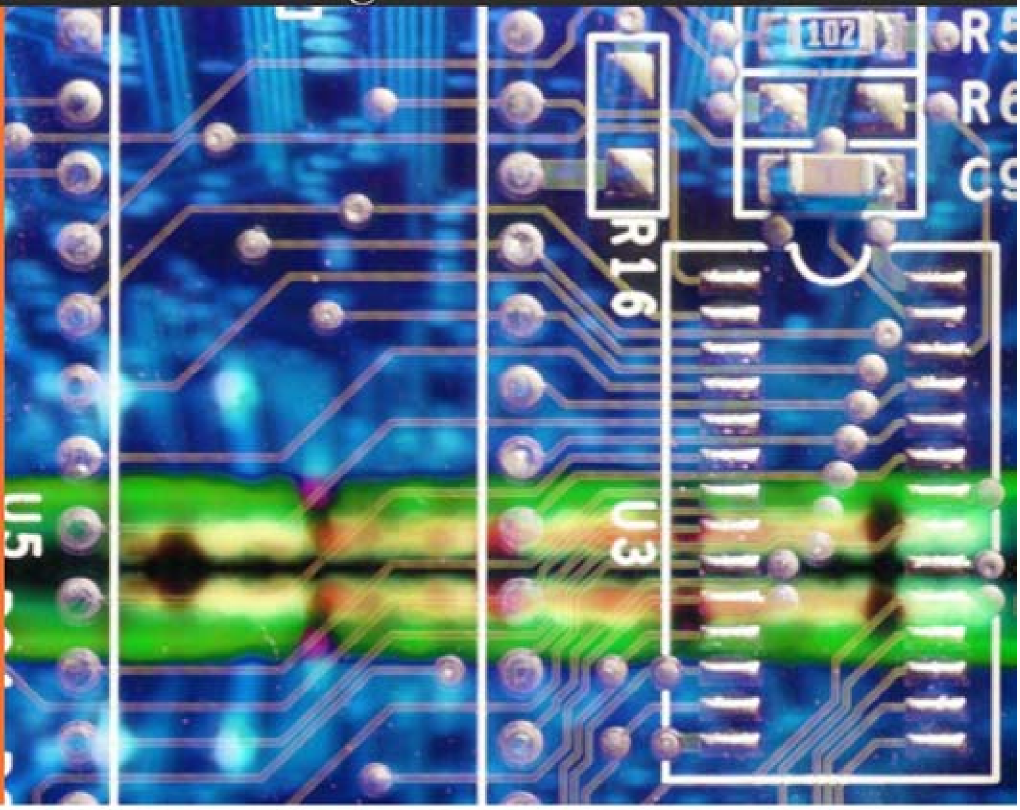
ADVANCED BROADCASTING ELECTRONICS

Broadcast Engineer's Handbooks

DIGITAL TECHNOLOGY



Digital TV broadcasting handbook

Broadcast Engineer's Handbooks

by abe

Digital TV broadcasting handbook

Digital TV Broadcasting Handbook
Edition 1-2004
Author: ***Roberto Valentin***
Author thanks ABE R&D team for support.

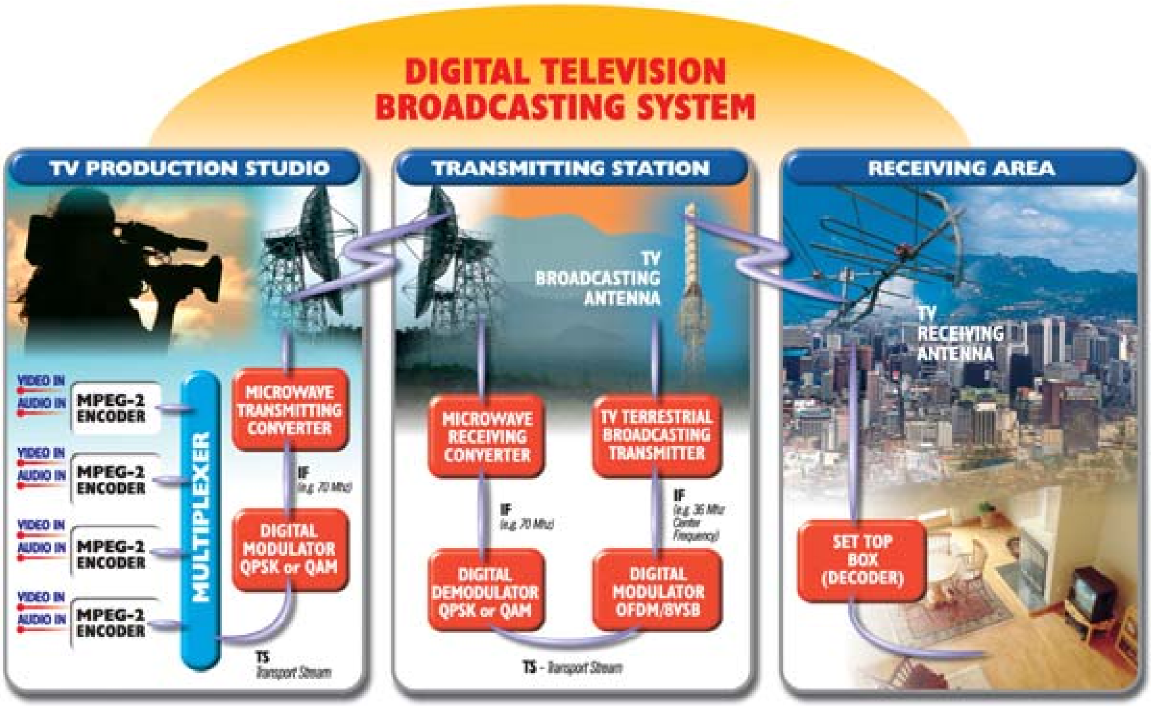
Digital TV Broadcasting Handbook

A new era in TV Broadcasting has dawned with the introduction of Digital Television Broadcasting.
Whether via Satellite and Cable or by Terrestrial Transmission to existing antennas, Digital TV is revolutionizing TV transmission.
Definitive standards have been carefully considered, debated and optimized through field trials and the various delivery platforms are now the subject of recognized ITU international standards.

The advantages of digital TV transmission, in comparison with analog, considering both microwave links and actual broadcasting, are notable and evident from the following:

- more TV programs may be transmitted in a given RF spectrum (typically at least four times as many, that is, in a single RF channel it is possible to broadcast 4 or more digital TV programs instead of a single analog one)
- lower transmission power will cover the same distance (that is, greater immunity to noise and interference)
- better picture/sound quality
- possibility of Isofrequency Terrestrial Broadcasting Networks, that is to have more transmitters in operation, broadcasting the same program, on the same frequency, covering adjacent areas. Practically, it is possible to use the same channel on large areas using more transmitters without having interference problems among them (OFDM emission with SFN – Single Frequency Network)
- possibility of mobile reception without having the typical problems of analog systems, that is double images, reflections, distortions, etc. (OFDM modulation)
- possibility for simultaneous transmission of auxiliary data

Example: Structure of a Digital TV Broadcasting multiprogram system



TV Production Studio:

The TV production studio generates more audio/video programs (in the example are 4) that are digitally codified according to the MPEG-2 standard and multiplexed (that is aggregated to make a single digital data stream called a Transport Stream).
The Transport Stream digitally modulates an IF (Intermediate Frequency) carrier (usually at 70MHz), according to the QPSK or QAM modulation scheme. The IF carrier is converted into microwave frequencies and transmitted to the broadcasting station directly (terrestrial microwave link) or through a satellite or terrestrial transponder.

Transmitting station:

The microwave received signal is converted at IF (Intermediate Frequency – 70MHz) and digitally demodulated so to get the Transport Stream that contains the four programs. The demodulator can eventually also decode the four programs, so as to have them separately available both in analog and/or in digital format, for other purposes (i.e., to be broadcasted with analog TV transmitters). The Transport Stream, at this point, digitally modulates an IF carrier (usually at 36 or 44MHz) according to the digital terrestrial broadcasting standard OFDM (DVB-T) or 8VSB (U.S. ATSC). The IF carrier is then converted to the VHF or UHF band, amplified and radiated through the broadcasting antenna, to be available in the receiving area.

Receiving area:

The digital broadcast signal is received through the viewer's antenna and feeds to a proper receiver / decoder (usually called set-top-box or IRD) connected to the TV set (functioning as a video / audio monitor).

MPEG-2 encoding and most Common profiles: 4:2:0 (MP@ML) and 4:2:2

Digital TV transmission needs digital audio and video signals. These signals may be originated digitally (with all-digital cameras and studio mixers) or, more often, they may be produced by digitally encoding available analog signals.

Uncompressed digital video and audio signals have a high data rate - typically one program requires a Bit Rate of 270Mbit/s. Normally, the Serial Digital Interface (SDI) is used for this type of signal (with 75 ohm BNC coaxial connectors).

If uncompressed data were to be transmitted as is, the occupied RF bandwidth would be much greater than in the analog case. It is necessary, therefore, to compress such data to a lower rate, making it suitable for transmission over microwave links and for distribution or broadcasting to viewers.

Ideally, data compression will not degrade the quality of the video or audio signals. The designated international coding standard for this purpose is MPEG-2 (Motion Picture Expert Group version 2) which is able to compress a TV program from 270Mbit/s to only 5 or 6Mbit/sec while maintaining excellent quality characteristics. (Compression to less than 4Mbit/s is possible but quality will be compromised.)

The following compression techniques are used to encode TV pictures:

- Human visual perception is more sensitive to luminance than chrominance. Less information (data) about the color is therefore transmitted.
- Adjacent areas within the picture often have pixels with the same luminance and chrominance values. During encoding these are combined so as to transmit less data.
- Only the differences between one picture frame and the next are transmitted. This process is carried out several times over a Group Of Pictures (GOP) before eventually transmitting a complete frame again.

So GOPs – Groups of Pictures – are made up from three different kinds of information frames:

- I-frame: the complete image or picture frame (the largest in terms of the data transmitted)
- P-frame: the differences between an actual and the previous I or P-frame (smaller than an I-frame)

- B-frame: the differences between the previous and the following I or P frames (the smallest frame, but which cannot be repeated too many times).

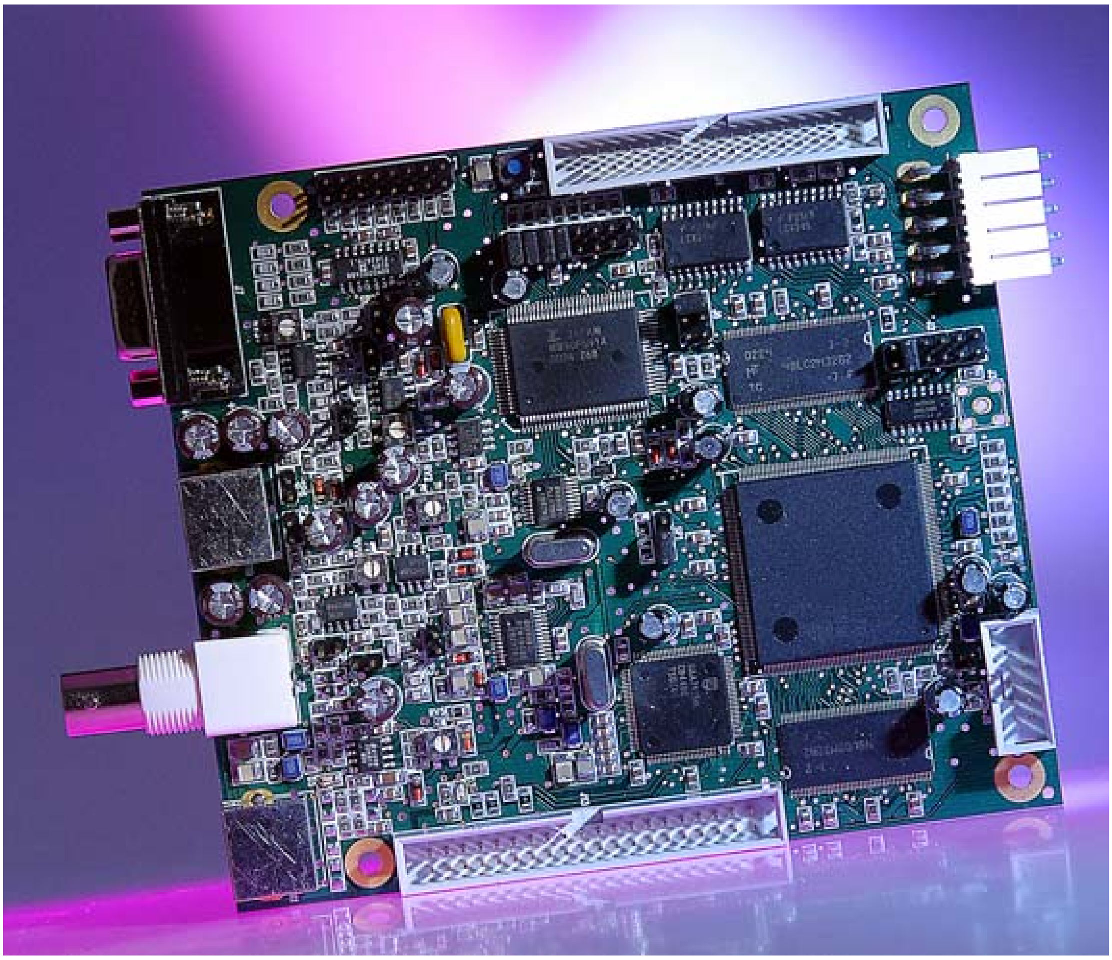


Figure 1 - MPEG-2 encoder board (ABE production)

Usually GOPs are constituted with one I-frame, some P-frames and, possibly, some B-frames. They should not be too long, because should an error occur, it would be perpetuated.
Furthermore, a decoder requires a complete picture (I-frame) to begin decoding, so has to wait for the start of a GOP.
One of the most usual and efficient GOP structures is 12 frames long and is constituted as follows: IBBPBBPBBPBB.
The most common encoding data profiles are 4:2:0 (Main Profile @ Main Level or MP@ML) and 4:2:2. We list below the properties, advantages and uses of each:

4:2:0 – The video is encoded with a ratio of 4 data elements for luminance to 2 for chrominance.

ADVANTAGES:

- This encoding ratio matches the visual perception characteristic
- Optimum performance, particularly for low Bit Rate transmission

USES:

- Broadcasting (the profile used in both terrestrial and satellite broadcasting)
- Contribution and Distribution networks
- Intra-studio links between analog and digital mixers

4:2:2 – The Video is encoded with a ratio of 4 data elements for luminance to 4 for chrominance

ADVANTAGES:

- Slightly better performance than 4:2:0 profile, but only when the Bit Rate is over 10MBit/s

USES:

- Intra-studio links between digital mixers

The above summary is supported in the respected EBU Technical Review, its autumn-1999 issue, reporting on a series of tests carried out by the Swedish Television Authority (SVT).
Comparative tests, according to ITU recommendations, were based on subjective evaluation by a group of observers, viewing digital TV pictures after encoding at 2, 3, 4 or 5Mbit/s. The results established that, for the each Bit Rate, 4:2:0 was preferable to 4:2:2 encoding.
Also, Mr. Al Kovalick (Technical Director of Pinnacle Systems Inc.) says in an article published by BroadcastPapers (©2001-2002), that a video sequence encoded with 4:2:0 at 10Mbit/s has the same quality when encoded using 4:2:2 profile, but at 13Mbit/s!
During our tests (at ABE Elettronica), we found some advantages using 4:2:2 encoding, but only at a Bit Rate over 10Mbit/s. It is preferable to use the 4:2:0 profile for Bit Rates under 10Mbit/s, especially for picture sequences with high motion content.
Please note that it is extremely difficult to detect quality differences on picture sequences encoded at over 10Mbit/s, since the quality is already so high that differences are very difficult to perceive.
The limited advantages of the 4:2:2 profile compared with 4:2:0, with Bit Rates over 10Mbit/s disappear if the source signals are analog in origin and converted to digital.
So, considering that it is now unusual (and expensive) to use Bit Rates of 15-20Mbit/s just for a single program, the encoding profile used is nearly always 4:2:0 (MP@ML).

We would like to mention some of the most common settings for an MPEG-2 encoder (4:2:0 profile):

- Video resolution: Full D1; 3/4 D1; 2/3 D1; 1/2 D1; SIF; QSIF (One has to choose the most appropriate setting, keeping in mind the available Bit Rate and required encoding quality according to the content).
- Resolution of displayed pictures: 720 x 576 pixels, max for PAL, and 720 x 480 pixels, max for NTSC. Higher resolution can produce better definition, but at the expense of a higher Bit Rate.
- Group of Pictures (GOP) structure: the number and sequence of encoded I, P, B frames.
- Encoding Bit Rate: up to 15MBit/s.
- Output Transport Stream Bit Rate: has to be equal to or higher than the total from the video and audio encoding, plus the data-tables. The difference between the real encoding Bit Rate and the output Transport Stream Bit Rate is made up by filling with null packets (bit-stuffing).
- Audio sampling frequency (32 or 44.1 or 48 kHz) and encoding Bit Rate: the higher the sampling frequency, the better the transmission quality, but the higher the necessary Bit Rate.
- Video, Audio and, possibly, Data PIDs (Program Identifiers): these have to be set avoiding duplication so as not to be in conflict with other PIDs with which they may be multiplexed.
- Filter settings: in the case of encoders with composite video input is possible to choose ‘comb’ or ‘notch’ filters to separate chrominance and luminance. Other kinds of filter are useful to reduce noise (for example, in the case of low Bit Rate to avoid transmitting noise instead of real pictures).

The above settings are only some of those available on a typical encoder. In practice, for example, on the MPEG-2 (4:2:0) encoder produced by ABE Elettronica, it is possible to modify some 200 parameters, although naturally many of these are inter-dependent. To help the user, the ABE encoder has four different factory set-up configurations and four others which are user configurable.

Finally, some considerations about testing MPEG-2 encoders:

- Encoding Quality, as such, is not easily measurable. The usual method of assessment is to make comparisons of picture sequences with subjective evaluation and/or expert viewing, rather than using the few tests and measuring sets available which may not match human quality perception.
- Supporting the chosen method of quality evaluation (but not replacing it), it is possible also to measure linear and non-linear video distortions and

noise. Keep in mind, however, that many of the results will depend on the encoder settings as well as on its ‘quality’ and detailed algorithms. Moreover, the effective luminance pass-band could be less than 3 MHz, depending on the filters used.

Transport Stream, interfaces (ASI/SPI) and multiplexing

In the Transport Stream (data stream containing video / audio / data program(s) to be carried from the generating/broadcasting equipment to the users/viewers), data have a constant bit rate and are organized in a continued sequence of "packets." These packets have fixed length of 188 bytes (204 bytes if data for Reed Solomon correction algorithm are present).

To maintain the bit rate of the Transport Stream constant, also when there are no data packets to be sent, valid packets with null content are generated and inserted (this procedure is called "Bit Stuffing"). These "null packets" will be recognized and eliminated during processing.

Each packet is composed of a header (that has a standard dimension of 4 bytes, except for particular cases), which includes a sync byte, the PID (Program Identifier – a number that identifies the video / audio / data program to which the packet is referred) and other information, followed by the "payload": the data of the real program to be "transported."

Commonly used Transport Stream interfaces are:

Synchronous Parallel Interface SPI
This interface is made by 11 contemporaneous signals: 8 data signals (Parallel Data Path), 1 clock signal, 1 synchronism signal (Psync) and 1 signal which identifies when valid data are transmitted (Dvalid).
The Bit Rate is variable (Max 108 Mbit/s on Data Path) and the standard connector for this interface is 25 pins.
Electrical levels may be LVDS (Low Voltage Differential Signal) for external, short connections, between different pieces of equipment or may be LVTTL (Low Voltage TTL) for short connections among the same equipment.

Asynchronous Serial Interface (ASI)
This is the most commonly used interface, which has a constant bit rate at 270 Mbit/s working on a single unbalanced coaxial line (75 Ohm impedance).
Its standard connector is BNC.
The difference between available Transport Stream Bit Rate and 270 Mbit/s is filled by stuffing bytes, which will be discarded during the deserialization process.
This interface is used for connections between different pieces of equipment, even when separated by long distances.

The **Multiplexer** is a device that aggregates several Transport Streams (coming, for example, from different encoders) into a single Transport Stream, which includes all the streams.

In addition, the Multiplexer (Re-Multiplexing function) can modify Transport Streams, adding data and tables (for example NIT, Network Information Table, into which it is possible to edit transmitted program's names that will appear to the user).

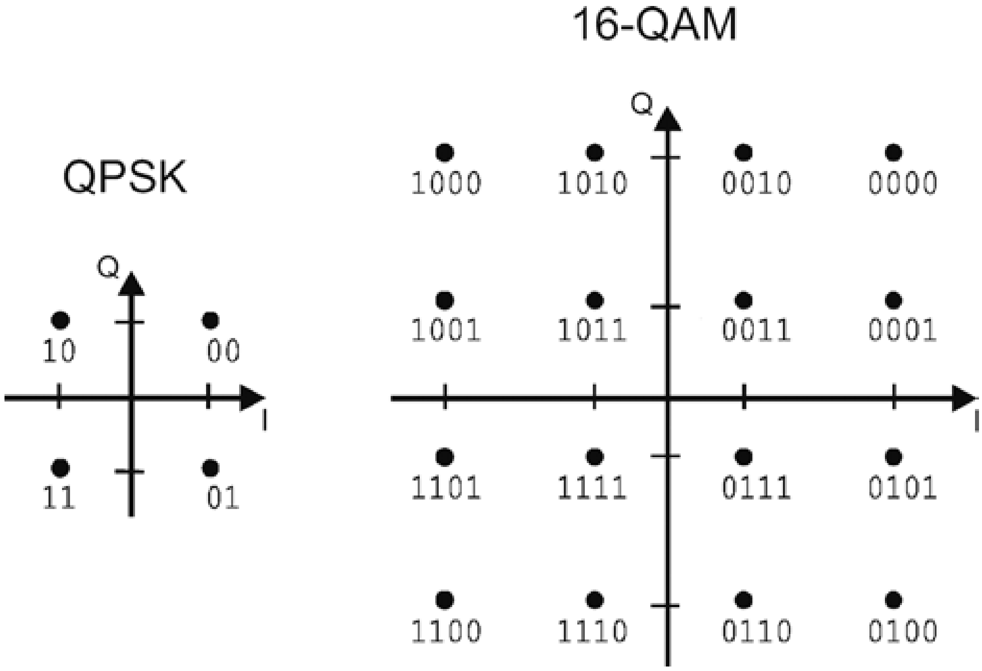
Some Multiplexer considerations, settings and tests:

- Multiplexer output Transport Stream Bit Rate must be set to be equal or greater to the sum of the input Transport Streams Bit Rate + data + tables.
- Data/tables to be inserted and/or modified may be several (NIT data, EIT Event Information Table that describe programs transmitted, etc.). ABE's multiplexers can add TELETEXT that, since it is not part of the video active lines, cannot be encoded by MPEG-2 encoders.
- If serious errors are present in the Transport Stream decoders will not work or will generate errors.
- To function, some decoders require data or tables (for example the NIT) in the Transport Stream, while for other decoders, these data aren't essential.
- For a correct and complete analysis of the Transport Stream there are dedicated instruments able to indicate errors or nonconformities (Transport Stream errors are classified with three priority levels - see ETSI technical report TR 101 290, ex ETR 290).

Digital modulations

The most used types of digital modulation are: **QPSK** (Quadrature Phase Shift Keying): It's a phase modulation, employed in the DVB-S standard for terrestrial microwave links and satellite contribution / distribution / broadcasting.
QAM (Quadrature Amplitude Modulation): It's a phase and amplitude modulation, used in DVB-C standard, for terrestrial microwave links, MMDS and CATV (cable television).
OFDM (Orthogonal Frequency Division Multiplexing – also called COFDM, that means Codified Orthogonal Frequency Division Multiplexing): It's composed by several carriers (2K=1705 carriers; 8K=6817 carriers), equally spaced in frequency, each one modulated QPSK or QAM. It's employed in the DVB-T standard for terrestrial broadcasting and mobile/ENG microwave links.
8VSB (8 Vestigial Side Band): It's an amplitude modulation with 8 amplitude levels and lateral vestigial side band partially cancelled.
It's used in the ATSC U.S. standard for terrestrial broadcasting.

Constellation examples of digital modulations



Representation of the constellation of possible positions of the carrier in the phase (angle) / amplitude (distance from the center), for QPSK and 16QAM modulation schemes. The numbers near each point of possible positioning of the carrier, indicate the transmitted Transport Stream bit sequence when the carrier has that position.

In all digital modulation schemes the carrier continuously moves on various predefined positions of phase and/or amplitude (called Symbols). Each position represents a bit sequence of the transmitted Transport Stream. The most used diagram that shows these phase/amplitude positions is the "constellation diagram."

- Each phase (and eventually amplitude) position of the carrier corresponds to 2 bit (QPSK modulation), 4 bit (16 QAM modulation) or 6 bit (64 QAM modulation).
- In the 8VSB modulation scheme, each amplitude position of the carrier corresponds to 3 bits.
- So, according to the modulation scheme employed, the Transport Stream's data are transmitted in sequences of 2, 3, 4 or 6 bits.
- The number of different positions of phase and/or amplitude that the carrier can have in the constellation diagram in one second is called the Symbol Rate.

The frequency of the data (Bit Rate) of the input stream (Transport Stream) of the digital modulators, depends on the modulation scheme employed (QPSK, 8VSB, 16QAM, 64QAM), on the Symbol Rate settings and on the quantity of data added by the modulator itself in order to correct the possible errors in the receiver (FEC - Forward Error Correction), that is, the Code Rate employed.

Please note that ABE modulators have an automatic Transport Stream adaptation function which can add null packets (Bit Stuffing) or, in some circumstances, remove them. In this way the usable Transport Stream Bit Rate adapts to that required by the Modulator Symbol and Code Rate settings.

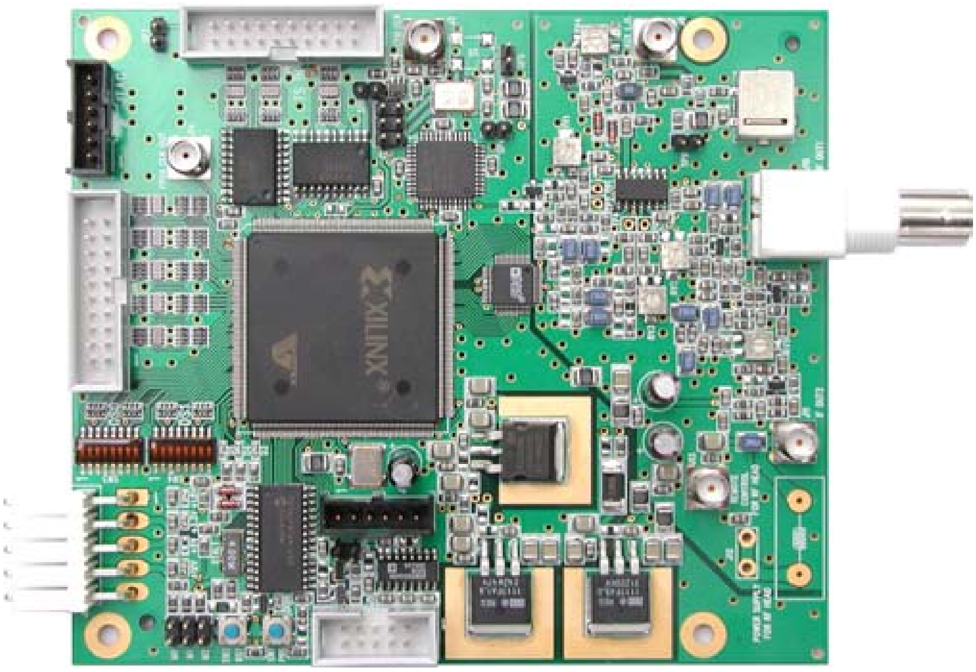


Figure 2 QPSK (DVB-S) and QAM (DVB-C) Modulator board - (ABE production)

Forward Error Correction (FEC) and choosing criteria

When data are transmitted to many users at the same time, it is essential to include, together with the programs' data, other data which, when reception is

disturbed, are able to correct errors that occur in the programs’ data (of course, up to a certain limit).
All TV broadcasting digital modulation standards use the Forward Error Correction encoding system “Reed Solomon” (RS).
This algorithm usually adds 16 data byte to each Transport Stream packet (188 bytes), for a total of 204 bytes.
Using this system it is possible to correct up to 8 non consecutive errors in each packet; it is practically guaranteed the error correction of a Bit Error Rate (BER) of 2 * 10 –4.
Some standards (i.e., DVB-S and DVB-T) consider RS algorithm not sufficient, therefore they add (in addition to RS) another correction system, much more powerful, called “Inner Code.”
Inner Code adds further correction data: for example it adds 1 correction bit every 7 data bits (Code Rate 7/8), or 1 correction bit every 2 data bits (Code Rate 2/3), and so on.
Low Code Rates (e.g., 7/8) allow lower possibilities of Forward Error Correction (the possibility of the receiver to correct eventual errors in the data), while larger Code Rates (e.g., 1/2) allow greater possibilities of FEC.

The following example shows the calculation of the input Transport Stream Bit Rate of a QPSK (DVB-S) modulator:

$$\text{Modulator Input usable Bit Rate} = \text{Symbol Rate} \underbrace{\times 2}_{\substack{\text{for QPSK}\\ \text{modulation}\\ \text{(2 bit per Symbol)}}} : \overbrace{8 \times 7}^{\text{example for code rate 7/8}} : \underbrace{204 \times 188}_{\substack{\text{for REED SOLOMON}\\ \text{encoding}}}$$

the difference is in the signal to noise/disturbances ratio: for example, with a 1/2 Code Rate the signal margin is around 4dB higher than using a 7/8 Code Rate, but the useful data bit rate is lower.

RF occupied bandwidth

A digitally modulated carrier Radio Frequency (RF) occupied bandwidth, essentially depends on two parameters: the transmitted Symbol Rate and the filtering (Roll-off factor / Shaping).
With QPSK and QAM modulation schemes, the occupied bandwidth (in MHz) is equal to the Symbol Rate (in MS/s) added by the Roll-off factor (%).
For example, using the QAM modulation scheme (DVB-C standard - Roll-off 15%), in order to transmit 6 MS/s, the occupied bandwidth will be 6MHz + 15% = 6.9 MHz.

Digital TV microwave links:

Specifications, advantages, differences and upgrading from analog operation

Digital microwave links have important advantages with respect to the analog ones; moreover, most ABE Microwave Links produced in recent years may easily be upgraded from analog operation to digital using QPSK modulation.
In this section we analyze the relevant design considerations.

IF Modulator

The 70MHz analog IF Modulator must be replaced by a suitable digital QPSK Modulator having the same 70MHz IF and compatible output level and impedances (e.g., the ABE DME 1000 that is also equipped with one or more MPEG-2 encoders and multiplexer).

IF/RF Bandwidth - Information Capacity

In relation to the standard analog Link emission mask, the digital Modulator may be set to produce a symbol rate up to approximately 15MS/s.
Using 15MS/s and a 7/8 Code Rate, the input transport stream Bit Rate to the QPSK Modulator is 24Mb/s; enough to accommodate four Video/Dual-Audio programs with excellent broadcast quality. This averages out at 6Mb/s per program.
It is possible to use higher Symbol Rates which are acceptable to the Link, but the occupied bandwidth will be greater. Using lower Symbol rates the occupied bandwidth will be lower (e.g., 7MHz); of course, in this case, the number of programs will be reduced.
As an alternative, using 16 QAM modulation scheme, it's possible to transmit in a 7 MHz bandwidth up to 22Mb/s.
16 QAM modulation is more "delicate" than QPSK and needs more linearity in the microwave link's conversion and amplification stages, a better local oscillator's phase noise, better signal to noise ratio and lower linear distortions in the path (amplitude/frequency, group delay, multi-path/selective fading).
So, when choosing, if there are no particular restrictions about occupied bandwidth, QPSK modulation scheme is preferable.

Phase / Amplitude Linearity

Analog Microwave Links normally employ saturated amplifiers and limiters. Digital modulation (and especially the QAM) requires linear amplification - without saturation or limiting. QPSK, among digital modulation schemes, is one of the least sensitive to linearity problems.

To achieve adequate linearity for QPSK digital modulation, it is best to work at 3dB less than the saturated RF power of the amplifiers - often referred to as 3dB back-off from the rated analog power - and to employ automatic gain control (AGC) circuits instead of limiters.
Nevertheless, when standard analog Links using saturated amplifiers and limiters are tested with QPSK modulation, the result is, generally, acceptable, even though the emission 'shoulders' are degraded, indicating higher intermodulation just outside the channel (this is the consequence of the amplification non linearity).

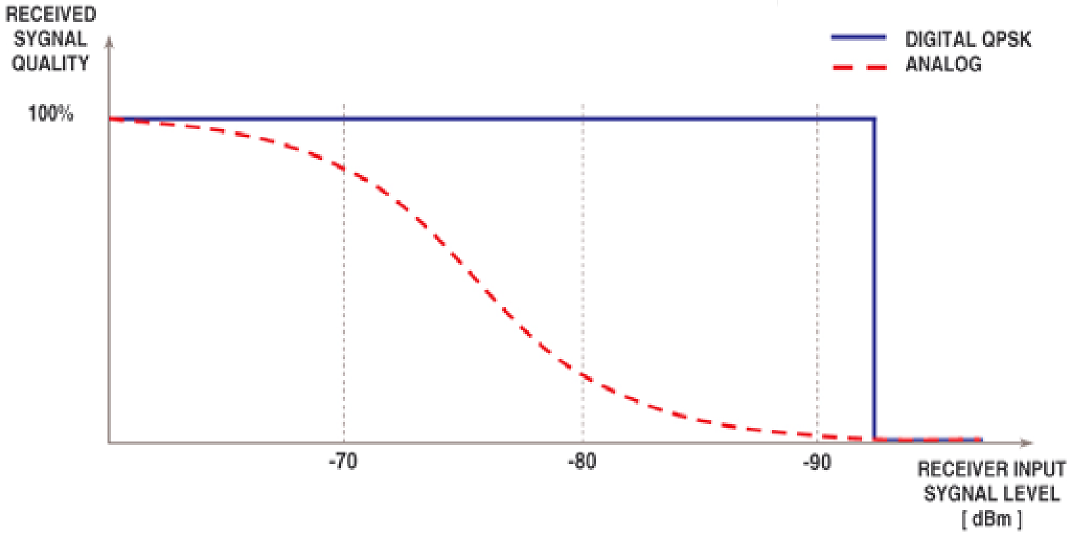
Local Oscillator Phase Noise - Frequency Stability

Generally speaking, the standard local oscillators employed in ABE analog Links have adequate phase noise and frequency stability limits to work with the digital QPSK modulation scheme.

IF Demodulator

The IF 70MHz analog Demodulator must be replaced by a suitable QPSK digital decoder, usually called an IRD (Integrated Receiver Decoder).
Digital receivers must have adaptive equalization systems, or, in any case, must have systems for correcting the problems created by the "selective fading" (dispersive/multipath fading) which, in contrast with the "flat fading," creates "holes" (notches) into the RF bandwidth that usually don't generate problems in analog links, but may make the digital received signal impossible to demodulate, even if the received level is sufficient or good.

Analog/digital to microwave link comparison



Link Performance Comparison

With an analog FM TV Link, as the level of received signal reduces below a certain point, the output quality deteriorates progressively, particularly the noise level. In practice, the minimum acceptable signal in fading conditions is about -70dBm. In contrast (see diagram), the output quality from a digital Link is not degraded and, moreover, remains constant as the received input signal level is reduced, right down to a threshold generally around -90dBm, below which the Link output is lost. The exact threshold level depends on various factors/settings: Code Rate, Symbol Rate, receiver noise figure etc.

Summarizing, digital operation confers many notable advantages:

- A single Link may be used to carry 4 (or more) video/audio program channels.
- Link performance does not degrade progressively as the receiver input reduces but remains constant down to a very low threshold.
- In practice - **and really remarkable** - the receiver input range is increased by around 20dB, allowing the Link to maintain high performance with increased fade-margin, or to use lower RF power or smaller antenna sizes to cover the same distance.

Digital TV broadcasting terrestrial transmitters:

Specifications, advantages, measurements, differences and upgrading from analog operation

ABE Elettronica has produced and supplied, during 25 years, thousands of TV transmitters to Broadcasting Stations worldwide.

Nowadays, even when ABE transmitters are not supplied as “digital,” thanks to all improvements introduced with the development of digital technologies, they are “digital ready,” meaning that they can be easily converted into digital transmitters with minimum economic and technical impact.

Here are main differences and possible modifications with respect to analog transmitters.

IF Modulator

The 36 or 44 MHz (center frequency of the bandwidth) IF analog Modulator has to be replaced by a suitable digital OFDM or 8VSB Modulator having the same IF and compatible output level and impedances (e.g., the ABE DME 1000 that is also equipped with one or more MPEG-2 encoders and multiplexer).

IF/RF Bandwidth – information capacity – considerations about settings – the guard interval

The occupied bandwidth (channel) is exactly the same of the analog transmitters: 6, 7 or 8 MHz.

The input Transport Stream Bit Rate with 8VSB transmitters is fixed at 19.28 Mb/s or depends on bandwidth setting (6,7 or 8 MHz), modulation scheme (QPSK, 16 or 64 QAM), Code Rate setting (from 1/2 to 7/8) and guard interval setting with OFDM transmitters (DVB-T standard) and may vary from around 4 to nearly 32 Mb/s.

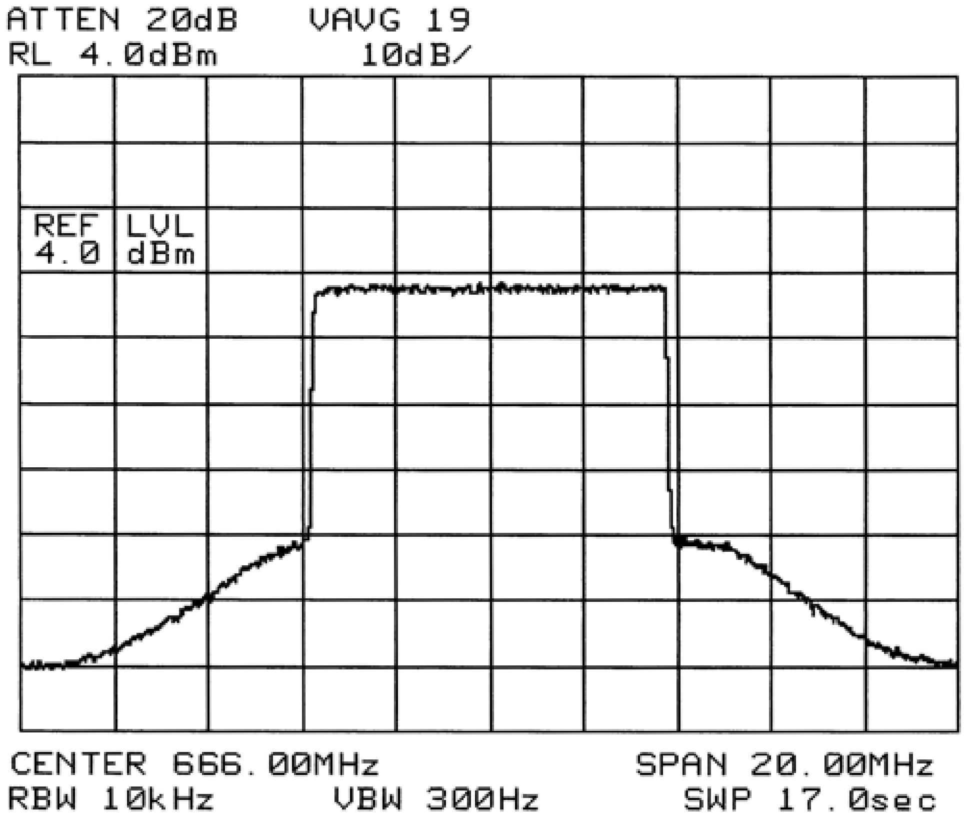
Since in a standard application it uses an input Bit Rate to the modulator of 19 to 24Mb/s, it is possible to transmit 4 video programs, each with double audio, with excellent broadcast quality, in a single TV channel (around 5 – 6 Mb/s per TV program).

Obviously the more complex modulation schemes (64 QAM) and the higher Code Rates (7/8) allow to transmit more data (that means more programs - higher bit rate), the transmission is more “delicate;” that is, it needs more linearity in the conversion and amplification stages of the transmitter, better phase noise in the local oscillators, better signal to noise ratio in the receivers and lower distortions in the connection (amplitude/frequency, group delay, multi-path/selective fading, etc.).

So, when choosing, if there is no particular need to transmit many programs, it is preferable to use the QPSK modulation scheme with low Code Rates, which are particularly suitable also for mobile reception.

The **guard interval** (available only for OFDM transmission) is the time interval during which the transmitter does not emit any essential signal after the emission of each symbol, to allow echoes (reflections of the transmitted signal, or other isofrequency emitted signals of the same network which arrive to the receiver with a certain delay) to extinguish themselves before transmitting the next symbol. This way, receivers will not be disturbed by possible symbols overlapping, which may make the received signal impossible to demodulate, even if the received level is sufficient or good.
Obviously, the longer the guard interval time, the greater the time allowed to extinguish echoes, but the lower the quantity of the data that can be transmitted (Bit Rate - programs number and/or quality).
The guard interval may be set from few microseconds to over 200 microseconds, in order to tolerate reflections/signals with different paths from a few Kilometres to around 70 Km.
When choosing 2K IFFT (OFDM modulation with 1705 carriers), since Symbol Rate is higher than the one with 8K IFFT (OFDM modulation with 6817 carriers), the possible guard intervals are shorter (because they are a fraction of the symbol time: 1/4, 1/8, 1/16, 1/32).

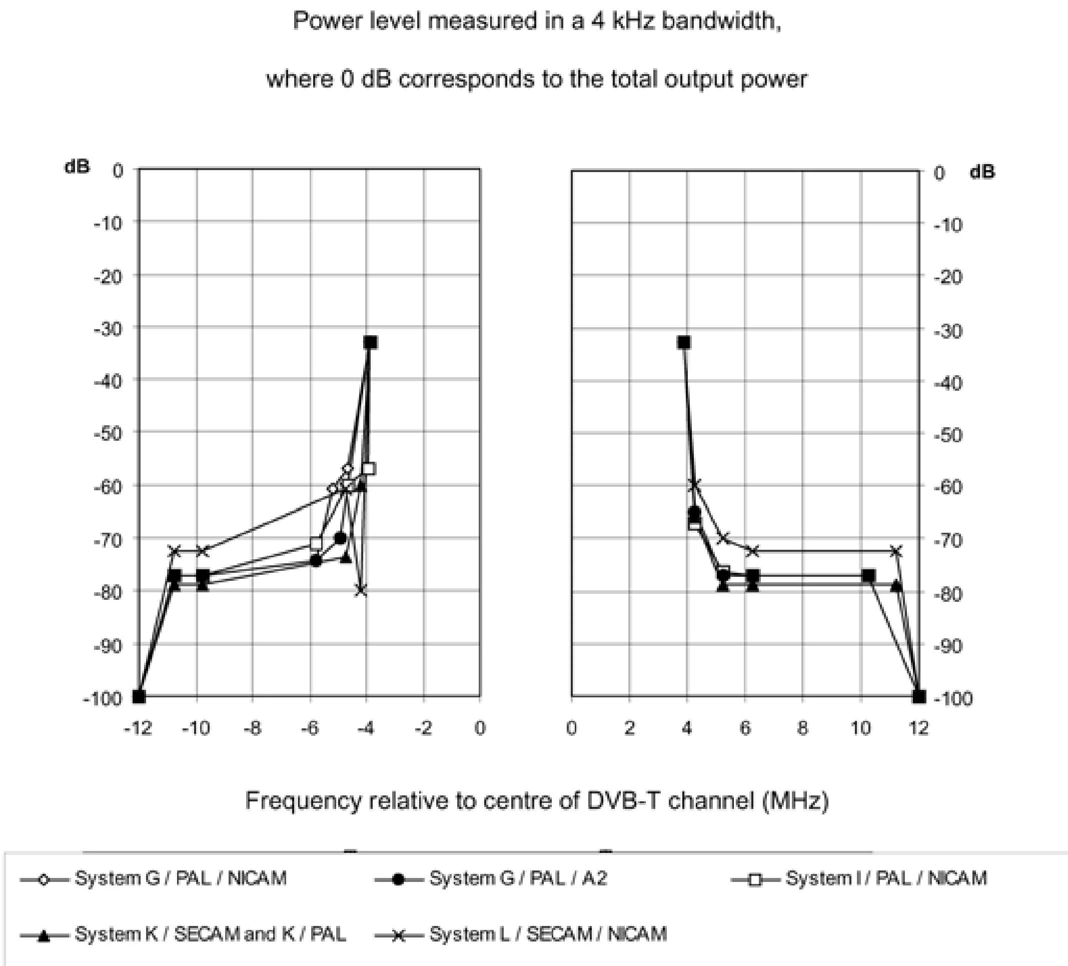
Output spectrum (before filtering) of an OFDM TV Transmitter (DVB-T)



Power amplifiers: phase and amplitude linearity – RF power and output spectrum measurements

Digital modulators (and especially OFDM) require extremely linear amplification.
ABE transmitters' power amplifiers are made with high efficiency and linear solid-state technology, thanks to last generation MOSFET and LDMOS and to the use of precorrection techniques.

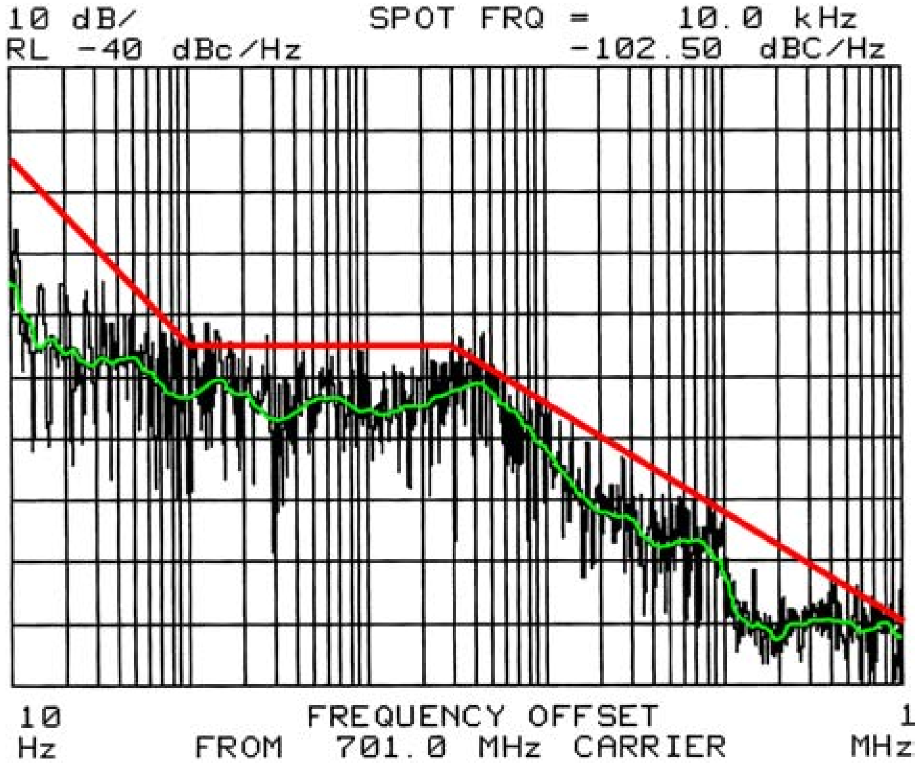
Output spectrum mask required for a DVB-T OFDM transmitter operating on a lower or a higher adjacent channel to a co-sited analog television transmitter (the specification is normally compiled using a proper filter at the output of the transmitter).



Output power, with digital modulation schemes, is defined and measured as "thermal" power (for measuring, you must use a specific or thermal wattmeter, otherwise measuring errors can be significant), and Peak to Average power Ratio (PAR) is considerable: around 8dB for 8VSB modulation and around 15/17dB for the OFDM modulation scheme (this can be limited to around 10dB, but to the detriment of other parameters, such as the Modulation Error Ratio – MER).

For this reason, the “analog” nominal power of the amplifiers / transmitters (video peak sync power, with combined amplification of video and audio carries) must be reduced of a percentage which usually is from 50% to 75% (-3 to -6dB).
Output RF spectrum specifications for digital transmitters are generally more strict than the analog ones, so it needs an adequate output filtering (normally it is used a 6 cavity filter).

ABE Local Oscillator phase noise plot (lower green track) with respect to the limit (mask) proposed by validate work group – AC 106 (higher red track)



Phase noise of local oscillators – frequency stability

In the digital TV transmitters, local oscillator phase noise must be very low, much lower than what is needed for analog TV transmission.
ABE has developed, for digital transmission, a series of synthesized local oscillators with excellent performance and low cost, which are now currently employed in the analog transmitters.
Please note that, especially with OFDM transmitters (up to 6817 carriers), the oscillator’s phase noise is added 6817 times into the emission channel!
The local oscillator’s low phase noise is crucial because otherwise it may cause deterioration of some qualitative parameters (first of all the Modulation Error Ratio – MER).
Frequency precision/stability (that for standard applications is required to be 500Hz) is a parameter that has great importance in case of OFDM transmission in SFN (Single Frequency Network – that is, a network of transmitters with a single frequency).

In this case all transmitters must be synchronized to a single reference signal: the GPS (Global Positioning System) has been chosen for this purpose.

ABE has developed a high stability frequency reference, locked to the GPS signal.

Digital TV transmitter's main measurements – MER

Frequency precision and stability, RF spectrum and output power: this handbook explains these measurements in the previous sections.

Exact parameters and masks may be obtained by consulting specific documentation (ETSI TR 101 290 - ex ETR 290 - and EN 300 744 for DVB-T, downloadable from the www.etsi.org web site; A/64A, A/53B and A54/A for ATSC, downloadable from the www.atsc.org web site).

MER

The Modulation Error Ratio (MER) may be considered the most important quality parameter in a digital transmitter (as the intermodulation is in an analog transmitter).

MER (expressed in dB) is a function of the ratio between the theoretical vector amplitude of a symbol and the amplitude of the shift vector from the theoretical position of the symbol in the constellation and the effective position, averaged for a certain number of symbols.

In other words, the symbol, in the constellation, should be in a certain point but, due to some problems (e.g., local oscillator phase noise, power amplifier compression, etc.), it is slightly shifted. MER is a function of the ratio between the amplitude of the vector that goes from the constellation's center to the ideal position of the symbol and the vector that goes from the theoretical to effective position of the symbol, averaged for a certain number of symbols.

Using other terms, MER indicates the precision of the constellation generated by the transmitter.

The higher the MER, the more precise the constellation generated from the transmitter and the lower the errors made by the receivers demodulating it.

In order to give some practical figures, it has to be considered that to demodulate a QPSK modulation scheme, MER cannot be lower than 5dB; for a 16QAM it needs to be at least 11dB MER and for a 64 QAM it needs to be at least 19dB MER.

Moreover it has to be considered that a commercial receiver generally cannot take advantage from MERs higher than 30dB; so a reasonable requirement for a MER value at the output of a transmitter, for OFDM/64QAM emission, can be 30/32dB.

Comparison between performances of digital and analog TV transmitters

With an analog TV transmitter, with amplitude modulation, as the level of received

signal reduces below a certain threshold, the video and audio quality deteriorates progressively. In practice, the minimum acceptable signal is about 0.5/1 mV.
In contrast, the audio and video quality of a digital receiver is not degraded and, moreover, remains constant as the received input signal level is reduced, right down to a threshold (generally around 20dB under analog threshold), below which the signal is lost.
The exact threshold level depends by various factors/settings: Code Rate, Symbol Rate, receiver noise figure etc.

Summarizing, digital operation confers many notable advantages:

- A single transmitter may be used to carry 4 (or more) video/audio/data program channels.
- Received signal quality is higher and does not degrade progressively as the receiver input reduces but remains constant down to a very low threshold (the receiver input range is increased by around 20dB with respect to the level needed for a good analog reception). So it is possible to use lower RF power or smaller antenna size to broadcast over the same area.
- The DVB-T standard allows mobile reception without the typical problems associated with analog systems (reflections, distortions, double images, and so on), and Single Frequency Network operation.

Notes:

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