for function control or storage).
baseband – original frequency band of an analogue or digital signal before modulation or after demodulation.
block – in JPEG and MPEG this designates the portion of an 8×8 pixel picture to which the DCT is applied.
Bluetooth – RF communication standard at 2.4 GHz for short distances (in the order of 10 m) and medium bit-rate (max. 1 Mb/s) allowing interconnection of up to seven participants forming a 'piconet'. Different piconets can communicate between each other by means of participants belonging to two piconets.

- burst errors** – multiple errors occurring in a short time with relatively long error-free periods in-between.
channel coding – addition of redundancy to a digital signal before transmission, allowing errors to be corrected at the receiving end (synonym: FEC).
comb filter – filter used in NTSC or PAL with 'teeth' corresponding to the stripes in the chrominance and luminance spectra for optimum separation of chrominance and luminance.
components video – colour video made of three elementary signals (e.g. RVB or YUV).
composite video – coded colour video using one signal only (NTSC, PAL or SECAM).
compression layer – in MPEG, designates the information at the output of the individual encoders (elementary streams, ESs).
constellation – simultaneous display in I/Q coordinates of the points representing all the possible states of a quadrature modulated signal (QAM, QPSK).
convolutional coding – 'inner' part of the channel coding for satellite and terrestrial transmissions, increasing the redundancy by providing two bitstreams from the original one; it corrects mainly random errors due to noise.
digitization – conversion of an analogue value into a (generally binary) number (synonym: analogue-to-digital conversion, ADC).
downlink – communication link from a satellite to earth station(s) or consumer receivers.
echo equalizer – device designed for cancellation or attenuation of the echoes introduced by transmission (cable transmission mainly).
encryption – encoding of information with a key to control its access.
energy dispersal – logic combination of a digital bitstream with a pseudo-random binary sequence (PRBS) to obtain an evenly distributed energy after modulation.
entropy coding – coding principle using variable length words to encode information elements depending on their probability of occurrence (synonym: variable length coding, VLC); the most well known method for VLC is the Huffman algorithm.
Eurocrypt – conditional access system mainly used with the D2-MAC standard.
flash EPROM – non-volatile, electrically erasable and rewritable memory (in blocks).
flicker – disturbing periodic variation of the luminance of a picture when its refresh frequency is too low (below 50 Hz).

frame (audio) – elementary period during which the psychoacoustical coding is performed (corresponds to 12 times 32 PCM samples); its duration varies from 8 to 12 ms depending on the sampling frequency; (DVB-T)–7 series of 68 OFDM symbols; four consecutive frames form a superframe.

granule – in MPEG audio (layer 2), designates a group of three consecutive sub-band samples (corresponds to 96 PCM samples).

interlaced scanning – scanning of a picture in two successive fields, one with odd lines and the other with even lines, in order to reduce by a factor of 2 the bandwidth required for a given resolution and a given refresh rate compared to a progressively scanned picture.

isochronous – in IEEE1394 terminology, designates the ‘quasi-synchronous’ mode used to transport time-critical data (real-time audio and video).

joint_stereo – MPEG audio mode exploiting the redundancy between left and right channels with two submodes (*MS_stereo*: coding of L + R and L – R; *intensity_stereo*: coding of common sub-band coefficients for high bands of L and R).

layer – in MPEG audio, defines the algorithm used for compression (there are three different layers).

layer – in MPEG video, corresponds to the hierarchical decomposition (from sequence to block).

letterbox – broadcast format used to transmit wide screen films (16/9 or more) on a standard 4/3 TV screen, leaving two horizontal black stripes at the top and bottom of the picture.

level – in MPEG-2, defines the spatial resolution of the picture to be coded.

line-locked clock – clock synchronized by a PLL loop to the line frequency of a video signal.

lossless compression – same as reversible coding (see this phrase; opposite: lossy compression).

lossy compression – compression process which discards some imperceptible or hardly perceptible information elements (opposite: lossless or reversible compression).

macroblock – picture area of 16×16 pixels used for motion estimation. A macroblock is made up of six blocks: 4 Y, 1 C_b and 1 C_r.

masking – occultation of the perception of a sound by a more powerful one at a near frequency (frequency masking) and/or time (temporal masking).

Mediahighway – middleware developed and promoted by Canal+ Technologies and used among others by ITV Digital.

motion estimation – determination of a motion vector allowing an area of a picture to be deduced from an area from a previous picture.

multicrypt – one of the conditional access options in DVB, based on a detachable CA module connected via the ‘common interface’ DVB-CI.

node – in IEEE1394 terminology, designates a participant connected to the bus.

OpenTV – middleware developed and promoted by the company OpenTV and used among others by BSkyB.

orthogonal sampling – sampling of a video signal by means of a clock locked to the line frequency in order to obtain samples with fixed positions on a rectangular grid.

orthogonality – property of a digitally modulated multiple carrier system when the spacing between consecutive carriers is equal to the inverse of the period of the modulating signal, so that the spectrum of any carrier presents zeroes for the central value of the neighbouring carriers (OFDM modulation).

padding – non-significant bits added to adjust the duration of an audio frame (padding bits), or non-significant stream added to adjust the bit-rate of a bitstream (padding stream).

payload – for MPEG-2 transport packets (188 bytes), this designates the ‘useful’ 184 bytes following the header.

peritel – 21 pin audio/video connector (also known as SCART plug or EUROCONNECTOR) used to interconnect audiovisual equipment (TV, VCR, set-top box, etc.).

pixel (or pel) abbreviation of ‘picture element’ – the smallest element of an imaging or display device. In digital TV, it corresponds to the visual representation of one sample of a digitized picture.

profile – in MPEG-2, defines the toolbox used for video encoding.

progressive scanning – scanning of all the lines of a picture in numerical order in only one frame containing all picture lines (type of scanning used for computer monitors).

psycho-acoustic model – mathematical model of the behaviour of the human auditive system, based, among other things, on the frequency and temporal masking effects.

puncture – operation consisting of taking only a part of the bits generated by the convolutional coding to reduce its redundancy, at the expense of a reduced robustness (used in DVB-S and DVB-T).

quantization – measurement of a quantity with a limited number of discrete values (distant from the quantization step), for instance in an analogue-to-digital conversion or a compression process.

- quantization noise** – noise introduced by an analogue-to-digital conversion process, mainly due to the uncertainty on the least significant bit (representing the quantization step).
- Reed–Solomon coding** – outer part of the DVB channel coding, which adds 16 parity bytes to the 188 byte packets and allows correction of up to 8 bytes per packet; it is denoted RS(204, 188, 8).
- reversible coding** – coding allowing the recovery of the exact original information by applying the reverse process (synonym: lossless coding; opposite: lossy coding).
- roll-off factor** – characteristic of the steepness of the filtering applied to a digital signal in order to limit its bandwidth, generally with a view to modulation.
- sampling** – periodic acquisition of the value of an analogue signal, generally with a view to converting it into a digital number.
- scalable profile** – MPEG-2 profile allowing different levels of quality to be obtained from the same bitstream (in terms of resolution for *spatially scalable profiles* or signal-to-noise ratio for *SNR scalable profiles*).
- scaling factor** – in MPEG audio, a 6-bit multiplying factor applied to each sub-band coefficient for the duration of a frame (giving a 128 dB range with 64 values in steps of 2 dB).
- sequence** – in MPEG, an uninterrupted series of GOP defined with the same basic parameters.
- set-top box** – popular denomination of an integrated receiver decoder (IRD).
- simulcast** – simultaneous transmission of a programme in two or more standards (e.g. PAL and DVB), generally during a transition period between these standards.
- simulcrypt** – principle consisting of sending ECM and EMM for more than one conditional access system for one programme, in order to allow reception by different types of decoder.
- slice** – in MPEG, a slice is a portion of the picture made up of horizontally consecutive macroblocks (most of the time a complete row). It is used for intra-frame addressing and resynchronization.
- source coding** – ensemble of coding operations aiming to reduce the quantity of information delivered by a source (synonym: compression).
- spectral efficiency** – ratio (in bits/s per Hz) between the bit-rate of a bitstream and the bandwidth occupied by the RF signal modulated by this bitstream.
- square pixels** – name given to the pixels obtained when sampling results in an equivalent resolution along the two axes of the picture (e.g. 640 × 480 for a 4/3 picture).

- sub-band sample** – in MPEG audio, output signal of one of the 32 sub-band filters (duration: 32 PCM samples, corresponding to 1 ms at 32 kHz sampling rate).
- symbol** – in a digital transmission, this is the modulating information element. The number of bits/symbol depends on the modulation type (e.g. 2 bits/symbol for QPSK, 6 for 64-QAM).
- symbol rate** – number of symbols transmitted per second.
- system layer** – in MPEG-1, designates the form of the packetized information after multiplexing.
- table** (MPEG-2 PSI and DVB-SI) – descriptive information necessary to access DVB transmissions or making this access easier.
- thresholding** – elimination of values below a given threshold (used in video compression to reduce the amount of information to be transmitted).
- transponder** – electronic device in a communication satellite receiving information from an earth station and re-sending it after frequency conversion and amplification to earth station(s) or consumers; there is generally one transponder per RF channel.
- transport packet** – packet of 188 bytes (4 header bytes + 184 payload bytes) forming the elementary blocks of an MPEG-2 transport bitstream to which the error correction is applied.
- uplink** – communication link from an earth station to a satellite, supplying the information to be rebroadcast by one or more transponder(s).
- zero IF** – (or direct conversion) – designates the concept of recent satellite tuners able to recover I and Q baseband signals by coherent demodulation directly at the incoming frequency, resulting in a 'zero IF' instead of the 480 MHz IF of the conventional 'superheterodyne' concept.

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- ISO 13818-1, 13818-2, 13818-3 (ISO/IEC) Coding of moving pictures and associated audio (MPEG-2 system, video, audio).
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Some useful Internet addresses

- | | |
|---------------|---|
| ASTRA: | http://www.astra.lu |
| ATSC: | http://www.atsc.org |
| Betaresearch: | http://www.betaresearch.de |
| Bluetooth: | http://www.bluetooth.org |
| Cable labs: | http://www.cablelabs.org |
| Canal Plus: | http://www.canalplustechnologies.com |

DVB:	http://www.dvb.org
EBU (UER):	http://www.ebu.ch
ETSI:	http://www.etsi.fr
Eurocablelabs:	http://www.eurocablelabs.com
EUTELSAT:	http://www.eutelsat.org
IEC (CEI):	http://www.iec.org
IEEE:	http://www.ieee.org
ISO:	http://www.iso.ch
Liberate:	http://www.liberate.com
MHP:	http://www.DVB-MHP.org
Open Cable:	http://www.opencable.com
Open TV:	http://www.opentv.com
SCTE:	http://www.scte.org
Tcomlabs:	http://www.tcomlabs.com
TV Anytime:	http://www.tv-anytime.org

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