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DIGITAL TELEVISION TRANSMISSION STANDARDS

A great deal of fear, uncertainty, and doubt can arise among engineers with an analog or radio-frequency (RF) background at the mere mention of digital transmission systems. Engineers sometimes fall into the trap of believing that digital systems are fundamentally different from their analog counterparts. As will be demonstrated, this is not the case. In concept, the transmission of digital television signals is no different than for analog television. The difference is in the details of implementation (hence the need for this book).

A block diagram of a typical broadcast transmission system is shown in Figure 1-1. This block diagram may, in fact, represent either an analog or a digital system. Major components include a transmitter comprising an exciter, power amplifier, and RF system components, an antenna with associated transmission line, and many receiving locations. Between the transmitter and receivers is the over-the-air broadcast transmission path. The input to the system is the baseband signal by which the RF carrier is modulated. In an analog system the baseband signal includes composite video and audio signals. In separate amplification, these modulate separate visual and aural carriers. If common amplification is used, the modulated signals are combined in the exciter and amplified together in the power amplifier. The combined signals are then transmitted together through the remainder of the link.

For a digital system, the conceptual block diagram most resembles common amplification. A single baseband signal modulates a carrier and is amplified in the transmitter, broadcast by means of the antenna, and received after propagating through the over-the-air link. The baseband signal is a composite digital data stream that may include video and audio as well as data. Since the method of modulation is also digital, the exciter used with the transmitter is also different. Beyond these details, the remainder of the system is fundamentally

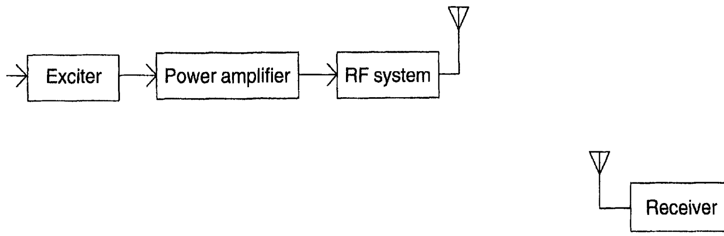

Figure 1-1. Broadcast transmission system.

the same, although there are further subtle differences in power measurement, tuning, control, and performance measurement, upconverters, power amplifiers, transmission lines, and antennas.

The similarities between digital and analog systems is also apparent when we consider the transmission channel. The ideal channel would transfer the modulated RF carrier from the modulator to the receiver with no degradation or impairment other than a reduction in the signal level and the signal-to-noise ratio. As a matter of fact, the real transmission channel is far from ideal. The signal may suffer linear and nonlinear distortions as well as other impairments in the transmitter and other parts of the channel. For analog television signals, these impairments are characterized in terms of noise, frequency response, group delay, luminance nonlinearity, differential gain, incidental carrier phase modulation (ICPM), differential phase, lower sideband reinsertion, and intermodulation distortion. For digital signals, linear distortions are also characterized in terms of frequency response and group delay. For nonlinear distortions, AM-to-AM and AM-to-PM conversion are the operative terms. In either case, the objective of good system design is to reduce these distortions to specified levels so that the channel may be as transparent as possible.

The antenna and transmission line may introduce some of the linear distortions. In most cases, these are relatively small compared to distortions introduced by the propagation path. This is especially true of matched coaxial transmission lines. Waveguides may introduce nontrivial amounts of group delay. Under some circumstances an antenna may introduce significant frequency response, nonlinear phase, and group delay distortion. Once the system design is finalized, however, there no attempt may be made to equalize distortions introduced by the transmission line or antenna.

The propagation path from the broadcast antenna to the receiver location may be the source of the most significant impairments. These impairments include noise and linear distortions resulting from reflections and other sources of multipath. Depending on specific site characteristics, the linear distortions may be severe. The impairments introduced by propagation effects vary from location to location and are also a function of time. Obviously, there is no practical means of equalizing these distortions at the transmitter. Any equalization to mitigate response and group delay introduced by the over-the-air path must be done in the receiver. The random noise introduced in the propagation path may be overcome at the transmitter only by increasing the average effective radiated power (AERP).

ATSC TERRESTRIAL TRANSMISSION STANDARD

At the time of this writing, the U.S. Federal Communications Commission (FCC), Canada, and South Korea have adopted the standard developed for digital television by the Advanced Television Systems Committee (ATSC). This standard, designated A/53, represents the results of several years of design, analysis, testing, and evaluation by many experts in industry and government. It promises to be a sound vehicle for digital television delivery for decades to come. The standard describes the system characteristics of the U.S. digital television system, referred to in this book as the ATSC or DTV system. The standard addresses a wide variety of subsystems required for originating, encoding, transporting, transmitting, and receiving of video, audio, and data by over-the-air broadcast and cable systems. The transmission system is a primary subject of this book, which is described in detail in Appendix D of the ATSC standard. The ATSC standard specifies a system designed to transmit high-quality digital video, digital audio, and data over existing 6-MHz channels. The system is designed to deliver digital information at a rate of 19.29 megabits per second (Mb/s).

The transmitter component affected most by the implementation of this standard is the exciter, although, only portions of the exciter need be affected. Figure 1-2 is a conceptual block diagram of a television exciter. As drawn, this block diagram could represent either an analog or a digital exciter. The first block, the modulator, represents composite video and audio processing and modulation in the case of analog television; for digital television, this block represents digital data processing or channel coding and modulation. (It is assumed that the reader is familiar with analog video and audio modulator functions; if not, refer to Chapter 6.2, "Television Transmitters," of the *NAB Engineering Handbook*, 9th edition.)

The second block, intermediate frequency (IF)-to-RF conversion, represents upconversion, IF precorrection and equalization, final amplification, and filtering. In principle, this block is the same for both analog and digital television signals in that the main purpose is to translate the IF to the desired RF channel. For the time being, the discussion will focus on processing the digital baseband signal prior to upconversion. To facilitate this, the nature of the input and output signals of the digital modulator block is first discussed.

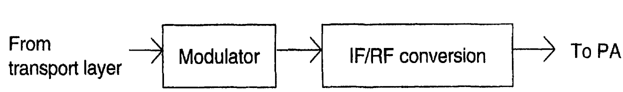

Figure 1-2. Block diagram of TV exciter.

The digital input signal to the ATSC transmission system is a synchronous serial MPEG[1]-2 transport stream at a constant data rate of 19.39... Mb/s. This serial data stream is comprised of 187-byte MPEG data packets plus a sync byte. The payload data rate is 19.2895. Mb/s. The payload may include encoded packets of digital video, digital audio, and/or data. The transport stream arrives at the exciter input on a single 75-Ω coaxial cable with a BNC input connector. The data clock is embedded with the payload data. Biphase mark coding is used. The data clock frequency error is specified to be less than ±54 Hz. The standard input level is 0.8 V ± 10% peak to peak as defined by the SMPTE Standard 310M-Synchronous Serial Interface for an MPEG-2 digital transport stream.

The output signal from the modulator block is an eight-level vestigial sideband modulated signal. Ordinarily, this is at some frequency intermediate to the baseband and RF channel frequency. The frequency, level, and other interface characteristics of the IF are generally dependent on the design choices made by the equipment manufacturer.

Figure 1-3 is a simplified block diagram of the signal processing functions required to convert the MPEG-2 transport stream to the eight-level vestigial sideband signal (8 VSB) required by the ATSC transmission system. The modulator may be viewed as performing two essential functions. The first function is channel coding. Among other things, the channel coder modifies the input data stream from the transport layer by adding information by which the receiver may detect and correct transmission errors. These are errors as a result of impairments introduced in the transmission channel. Without channel coding, the receiver would be unable to decode and display the signal properly except at receive sites with a very high signal-to-noise ratio and a minimum of multipath. The second block in Figure 1-3 is the modulator proper. It is in this block that an IF signal is modulated with the channel-coded data stream to produce the 8 VSB signal required for terrestrial over-the-air transmission.

A block diagram of the channel coder is shown in Figure 1-4. Six major functions are performed in the channel coder: data randomizing, Reed–Solomon (R/S) coding, data interleaving, trellis coding, sync insertion, and pilot signal insertion.

The incoming data from the transport stream are first randomized. This process exclusive-ORs the data bytes with a pseudorandom binary sequence locked to the data frame. The purpose of randomization is to assure that the data spectrum is uniform throughout the 6-MHz channel, even when the data are constant.

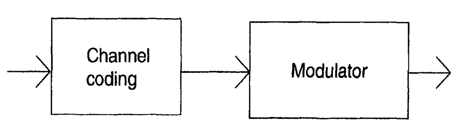

Figure 1-3. DTV modulator.

[1] Motion Pictures Expert Group.

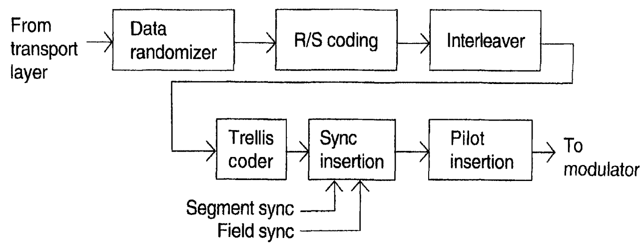

Figure 1-4. DTV channel coding. (From ATSC DTV Standard A/53, Annex D; used with permission.)

This pseudorandom sequence is generated in a 16-bit shift register with nine feedback taps. A complementary derandomizer is provided in the receiver to recover the original data sequence. Randomizing is not applied to the sync byte of the transport packet.

The next step is R/S coding. This is a forward error correction (FEC) code designed to protect against noise bursts. In this code, 20 parity bytes are added to each data block or 187-byte data packet. The R/S code selected is capable of correcting up to 10-byte errors per data block. Because of the additional bytes, the clock and data rate is necessarily increased from 19.39 Mb/s to 21.52 Mb/s. As with randomization, R/S coding is not applied to the sync bytes.

After R/S coding, the data structure is formatted into data bytes and segments, fields, and frames as defined in Figure 1-5. A data field is comprised of 312 data segments plus a sync segment, for a total of 313 segments. A data frame is comprised of two data fields, or 626 segments. The R/S coded data are interleaved to provide additional error correction. This process spreads the data bytes from several R/S packets over a much longer period of time so that a very long burst of noise is required to overrun the capability of the R/S code. A total of 87 R/S packets are processed in the interleaver.

Trellis coding, another error correction code, follows the R/S interleaver. The purpose of this code differs from the R/S code in that it has the effect of improving the signal-to-noise ratio (S/N) threshold in the presence of thermal or white noise. It is termed a $\frac{2}{3}$-rate code because every other input bit is encoded to 2 output bits; the alternate bit is not encoded. Thus the output of the trellis coder is a parallel bus of 3 bits for every 2 input bits. The trellis-coded data are interleaved with a 12-symbol code interleaver. The data rate at the output of the trellis coder is increased by a ratio of $\frac{3}{2}$, to 32.28 Mb/s. Taken together, the output bits of the trellis coder comprise the 3-bit symbols. These symbols (-7, -5, -3, -1, 1, 3, 5, 7) are the eight levels of the VSB modulator. The symbol rate is one-third that of the trellis-coded data rate, or 10.76 symbols/s.

The spectral efficiency, η_s, is the ratio of the encoded data rate to the channel bandwidth:

$$\eta_s = \frac{32.28}{6} = 5.38 \text{ bps/Hz}$$

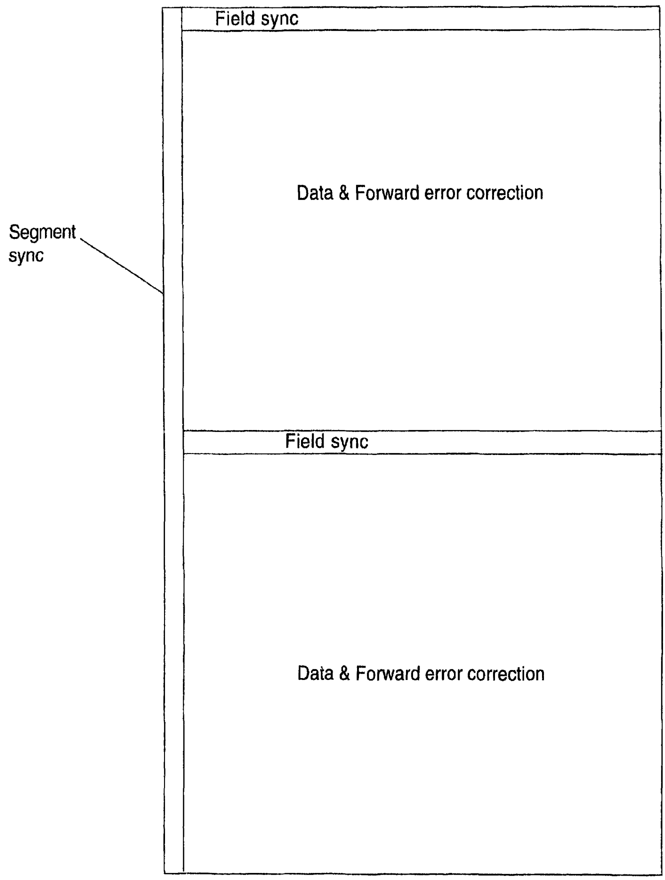

Figure 1-5. Data frame structure for the ATSC system. (From ATSC DTV Standard A/53, Annex D; used with permission.)

This is a consequence of using 3 bits per symbol to create the eight VSB levels ($M = 8$) and the excess bandwidth of the Nyquist filter ($\alpha_N = 0.1152$). Using these parameters, the spectral efficiency may be computed by

$$\eta_s = \frac{2 \log_2 M}{1 + \alpha_N} \qquad \text{bps/Hz}$$

which also results in 5.38 bps/Hz.

A data segment is comprised of the equivalent of the data from one R/S transport packet plus FEC code and data segment sync as shown in Figure 1-6. Actually, the data come from several R/S packets because of interleaving.

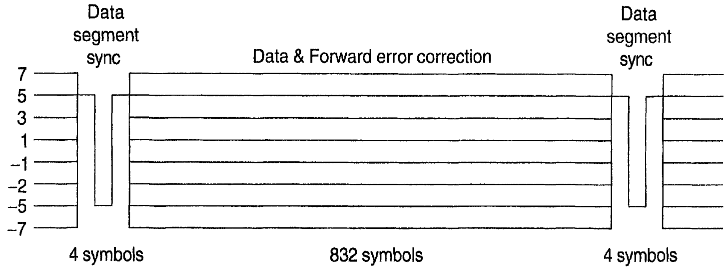

Figure 1-6. Data segment for the ATSC system. (From ATSC DTV Standard A/53, Annex D; used with permission.)

Since each R/S packet is 207 bytes in length, a data segment is 208 bytes $(207 + 1)$. At 8 bits per byte and 3 bits per symbol, the data segment is 208 bytes $\times$ 8 bits/byte $\times$ $\frac{3}{2}/3$ bits/symbol, or 832 symbols in length, 828 of which are FEC coded data; the remaining four are segment sync symbols. There are 3×832, or 2496 bits per segment, 2484 of which are data and 12 of which are segment syncs. For the data rate of 32.28 Mb/s the time per bit is 31 ns. Thus the time per segment is 2496×31, or 77.3 μs, and the segment rate, $f_{seg} = 12.94$ data segments per second. With 313 segments per field, the field time is 313×77.3 μs, or 24.2 ms, and the field rate is 41.3 kHz. The frame rate, f_{frame}, is one-half the field rate, or 20.66 kHz.

Following the trellis coding, field and segment sync symbols are inserted. The structure of the data field sync segment is defined in Figure 1-7. As with the data segments, the field sync segment is 832 symbols in length. Each symbol is binary encoded as either + or –5. Four data segment sync symbols replace the MPEG sync byte. These are followed by a series of pseudorandom number (PN) sequences of length 511, 63, 63, and 63 symbols, respectively. The PN63 sequences are identical, except that the middle sequence is of opposite sign in every other field. This inversion allows the receiver to recognize the alternate data fields comprising a frame.

The PN63 sequences are followed by a level identification sequence consisting of 24 symbols. The last 104 symbols of the field sync segment are reserved; 92 of these symbols may be a continuation of the PN63 sequence. The last 12 of these symbols are duplicates of the last 12 symbols of the preceding data segment.

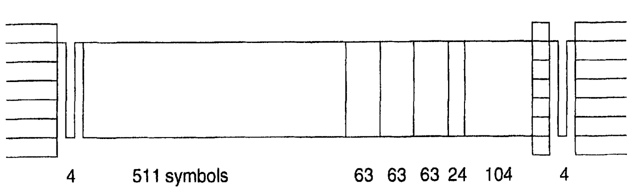

Figure 1-7. Field sync for the ATSC system. (From ATSC DTV Standard A/53, Annex D; used with permission.)

In addition to providing a means of synchronizing the receiver to the formatted data, the sync segments serve as training signals for the receiver equalizer. The equalizer improves the quality of the received signal by reducing linear distortions. This is analogous to ghost reduction due to multipath in analog systems. Since the sync sequences are known repetitive signals, the equalizer taps may be adjusted to reproduce these sequences with a minimum of distortion. The taps, thus adjusted, reduce distortion of the received data. The sync segments may also be used for diagnostic purposes.

The data field and frame structure has the familiar appearance of the field and frame structure of analog television. However, it should not be assumed that a data field corresponds to a video field. Each data field may include video, audio, or other data, so there is generally no correspondence between data fields and video fields.

VESTIGIAL SIDEBAND MODULATION

Vestigial sideband modulation may be accomplished in either the analog or the digital domain. Manufacturers have generally developed their own modulation schemes, some of which may be proprietary. Since the purpose of this book is to describe the principles of digital television transmission, a generic modulator using analog circuitry is presented.

Such a modulator is illustrated in Figure 1-8. The signal (i.e., the 3-bit multilevel symbols or pulses from the output of the trellis coder) is divided equally to form in-phase (I) and quadrature (Q) paths at the input to the modulator. The pulses are then shaped to minimize intersymbol interference. This pulse shaping is accomplished in a Nyquist filter. This is a low-pass linear-phase filter with flat amplitude response over most of its passband. At the upper and lower band edges, the filter response transitions to the stopband by means of skirts with a root-raised-cosine shape. The steepness of the skirts is determined by the shape factor, α_N. For the ATSC system, α_N is specified to be 0.1152. The Nyquist filter multiplies the shaped signals by either $\sin(\pi t/2T)$ or $\cos(\pi t/2T)$, where T is the symbol time.

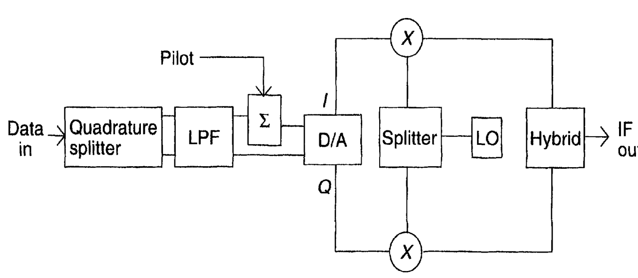

Figure 1-8. Typical digital modulator.

The shaped I and Q signals are now presented to digital-to-analog (D/A) converters in each of the I and Q channels. The I and Q signals are each multiplied by equal levels of the local oscillator (LO) signal. For the Q path, the LO signal is 90° out of phase with respect to the LO signal for the I path. These signals are then summed in a two-way power combiner to produce the IF output. The resulting spectrum contains only one of the sidebands of the modulated signals and the carrier is suppressed. Thus this modulation technique is called vestigial sideband. A pilot signal is inserted in the I path of the modulator. By adding a small direct-current (dc) offset of 1.25 V to all of the encoded symbols (including sync), a tone at the same frequency as the suppressed carrier is generated in the output of the VSB modulator. The presence of the pilot adds very little power (only 0.3 dB) to the modulated signal, but it is important in that it enables receiver tuning under conditions of severe noise and interference. It also speeds carrier recovery and, therefore, data acquisition in the receiver. It is apparent that the quality of the IF output is dependent on the stability of both the incoming data and the LO.

At this point in the system, the complete DTV signal has been generated, consisting of eight amplitude levels, four positive and four negative. The signal is often displayed in a two-dimensional I–Q or constellation diagram, as shown in Figure 1-9. This is a graphical representation of the orthogonal I and Q components of the modulated waveform, plotted in X–Y or rectangular

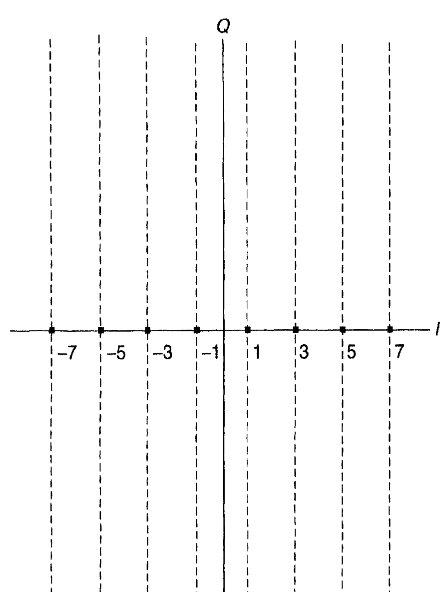

Figure 1-9. I–Q diagram for 8 VSB signal.

coordinates, where the X and Y axes are called the I and Q axes, respectively. Each point in the I–Q diagram represents a specific amplitude and phase of the RF carrier. For 8 VSB, information is carried only by the I component, for which the distinct levels of 8 VSB are plotted on the horizontal axis. Although a quadrature component is present and is displayed in the direction of the Q-axis, there are no distinct levels associated with the Q component and no information conveyed.

The modulated signal occupies 6 MHz of total bandwidth by virtue of the vestigial sideband modulation scheme. The spectrum of the modulated signal is shown in Figure 1-10. The energy is spread uniformly throughout most of the channel. At both the upper and lower band edges, the spectrum is shaped in accordance with the root-raised-cosine or Nyquist filter. A complementary root-raised-cosine filter of the same shape is included in the receiver so that the system response is a raised cosine function.

The 3-dB bandwidth of the resulting transmitted spectrum is 5.38 MHz. At the RF channel frequency, the pilot is located at the lower 3-dB point, 0.31 MHz above the lower edge of the channel. The pilot is the same frequency as the suppressed carrier. (The pilot may be at the opposite end of the spectrum at the IF.) The DTV pilot is offset from the NTSC visual carrier to minimize DTV-to-NTSC cochannel interference. The remainder of the system exists for the purposes of upconverting to the desired channel, amplifying to the required power level, and radiating the on-channel signal.

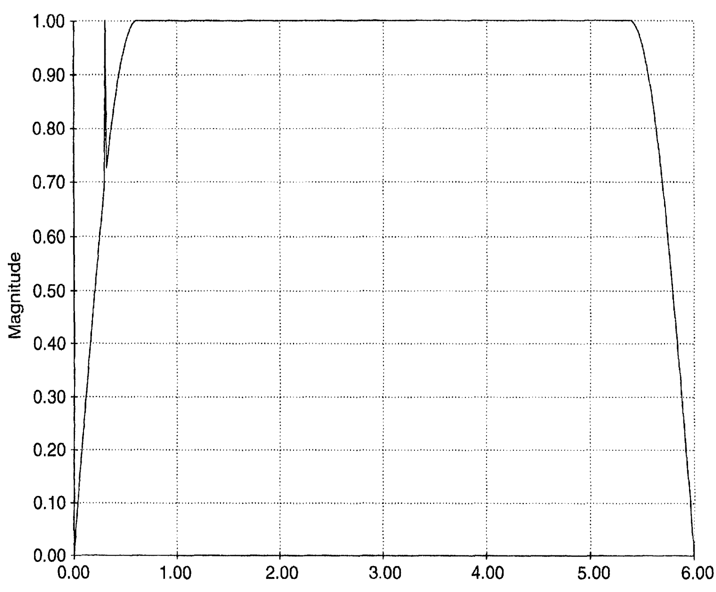

Figure 1-10. Transmitted spectrum, 8 VSB.

DVB-T TRANSMISSION STANDARD

The European Telecommunications Standards Institute has adopted a set of standards for digital broadcasting of television, sound, and data services. Standards have been adopted for satellite, cable, and terrestrial signal delivery. The standard for terrestrial transmission, ETS 300 744, is designated Digital Video Broadcast–Terrestrial (DVB-T). This standard describes a baseline transmission system for digital broadcasting of television. At the time of this writing, it has been adopted by the 15 members of the European Union, Australia, and New Zealand. It is similar in many respects to the U.S. DTV standard. However, there are also important and significant differences in both channel coding and modulation.

The DVB-T standard specifies a system designed to transmit high-quality digital video, digital audio, and data over existing 7- or 8-MHz channels. The system is designed to deliver digital information at rates from 4.98 to 31.67 Mb/s. Although there are many similarities with the ATSC standard in the transport layer and channel coding, a significant difference is in the type of modulation used. Coded orthogonal frequency-division multiplex (COFDM) has been selected for DVB-T, in part due to the unique requirements of European broadcasting stations and networks. Single-frequency networks (SFN) are used extensively in Europe to more effectively use the channels available; COFDM is seen as best suited to this requirement. In a SFN, all stations broadcasting a particular program do so on the same channel, each being synchronized to precisely the same reference signal and having common baseband timing. A receiver tuned to this channel may receive signals from one or more stations simultaneously, each with a different delay. Under multipath conditions, the signal strength from each station may vary with time. The guard intervals and equalization built into the COFDM system facilitate effective reception under these conditions. The guard interval may be selected from $\frac{1}{32}$ to $\frac{1}{4}$ the duration of the active symbol time, so that the total symbol duration is from $1\frac{1}{32}$ to $1\frac{1}{4}$ the active symbol time.

As with the ATSC standard, the transmitter assembly most affected by the transition to digital broadcast is the exciter, with the major changes required being baseband processing and modulation. Thus the focus of this discussion is on the modulator block. The nature of the input and output signals is discussed first.

In common with DTV in the United States, the digital input signal to the DVB-T transmission system is a MPEG-2 synchronous transport stream comprised of 187-byte MPEG data packets plus a sync byte. The payload may include encoded packets of digital video, digital audio, and/or data. The parallel transport stream connector at the modulator input is a DB25 female connector. The data clock line is separate from the payload data lines.

The output signal from the modulator block is a COFDM signal. Ordinarily, this is generated at some frequency intermediate to the baseband and RF channel frequency. The frequency, signal level, and other interface characteristics of the IF are generally dependent on design choices made by the equipment manufacturer.

As in the ATSC system, the modulator may be viewed as performing the functions of channel coding and modulation proper. The functions performed in the channel coder include energy dispersal or data randomization, outer or R/S coding, outer interleaving, inner or trellis coding, and interleaving. The modulator functions include mapping, frame adaptation, and pilot insertion.

The incoming data from the transport stream are first dispersed or randomized. A complementary derandomizer is provided in the receiver to recover the original data sequence. As with the DTV system, randomizing is not applied to the sync byte of the transport packet.

The next step is outer or R/S coding. The details of the code selected differ from those of the DTV standard in that it is capable of correcting up to only eight byte errors per data block. In this code, 16 parity bytes are added to each sync and data block or 188-byte packet. The R/S coded data are interleaved to provide additional error correction.

Convolutional coding and interleaving follow the R/S interleaver. The DVB-T system allows for a range of punctured convolutional codes. Selection of the code rate is based on the most appropriate level of error correction for a given service and data rate. Punctured rates of $\frac{2}{3}$, $\frac{3}{4}$, $\frac{5}{6}$, or $\frac{7}{8}$ are derived from the $\frac{1}{2}$-rate mother code. Interleaving consists of both bitwise and symbol interleaving. Bit interleaving is performed only on the useful data.

The purpose of the symbol interleaver is to map bits on to the active OFDM carriers. Detailed operation of the symbol interleaver depends on the number of carriers generated, whether 2048 (2^{11}) in the 2k mode or 8192 (2^{13}) in the 8k mode. Some of the carriers are used to transmit reference information for signaling purposes (i.e., to select the parameters related to the transmission mode). The number of carriers available for data transmission is 1705 in the 2k mode or 6817 in the 8k mode. The overall bit rate available for data transmission is not dependent on the mode but on the choice of modulation used to map data on to each carrier.

The OFDM modulator follows the inner coding and interleaving. This involves computing an inverse discrete fourier transform (IDFT) to generate multiple carriers and quadrature modulation. The transmitted signal is organized in frames, each frame having a duration of T_F and consisting 68 OFDM symbols. The symbols are numbered from 0 to 67, each containing data and reference information. In addition, an OFDM frame contains pilot cells and transmission parameter signaling (TPS) carriers. The pilot signals may be used for frame, frequency, and time synchronization, channel estimation, and transmission mode identification. TPS is used to select the parameters related to channel coding and modulation.

The many separately modulated carriers may employ any one of three square constellation patterns: quadrature-phase shift keying (QPSK, 2 bits per symbol), 16-constellation-point quadrature amplitude modulation (16 QAM, 4 bits per symbol), or 64 QAM (6 bits per symbol). By selecting different levels of QAM in conjunction with different inner code rates and guard interval ratios, bit rate may be traded for ruggedness. For example, QPSK with a code rate of $\frac{1}{2}$ and a guard interval ratio of $\frac{1}{4}$ is much more rugged than 64 QAM with a code rate of $\frac{5}{6}$ and $\frac{1}{32}$ guard interval ratio. However, the available data rate is much less. The 8k mode has the longest available guard interval, making it the best choice for single-frequency networks with widely separated transmitters.

Hierarchical transmission is also a feature of the DVB-T standard. The incoming data stream is divided into two separate streams, a low- and a high-priority stream, each of which may be transmitted with different channel coding and with different modulation on the subcarriers. This allows the broadcaster to make different trade-offs of bit rate and ruggedness for the two streams.

The power spectral density of the modulated carriers is the sum of the power spectral density of the individual carriers. The upper portion of the transmitted spectrum for an 8-MHz channel is shown in Figure 1-11, plotted relative to the channel center frequency. Overall, the energy is spread nearly uniformly throughout most of the channel. However, since the symbol time is larger than the inverse of the carrier spacing, the spectral density is not constant. The center frequency of the DVB-T channels is the same as the current European analog ultrahigh frequency (UHF) channels. The minimum carrier-to-noise (C/N) ratio is dependent, among other parameters, on modulation and inner code rate. As with the DTV system, there is no visual, chroma, or aural carrier frequencies as in analog TV.

The complete DVB-T signal has been generated at the output of the modulator. The remainder of a transmitting system exists for the purposes of upconverting

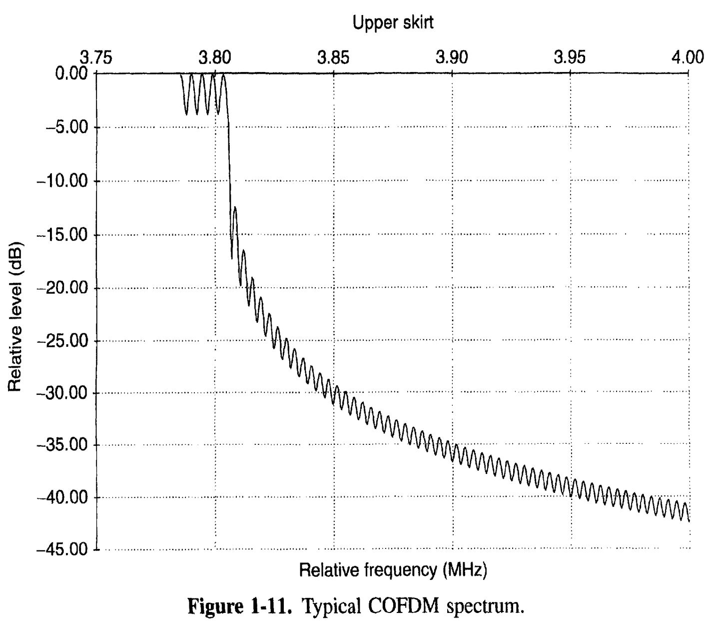

Figure 1-11. Typical COFDM spectrum.

to the desired channel, amplifying to the required power level, and radiating the on-channel signal.

ISDB-T TRANSMISSION STANDARD

Japan's Digital Broadcasting Experts Group (DiBEG) has developed a standard for digital broadcasting of television, sound, and data services, designated integrated services digital broadcasting (ISDB). Standards have been developed for delivery of satellite, cable, and terrestrial signals. These standards include a description of a baseline transmission system that provides for digital broadcasting of television, including channel coding and modulation. The transmission standard for terrestrial digital television is similar in many respects to the DVB-T standard. It is entitled Integrated Services Digital Broadcasting–Terrestrial (ISDB-T).[2] A key difference with respect to DVB-T is the use of band-segmented transmission–OFDM (BST-OFDM). This is a data segmentation approach that permits the service bandwidth to be allocated to various services, including data, radio, standard definition television (SDTV), and high-definition television (HDTV) in a flexible manner. It is planned that digital television will be launched in Japan after 2003.

The ISDB-T standard specifies a system designed to transmit over existing 6-, 7-, or 8-MHz channels. The system is designed to deliver digital information at data rates from 3.561 to 30.980 Mb/s.

In common with the other world standards, the digital input signal to the ISDB-T transmission system is a MPEG-2 synchronous transport comprised of 187-byte MPEG data packets plus a sync byte. The payload may include encoded packets of digital video, digital audio, text, graphics, and data. In addition, transmission and multiplex control (TMCC) is defined for hierarchical transmission. To make use of the band-segmenting feature, the data stream is remultiplexed and arranged into data groups, each representing all or part of a program or service. After channel coding, these data groups become OFDM segments. Each OFDM segment occupies $\frac{1}{14}$ of the channel bandwidth. This arrangement allows for both broadband and narrowband services.

For example, a single HDTV service might occupy 12 of the OFDM segments, with the thirteenth used for sound and data.[3] Alternatively, multiple SDTV programs might occupy the 12 OFDM segments. A maximum of three OFDM segment groups or hierarchical layers may be accommodated at one time. For the narrowband services, a small, less expensive narrowband receiver may be used. The OFDM segment in the center of the channel is dedicated to such narrowband or partial reception services. Obviously, a receiver decoding a single OFDM segment receives only a portion of the original transport stream.

[2] "Channel Coding, Frame Structure, and Modulation Scheme for Terrestrial Integrated Services Digital Broadcasting (ISDB-T)," *ITU Document 11A/Jxx-E*, March 30, 1999.

[3] The upper and lower channel edges occupy the bandwidth of the remaining OFDM segment.

ISDB-T has many features in common with DVB-T. Both inner and outer FEC codes are applied to the data. The resulting data stream modulates multiple orthogonal carriers. Thus both standards make use of COFDM. The guard interval may be selected from $\frac{1}{32}$ to $\frac{1}{4}$ the duration of the active symbol time. As in Europe, Japan will use this approach to increase the number of available channels by means of SFNs. The R/S code is capable of correcting up to eight-byte errors per data block. A total of 16 parity bytes are added to each sync and data block. The system also allows for a range of punctured convolutional codes. Selection of the code rate is based on the most appropriate level of error correction for a given service or data rate. Code rates of $\frac{2}{3}$, $\frac{3}{4}$, $\frac{5}{6}$, or $\frac{7}{8}$ are derived from a $\frac{1}{2}$-rate mother code.

Despite similarities, there are differences in implementation of the channel coding. As shown in Figure 1-12, the order of R/S coding and energy dispersal are interchanged with the order used in either the DTV and DVB-T systems. The R/S coding is applied to the data as they emerge from the remultiplexed transport stream, including any null packets. Energy dispersal, delay adjustment, bytewise interleaving, and trellis coding are then applied in that order to each data group separately. This permits the length of the interleaving, code rate for the inner FEC code, and signal constellation to be selected independently for each hierarchical layer. The null packets at the output of the R/S coder are removed. The delay resulting from the bytewise interleaving differs for each layer, depending on the channel coding and modulation. To compensate, a delay adjustment is inserted prior to the interleaver.

The OFDM modulator follows the inner coding and interleaving. This involves computing an IDFT to generate multiple carriers and quadrature modulation. The number of carriers available ranges from 1405 to 5617 for all channel bandwidths depending on the transmission mode. Of these, the number of carriers available for data transmission ranges from 1249 to 4993. Obviously, the carrier spacing is increased for the wider channels for a given number of carriers. The information bandwidth is approximately 5.6, 6.5, and 7.4 MHz for the 6-, 7-, and 8-MHz channels, respectively.

The transmitted signal is organized in frames. However, the frame duration is not fixed as in the DVB-T standard. Rather, the frame duration depends on the

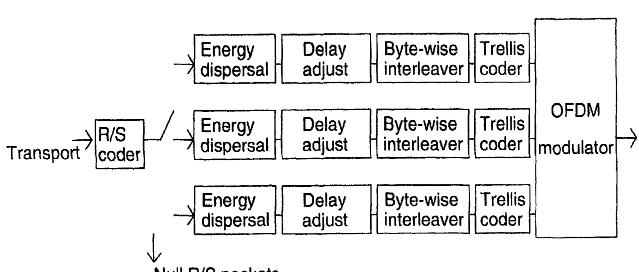

Figure 1-12. Block diagram of ISDB-T channel coding.

TABLE 1-1. Frame Duration (ms) versus Mode and Guard Interval Ratio

Guard Interval Ratio	Mode 1	Mode 2	Mode 3
$\frac{1}{4}$	64.26	128.52	257.04
$\frac{1}{8}$	57.83	115.67	231.34
$\frac{1}{16}$	54.62	109.24	218.46
$\frac{1}{32}$	53.01	106.03	212.06

transmission mode and the length of the guard interval. These relationships are summarized in Table 1-1. The frame duration doubles from mode 1 to mode 2 and doubles again from mode 2 to mode 3. Modes 1, 2, and 3 are defined for 108, 216, and 432 OFDM carriers per segment, respectively. The packets comprising the frames are numbered consecutively and contain the payload data as well as the information necessary for operation of the broadcast system. Scattered and continual pilot signals are available for frequency synchronization and channel estimation. The TMCC subchannels carry packets with information on the transmission parameters. Auxiliary subchannels carry ancillary information for network operation.

The many separately modulated carriers may employ the same three constellation patterns as provided in the DVB-T standard-QPSK, 16 QAM or 64 QAM. In addition, differential quadrature-phase shift keying (DQPSK) is available.

With modulation complete, the complete ISDB-T signal has been generated. The remainder of the transmitting system exists for the purposes of upconverting to the desired channel, amplifying to the required power level, and radiating the on-channel signal.

CHANNEL ALLOCATIONS

The DVB-T system operates in 7- and 8-MHz channels primarily within the existing European UHF spectrum (bands IV and V), although implementation guidelines have been published for bands I and III as well.[4] In Japan, 6-MHz channels are to be used. In general, the availability of spectrum varies from country to country; in virtually every case a scarcity exists. As in the United States, the tendency is to move terrestrial television to the UHF spectrum to free other frequencies for other uses. A common solution in Europe, Japan, and other countries is to use regional and/or national SFNs. This approach allows for the broadcast of just a few programs to a high percentage of the target area using a minimum number of channels. Generally, the digital services will coexist

[4] *Implementation Guidelines for DVB-T: Transmission Aspects,* European Telecommunications Standards Institute, April 1997.

with existing analog services for some extended period of time (say, 15 years), after which the analog service will be discontinued. Where possible, existing transmitter sites, antennas, and towers are expected to be used. To maximize the possibility of viewers using existing receiving antennas, assignment of channels near the existing analog channels with the same polarization is desirable.

In the United States, the DTV system operates in 6-MHz channels in portions of both the very high frequency (VHF) and UHF spectrum. A core spectrum is defined which includes a total bandwidth of 294 MHz, extending from channel 2 through channel 51. Because of the limited availability of spectrum and the need to minimize interference, some broadcasters are assigned spectrum outside the core at frequencies as high as channel 69 during the transition period. Use of channel 6 is minimized due to the potential for interference with the lower FM frequencies. The use of Channels 3 and 4 in the same market is minimized to facilitate use of cable terminal devices, which may operate on either of these channels. Channel 37 is reserved for radio astronomy. Where it is necessary to use adjacent channels in the same market, the NTSC and DTV stations are colocated, if possible. The licensees for the adjacent channels are required to lock the DTV and NTSC carrier frequencies to a common reference frequency to protect the NTSC from excessive interference.

The U.S. Telecommunications Act of 1996 provides that initial eligibility for an advanced television license is limited to existing broadcasters with the condition that they eventually relinquish either the current analog channel or the new digital channel at the end of the transition period. The purpose of this provision is, in part, to promote spectrum efficiency and rapid recovery of spectrum for other purposes. Consequently, DTV was introduced in the United States by assigning existing broadcasters with a temporary channel on which to operate a DTV station during the transition period, which will extend to 2006. It is planned that 78-MHz of spectrum will be recovered at the end of the transition period; it is also planned that 60 MHz in channels 60 to 69 will be recovered earlier.

If in the future channels 2 to 6 prove to be acceptable for transmission of DTV, the core spectrum may be redefined to be channels 2 to 46.[5] Those stations operating outside the core spectrum during the transition will be required to move their DTV operations to a channel inside the core when one becomes available. Broadcasters whose existing NTSC channel is in the core spectrum could move their DTV operations to this channel in the future. Broadcasters whose NTSC and DTV channels are in the core spectrum could chose which of those will be their permanent DTV channel.

It is evident that broadcast engineers in the United States will face the challenge of transmission system design and operation for the foreseeable future. Many important decisions must be made for initial systems during the transition period. Many of these systems will continue to operate without the need for major changes after the transition period ends. Others systems, however, may require major changes to accommodate the channel shifts required by the FCC.

[5] *FCC 6th Report and Order DTV Allocations*, Appendix D, April 22, 1997, p. D-11.

ANTENNA HEIGHT AND POWER

In the United States, the antenna height above average terrain (HAAT) and AERP for DTV stations operated by existing licensees is designed to provide equivalent noise-limited coverage to a distance equal to the present NTSC grade B service contour. The maximum permissible power for new DTV stations in the UHF band is 316 kW. The maximum antenna height is 2000 ft above average terrain. For HAATs below this value, higher AERP is permitted to achieve equivalent coverage. The maximum AERP is 1000 kW regardless of HAAT. The minimum AERP for UHF is 50 kW. Power allocations for VHF range from 200 W to slightly more than 20 kW.

MPEG-2

Although the source encoding and transport layer are distinct from the transmission system, they are closely associated. It is therefore important that the transmission system engineer have an understanding of MPEG-2. The following discussion is a cursory overview; for more details, the interested reader is referred to ATSC A/53 or the Implementation Guidelines for DVB-T, which point to additional documents.

In accordance with the International Telecommunications Union, Radio Sector (ITU-R) digital terrestrial broadcast model, the transport layer supplies the data stream to the RF/transmission system. This is illustrated in Figure 1-13. Since there is no error protection in the transport stream, compatible forward error correction codes are supplied in the transmission layer as already described.

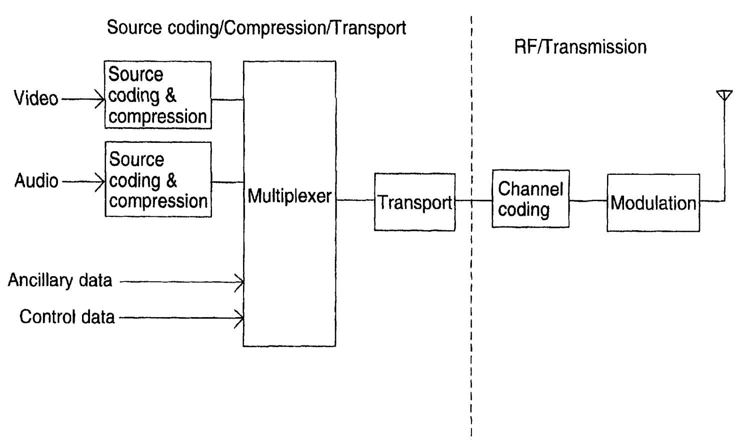

Figure 1-13. Digital television broadcast model. (From ATSC DTV Standard A/53, Annex D; used with permission.)

MPEG-2 refers to a set of four standards adopted by the International Standards Organization (ISO). Together, these standards define the syntax for the source coding of video and audio and the packetization and multiplexing of video, audio, and data signals for the DTV, DVB-T, and ISDB-T systems. MPEG-2 defines the protocols for digital compression of the video and audio data. These video coding "profiles" allow for the coding of four source formats, ranging from VCR quality to full HDTV, each profile requiring progressively higher bit rates. Several compression tools are also available, each higher level being of increased sophistication. The sophistication of each level affects the video quality and receiver complexity for a given bit rate. In general, the higher the bit rate, the higher the video and audio quality. Tests indicate that studio-quality video can be achieved with a bit rate of about 9 Mb/s. Consumer-quality video can be achieved with a bit rate ranging from 2.5 to 6 Mb/s, depending on video content.

Audio compression takes advantage of acoustic masking of low-level sounds at nearby frequencies by coding these at low data rates. Other audio components that cannot be heard are not coded. The result is audio quality approaching that of a compact disk at a relatively low data rate. The transport format and protocol are based on a fixed-length packet defined and optimized for digital television delivery. Elementary bit streams from the audio, video, and data encoders are packetized and multiplexed to form the transport bit stream. Complementary recovery of the elementary bit streams is made at the receiver.

The transport stream is designed to accommodate a single HDTV program or several standard definition programs, depending on the broadcaster's objectives. Even in the case of HDTV, multiple data sources are multiplexed, with the multiplexing taking place at two distinct levels. This is illustrated in Figure 1-14. In the first level, program bit streams are formed by multiplexing packetized elementary streams from one or more sources. These packets may be coded video, coded audio, or data. Each of these contain timing information to assure that each is decoded in proper sequence.

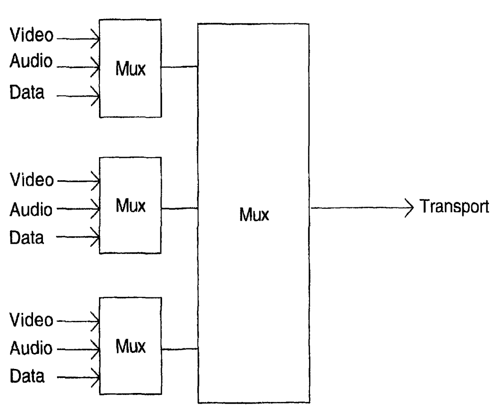

Figure 1-14. MPEG-2 multiplexing.

A typical program might include video, several audio channels, and multiple data streams. In the second level of multiplexing, many single programs are combined to form a system of programs. The content of the transport stream may be varied dynamically depending on the information content of the program sources. If the bit rate of the multiplexed packets is less than the required output bit rate, null packets are inserted so that the sum of the bit rates matches the constant bit rate output requirement. All program sources share a common clock reference. The transport stream must include information that describes the contents of the complete data stream and access control information, and may include internal communications data. Scrambling for the purpose of conditional access and teletext data may also be accommodated. An interactive program guide and certain system information may be included.

As implemented in the ATSC system, the video and audio sampling and transport encoders are frequency locked to a 27-MHz clock. The transport stream data rate and the symbol rate are related to this clock. If the studio and transmitter are colocated, the output of the transport stream may be connected directly to the transmitter. In many cases, the transport stream will be transmitted via a studio-to-transmitter link (STL) to the main transmitter site. This requires demodulation and decoding of the STL signal to recover the transport stream prior to modulation and coding in the DTV, DVB-T, or ISDB-T transmitter.