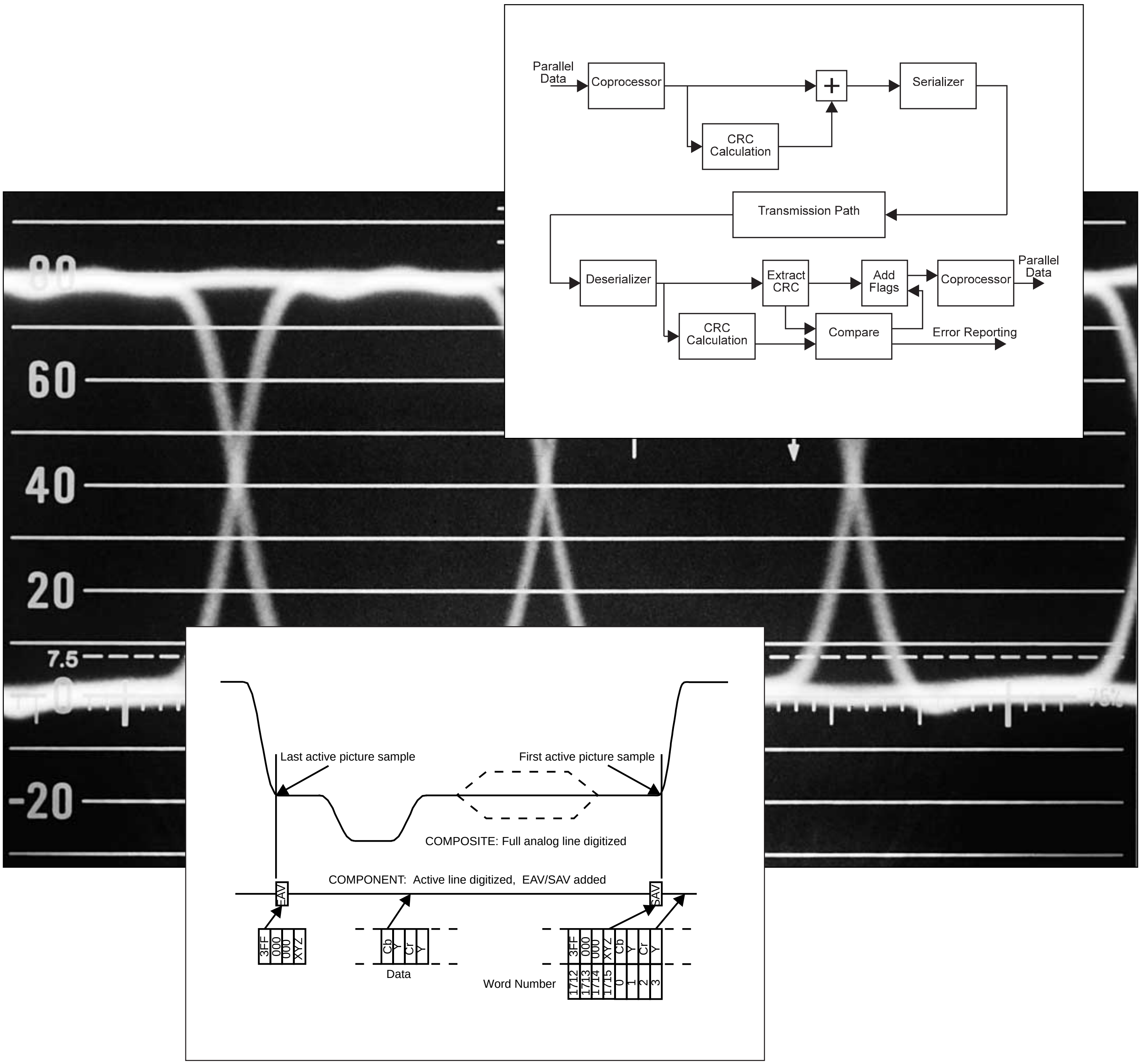
A Guide to Digital Television Systems and Measurements



A Guide to Digital Television Systems and Measurements

Acknowledgment

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Contents

1. Introduction

New methods of test and measurement are required to determine the health of the digital video transmission system and the signals it carries. Analysis methods for the program video and audio signals are well known. However, measurements for the serial form of the digitized signal differ greatly from those used to analyze the familiar baseband video signal.

This guide addresses many topics of interest to designers and users of digital television systems. You'll find information on embedded audio (Section 3), system timing issues (Section 4), and system timing measurements (Section 6). Readers who are already familiar with digital television basics may wish to start with Section 4, System Hardware and Issues. Discussion of specific measurement topics begins in Section 6.

The following major topics are covered in this booklet:

1. A definition of digital television including digital basics, video standards, conversions between video signal forms, and digital audio formats and standards.
2. Considerations in selecting transmission system passive components such as cable, connectors and related impedance matching concerns are discussed as well as continued use of the passive loop-through concept. Much of the transmission system methods and hardware already in place may be used for serial digital.
3. New and old systems issues that are particularly important for digital video are discussed, including equalization, reclocking, and system timing.
4. Test and measurement of serial digital signals can be characterized in three different aspects: usage, methods, and operational environment. Types of usage include: designer's workbench, manufacturing quality assurance, user equipment evaluation, system installation and acceptance testing, system and equipment maintenance, and perhaps most important, operation.
5. Several measurement methods are covered in detail, including serial waveform measurements, definition and detection of errors, jitter effects and measurements, and system testing with special test signals.

Component and Composite (analog)

Image origination sources, such as cameras and telecines, internally produce color pictures as three, full bandwidth signals – one each for Green, Blue and Red. Capitalizing on human vision not being as acute for color as it is for brightness level, television signals are generally transformed into luminance and color difference signals as shown in Figure 2-1. Y, the luminance signal, is derived from the GBR component colors, based on an equation such as:

$$Y = 0.59G + 0.30R + 0.11B$$

The color difference signals operate at reduced bandwidth, typically one-half of the luminance bandwidth. In some systems, specifically NTSC, the color difference signals have even lower and unequal bandwidths. Component signal formats and voltage levels are not standardized for 525-line systems, whereas there is an EBU document (EBU N-10) for 625 line systems. Details of component analog systems are fully described in another booklet from Tektronix, **Solving the Component Puzzle**. It's important to note that signal level values in the color difference domain allow combinations of Y, B-Y, R-Y that will be out of legal gamut range when converted to GBR. Therefore, there's a need for gamut checking of color difference signals when making operational adjustments for either the analog or digital form of these signals.

Most television signals today are two field-per-frame "interlaced." A frame contains all the scan lines for a picture, 525 lines in 30 frame/second applications and 625 lines in 25 frame/second applications. Interlacing increases the temporal resolution by providing twice as many fields per second, each one with only half the lines. Every other picture line is represented in the first field and the others are filled in by the second field.

Another way to look at interlace is that it's bandwidth reduction. Display rates of 50 or more presentations per second are required to eliminate perceived flicker. By using two-field interlace, the bandwidth for transmission is reduced by a factor of two. Interlace does produce artifacts in pictures with high vertical information content. However, for television viewing, this isn't generally considered a problem when contrasted to the cost of twice the bandwidth. In computer display applications, the artifacts of interlace are unacceptable so progressive scanning is used, often with frame rates well above 60 Hz. As television (especially high definition) and computer applications become more integrated there will be a migration from interlace to progressive scanning for all applications.

Further bandwidth reduction of the television signal occurs when it's encoded into composite PAL or NTSC as shown in Figure 2-2. NTSC is defined for studio operation by SMPTE 170M which formalizes and updates the never fully approved, RS 170A. Definitions for PAL and NTSC (as well as SECAM) can be found in ITU-R (formally CCIR) Report 624. Where each GBR signal may have a 6 MHz bandwidth, color difference signals would typically have Y at 6 MHz and each color difference signal at 3 MHz; however, a composite signal is one channel of 6 MHz or less. The net result is a composite 6 MHz channel carrying color fields at a 60 per second rate that in its uncompressed, progressive scan form, would have required three 12 MHz channels for a total band-

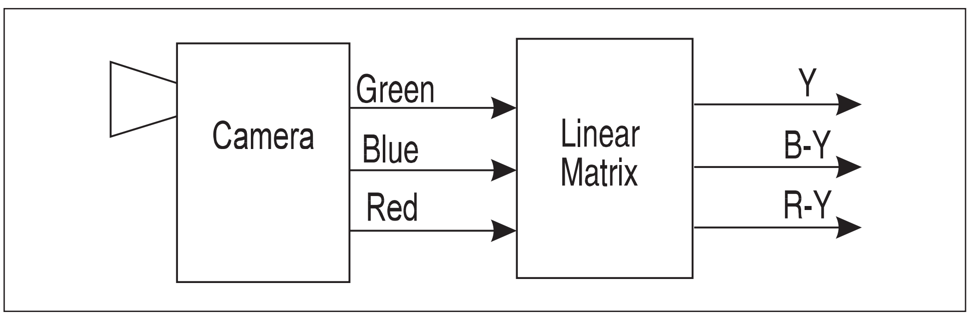

Figure 2-1. Component signals.

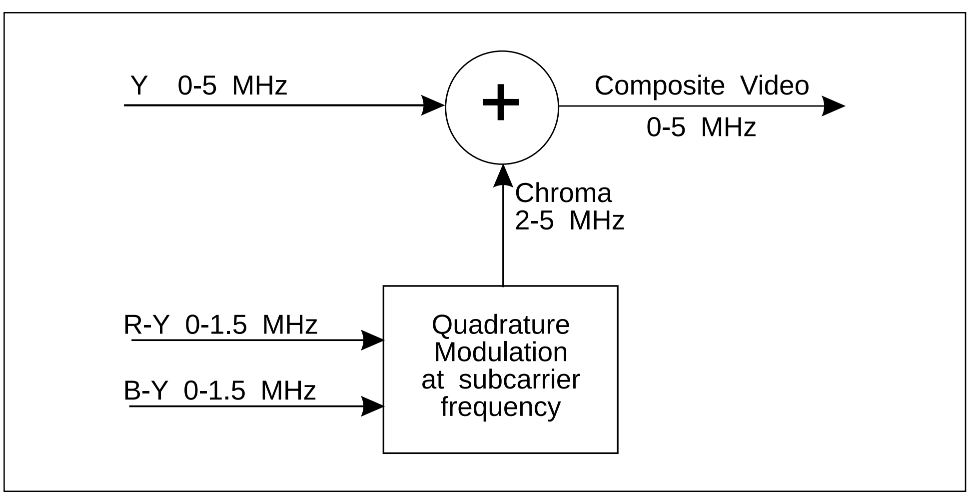

Figure 2-2. Composite encoding.

width of 36 MHz. So, data compression is nothing new; digital just makes it easier.

For composite NTSC signals, there are additional gamut considerations when converting from the color difference domain. NTSC transmitters will not allow 100 percent color amplitude on signals with high luminance level (such as yellow). This is because the transmitter carrier will go to zero for signals greater than about 15 percent above 1 volt. Hence, there's a lower gamut limit for some color difference signals to be converted to NTSC for RF transmission.

As part of the encoding process, sync and burst are added as shown in Figure 2-3. Burst provides a reference for decoding back to components. The phase of burst with respect to sync edge is called SCH phase, which must be carefully controlled in most studio applications. For NTSC, 7.5 IRE units of setup is added to the luminance signal. This poses some conversion difficulties, particularly when decoding back to component. The problem is that it's relatively easy to add setup; but removing it when the amplitudes and timing of setup in the composite domain are not well controlled can result in black level errors and/or signal disturbances at the ends of the active line.

Sampling and Quantizing

The first step in the digitizing process is to "sample" the continuous variations of the analog signal as shown in Figure 2-4. By looking at the analog signal at discrete time intervals, a sequence of voltage samples can be stored, manipulated and later, reconstructed.

In order to recover the analog signal accurately, the sample rate must be rapid enough to avoid missing important information. Generally, this requires the sampling frequency to be at least twice the highest analog frequency. In the real world, the frequency is a little higher than twice. (The Nyquist Sampling Theorem says that the interval between successive samples must be equal to or less than one-half of the period of the highest frequency present in the signal.)

The second step in digitizing video is to "quantize" by assigning a digital number to the voltage levels of the sampled analog signal – 256 levels for 8-bit video, 1024 for 10-bit video, and up to several thousand for audio.

To get the full benefit of digital, 10-bit processing is required. While most current tape machines are 8-bit, SMPTE 125M calls out 10-bit as the interface standard. Processing at less than 10 bits may cause truncation and rounding artifacts, particularly in electronically generated pictures. Visible defects will be revealed in the picture if the quantization levels are too coarse (too few levels). These defects can appear as "contouring" of the picture. However, the good news is that random noise and picture details present in most live video signals actually help conceal these contouring effects by adding a natural randomness to them. Sometimes the number of quantizing levels must be reduced; for example, when the output of a 10-bit processing device feeds an 8-bit recorder. In this case, contouring effects are minimized by deliberately adding a small amount of random noise (dither) to the signal. This technique is known as randomized rounding.

Digital Video Standards

Although early experiments with digital technology were based on sampling the composite (NTSC or PAL) signal, it was realized that for the highest quality operation, component processing was necessary. The first digital standards were component. Interest in composite digital was revived when Ampex and Sony announced a composite digital recording format, which became known as

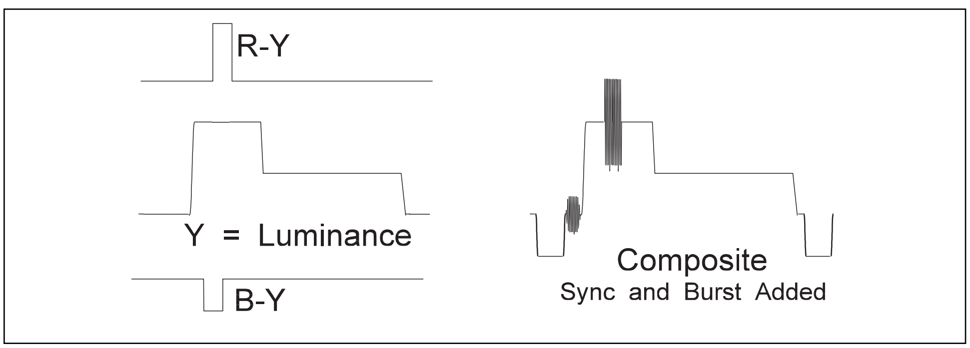

Figure 2-3. Color difference and composite waveforms.

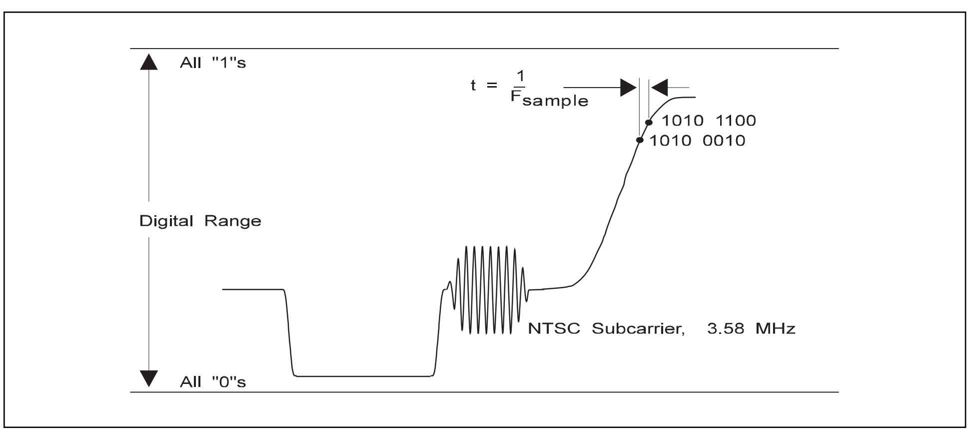

Figure 2-4. Sampling an analog signal.

D-2. Initially, these machines were projected as analog in/out devices for use in existing analog NTSC and PAL environments; digital inputs and outputs were provided for machine-to-machine dubs. However, the post-production community recognized that greater advantage could be taken of the multi-generation capability of these machines if they were used in an all digital environment.

Rec. 601. Recommendation ITU-R BT.601 (formerly CCIR Recommendation 601) is not a video interface standard, but a sampling standard. Rec. 601 evolved out of a joint SMPTE/EBU task force to determine the parameters for digital component video for the 525/59.94 and 625/50 television systems. This work culminated in a series of tests sponsored by SMPTE in 1981, and resulted in the well known CCIR Recommendation 601. This document specified the sampling mechanism to be used for both 525 and 625 line signals. It specified orthogonal sampling at 13.5 MHz for luminance, and 6.75 MHz for the two color difference signals C_B and C_R, which are scaled versions of the signals B-Y and R-Y.

The sampling structure defined is known as "4:2:2." This nomenclature is derived from the days when multiples of NTSC subcarrier were being considered for the sampling frequency. This approach was abandoned, but the use of "4" to represent the luminance sampling frequency was retained. The task force mentioned above examined luminance sampling frequencies from 12 MHz to 14.3 MHz. They selected 13.5 MHz as a compromise because the submultiple 2.25 MHz is a factor common to both 525- and 625-line system. Some extended definition TV systems use a higher resolution format called 8:4:4 which has twice the bandwidth of 4:2:2.

Parallel component digital. Rec. 601 described the sampling of the signal. Electrical interfaces for the data produced by this sampling were standardized separately by SMPTE and the EBU. The parallel interface for 525/59.94 was defined by SMPTE as SMPTE Standard 125M (a revision of the earlier RP-125), and for 625/50 by EBU Tech 3267 (a revision of the earlier EBU Tech 3246). Both of these were adopted by CCIR and are included in Recommendation 656, the document defining the hardware interface.

The parallel interface uses eleven twisted pairs and 25-pin "D" connectors. (The early documents specified slide locks on the connectors; later revisions changed the retention mechanism to 4/40 screws.) This interface multiplexes the data words in the sequence C_B, Y, C_R, Y, C_B…, resulting in a data rate of 27 Mwords/s. The timing sequences SAV and EAV were added to each line to represent Start of Active Video and End of Active Video. The digital active line contains 720 luminance samples, and includes space for representing analog blanking within the active line.

Rec. 601 specified eight bits of precision for the data words representing the video. At the time the standards were written, some participants suggested that this may not be adequate, and provision was made to expand the interface to 10-bit precision. Ten-bit operation has indeed proved beneficial in many circumstances, and the latest revisions of the interface standard provide for a 10-bit interface, even if only eight bits are used. Digital-to-analog conversion range is chosen to provide headroom above peak white and footroom below black as shown in Figure 2-5. Quantizing levels for black and white are selected so the 8-bit levels with two "0"s added will have the same values as the 10-bit levels. Values 000 to 003 and 3FF to 3FC are reserved for synchronizing

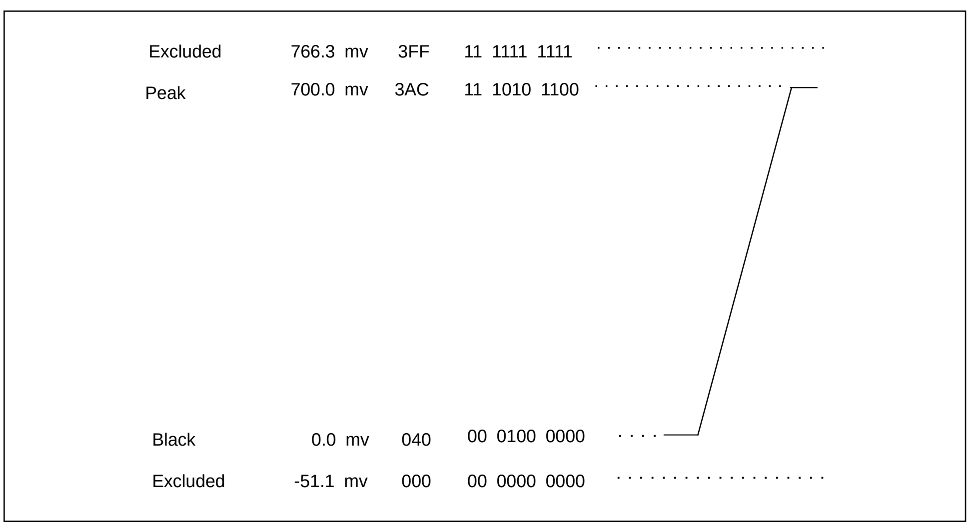

Figure 2-5. Luminance quantizing.

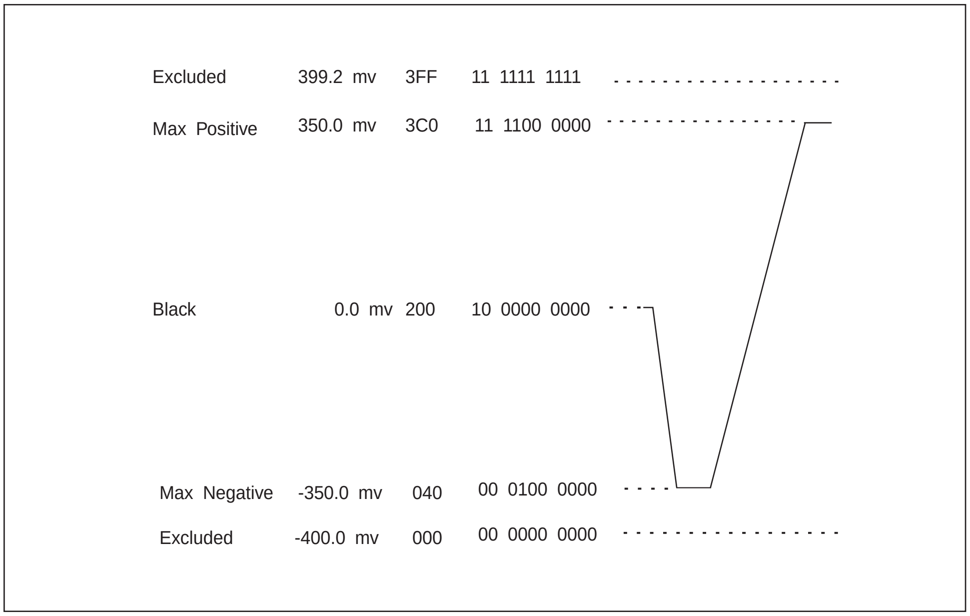

Figure 2-6. Color difference quantizing.

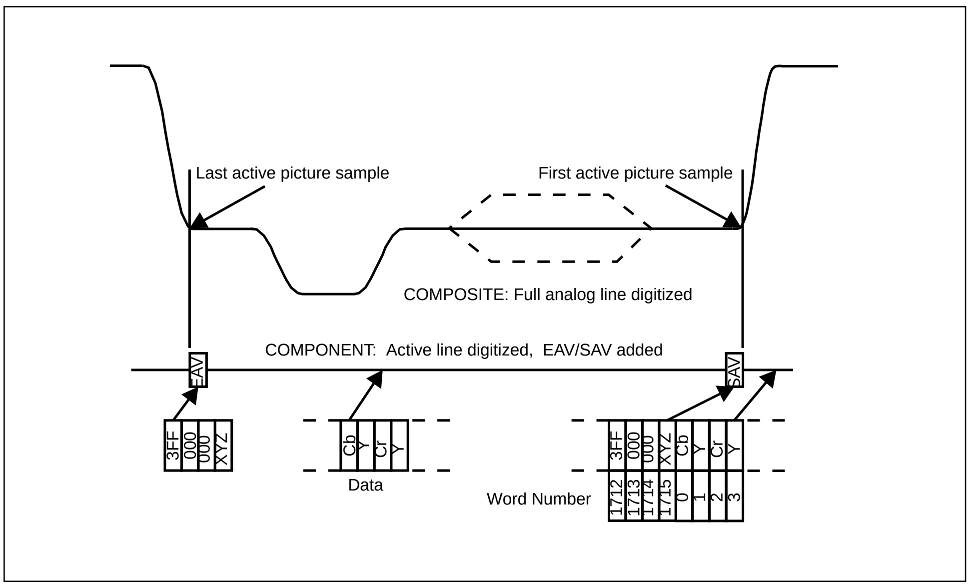

Figure 2-7. Digital horizontal line.

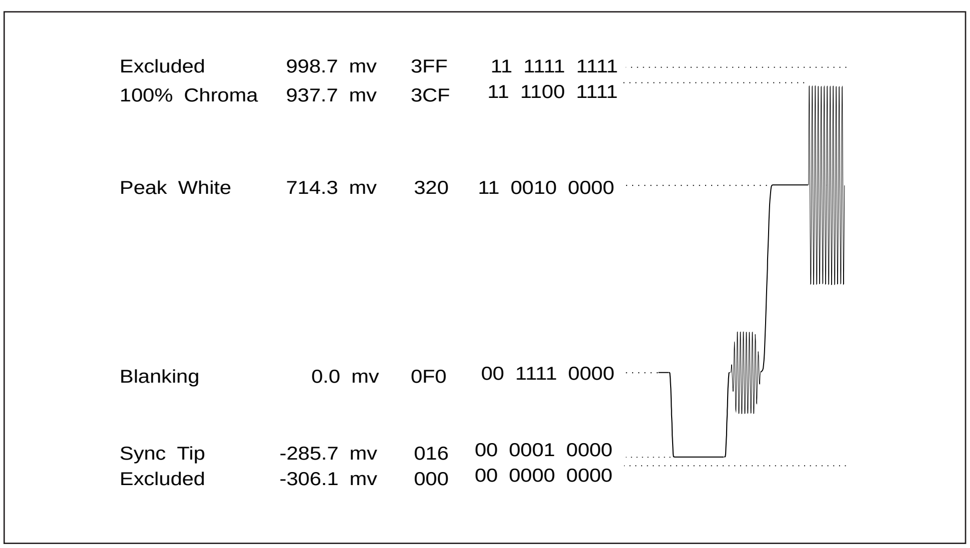

Figure 2-8. NTSC quantizing.

purposes. Similar factors determine the quantizing values for color difference signals as shown in Figure 2-6.

Figure 2-7 shows the location of samples and digital words with respect to an analog horizontal line. Because the timing information is carried by EAV and SAV, there is no need for conventional synchronizing signals, and the horizontal intervals (and the active line periods during the vertical interval) may be used to carry ancillary data. The most obvious application for this data space is to carry digital audio, and documents are being prepared by SMPTE to standardize the format and distribution of the audio data packets.

Rec. 601/656 is a well proven technology with a full range of equipment available for production and post production. Generally, the parallel interface has been superseded by a serial implementation, which is far more practical in larger installations. Rec. 601 provides all of the advantages of both digital and component operation. It's the system of choice for the highest possible quality in 525- or 625-line systems.

Parallel composite digital. The composite video signal is sampled at four times the (NTSC or PAL) subcarrier frequency, giving nominal sampling rates of 14.3 MHz for NTSC and 17.7 MHz for PAL. The interface is standardized for NTSC as SMPTE 244M and, at the time of writing, EBU documentation is in process for PAL. Both interfaces specify ten-bit precision, although D-2 and D-3 machines record only eight bits to tape. Quantizing of the NTSC signal (shown in Figure 2-8) is defined with a modest amount of headroom above 100% bars, a small footroom below sync tip and the same excluded values as for component.

PAL composite digital has been defined to minimize quantizing noise by using a maximum amount of the available digital range. As can be seen in Figure 2-9, the peak analog values actually exceed the digital dynamic range, which might appear to be a mistake. Because of the specified sampling axis, reference to subcarrier and the phase of the highest luminance level bars (such as yellow), the samples never exceed the digital dynamic range. The values involved are shown in Figure 2-10.

Like the component interface, the composite digital active line is long enough to accommodate the analog active line and the analog blanking edges. Unlike the component interface, the composite interface transmits a digital representation of conventional sync and burst during the horizontal blanking interval. A digital representation of vertical sync and equalizing pulses is also transmitted over the composite interface.

Composite digital installations provide the advantages of digital processing and interface, and particularly the multi-generation capabilities of digital recording. However, there are some limitations. The signal is composite and does bear the footprint of NTSC or PAL encoding, including the narrowband color information inherent to these coding schemes. Processes such as chroma key are generally not satisfactory for high quality work, and separate provision must be made for a component derived keying signal. Some operations, such as digital effects, require that the signal be converted to component form for processing, and then re-encoded to composite. Also, the full excursion of the composite signal has to be represented by 256 levels on 8-bit recorders. Nevertheless, composite digital provides a more powerful environment than analog for NTSC and PAL installations and is a very cost effective solution for many users.

As with component digital, the parallel composite interface uses a multi-pair cable and 25-pin "D" connectors. Again, this has proved to be satisfactory for small and medium sized installations, but practical implementation of a large system requires a serial interface.

16:9 widescreen component digital signals. Many television markets around the world are seeing the introduction of delivery systems for widescreen (16:9 aspect ratio) pictures. Some of these systems, such as MUSE in Japan and the future ATV system in the USA, are intended for high-definition images. Others, such as PALplus in Europe will use 525- or 625-line pictures. Domestic receivers with 16:9 displays have been introduced in many markets, and penetration will likely increase significantly over the next few years. For most broadcasters, this creates the need for 16:9 program material.

Two approaches have been proposed for the digital representation of 16:9 525- and 625-line video. The first method retains the sampling frequencies of the Rec. 601 standard for 4:3 images (13.5 MHz for luminance). This "stretches" the pixel represented by each data word by a factor of 1.33 horizontally, and results in a loss of 25 percent of the horizontal spatial resolution when compared to Rec. 601 pictures. For some applications, this method still provides adequate resolution, and it has the big advantage that much

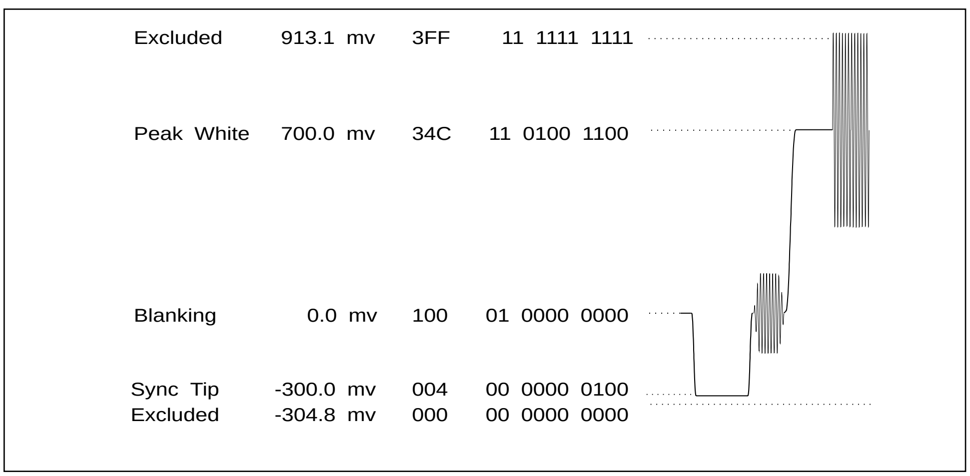

Figure 2-9. PAL quantizing.

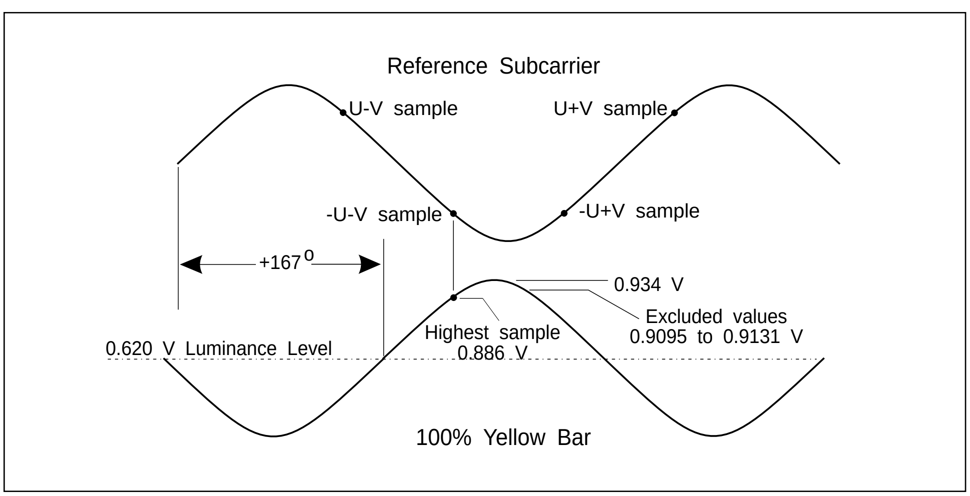

Figure 2-10. PAL yellow bar sampling.

existing Rec. 601 equipment can be used.

A second method maintains spatial resolution where pixels (and data words) are added to represent the additional width of the image. This approach results in 960 luminance samples for the digital active line (compared to 720 samples for a 4:3 picture, and to 1920 samples for a 16:9 high definition picture). The resulting sampling rate for luminance is 18 MHz. This system provides the same spatial resolution as Rec. 601 4:3 images; but existing equipment designed only for 13.5 MHz cannot be used. The additional resolution of this method may be advantageous when some quality overhead is desired for post production or when up-conversion to a high-definition system is contemplated.

SMPTE 267M provides for both the 13.5 MHz and 18 MHz systems. Because 16:9 signals using 13.5 MHz sampling are electrically indistinguishable from 4:3 signals, they can be conveyed by the SMPTE 259M serial interface at 270 Mb/s. It's planned that SMPTE 259M (discussed below) will be revised to provide for serial transmission of 18 MHz sampled signals, using the same algorithm, at a data rate of 360 Mb/s.

Serial Digital Video

Parallel connection of digital equipment is practical only for relatively small installations, and there's a clear need for transmission over a single coaxial cable. This is not simple as the data rate is high, and if the signal were transmitted serially without modification, reliable recovery would be very difficult. The serial signal must be modified prior to transmission to ensure that there are sufficient edges for reliable clock recovery, to minimize the low-frequency content of the transmitted signal, and to spread the transmitted energy spectrum so that radio-frequency emission problems are minimized.

In the early 1980s, a serial interface for Rec. 601 signals was recommended by the EBU. This interface used 8/9 block coding and resulted in a bit rate of 243 Mb/s. Its interface did not support ten-bit precision signals, and there were some difficulties in producing reliable, cost effective, integrated circuits. The block coding based interface was abandoned and has been replaced by an interface with channel coding that uses scrambling and conversion to NRZI. The serial interface has been standardized as SMPTE 259M and EBU Tech. 3267, and is defined for both component and composite signals including embedded digital audio.

Conceptually, the serial digital interface is much like a carrier system for studio applications. Baseband audio and video signals are digitized and combined on the serial digital "carrier" as shown in Figure 2-11. (It's not strictly a carrier system in that it's a baseband digital signal, not a signal modulated on a carrier.) The bit rate (carrier frequency) is determined by the clock rate of the digital data: 143 Mb/s for NTSC, 177 Mb/s for PAL and 270 Mb/s for Rec. 601 component digital. The widescreen (16:9) component system defined in SMPTE 267 will produce a bit rate of 360 Mb/s.

Parallel data representing the samples of the analog signal is processed as shown in Figure 2-12 to create the serial digital data stream. The parallel clock is used to load sample data into a shift register, and a times-ten multiple of the parallel clock shifts the bits out, LSB (least significant bit) first, for each 10-bit data word. If only 8 bits of data are available at the input, the serializer places

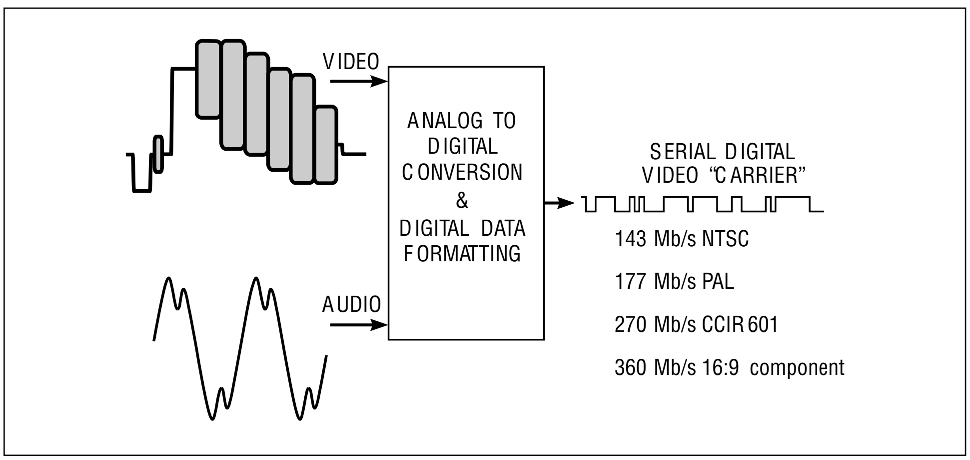

Figure 2-11. The carrier concept.

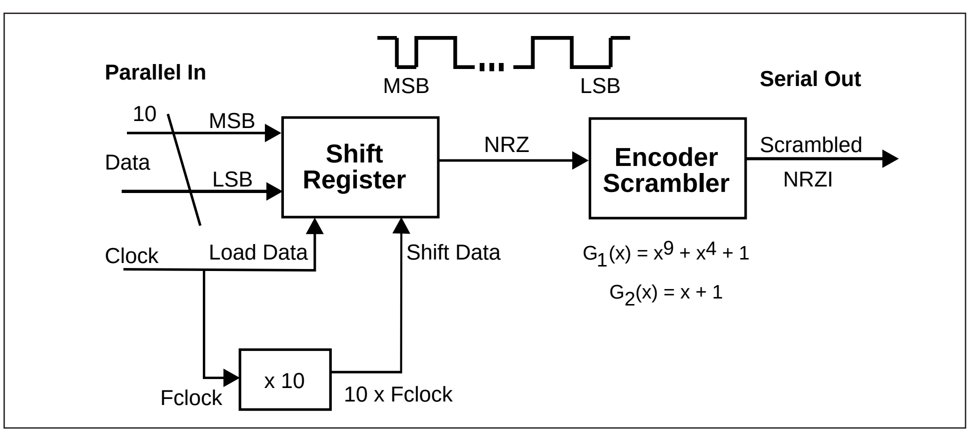

Figure 2-12. Parallel-to-serial conversion.

zeros in the two LSBs to complete the 10-bit word. Component signals do not need further processing as the SAV and EAV signals on the parallel interface provide unique sequences that can be identified in the serial domain to permit word framing. If ancillary data such as audio has been inserted into the parallel signal, this data will be carried by the serial interface. The serial interface may be used with normal video coaxial cable.

The conversion from parallel to serial for composite signals is somewhat more complex. As mentioned above, the SAV and EAV signals on the parallel component interface provide unique sequences that can be identified in the serial domain. The parallel composite interface does not have such signals, so it is necessary to insert a suitable timing reference signal (TRS) into the parallel signal before serialization. A diagram of the serial digital NTSC horizontal interval is shown in Figure 2-13. The horizontal interval for serial digital PAL would be similar except that the sample location is slightly different on each line, building up two extra samples per field. A three-word TRS is inserted in the sync tip to enable word framing at the serial receiver, which should also remove the TRS from the received serial signal.

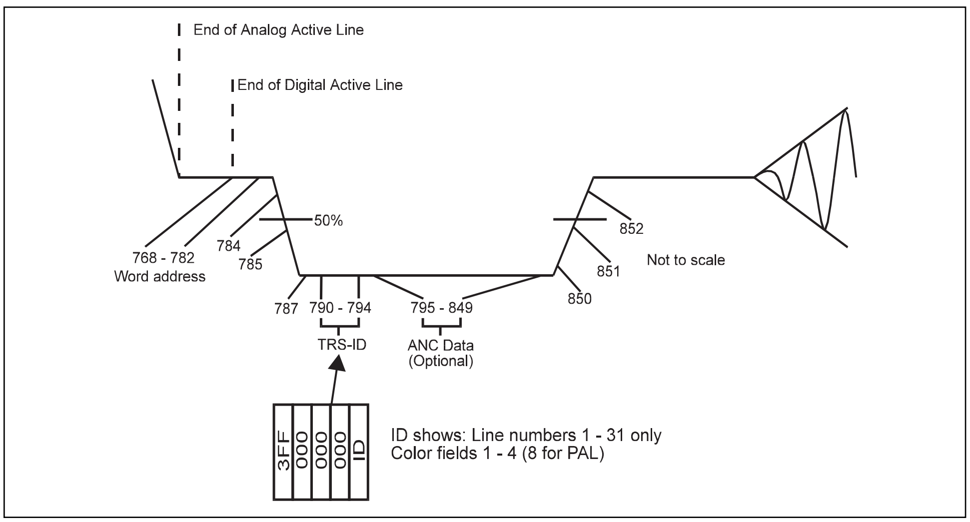

Figure 2-13. NTSC horizontal interval.

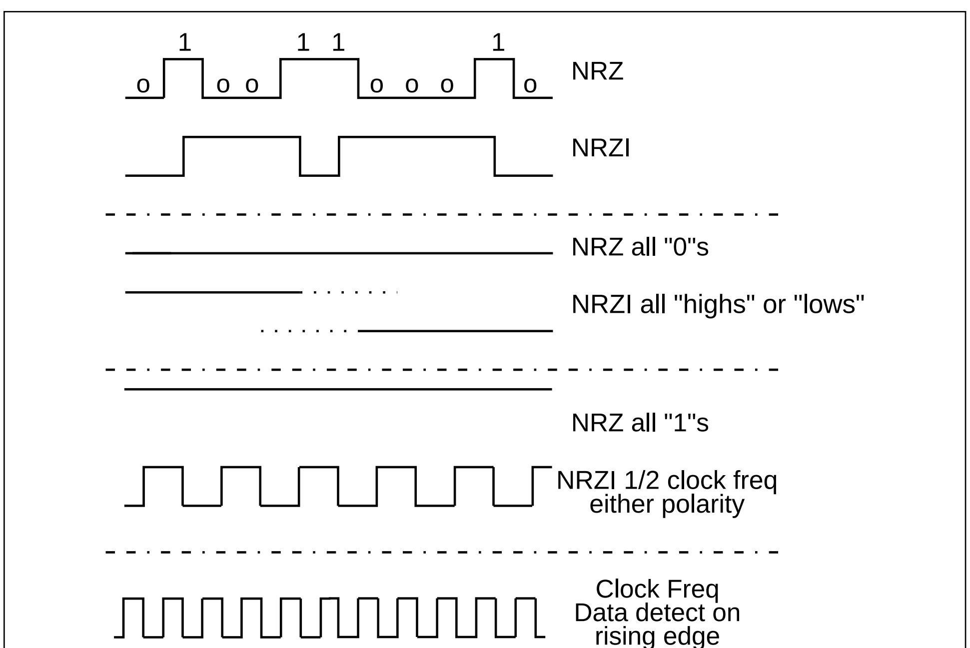

Figure 2-14. NRZ and NRZI relationship.

The composite parallel interface does not provide for transmission of ancillary data, and the transmission of sync and burst means that less room is available for insertion of data. Upon conversion from parallel to serial, the sync tips may be used. However, the data space in NTSC is sufficient for four channels of AES/EBU digital audio. Ancillary data such as audio may be added prior to serialization, and this would normally be performed by the same co-processor that inserts the TRS.

Following the serialization of the parallel information the data stream is scrambled by a mathematical algorithm then encoded into NRZI (non-return to zero inverted) by a concatenation of the following two functions:

$$G_1(X) = X^9 + X^4 + 1$$

$$G_2(X) = X + 1$$

At the receiver the inverse of this algorithm is used in the deserializer to recover the correct data. In the serial digital transmission system, the clock is contained in the data as opposed to the parallel system where there's a separate clock line. By scrambling the data, an abundance of transitions is assured as required for clock recovery. The mathematics of scrambling and descrambling lead to some specialized test signals for serial digital systems that are discussed later in this guide.

Encoding into NRZI makes the serial data stream polarity insensitive. NRZ (non return to zero, an old digital data tape term) is the familiar circuit board logic level, high is a "1" and low is a "0." For a transmission system it's convenient to not require a certain polarity of the signal at the receiver. As shown in Figure 2-14, a data transition is used to represent each "1" and there's no transition for a data "0." The result's that it's only necessary to detect transitions; this means either polarity of the signal may be used. Another result of NRZI encoding is that a signal of all "1"s now produces a transition every clock interval and results in a square wave at one-half the clock frequency. However, "0"s produce no transition, which leads to the need for scrambling. At the receiver, the rising edge of a square wave at the clock frequency would be used for data detection.

Rate Conversion – Format Conversion

When moving between component digital and composite digital in either direction, there are two steps: the actual encoding or decoding, and the conversion of the sample rate from one standard to the other. The digital sample rates for these two formats are different: 13.5 MHz for component digital and 14.3 MHz for NTSC composite digital (17.7 MHz for PAL). This second step is called "rate conversion." Often the term rate conversion is used to mean both encoding/decoding and resampling of digital rates. Strictly speaking, rate conversion is taking one sample rate and making another sample rate out of it. For our purposes, we'll use the term "format conversion" to mean both the encode/decode step and resampling of digital rates. The format conversion sequence depends on direction. For component to composite, the usual sequence is rate conversion followed by encoding. For composite to component, the sequence is decoding followed by rate conversion. See Figure 2-15.

It's much easier to do production in component because it isn't necessary to wait for every color framing boundary (four fields for NTSC and eight for PAL) to match two pieces of video together; instead, they can be matched every two fields (motion frame). This provides four times the opportunity. Additionally, component is a higher quality format since luminance and chrominance are handled separately. To the extent possible, origination for component environment production should be accomplished component. A high quality composite-to-component format converter provides a reasonable alternative.

After post-production work, component digital often needs to be converted to composite digital. Sources that are component digital may be converted for input to a composite digital switcher or effects machine. Or a source from a component digital telecine may be converted for input to a composite digital suite. Furthermore, tapes produced in component digital may need to be distributed or archived in composite digital.

The two major contributors to the quality of this process are the encoding or decoding process and the sample rate conversion. If either one is faulty, the quality of the final product suffers. In order to accurately change the digital sample rate, computations must be made between two different sample rates and interpolations must be calculated between the physical location of the source pixel data and the physical location of destination pixel data. Of the 709,379 pixel locations in a PAL composite digital frame, all except one must be mapped (calculated). In order to accomplish this computationally intense conversion, an extremely accurate algorithm must be used. If the algorithm is accurate enough to deal with PAL conversions, NTSC conversions using the same algorithm are simple. To ensure quality video, a sophisticated algorithm must be employed and the hardware must produce precise coefficients and minimize rounding errors.

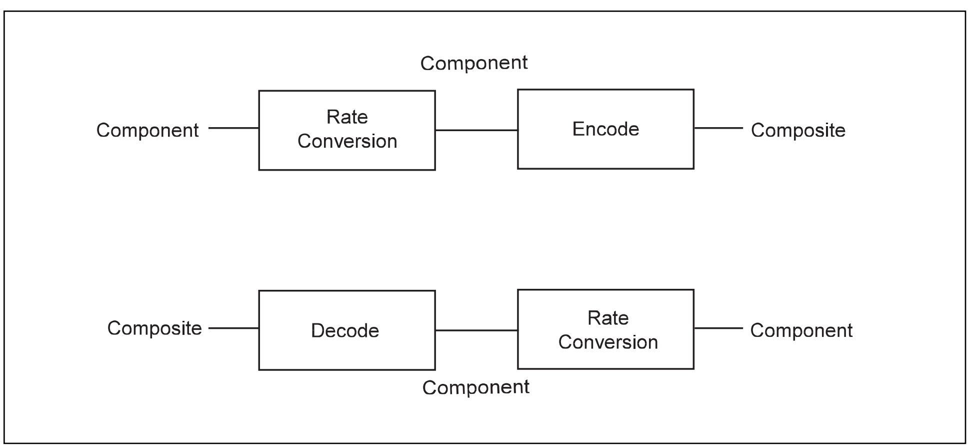

Figure 2-15. Format conversion.

3. Digital Audio

AES/EBU Audio Data Format

AES digital audio (also known as AES/EBU) conforms to the specification AES3 (ANSI 4.40) titled "AES recommended practice for digital audio engineering – Serial transmission format for two-channel linearly represented digital audio data." AES/EBU digital audio is the result of cooperation between the Audio Engineering Society and the European Broadcasting Union.

When discussing digital audio, one of the important considerations is the number of bits per sample. Where video operates with 8 or 10 bits per sample, audio implementations range from 16 to 24 bits to provide the desired dynamic range and signal-to-noise ratio (SNR). The basic formula for determining the SNR for digital audio is:

SNR = (6.02 * n) + 1.76

where "n" is the number of bits per sample

For a 16-bit system, the maximum theoretical SNR would be (6.02 * 16) + 1.76 = 98.08 dB; for an 18-bit system, the SNR would be 110.2 dB; and for a 20-bit device, 122.16 dB. A well-designed 20-bit ADC probably offers a value between 100 and 110 dB. Using the above formula for an SNR of 110 dB, this system has an equivalent 18.3-bit resolution.

An AES digital audio signal always consists of two channels which may be distinctly separate audio material or stereophonic audio. There's also a provision for single-channel operation (monophonic) where the second digital data channel is either identical to the first or has data set to logical "0." Formatting of the AES data is shown in Figure 3-1. Each sample is carried by a sub-frame containing: 20 bits of sample data, 4 bits of auxiliary data (which may be used to extend the sample to 24 bits), 4 other bits of data and a preamble. Two sub-frames make up a frame which contains one sample from each of the two channels.

Frames are further grouped into 192-frame blocks which define the limits of user data and channel status data blocks. A special preamble indicates the channel identity for each sample (X or Y preamble) and the start of a 192-frame block (Z preamble). To minimize the direct-current (DC) component on the transmission line, facilitate clock recovery, and make the interface polarity insensitive, the data is channel coded as biphase-mark. The preambles specifically violate the biphase-mark rules for easy recognition and to ensure synchronization. When digital audio is embedded in the serial digital video data stream, the start of the 192-frame block is indicated by the so-called "Z" bit which corresponds to the occurrence of the Z-type preamble.

The validity bit indicates whether the audio sample bits in the sub-frame are suitable for conversion to an analog audio signal. User data is provided to carry other information, such as time code. Channel status data contains information associated with each audio channel. There are three levels of implementation of the channel status data: minimum, standard, and enhanced. The standard implementation is recommended for use in professional television applications, hence the channel status data will contain information about signal emphasis, sampling frequency, channel mode (stereo, mono, etc.), use of auxiliary bits (extend audio data to 24 bits or other use), and a CRC (cyclic redundancy code) for error checking of the total channel status block.

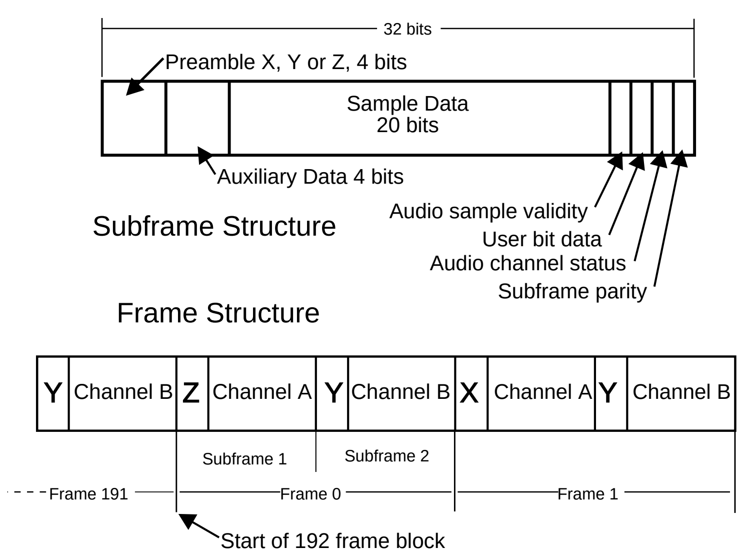

Figure 3-1. AES audio data formatting.

Embedded Audio

One of the important advantages of SDI (Serial Digital Interconnect) is the ability to embed (multiplex) several channels of digital audio in the digital video. This is particularly useful in large systems where separate routing of digital audio becomes a cost consideration and the assurance that the audio is associated with the appropriate video is an advantage. In smaller systems, such as a post production suite, it's generally more economical to maintain separate audio thus eliminating the need for numerous mux (multiplexer) and demux (demultiplexer) modules. In developing the SDI standard, SMPTE 259M, one of the key considerations was the ability to embed at least four channels of audio in the composite serial digital signal which has limited ancillary data space. A basic form of embedded audio for composite video is documented in SMPTE 259M and there's a significant amount of equipment in the field using this method for both composite and component signals. As engineers were using the SMPTE 259M specifications for their designs, it became clear that a more definitive document was needed. A draft proposed standard is being developed for embedded audio which includes such things as distribution of audio samples within the available composite digital ancillary data space, the ability to carry 24-bit audio, methods to handle non-synchronous clocking and clock frequencies other than 48 kHz, and specifications for embedding audio in component digital video.

Ancillary data space. Composite digital video allows ancillary data only in its serial form and then only in the tips of synchronizing signals. Ancillary data is not carried in parallel composite digital as parallel data is simply a digital representation of the analog signal. Figures 3-2 and 3-3 show the ancillary data space available in composite digital signals which includes horizontal sync tip, vertical broad pulses, and vertical equalizing pulses.

A small amount of the horizontal sync tip space is reserved for TRS-ID (timing reference signal and identification) which is required for digital word framing in the deserialization process. Since composite digital ancillary data is only available in the serial domain, it consists of 10-bit words, whereas both composite and component parallel interconnects may be either 8 or 10 bits. A summary of composite digital ancillary data space for NTSC is shown in Table 3-1. Thirty frames/second will provide 9.87 Mb/second of ancillary data which is enough for four channels of embedded audio with a little capacity left over. PAL has slightly more ancillary data space due to the higher sample rate; however

Table 3-1. NTSC Digital Ancillary Data Space

Pulse Tip	Words	Per Frame	Total
Horizontal	55	507	27,885
Equalizing	21	24	504
Broad	376	12	4,512
	Total 10-bit words/frame		32,901

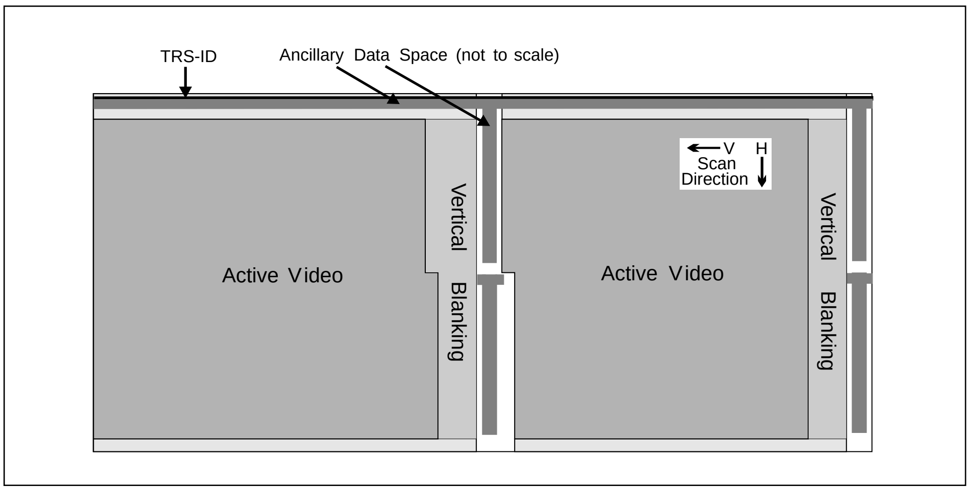

Figure 3-2. Composite ancillary data space.

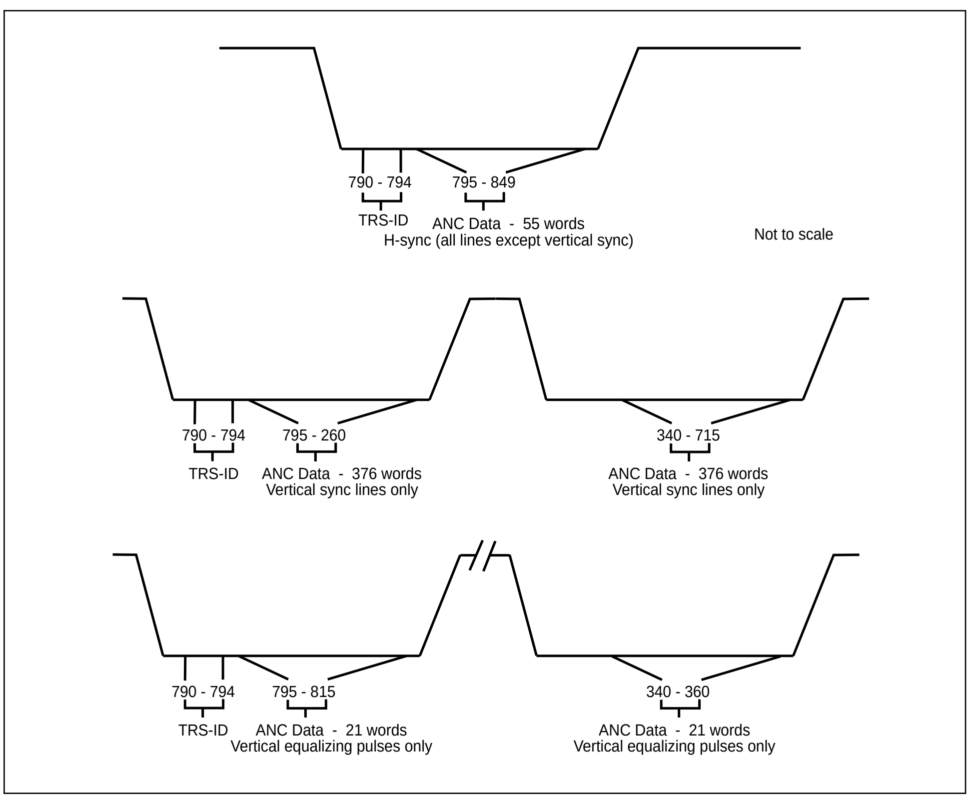

Figure 3-3. NTSC available ancillary data words.

embedded audio for digital PAL is rarely used.

There's considerably more ancillary data space available in component digital video as shown in Figure 3-4. All of the horizontal and vertical blanking intervals are available except for the small amount required for EAV (end of video) and SAV (start of video) synchronizing words.

The ancillary data space has been divided into two types HANC (horizontal ancillary data) and VANC (vertical ancillary data) with the SMPTE defining the use of HANC and the EBU defining the use of VANC. Word lengths are 10-bit for HANC specifically so it can carry embedded audio in the same format as for composite digital and 8-bit for VANC specifically so it can be carried by D-1 VTRs (on those vertical interval lines that are recorded). Total data space is shown in Table 3-2. Up to 16 channels of embedded audio are specified for HANC in the proposed standard and there's room for significantly more data.

Table 3-2. Component Digital Ancillary Data Space

525 line	Words	Per Frame	Total	Rate Mb/s
HANC	268	525	140,700	42.2
VANC	1440	39	56,160	13.5
		Total words	198,860	55.7
625 line	**Words**	**Per Frame**	**Total**	**Rate Mb/s**
HANC	282	625	176,250	44.1
VANC	1440	49	70.560	14.1
		Total words	246,810	58.2

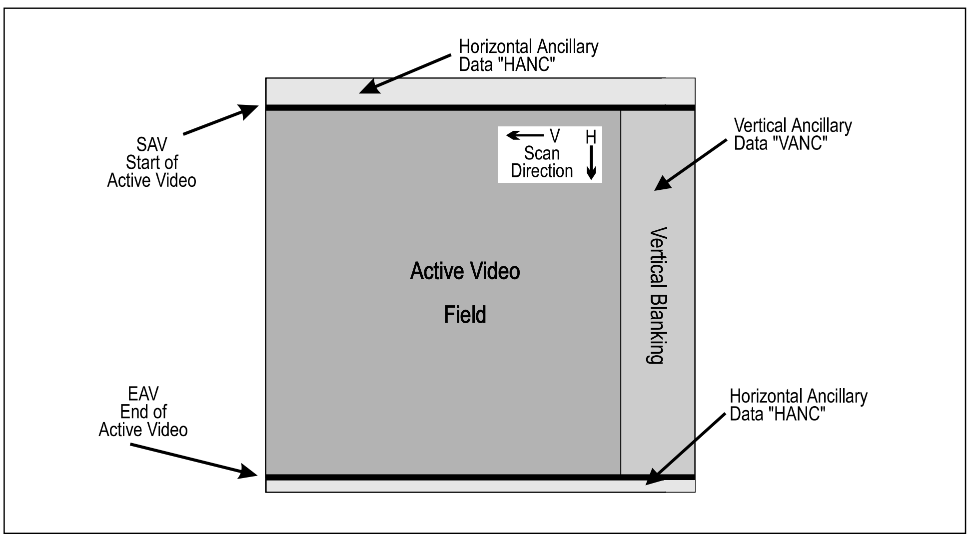

Figure 3-4. Component ancillary data space.

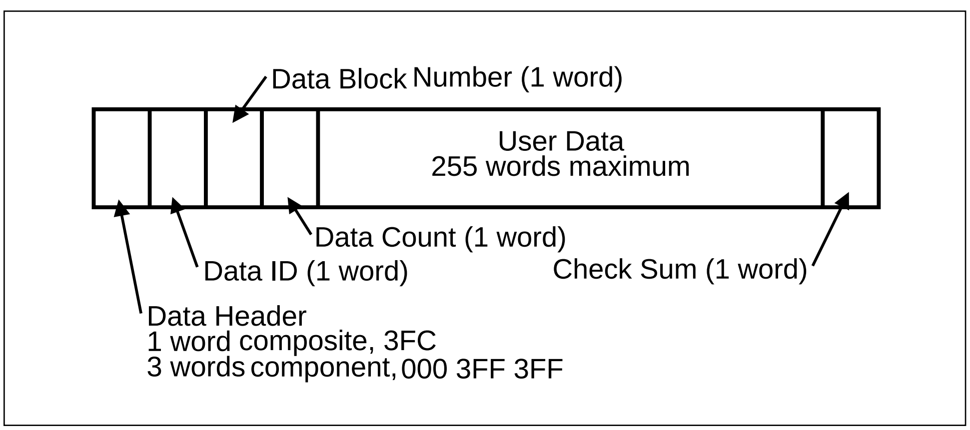

Figure 3-5. Ancillary data formatting.

Not all of the VANC data space is available. For instance, the luminance samples on one line per field are reserved for DVITC (digital vertical interval time code) and the chrominance samples on that line may be devoted to video index. Also, it would be wise to avoid using the vertical interval switch line and perhaps, the subsequent line where data might be lost due to relock after the switch occurs.

Ancillary data formatting. Ancillary data is formatted into packets prior to multiplexing it into the video data stream as shown in Figure 3-5. Each data block may contain up to 255 user data words provided there's enough total data space available to include five (composite) or seven (component) words of overhead. For composite digital, only the vertical sync (broad) pulses have enough room for the full 255 words. Multiple data packets may be placed in individual ancillary data spaces, thus providing a rather flexible data communications channel.

At the beginning of each data packet is a header using word values that are excluded for digital video data and reserved for synchronizing purposes. For composite video, a single header word of 3FCh is used whereas for component video, a three-word header 000h 3FFh 3FFh is used. Each type of data packet is identified with a different Data ID word. Several different Data ID words are defined to organize the various data packets used for embedded audio. The Data Block Number (DBN) is an optional counter that can be used to provide sequential order to ancillary data packets allowing a receiver to determine if there's missing data. As an example, with embedded audio the DBN may be used to detect the occurrence of a vertical interval switch, thereby allowing the receiver to process the audio data to remove the likely transient "click" or "pop." Just prior to the data is the Data Count word indicating the amount of data in the packet. Finally, following the data is a checksum which is used to detect errors in the data packet.

Basic embedded audio. Embedded audio defined in SMPTE 259M provides four channels of 20-bit audio data sampled at 48 kHz with the sample clock locked to the television signal. Although specified in the composite digital part of the standard, the same method is also used for component digital video.

This basic embedded audio corresponds to Level A in the proposed embedded audio standard. Other levels of operation provide more channels, other sampling frequencies, and additional information about the audio data. Basic embedded audio data packet formatting derived from AES audio is shown in Figure 3-6.

Two AES channel-pairs are shown as the source; however it's possible for each of the four embedded audio channels to come from a different AES signal (specifically implemented in some D-1 digital VTRs). The Audio Data Packet contains one or more audio samples from up to four audio channels. 23 bits (20 audio bits plus the C, U, and V bits) from each AES sub-frame are mapped into three 10-bit video words (X, X+1, X+2) as shown in Table 3-3.

Bit-9 is always not Bit-8 to ensure that none of the excluded word values (3FFh-3FCh or 003h-000h) are used. The Z-bit is set to "1" corresponding to the first frame of the 192-frame AES block. Channels of embedded audio are essentially independent (although they,re always transmitted in pairs) so the Z-bit is set to a "1" in each channel even if derived from the same AES source. C, U, and V bits are mapped from the AES signal; however the parity bit is not the AES parity bit. Bit-8 in word X+2 is even parity for bits 0-8 in all three words.

There are several restrictions regarding distribution of the audio data packets although there's a "grandfather clause" in the proposed standard to account for older equipment that may not observe all the restrictions. Audio data packets are not transmitted in the horizontal ancillary data space following the normal vertical interval switch as defined in RP 168. They're also not transmitted in the ancillary data space desig-

Table 3-3. Embedded Audio Bit Distribution

Bit	X	X + 1	X + 2
b9	not b8	not b8	not b8
b8	aud 5	aud 14	Parity
b7	aud 4	aud 13	C
b6	aud 3	aud 12	U
b5	aud 2	aud 11	V
b4	aud 1	aud 10	aud 19 (msb)
b3	aud 0	aud 9	aud 18
b2	ch bit-1	aud 8	aud 17
b1	ch bit-2	aud 7	aud 16
b0	Z-bit	aud 6	aud 15

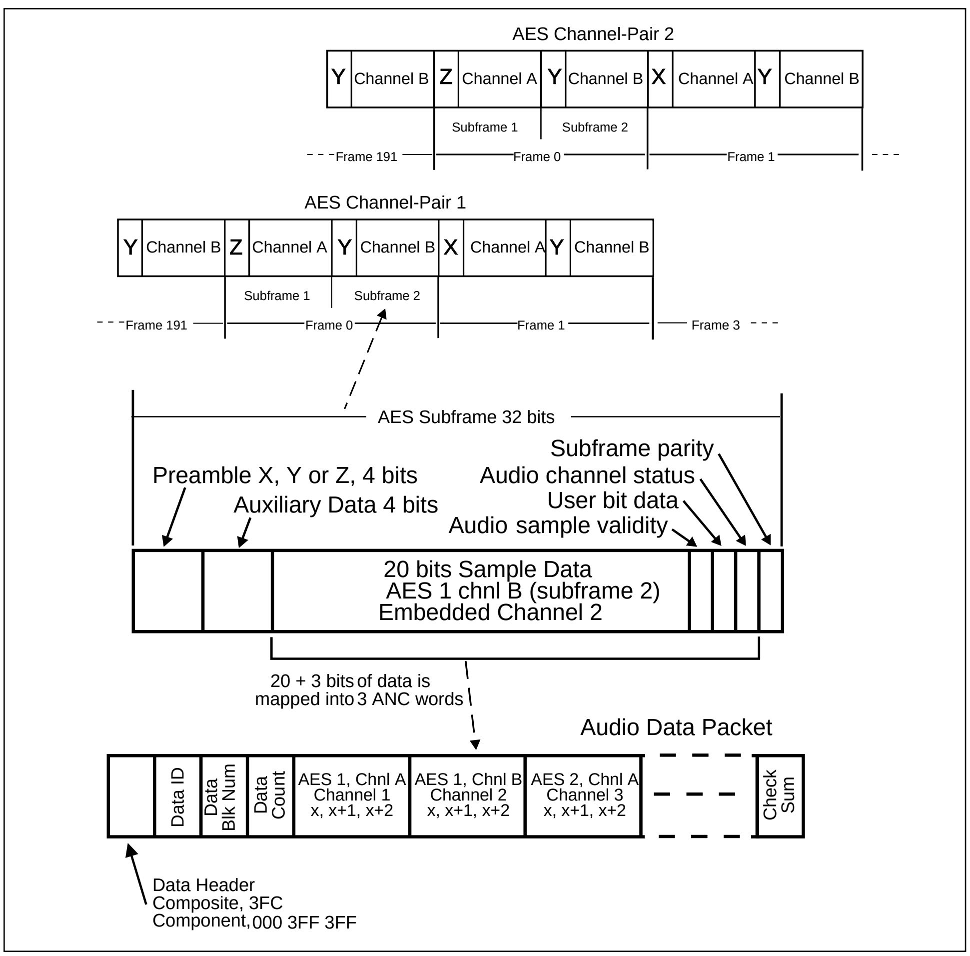

Figure 3-6. Basic embedded audio.

nated for error detection checkwords defined in RP 165. For composite digital video, audio data packets are not transmitted in equalizing pulses. Taking into account these restrictions “data should be distributed as evenly as possible throughout the video field.” The reason for this last statement is to minimize receiver buffer size which is an important issue for transmitting 24-bit audio in composite digital systems. For basic, Level A, this results in either three or four audio samples per channel in each audio data packet.

Extended embedded audio. Full-featured embedded audio defined in the proposed standard includes:

- Carrying the 4 AES auxiliary bits (which may be used to extend the audio samples to 24-bits)
- Allowing non-synchronous clock operation
- Allowing sampling frequencies other than 48 kHz
- Providing audio-to-video delay information for each channel
- Documenting Data IDs to allow up to 16 channels of audio in component digital systems
- Counting “audio frames” for 525 line systems.

To provide these features, two additional data packets are defined. Extended Data Packets carry the 4 AES auxiliary bits formatted such that one video word contains the auxiliary data for two audio samples as shown in Figure 3-7. Extended data packets must be located in the same ancillary data space as the associated audio data packets and must follow the audio data packets.

The Audio Control Packet (shown in Figure 3-8.) is transmitted once per field in the second horizontal ancillary data space after the vertical interval switch point. It contains information on audio frame number, sampling frequency, active channels, and relative audio-to-video delay of each channel. Transmission of audio control packets is optional for 48 kHz synchronous operation and required for all other modes of operation (since it contains the information as to what mode is being used).

Audio frame numbers are an artifact of 525 line, 29.97 frame/second operation. In that system there are exactly 8008 audio samples in exactly five frames, which means there’s a non-integer number of samples per frame. An audio frame sequence is the number of frames for an integer number of samples (in this case five) and the audio frame number indicates where in the sequence a particular frame belongs. This is important when switching between sources because certain equipment (most notably digital VTRs) require consistent synchronous operation to prevent buffer over/under

Table 3-4. Data IDs For Up To 16-channel Operation

	Audio Channels	Audio Data Packet	Extended Data Packet	Audio Control Packet
Group 1	1-4	1FF	1FE	1EF
Group 2	5-8	1FD	2FC	2EE
Group 3	9-12	1FB	2FA	2ED
Group 4	13-16	2F9	1F8	1EC

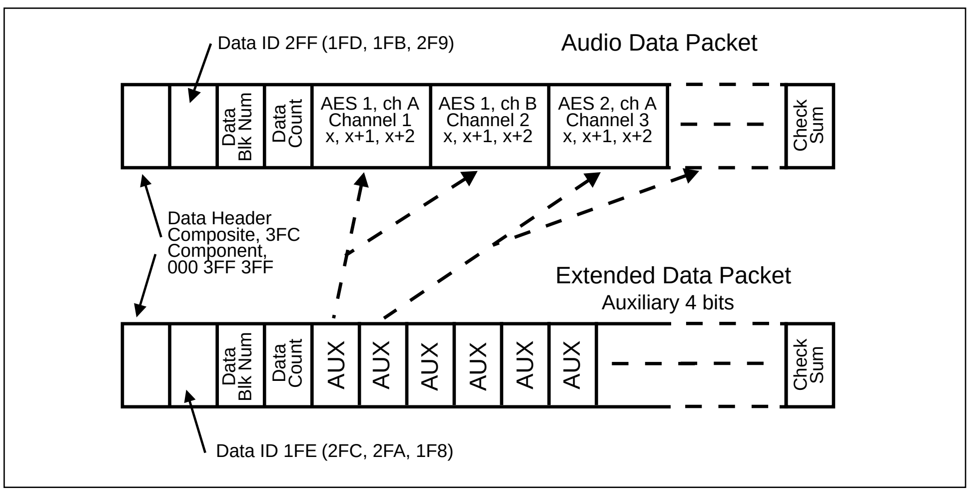

Figure 3-7. Extended embedded audio.

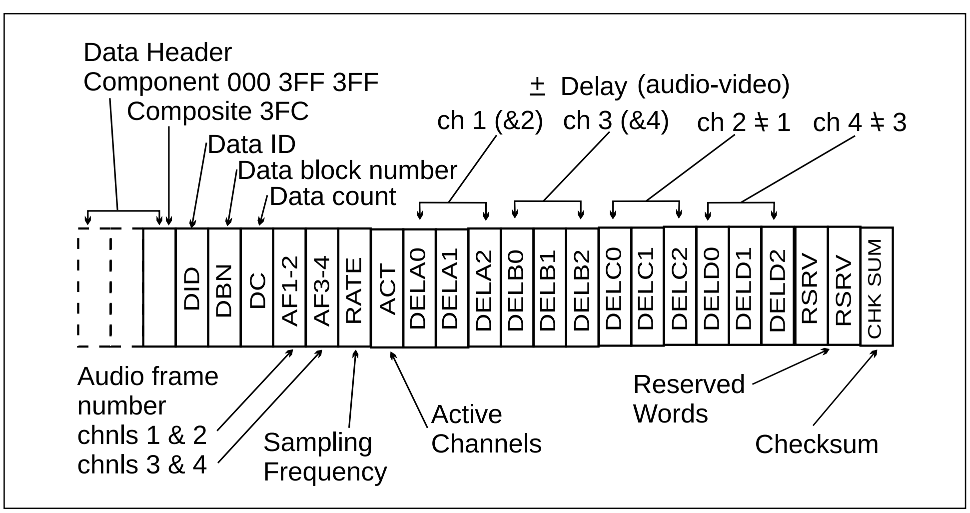

Figure 3-8. Audio control packet formatting.

flow. Where frequent switching is planned, receiving equipment can be designed to add or drop a sample following a switch in the four out of five cases where the sequence is broken. The challenge in such a system is to detect that a switch has occurred. This can be facilitated by use of the data block number in the ancillary data format structure and by including an optional frame counter with the unused bits in the audio frame number word of the audio control packet.

Audio delay information contained in the audio control packet uses a default channel-pair mode. That is, delay-A (DELA0-2) is for both channel 1 and channel 2 unless the delay for channel 2 is not equal to channel 1. In that case, the delay for channel 2 is located in delay-C. Sampling frequency must be the same for each channel in a pair, hence the data in "ACT" provides only two values, one for channels 1 and 2 and the other for channels 3 and 4.

In order to provide for up to 16 channels of audio in component digital systems the embedded audio is divided into audio groups corresponding to the basic four-channel operation. Each of the three data packet types are assigned four Data IDs as shown in Table 3-4.

Receiver buffer size. In component digital video, the receiver buffer in an audio demultiplexer is not a critical issue since there's much ancillary data space available and few lines excluding audio ancillary data. The case is considerably different for composite digital video due to the exclusion of data in equalizing pulses and, even more important, the data packet distribution required for extended audio. For this reason the proposed standard requires a receiver buffer of 64 samples per channel with a grandfather clause of 48 samples per channel to warn designers of the limitations in older equipment. In the proposed standard, Level A defines a sample distribution allowing use of a 48 sample-per-channel receiver buffer while other levels generally require the use of the specified 64 sample buffer. Determination of buffer size is analyzed in this section for digital NTSC systems, however the result is much the same for digital PAL systems.

Synchronous sampling at 48 kHz provides exactly 8008 samples in five frames which is $8008 \div (5 \times 525) = 3.051$ samples per line. Therefore most lines may carry three samples per channel while some should carry four samples per channel. For digital NTSC, each horizontal sync tip has space for 55 ancillary data words. Table 3-5 shows how those words are used for four-channel, 20-bit and 24-bit audio (overhead includes header, data ID, data count, etc.). Since it would require 24 more video words to carry one additional sample per channel for four channels of 24-bit audio and that would exceed the 55-word space, there can be no more than three samples per line of 24-bit audio.

Table 3-5. Maximum Samples Per NTSC Sync Tip

4-Channels	20-bit Audio 3-samples	20-bit Audio 4-samples	24-bit Audio 3-samples
Audio Data			
Overhead	5	5	5
Sample data	36	48	36
Extended Data			
Overhead			5
Auxiliary data			6
TOTAL	41	53	47

A receiver sample buffer is needed to store samples necessary to supply the continuous output required during the time of the equalizing pulses and other excluded lines (the vertical interval switch line and the subsequent line). When only 20-bit audio is required the "even distribution of samples" dictates that some horizontal lines carry four samples per channel resulting in a rather modest buffer size of 48 samples or less depending on the type of buffer. Using only the first broad pulse of each vertical sync line, four or five samples in each broad pulse will provide a satisfactory, even distribution. This is known as Level A. However, if 24-bit audio is used or if the sample distribution is to allow for 24-bit audio even if not used, there can be no more than three samples per horizontal ancillary data space for a four-channel system. The first broad pulse of each vertical sync line is required to carry 16 or 17 samples per channel to get the best possible "even distribution." This results in a buffer requirement of 80 samples per channel, or less, depending on the type of buffer. To meet the 64 sample-per-channel buffer, a smart buffer is required.

There are two common types of buffer that are equivalent for the purpose of understanding requirements to meet the proposed standard. One is the FIFO (first in first out) buffer and the other is a circular buffer. Smart buffers have the following synchronized load abilities:

- FIFO Buffer – hold off "reads" until a specific number of samples are in the buffer (requires a counter) AND neither read or write until a specified time (requires vertical sync).
- Circular Buffer – set the read address to be a certain number of samples after the write address at a spec-

ified time (requires vertical sync).

Using a smart buffer the minimum sizes are 24 samples per channel for 20-bit operation and 40 samples per channel for 24-bit operation. Because of the different requirements for the two types of sample distribution, the minimum smart buffer size for either number of bits per sample is 57 samples. Therefore, a 64 sample-per-channel smart buffer should be adequate.

In the case of a not-so-smart buffer, read and write addresses for a circular buffer must be far enough apart to handle either a build up of 40 samples or a depletion of 40 samples. This means the buffer size must be 80 samples for 24-bit audio or for automatic operation of both sample sizes. Therefore, a 64 sample-per-channel smart buffer is required to meet the specifications of the proposed standard.

Systemizing AES/EBU audio. Serial digital video and audio are becoming commonplace in production and post-production facilities as well as television stations. In many cases, the video and audio are married sources; and it may be desirable to keep them together and treat them as a single serial digital data stream. This has, for one example, the advantage of being able to keep the signals in the digital domain and switch them together with a serial digital video routing switcher. In the occasional instances where it's desirable to break away some of the audio sources, the digital audio can be demultiplexed and switched separately via an AES/EBU digital audio routing switcher.

At the receiving end, after the multiplexed audio has passed through a serial digital routing switcher, it may be necessary to extract the audio from the video so that editing, audio sweetening, or other processing can be accomplished. This requires a demultiplexer that strips off the AES/EBU audio from the serial digital video. The output of a typical demultiplexer has a serial digital video BNC as well as connectors for the two-stereo-pair AES/EBU digital audio signals.

4. System Hardware and Issues

Cable Selection

The best grades of precision analog video cables exhibit low losses from very low frequencies (near DC) to around 10 MHz. In the serial digital world, cable losses in this portion of the spectrum are of less consequence but still important. It's in the higher frequencies associated with transmission rates of 143, 177, 270, or 360 Mb/s where losses are considerable. Fortunately, the robustness of the serial digital signal makes it possible to equalize these losses quite easily. So when converting from analog to digital, the use of existing, quality cable runs should pose no problem.

The most important characteristic of coaxial cable to be used for serial digital is its loss at 1/2 the clock frequency of the signal to be transmitted. That value will determine the maximum cable length that can be equalized by a given receiver. It's also important that the frequency response loss in dB be approximately proportional to $1/\sqrt{f}$ down to frequencies below 5 MHz. Some of the common cables in use are the following:

PSF 2/3	UK
Belden 8281	USA and Japan
F&G 1.0/6.6	Germany

These cables all have excellent performance for serial digital signals. Cable manufacturers are seizing the opportunity to introduce new, low-loss, foam dielectric cables specifically designed for serial digital. Examples of alternative video cables are Belden 1505A which, although thinner, more flexible, and less expensive than 8281, has a higher performance at the frequencies that are more critical for serial digital signals. A recent development specifically for serial digital video is Belden 1694A with lower loss than 1505A.

Connectors

Until recently, all BNC connectors used in television had a characteristic impedance of 50 ohms. BNC connectors of the 75-ohm variety were available, but were not physically compatible with 50-ohm connectors. The impedance "mismatch" to coax is of little consequence at analog video frequencies because the wavelength of the signals is many times longer than the length of the connector. But with serial digital's high data rate (and attendant short wavelengths), the impedance of the connector must be considered. In general, the length of a simple BNC connector doesn't have significant effect on the serial digital signal. Several inches of 50-ohm connection, such as a number of barrels or short coax, would have to be used for any noticeable effect. In the specific case of a serial transmitter or receiver, the active devices associated with the chassis connector need significant impedance matching. This could easily include making a 50-ohm connector look like 75 ohms over the frequency band of interest. Good engineering practice tells us to avoid impedance mismatches and use 75-ohm components wherever possible and that rule is being followed in the development of new equipment for serial digital.

Patch Panels

The same holds true for other "passive" elements in the system such as patch panels. In order to avoid reflections caused by impedance discontinuities, these elements should also have a characteristic impedance of 75 ohms. Existing 50-ohm patch panels will probably be adequate in many instances, but new installations should use 75-ohm patch panels. Several patch panel manufacturers now offer 75-ohm versions designed specifically for serial digital applications.

Terminations and Loop-throughs

The serial digital standard specifies transmitter and receiver return loss to be greater than 15 dB up to 270 MHz. That is, terminations should be 75 ohms with no significant reactive component to 270 MHz. Clearly this frequency is related to component digital, hence a lower frequency might be suitable for NTSC and that may be considered in the future. As can be seen by the rather modest 15 dB specification, return loss is not a critical issue in serial digital video systems. It's more important at short cable lengths where the reflection could distort the signal rather than at long lengths where the reflected signal receives more attenuation.

Most serial digital receivers are directly terminated in order to avoid return loss problems. Since the receiver active circuits don't look like 75 ohms, circuitry must be included to provide a good resistive termination and low return loss. Some of today's equipment does not provide return loss values meeting the specification for the higher frequencies. This hasn't caused any known system problems; however, good engineering practice should prevail here as well.

Active loop-throughs are very common in serial digital equipment because they're relatively simple and have signal regeneration qualities similar to a reclocking distribution amplifier. Also, they provide isolation between input and output. However, if the equipment power goes off for any reason, the connection is broken. Active loop-throughs also have the same need for care in circuit impedance matching as mentioned above.

Passive loop-throughs are both possible and practical. Implementations used in serial digital waveform monitors have return loss greater that 25 dB up to the clock frequency. This makes it possible to monitor the actual signal being received by the operational equipment or unit-under-test and not substitute the monitor receiver.

The most important use of passive loop-throughs is for system diagnostics and fault finding where it's necessary to observe the signal in the troubled path. If a loop-through is to be used for monitoring, it's important that it be passive loop-through for two reasons. First, serial transmitters with multiple outputs usually have separate active devices on each output, therefore monitoring one output doesn't necessarily indicate the quality of another output as it did in the days of resistive fan-out of analog signals. Second, when an active loop-through is used, loss of power in the monitoring device will turn off the signal. This would be disastrous in an operational situation. If an available passive loop-through isn't being used, it's important to make sure the termination is 75 ohms with no significant reactive component to at least the clock frequency of the serial signal. That old, perhaps "precision," terminator in your tool box may not do the job for serial digital.

Digital Audio Cable and Connector Types

AES/EBU digital audio has raised some interesting questions because of its characteristics. In professional applications, balanced audio has traditionally been deemed necessary in order to avoid hum and other artifacts. Usually twisted, shielded, multiconductor audio cable is used. The XLR connector was selected as the connector of choice and is used universally in almost all professional applications. When AES/EBU digital audio evolved, it was natural to assume that the traditional analog audio transmission cable and connectors could still be used. The scope of AES3 covers digital audio transmission of up to 100 meters, which can be handled adequately with shielded, balanced twisted-pair interconnection.

Because AES/EBU audio has a much wider bandwidth than analog audio, cable must be selected with care. The impedance, in order to meet the AES3 specification, requires 110-ohm source and load impedances. The standard has no definition for a bridging load, although loop-through inputs, as used in analog audio, are theoretically possible. Improperly terminated cables may cause signal reflections and subsequent data errors.

The relatively high frequencies of the AES/EBU signals cannot travel over twisted-pair cables as easily as analog audio. Capacitance and high-frequency losses cause high-frequency rolloff. Eventually, signal edges become so rounded and amplitude so low that the receivers can no longer tell the "1"s from the "0"s. This makes the signals undetectable. Typically, cable lengths are limited to a few hundred feet. XLR connectors are also specified. Since AES/EBU digital audio has frequencies up to about 6 MHz, there've been suggestions to use unbalanced coaxial cable with BNC connectors for improved performance in existing video installations and for transmissions beyond 100 meters.

There are two committees defining the use of BNC connectors for AES/EBU digital audio. The Audio Engineering Society has developed AES3-ID and SMPTE is developing a similar recommended practice. These recommendations will use a 110-ohm balanced signal instead of 75-ohm unbalanced and reduce the 3- to 10-volt signal to 1 volt. The resulting signal now has the same characteristics as analog video, and traditional analog video distribution amplifiers, routing switchers, and video patch panels can be used. Costs of making up the coaxial cable with BNCs is also less than multiconductor cable with XLRs. Tests have indicated that EMI emissions are also reduced with coaxial distribution.

Signal Distribution, Reclocking

Although a video signal may be digital, the real world through which that signal passes is analog. Consequently, it's important to consider the analog distortions that affect a digital signal. These include frequency response rolloff caused by cable attenuation, phase distortion, noise, clock jitter, and baseline shift due to AC coupling. While a digital signal will retain the ability to communicate its data despite a certain degree of distortion, there's a point beyond which the data will not be recoverable. Long cable runs are the main cause of signal distortion. Most digital equipment provides some form of equalization and regeneration at all inputs in order to compensate for cable runs of varying lengths. Considering the specific case of distribution amplifiers and routing switchers, there are several approaches that can be used: wideband amplifier, wideband digital amplifier, and regenerating digital amplifier.

Taking the last case first, regeneration of the digital signal generally means to recover data from an incoming signal and to retransmit it with a clean waveform using a stable clock source. Regeneration of a digital signal allows it to be transmitted farther and sustain more analog degradation than a signal which already has accumulated some analog distortions. Regeneration generally uses the characteristics of the incoming signal, such as extracted clock, to produce the output. In serial digital video, there are two kinds of regeneration: serial and parallel.

Serial regeneration is simplest. It consists of cable equalization (necessary even for cable lengths of a few meters or less), clock recovery, data recovery, and re-transmission of the data using the recovered clock. A phase locked loop (PLL) with an LC (inductor/capacitor) or RC (resistor/capacitor) oscillator regenerates the serial clock frequency, a process called reclocking.

Parallel regeneration is more complex. It involves three steps: deserialization; parallel reclocking, usually using a crystal-controlled time base; and serialization.

Each form of regeneration can reduce jitter outside its PLL bandwidth, but jitter within the loop bandwidth will be reproduced and may accumulate significantly with each regeneration. (For a complete discussion of jitter effects and measurements see Section 8.) A serial regenerator will have a loop bandwidth on the order of several hundred kilohertz to a few megahertz. A parallel regenerator will have a much narrower bandwidth on the order of several Hertz. Therefore, a parallel regenerator can reduce jitter to a greater extent than a serial regenerator, at the expense of greater complexity. In addition, the inherent jitter in a crystal controlled timebase (parallel regenerator) is much less than that of an LC or RC oscillator timebase (serial regenerator). Serial regeneration obviously cannot be performed an unlimited number of times because of cumulative PLL oscillator jitter and the fact that the clock is extracted from the incoming signal. Serial regeneration can typically be performed several dozen times before a parallel regeneration is necessary. Excessive jitter build up eventually causes a system to fail; jitter acceptable to a serial receiver must still be considered and handled from a system standpoint which will be discussed later.

Regeneration, where a reference clock is used to produce the output, can be performed an unlimited number of times and will eliminate all the jitter built up in a series of regeneration operations. This type of regeneration happens in major operational equipment such as VTRs, production switchers (vision mixers) or special effects units that use external references to perform their digital processing. So the bottom line is, regeneration (or reclocking) can be perfect, only limited by economic considerations.

A completely different approach to signal distribution is the use of wideband analog routers, but there are many limitations. It's true that wideband analog routers (100 MHz plus) will pass a 143 Mb/s serial digital signal. However, they're unlikely to have the performance required for 270 or 360 Mb/s. With routers designed for wideband analog signals, the $1/\sqrt{f}$ frequency response characteristics will adversely affect the signal. Receivers intended to work with SMPTE 259M signals expect to see a signal transmitted from standard source and attenuated by coaxial cable with frequency response losses. Any deviation from 6 dB/octave rolloff over a bandwidth of 1 MHz up to the clock frequency will cause improper operation of the automatic equalizer in the serial receiver. In general, this will be a significant problem because the analog router is designed to have flat response up to a certain bandwidth and then roll off at some slope that may, or may not, be 6 dB/octave. It's the difference between the in-band and out-of-band response that causes the problem.

In between these two extremes are wideband serial digital routers that don't reclock the signal. Such non-reclocking routers will generally perform adequately at all clock frequencies for the amount of coax attenuation given in their specifications. However, that specification will usually be shorter than for reclocking routers.

System Timing

The need to understand, plan, and measure signal timing has not been eliminated in digital video, it's just moved to a different set of values and parameters. In most cases, precise timing requirements will be relaxed in a digital environment and will be measured in microseconds, lines, and frames rather than nanoseconds. In many ways, distribution and timing is becoming simpler with digital processing because devices that have digital inputs and outputs can lend themselves to automatic input timing compensation. However, mixed analog/digital systems place additional constraints on the handling of digital signal timing.

Relative timing of multiple signals with respect to a reference is required in most systems. At one extreme, inputs to a composite analog production switcher must be timed to the nanosecond range so there will be no subcarrier phase errors. At the other extreme, most digital production switchers allow relative timing between input signals in the range of one horizontal line. Signal-to-signal timing requirements for inputs to digital video can be placed in three categories:

1. Digital equipment with automatic input timing compensation.

2. Digital equipment without automatic input timing compensation.
3. Digital-to-analog converters without automatic timing compensation.

In the first case, the signals must be in the range of the automatic compensation, generally about one horizontal line. For system installation and maintenance it's important to measure the relative time of such signals to ensure they're within the automatic timing range with enough headroom to allow for any expected changes. Automatic timing adjustments are a key component for digital systems; but they cannot currently be applied to a serial data stream due to the high digital frequencies involved. A conversion back to the parallel format for the adjustable delay is necessary due to the high data rates. Therefore, automatic input timing is not available in all equipment.

For the second case, signals are nominally in-time as would be expected at the input to a routing switcher. SMPTE recommended practice RP 168, which applies to both 525 and 625 line operation, defines a timing window on a specific horizontal line where a vertical interval switch is to be performed. See Figure 4-1 which shows the switching area for field 1. (Field 2 switches on line 273 for 525 and line 319 for 625.) The window size is 10 µs which would imply an allowable difference in time between the two signals of a few microseconds. A preferred timing difference is generally determined by the reaction of downstream equipment to the vertical interval switch. In the case of analog routing switchers, a very tight tolerance is often maintained based on subcarrier phase requirements. For serial digital signals microseconds of timing difference may be acceptable. At the switch point, clock lock of the serial signal will be recovered in one or two digital words (10 to 20 serial clock periods). However, word framing will, most likely, be lost causing the signal value to assume some random value until the next synchronizing data (EAV) resets the word framing and produces full synchronization of the signal at the receiver. A signal with a random value for part of a line is clearly not suitable on-line to a program feed; however downstream equipment can be designed to blank the undesired portion of the line. In the case where the undesired signal is not eliminated, the serial signal timing to the routing switcher will have to be within one or two nanoseconds so the disturbance doesn't occur. Clearly, the downstream blanking approach is preferred.

Finally, the more difficult case of inputs to DACs without timing compensations may require the same accuracy as today's analog systems – in the nanosecond range. This will be true where further analog processing is required, such as used in analog production switchers (vision mixers). Alternately, if the analog signal is simply a studio output, this high accuracy may not be required.

Automatic timing adjustments are a key component for digital systems; but they cannot currently be applied to a serial data stream due to the high digital frequencies involved. A conversion back to the parallel format for the delay is necessary due to the high data rates. There's a need for a variety of adjustable delay devices to address these new timing requirements. These include a multiple-line delay device and a frame delay device. A frame delay device, of course, would result in a video delay that could potentially create some annoying problems with regard to audio timing and the passing of time code through the system. Whenever audio and video signals are processed through different paths, a potential exists for differential audio-to-video delay. The differential delay caused by embedding audio is insignificant (under 1 ms). However, potential problems occur where frame delays are used and the video and audio follow separate paths.

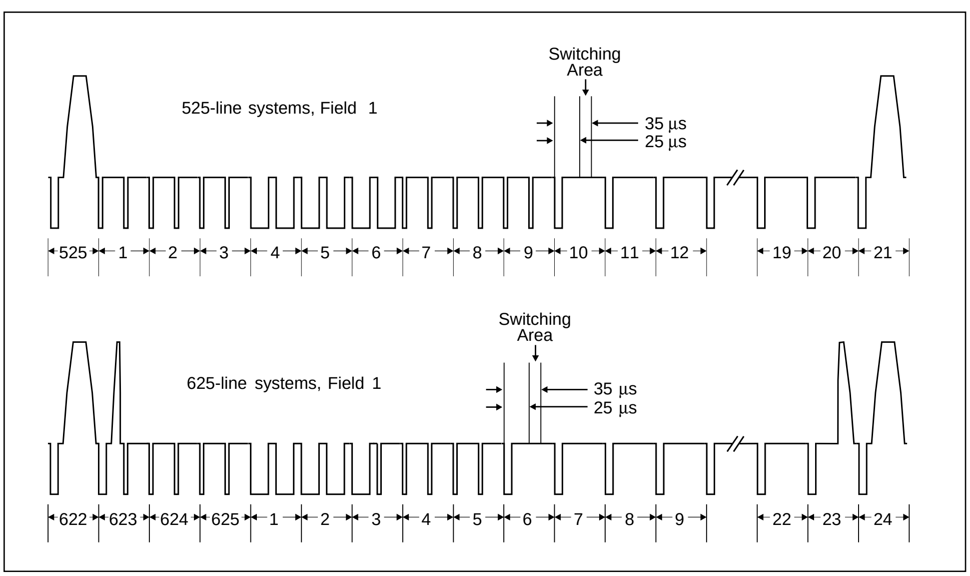

Figure 4-1. Vertical interval switching area.

5. Monitoring and Measurements

Test and measurement of serial digital signals can be characterized in three different aspects: usage, methods, and operational environment. Types of usage would include: designer's workbench, manufacturing quality assurance, user equipment evaluation, system installation and acceptance testing, system and equipment maintenance, and perhaps most important, operation.

Requirements for a basic operational monitor include display of the program signals carried by the digital signal with features and accuracy consistent with today's analog baseband signal monitors. An operational monitor should also include information about the serial digital signal itself, such as data available, bit errors, and data formatting errors. A display of the actual serial waveform is not required. Due to the robust nature of digital video signals a reduction in the number of operational monitors may be possible. However, monitoring of the program signal waveform is required at all locations where an operator or equipment has the ability to change program parameters. The Tektronix WFM 601 Series of serial-component monitors is shown in Figure 5-1.

Figure 5-1. The WFM 601 Series of serial-component monitors.

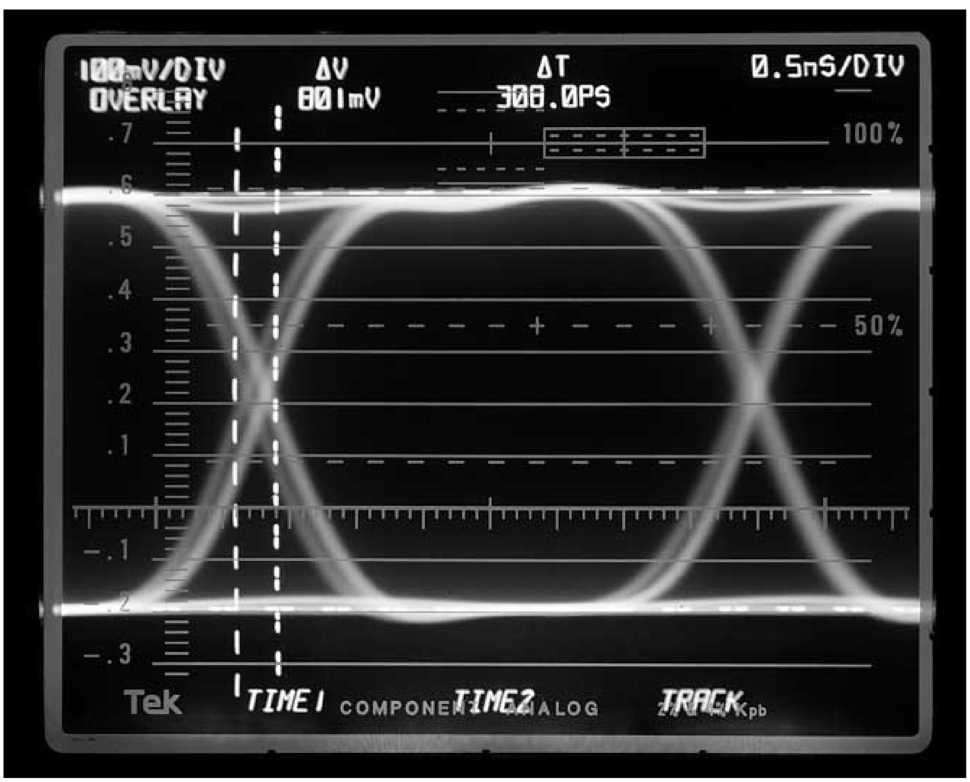

Figure 5-2. Eye-pattern display on WFM601M.

Methods for technical evaluation cover several usage areas that have overlapping requirements. In addition to the traditional television system measurements there's a new dimension for test and measurement – to quantify the various parameters associated directly with the serial waveform. The result is several categories of monitoring and measurement methods to be considered: program signal analysis, data analysis, format verification, transmitter/receiver operation, transmission hardware, and fault reporting. An equivalent-time sampling display of the serial waveform is displayed on the Tektronix WFM601M Serial Component Monitor in Figure 5-2.

Program signal measurements are essentially the baseband video and audio measurements that have been used for years. An important aspect to these measurements is that the accuracy of signal representation is limited by the number of bits-per-sample. In analog systems there's been a small amount of digital data present in video signals in the form of Vertical

Interval Time Code (VITC); but the serial data stream has much more capacity for data other than video. Hence, more complete measurement methods are desirable.

Digital signal waveform analysis and its effect on transmitter/receiver operation is a new requirement in test and measurement for television facilities. Rather than acquire special equipment for interpreting this high speed waveform, appropriate capabilities are being added to traditional television test equipment, helping in the economic transition to serial digital television. Testing of passive transmission components (coax, patch panels, etc.) is similar to that used with baseband systems except that much wider bandwidths must be considered.

The third aspect to test and measurement is whether it's in-service or out-of-service. All operational monitoring must be in-service, which means that monitors must be able to give the operator information about the digital signal that contains active program material as well as the program signal itself. If there are problems to be solved, discovering those with an intermittent nature will also require in-service measurements.

Because of the well known crash-point type of failure for digital systems, out-of-service testing is also especially important. In order to know how much headroom is available it's necessary to add stressing parameters to the digital signal in measured amounts until it does crash, which is certainly not acceptable in operational situations.

The next few sections deal with the details of specific measurement techniques covering monitoring, measurements, in-service testing, and out-of-service testing.

6. Measuring the Serial Signal

Waveform Measurements

When viewed on an appropriate scope or monitor, several time sweeps (overlaid by the CRT persistence or digital sample memory) produces a waveform that follows a number of different paths across the screen. The different paths are due to the fact that the digits in the serial stream vary based on the data (high or low states, with or without change at possible transition times). The waveform that results is known as an eye pattern (with two "eyes" shown in Figure 6-1 and displayed on a monitor as shown on the WFM601M Serial Component Monitor in Figure 5-2).

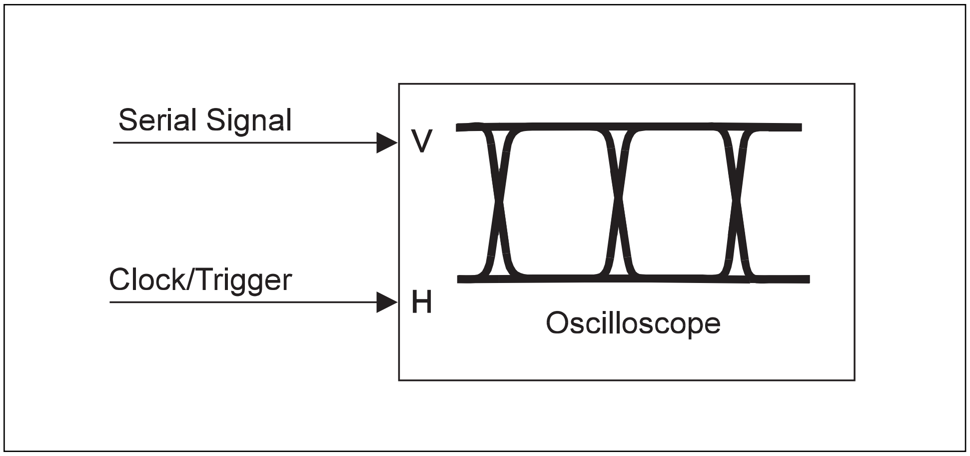

Figure 6-1. Eye pattern display.

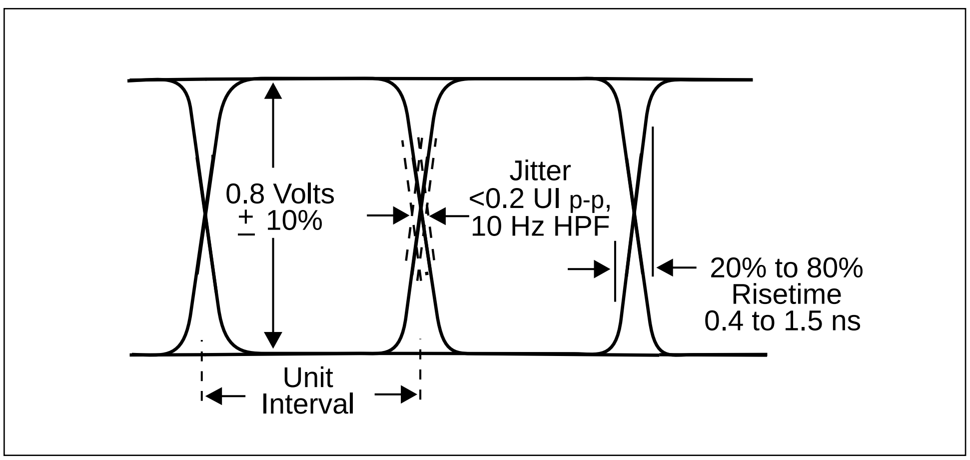

Figure 6-2. Serial signal specifications.

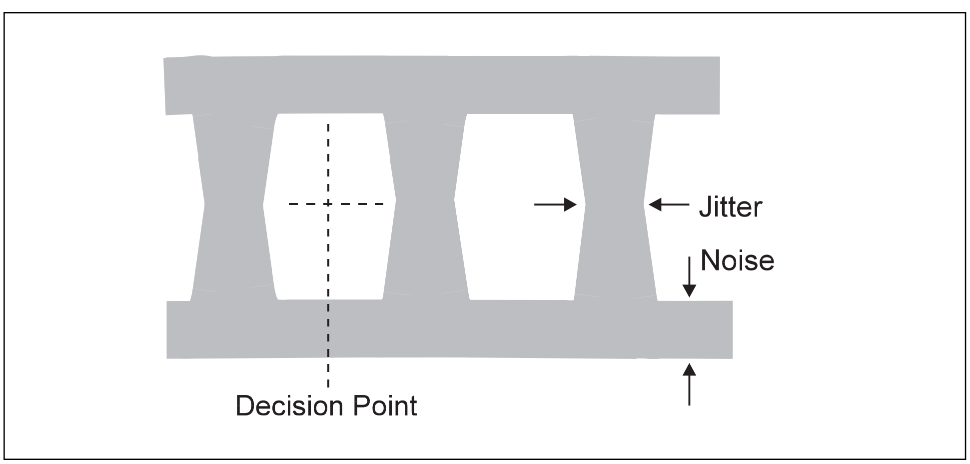

Figure 6-3. Data recovery.

Analog measurements of the serial digital waveform start with the specifications of the transmitter output as shown in Figure 6-2. Specifications to be measured are amplitude, risetime, and jitter, which are defined in the serial standard, SMPTE 259M. Frequency, or period, is determined by the television sync generator developing the source signal, not the serialization process. A unit interval (UI) is defined as the time between two adjacent signal transitions, which is the reciprocal of clock frequency. The unit interval is 7.0 ns for NTSC, 5.6 ns for PAL, and 3.7 ns for component 525 or 625.

A serial receiver determines if the signal is a "high" or a "low" in the center of each eye, thereby detecting the serial data. As noise and jitter in the signal increase through the transmission channel, certainly the best decision point is in the center of the eye (as shown in Figure 6-3) although some receivers select a point at a fixed time after each transition point. Any effect which closes the eye may reduce the usefulness of the received signal.

In a communications system with forward error correction, accurate data recovery can be made with the eye nearly closed. With the very low error rates required for correct transmission of serial digital video (see Section 7) a rather large and clean eye opening is required after receiver equalization. This is because the random nature of the processes that close the eye have statistical "tails" that would cause an occasional, but unacceptable error. Jitter effects that close the eye are discussed in Section 8.

Amplitude is important because it affects maximum transmission distance; too large can be a problem as well as the more obvious too small. Some receiver equalizers depend on amplitude to estimate cable length and setting of the equalizer significantly affects resulting noise and jitter.

Precise measurement of the serial transmitter waveform requires a scope with a 1 GHz bandwidth because of the serial signal's 1 ns risetime. However, amplitude measurements may be made with lower bandwidth scopes (300 to 500 MHz bandwidth). Monitoring quality risetime measurements can be made with television test equipment using equivalent-time sampling at the lower bandwidths. Risetime measurements are made from the 20% to 80% points as appropriate for ECL logic devices. Since the serial signal has approximately a 1 ns risetime the measurement must be adjusted per the following formula:

$$T_a = \sqrt{(T_m^2 - 0.5\ T_s^2)}$$

Where:

T_a = actual risetime

T_m = measured risetime

T_s = risetime of the scope

The factor of 0.5 adjusts for the fact that the scope risetime specification is for the 10% to 90% points. As an example: with a scope 10 to 90% risetime of 1.0 ns, a measured risetime of 1.2 ns would indicate an actual serial waveform 20 to 80% risetime of 0.97 ns, and a measured 1.6 ns would indicate 1.44 ns actual risetime.

Measuring Serial Digital Video Signal Timing

Relative timing between serial digital video signals that are within an operational range for use in studio equipment may vary from several nanoseconds to a few television lines. Measurement of the timing differences in operational signal paths may be accomplished using the Active Picture Timing Test Signal available from the TSG 422 Digital Component Generator in conjunction with the timing cursors and line select of the WFM 601 series of serial component waveform monitors.

Figure 6-4 shows a representation of a picture monitor display of the Active Picture Timing Test Signal (previously known as the Digital Blanking Test Signal). The first and last full active analog lines have a white (luminance-only) bar with nominal blanking edges. Depend-

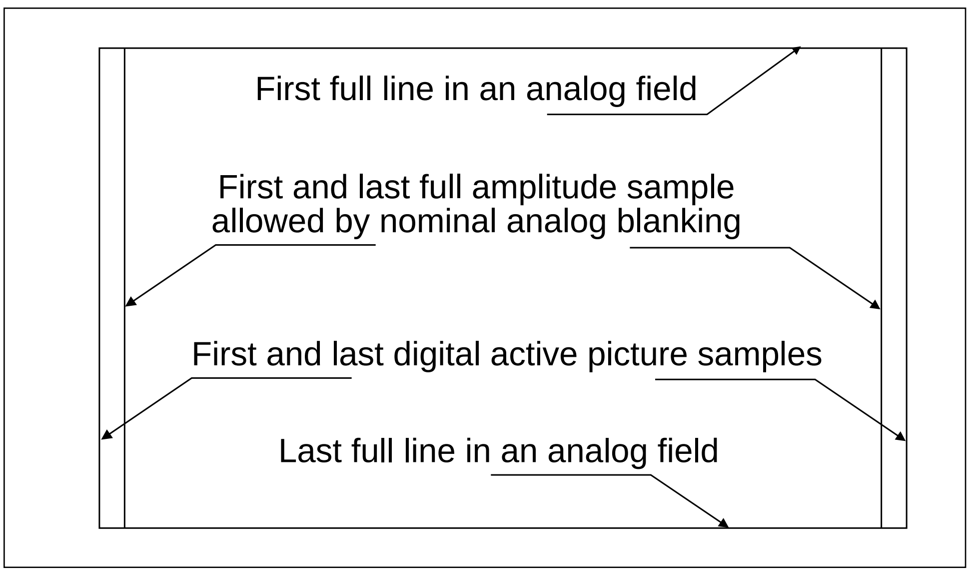

Figure 6-4. Picture monitor display of the Active Picture Timing Test signal.

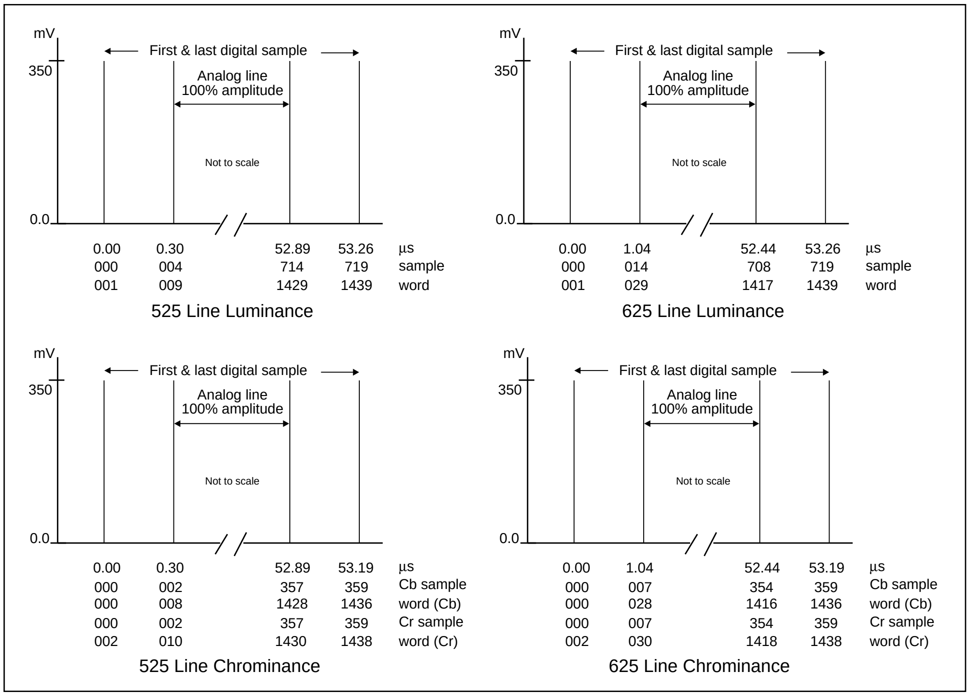

Figure 6-5. Locations of half-amplitude samples.

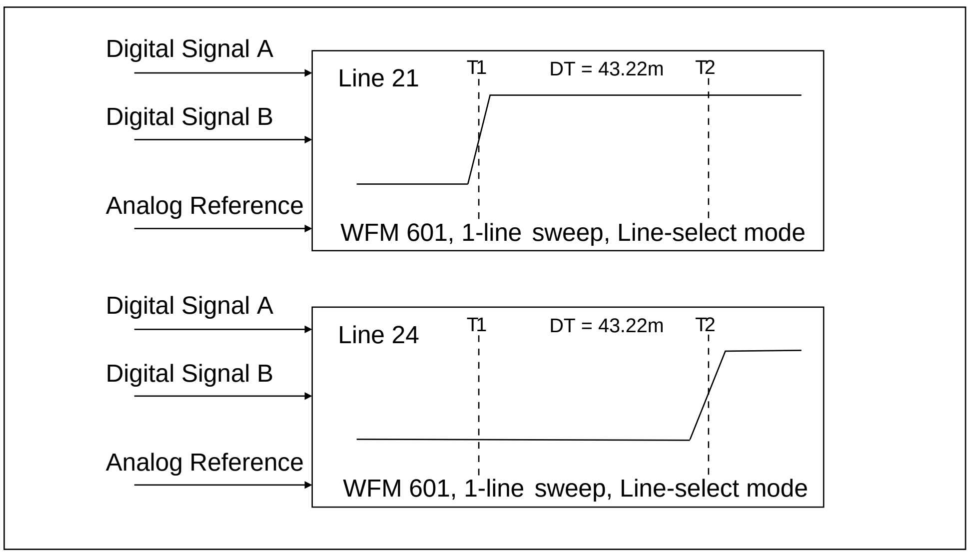

Figure 6-6.Timing measurement, large time difference.

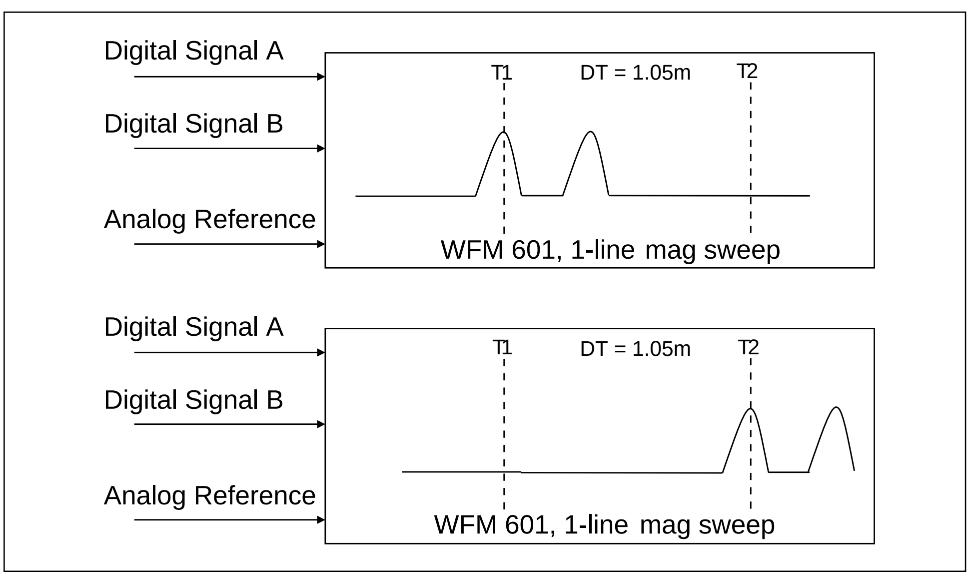

Figure 6-7.Timing measurement, small time difference.

ing on the field, these may or may not be the first and last full digital active lines. Use of the specific full-lines shown below ensure that the complete signal will be visible after digital-to-analog conversion. It's the responsibility of the converter to make half-lines where appropriate. The lines with the luminance white bar are:

525-line signals:	Lines 21, 262, 284, and 525
625-line signals:	Lines 24, 310, 336, and 622

All other active picture lines have a one-sample, half-amplitude word in four locations: first and last active digital sample, first and last sample at the 100% point of a nominal white bar. Figure 6-5 shows the specific location of each half-amplitude word. Note: the last active chrominance sample on each line is co-sited with luminance sample #718 whereas the last luminance sample is #719.

To make a measurement of large relative timing between two digital signals, set up the WFM 601 as shown in Figure 6-6. Alternately, select each input in the line select mode and find the 50% point of the white bar. The line number and delta-T measurement give the total time difference. For higher resolution measurement of small time differences, use the same set up to view the time of the double pulses at the start of the active line as shown in Figure 6-7. Although the figure shows nice clean pulses, considerable ringing will be seen due to the non-nyquist nature of the single-word excursion. As with other frequently used setups, these can be stored as one of the nine WFM 601 presets, to be recalled instantly when needed.

7. Definition and Detection of Errors

In our application of computers and digital techniques to automate many tasks, we've grown to expect the hardware and software to report problems to an operator. This is particularly important as systems become larger, more complex, and more sophisticated. It's reasonable to expect digital television systems to provide similar fault reporting capabilities.

Television equipment has traditionally had varying levels of diagnostic capabilities. With the development of digital audio and video signals it's now possible to add a vital tool to system fault reporting; confirmation of signal integrity in studio interconnection systems. To understand and use this tool, it's necessary to delve into the technology of digital error detection. The combination of the serial-digital video interconnect method, the nature of television systems, and typical digital equipment design lead to the use of error measurement methods that are economical yet specialized for this application.

Serial digital video operates in a basically noise-free environment, so traditional methods of measuring random error rates are not particularly useful. Examination of the overall system characteristics leads to the conclusion that a specialized burst error measurement system will provide the television studio engineer with an effective tool to monitor and evaluate error performance of serial-digital video systems.

Definition of Errors

In the testing of in-studio transmission links and operational equipment a single error is defined as one data word whose digital value changes between the signal source and the measuring receiver. This is significantly different than the analog case where a range of received values could be considered to be correct. Application of this definition to transmission links is straight forward. If any digital data word values change between transmitter and receiver there's an error. For operational studio equipment, the situation is a little more complicated and needs further definition.

Equipment such as routing switchers, distribution amplifiers and patch panels should, in general, not change the data carried by the signal; hence, the basic definition applies. However, when a routing switcher changes the source selection there will be a short disturbance. This should not be considered an error although the length of the disturbance is subject to evaluation and measurement. SMPTE Recommended Practice RP 168 specifies the line in the vertical interval where switching is to take place, which allows error measurement equipment to ignore this acceptable error.

There are various types of studio equipment that would not normally be expected to change the signal. Examples are frame synchronizers with no proc amp controls, production switchers in a straight-through mode, and digital VTRs in the E-to-E mode. All have the capability (and are likely to) replace all of the horizontal and part of the vertical blanking intervals. In general, replacing part of the television signal will destroy the error-free integrity of the signal. This is true for both component and composite signals as there could be ancillary data in the replaced sections of the signal. There's an even stronger case for composite signals as the sync areas are not a strictly defined set of digits, hence may vary within the limits of the analog signal standard.

Considering the possible replacement of blanking areas by some equipment and knowing that the active picture sample locations are well defined by the various digital standards, it's possible to measure Active Picture Errors separately from Full Field Errors. The Active Picture Error concept takes care of blanking area replacements. However, there is a more insidious possibility based on the television design engineer's historical right to modify the blanking edges. Most digital standards were designed with digital blanking narrower than analog blanking to ensure appropriate edge transitions in the analog domain. (Large amplitude edge transitions within one sample period caused by time truncating the blanking waveform can create out-of-band spectral components that can show up as excessive ringing after digital-to-analog conversion and filtering.) What happens in some equipment is that engineers have modified the samples representing the analog blanking edge in order to provide a desired transition; therefore, error measurements through such equipment even using the Active Picture concept will not work.

With VTRs, there's a further limit to measurement of errors. The original full bit rate VTR formats (D-1, D-2, and D-3) are designed with significant amounts of forward error correction and work very well. However, it's the nature of the tape recording and reproduction process that some errors will not be corrected. Generally the error correction system identifies the uncorrected data and sophisticated error concealment systems can be applied to make a virtually perfect picture. The systems are so good that tens of generations are possible with no human-noticeable defect. Although the error concealment is excellent, it's almost certain that error measurement systems (that by definition require perfect data reproduction) would find errors in many completely acceptable fields. New VTR formats that use bit-rate reduction will have an additional potential source of small but acceptable data value changes. The data compression and decompression schemes are not absolutely lossless, resulting in some minor modification of the data. Therefore, all error measurement methods external to a digital VTR are not meaningful because of potential blanking edge adjustment, error concealment methods and, in some equipment, the use of data-rate reduction. The good news is that various types of error rate and concealment rate data are available inside the VTR and standards for reporting that information are being developed.

Table 7-1. Error Frequency and Bit Error Rates

Time Between Errors	NTSC 143 Mb/s	PAL 177 Mb/s	Component 270 Mb/s
1 television frame	2×10^{-7}	2×10^{-7}	1×10^{-7}
1 second	7×10^{-9}	6×10^{-9}	4×10^{-9}
1 minute	1×10^{-10}	9×10^{-11}	6×10^{-11}
1 hour	2×10^{-12}	2×10^{-12}	1×10^{-12}
1 day	8×10^{-14}	7×10^{-14}	4×10^{-14}
1 week	1×10^{-14}	9×10^{-15}	6×10^{-15}
1 month	3×10^{-15}	2×10^{-15}	1×10^{-15}
1 year	2×10^{-16}	2×10^{-16}	1×10^{-16}
1 decade	2×10^{-17}	2×10^{-17}	1×10^{-17}
1 century	2×10^{-18}	2×10^{-18}	1×10^{-18}

Table 7-2. Error Rate as a Function of SNR for Serial Digital NTSC

Time Between Errors	BER	SNR (dB)	SNR (volts ratio)
1 microsecond	7×10^{-3}	10.8	12
1 millisecond	7×10^{-6}	15.8	38
1 television frame	2×10^{-7}	17.1	51
1 second	7×10^{-9}	18.1	64
1 minute	1×10^{-10}	19.0	80
1 day	8×10^{-14}	20.4	109
1 month	3×10^{-15}	20.9	122
1 century	2×10^{-18}	21.8	150

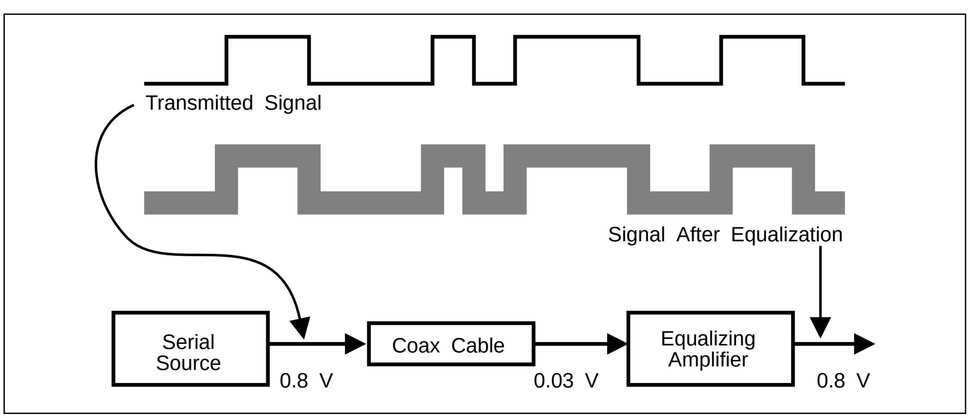

Figure 7-1. Serial transmission system.

Quantifying Errors

Most engineers are familiar with the concept of Bit Error Rate (BER), which is the ratio of bits in error to total bits. As an example, the 10-bit digital component data rate is 270 Mb/s. If there were one error per frame the BER would be 30/(270 x 106) = 1.11×10^{-7} for 525 line systems or 25/(270 x 106) = 0.93 x 10^{-7} for 625 line systems. Table 7-1 shows the BER for one error over different lengths of time for various television systems. BER is a useful measure of system performance where the signal-to-noise ratio at the receiver is such that noise-produced random errors occur.

As part of the serial digital interconnect system, scrambling is used to lower DC content of the signal and provide sufficient zero crossings for reliable clock recovery. It's the nature of the descrambler that a single bit error absolutely causes an error in two words (samples) and has a 50-percent probability of the error in one of the words being in the most, or next to the most, significant bit. Therefore, an error rate of 1 error/frame will be noticeable by a reasonably patient observer. If it's noticeable, it's unacceptable; but it's even more unacceptable because of what it tells us about the operation of the serial transmission system.

Nature of Studio Digital Video Transmission Systems

Specifications for sources of digital video signals are defined by SMPTE 259M. Although the specifications do not include a signal-to-noise ratio (SNR) typical values would be 40 dB or greater at the transmitter. Errors will occur if the SNR at some location in the system reaches a low enough value, generally in the vicinity of 20 dB. Figure 7-1 is a block diagram of the basic serial transmitter and receiver system. An intuitive method of testing the serial system is to add cable – a straight-forward method of lowering the SNR. Since coax itself is not a significant noise source it's the noise figure of the receiver that determines the operating SNR. Assuming an automatic equalizer in the receiver, eventually, as more cable is added, the signal level due to coax attenuation causes the SNR in the receiver to be such that errors occur.

Based on the scrambled NRZI channel code used and assuming gaussian distributed noise, a calculation using the error-function gives the theoretical values shown in Table 7-2. The calibration point for this calculation is based on the capabilities of the serial-digital interface. In the proposed serial digital standard it's stated that the expected operational distance is through a length of coax that attenuates a frequency of 1/2 the clock rate by up to 30 dB. That is, receivers may be designed with less or more capability, but the 30 dB value is considered to be realizable. The data in Table 7-2 for NTSC serial digital transmission shows that a 4.7 dB increase in SNR changes the result from 1 error-per-frame to 1 error-per-century. For NTSC the calibration point for the calculation is 400 meters of Belden 8281 coax. (Various types of coax can be used provided they have a frequency response reasonably meeting the $\sqrt{f}$ characteristic described in the proposed standard. It's the 30 dB-of-loss point that's critical.)

This same theoretical data can be expressed in a different manner to show error rates as a function of cable length as demonstrated in Table 7-3 and shown graphically in Figure 7-2. The graph makes it very apparent that there's a sharp knee in the cable length vs. error rate curve. Eighteen additional meters of cable (5 percent of the total) moves operation from the knee to completely unacceptable while 50 fewer meters of cable (12 percent of the total length) moves operation to a reasonably safe, 1 error/month. Similar results will be obtained for other standards where the calibration point for the calculation is 360 meters for PAL and 290 meters for component. Cable length changes required to maintain headroom 'scales proportionally as shown in Figure 7-3.

Good engineering practice would suggest a 6-dB margin or 80 meters of cable, hence a maximum operating length of about 320 meters in an NTSC system where the knee of the curve is at 400 meters. At that operating level, there should never be any errors (at least not in our lifetime).

Practical systems includes equipment that doesn't necessarily completely reconstitute the signal in terms of SNR. That is, sending the signal through a distribution amplifier or routing switcher may result in a completely useful but not completely standard signal to be sent to a receiving device. The non-standardness could be both jitter and noise, but the sharp-knee characteristic of the system would remain, occurring at a different amount of signal attenuation. Use of properly equalized and re-clocked distribution and routing equipment at intervals with adequate headroom provides virtually

Table 7-3. Error Rate as a Function of Cable Length Using 8281 Coax in for Serial Digital NTSC

Time Between Errors	BER	Cable Length (meters)	Attenuation (dB) at 1/2 Clock Freq
1 microsecond	7×10^{-3}	484	36.3
1 millisecond	7×10^{-6}	418	31.3
1 television frame	2×10^{-7}	400	30.0
1 second	7×10^{-9}	387	29.0
1 minute	1×10^{-10}	374	28.1
1 day	8×10^{-14}	356	26.7
1 month	3×10^{-15}	350	26.2
1 century	2×10^{-18}	≤338	25.3

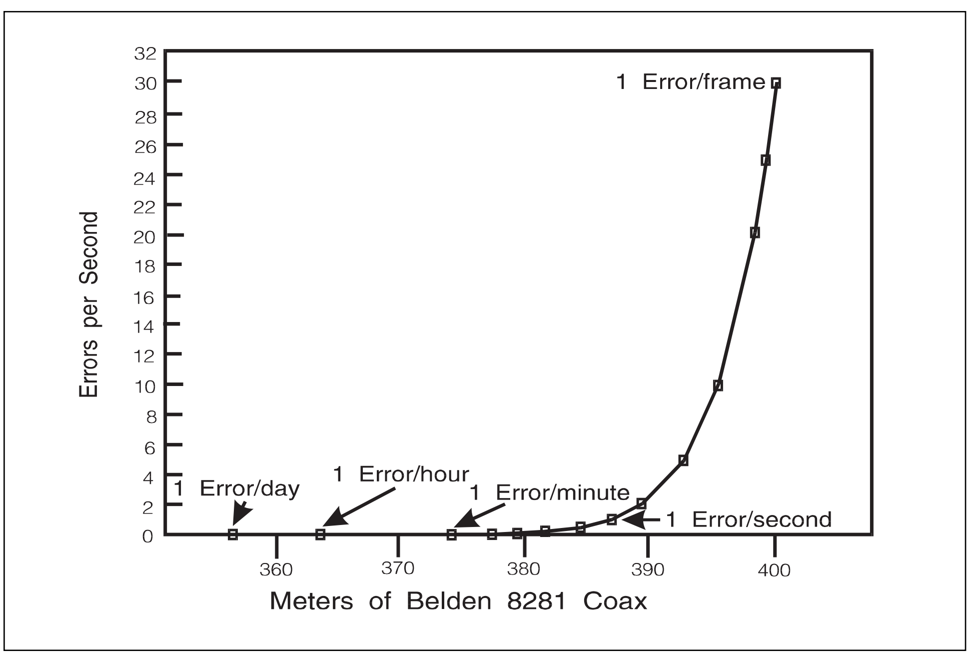

Figure 7-2. Calculated NTSC bit-error rate.

unlimited total transmission distances.

Measuring Bit Error Rates (BER)

BER measurements may be made directly using equipment specifically designed for that purpose. Unfortunately, in a properly operating system, say with 6 dB of headroom, there's no BER to measure. This is because the serial digital television system generally operates in an environment that is free from random errors. If errors are never going to occur (as implied by Table 7-2) with 6 dB of headroom, what's there to measure? A more common problem will be burst errors due to some sort of interfering signal such as a noise spike that occurs at intermittent intervals spaced far apart in time. Another source could be crosstalk that might come and go depending on what other signals are being used at a particular time. Also, there's the poor electrical connection at an interface that would cause noise only when it's mechanically disturbed.

Because of the intermittent nature of burst errors, data recording and communications engineers have defined another error measurement – the Errored Second. The following example demonstrates the benefits of the errored second. Suppose a burst error causes 10,000 errors in two frames of video. A BER measurement made for 1 minute would indicate a 1×10^{-6} BER and a measurement made for 1 day would indicate a 8×10^{-9} BER. Whereas an errored second measurement could indicate that there was one second in error at a time 3 hours, 10 minutes and 5 seconds ago. The errored second method is clearly a more useful measurement in this case.

A significant advantage of errored seconds vs. a straight BER measurement is that it's a better measure of fitness for service of links that are subject to burst errors. The serial digital video system fits this category because television images are greatly disturbed by momentary loss in synchronization. A BER measurement could give the same value for a single, large burst as it does for several shorter scattered bursts. But if several of the shorter bursts each result in momentary sync failure, the subjective effect will be more damage to the viewed picture than caused by the single burst. Errored seconds, and its inverse, error free seconds, do a good job of quantifying this.

For use in serial-digital television systems, there are several disadvantages to direct measurement of BER:

1. It must be an out-of-service test because traditional BER measurements use one of several defined pseudo-random sequences at various bit rates.
2. None of the sequences are particularly similar to the serial-digital video bit stream; hence, some television equipment will not process the test set bit patterns.
3. Consider a system operating with a reasonable 6 dB of SNR margin with respect to 1 error/frame. Errors due to random noise will occur once every hundred years or so. A BER measurement will not be possible.
4. BER measurements do not provide meaningful data when the system-under-test is basically noise free but potentially subject to burst errors.
5. Test sets for BER measurement are expensive considering their limited application in a television system.

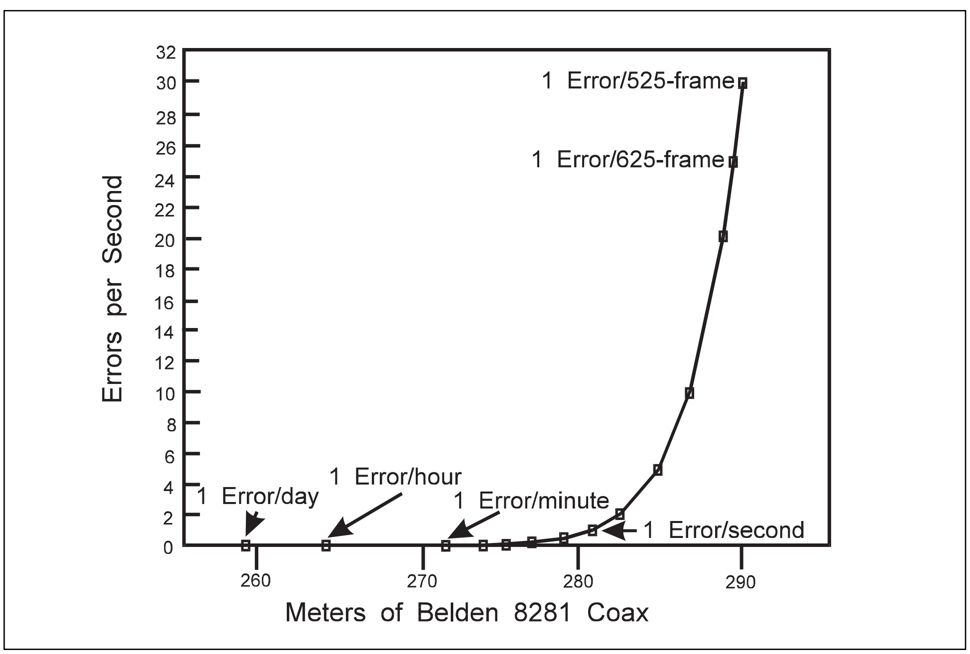

Figure 7-3. Calculated 525/625 component bit-error rate.

An Error Measurement Method for Television

Tektronix has developed an error detection system for digital television signals and has placed the technical details in the public domain to encourage other manufacturers to use the method. This method has proven to be a sensitive and accurate way to determine if the system is operating correctly and it has been approved for standardization by SMPTE as Recommended Practice RP 165. Briefly, the Error Detection and Handling (EDH) concept is based on making Cyclic Redundancy Code (CRC) calculations for each field of video at the serializer as shown in Figure 7-4. Separate CRCs for the full field and active picture, along with status flags, are then sent with the other serial data through the transmission system. The CRCs are recalculated at the deserializer and, if not identical to the transmitted values, an error is indicated. Typical error detection data will be presented as errored seconds over a period of time, and time since the last errored second.

In normal operation of the serial-digital interface, there will be no errors to measure. What's of interest to the television engineer is the amount of headroom that's available in the system. That is, how much stressing could be added to the system before the knee of the error rate curve or crash point is reached. As an out-of-service test, this can be determined by adding cable or other stressing method until the onset of errors. Since it's an out-of-service test, either a BER test set or RP 165 could be used. There are, however, many advantages to using the RP 165 system:

1. The CRC data is part of the serial-digital television signal, thereby providing a meaningful measure of system performance.
2. RP 165 can be used as an in-service test to automatically and electronically pinpoint any system failures.
3. For out-of-service testing, RP 165 is sufficiently sensitive to accurately define the knee of the error rate curve during stressing tests.
4. Where there are errors present, RP 165 provides the information necessary to determine errored seconds, which is more useful than bit-error rate.
5. Facility is provided for measuring both full-field and active-picture errors. Optional status flags are also available to facilitate error reporting.
6. CRC calculation can be built into all serial transmitters and receivers for a very small incremental cost. With error information available from a variety of television equipment, the results can then be routed to a central collection point for overall system diagnostics.

Composite digital systems already must have a coprocessor to handle special timing signals required in the serial data and not allowed in the parallel data. Adding basic CRC capabilities to the coprocessor costs close to nothing. For component systems that do not have the differences between serial and parallel, there's a small incremental cost which should be a fraction of a percent of the total cost in equipment where it's appropriate to use this method.

System Considerations

In a large serial digital installation the need for traditional waveform monitoring can be reduced by the systematic application of error detection methods. At signal source locations, such as cameras, DVEs and VTRs, where operational controls can affect the program signal, it will continue to be important to verify the key program signal parameters using waveform monitors with serial digital input capabilities. The results of certain technical operations, such as embedded audio mux/demux, serial link transmission, and routing switcher I/O, can be monitored with less sophisticated digital-only equipment provided the signal data integrity is verified. It's the function of the RP 165 system to provide that verification.

For equipment where the digital signal remains in the serial domain, such as routing switchers or distribution amplifiers, it's not economical to provide the CRC calculation function. Therefore, a basic error detection system will be implemented as

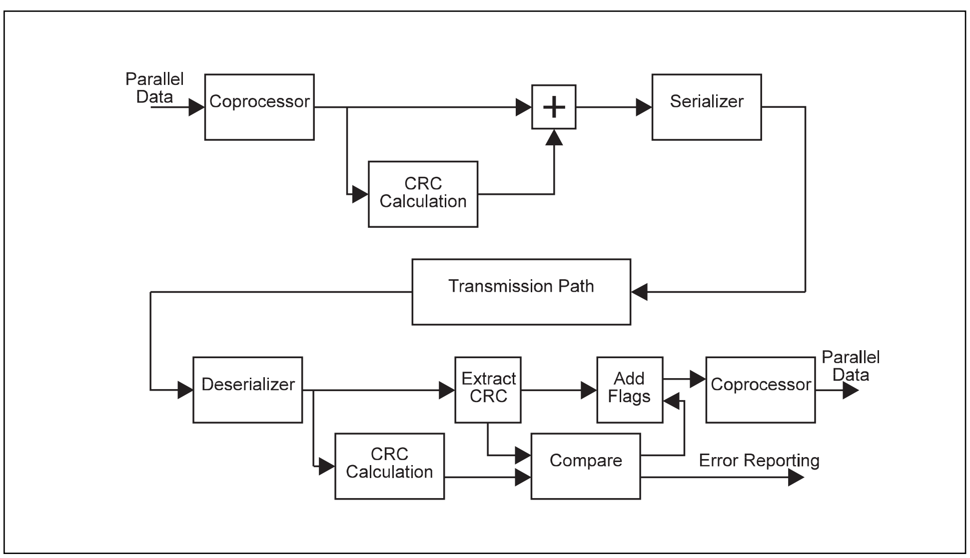

Figure 7-4. Error detection concept.

shown in Figure 7-5. Equipment that processes the signal in the parallel domain, such as VTRs, DVEs or production switchers, should provide the transmit and receive CRC calculation necessary for error detection. That equipment can then report errors locally and/or to a central diagnostics computer. Routing switchers and other equipment operating in the serial domain will have their signal path integrity verified by the error detection system.

To emphasize the importance of including the CRC calculation in parallel-domain processing equipment, consider the block diagram in Figure 7-6. Since the signal source doesn't have an internal CRC data generator one might consider a black box to provide that function. Unfortunately, the cost of deserializing and reserializing makes the cost prohibitively high so such a system is not recommended. At a receiver that doesn't calculate the CRC, it's possible, and not unreasonable, to provide a monitor with either passive or active loop-through that determines the signal integrity. The drawback of this system is that it's the monitor's receiver that's being used to process the signal, not the actual destination equipment. It's possible that different receiver characteristics of the two pieces of equipment would provide misleading results. If the receiver being tested has an active loop-through, it's useful to place the error detecting monitor at that output. Errors detected would most likely be due to the first receiver, that's the unit being tested, eliminating the difference in receiver sensitivities.

To extend the idea of system diagnostics beyond simple digital data error detection, it would be useful to provide an equipment fault reporting system. SMPTE 269M documents a single contact closure fault reporting system and discussions have begun on computer data communications protocols for more sophisticated fault reporting. Error detection is an important part of system diagnostics as shown in Figure 7-7. Information relating to signal transmission errors are combined with other equipment internal diagnostics to be sent to a central computer. A large routing switcher can economically support internal diagnostics. Using two serial receivers a polling of serial-data errors could be implemented. The normal internal bus structure could be used to send serial data to a receiver to detect input errors and a special output bus could be used to ensure that no errors were created within the large serial domain routing switcher.

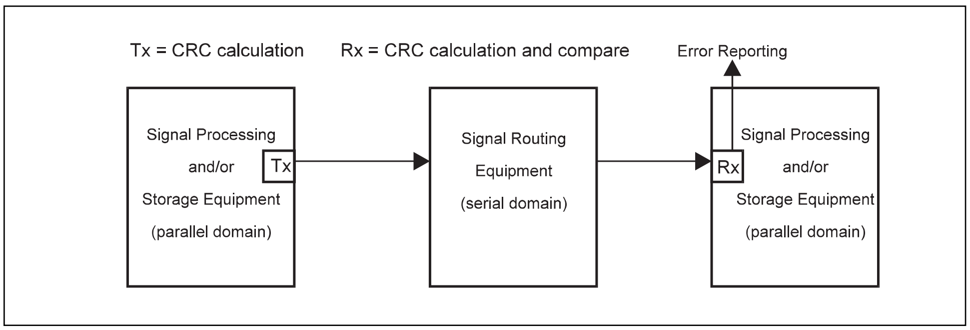

Figure 7-5. Using error detection (basic).

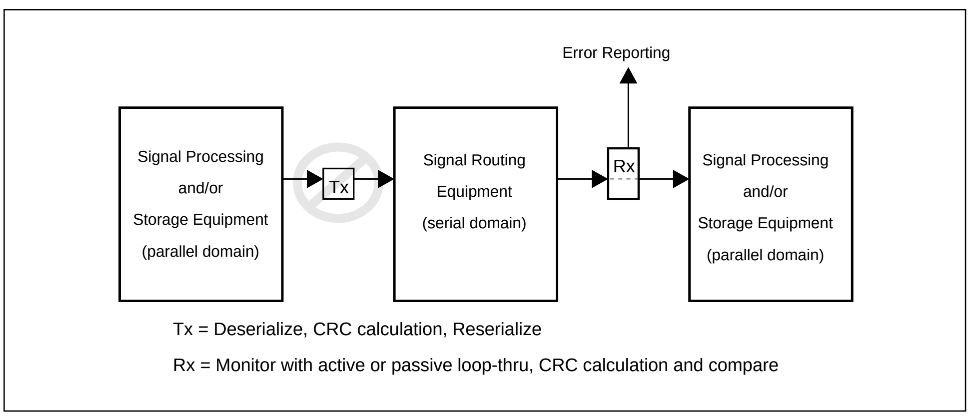

Figure 7-6. Using error detection (alternate).

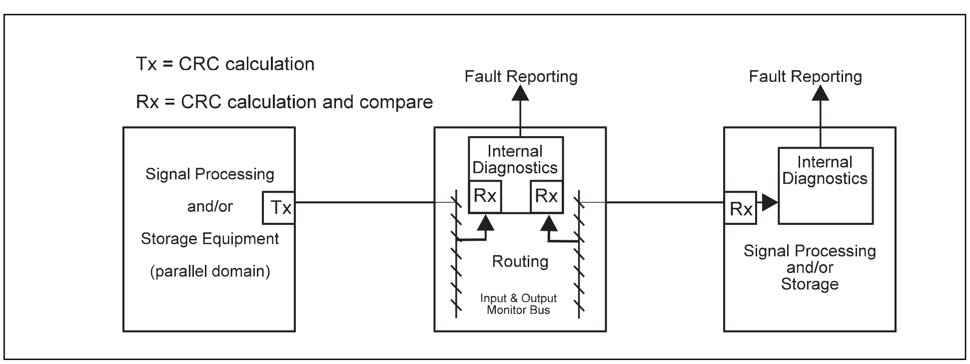

Figure 7-7. System fault reporting.

8. Jitter Effects and Measurements

Digital transmission systems can operate with a considerable amount of jitter; in fact, more jitter than would be tolerated in the program signal represented by the digits. This section looks at the reasons behind this separation of jitter effects – the amount of jitter that may be found in different digital transmission systems and how to measure the jitter in serial transmission systems.

The jitter specification in the standard for the serial-digital video signal is "the timing of the rising edges of the data signal shall be within +0.25 ns of the average timing of rising edges, as determined over a period of one line." As of the first publication of the standard in February 1992, there's a note to the specification which states "this specification is tentative with further work in progress to determine the measurement method." In fact, both the specification and its measurement method are the subject of SMPTE engineering committee work.

In parallel component transmission, jitter specifications for the 27 MHz clock signal are stated as "the peak-to-peak jitter between rising edges shall be within 3 ns of the average time of the rising edge computed over at least one field." For NTSC composite, the specification is 5 ns.

Digital audio per the AES standard is a serial signal that is a multiplex of samples from two audio channels and some ancillary data (see digital audio formats in Section 3). For television, the preferred audio sampling rate is 48 kHz, clock-locked to video as required for digital VTRs. There are 32 bits of data associated with each audio sample, which produces a 2-channel (one serial stream) data rate of 3.07 Mb/s. Since the channel coding scheme (biphase mark) inserts a transition for each clock, the resulting transition rate of the serial signal can be 6.14 MHz, giving an eye pattern unit interval of 163 ns. For AES audio, the jitter specification is "data transitions shall occur within +20 ns of an ideal jitter-free clock."

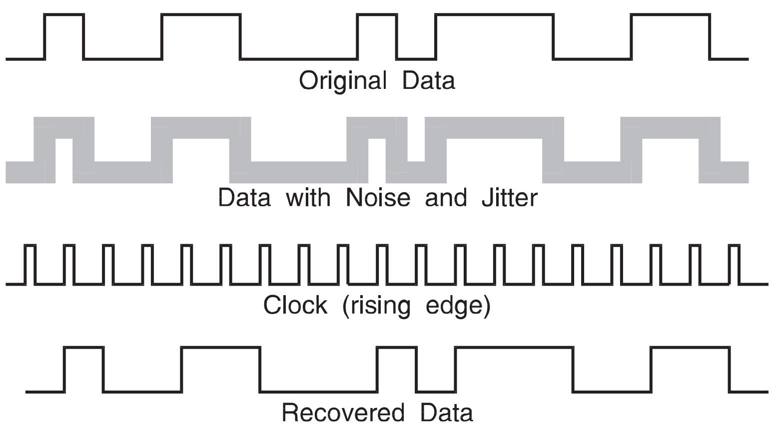

Figure 8-1. Data recovery with a noise-free clock.

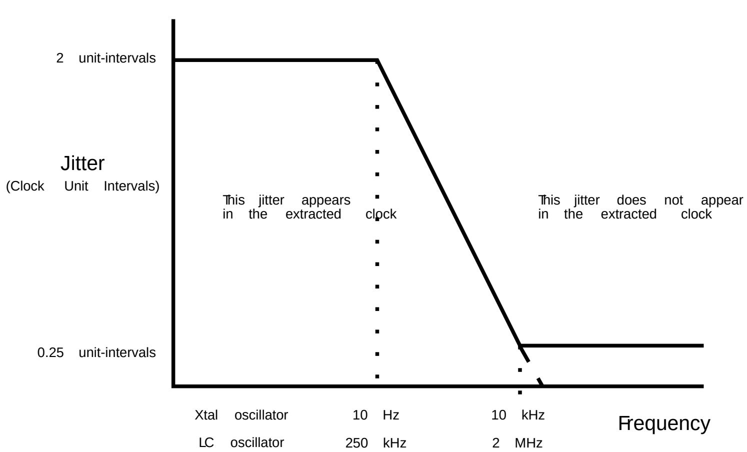

Figure 8-2. Receiver jitter sensitivity.

Jitter In Serial Digital Signals

Jitter, noise, amplitude changes, and other distortions to the serial-digital signal can occur as it's processed by distribution amplifiers, routing switchers, and other equipment that operate on the signal exclusively in its serial form. Figure 8-1 shows how the recovered signal can be as perfect as the original if the data is detected with a jitter-free clock. As long as the noise and jitter don't exceed the threshold of the detection circuits (that's, the eye is sufficiently open) the data will be perfectly reconstructed.

In a practical system, the clock is extracted from the serial bit stream and contains some of the jitter present in the signal. Jitter in the clock can be a desirable feature to the extent that the jitter helps position the clock edge in the middle of the eye. Large amounts of low-frequency jitter can be tolerated in serial receivers if the clock follows the eye opening location as its position varies with time. An example of receiver jitter sensitivity is shown in Figure 8-2 where the solid line represents the amount of jitter that would cause data errors at the receiver. For low jitter frequencies, eye location variations of up to several unit intervals will not defeat data extraction. As the jitter frequency increases, the receiver tolerance decreases to a value of about 0.25 unit intervals. The break points for this tolerance curve depend on the type of phase-locked loop circuitry used in clock extraction.

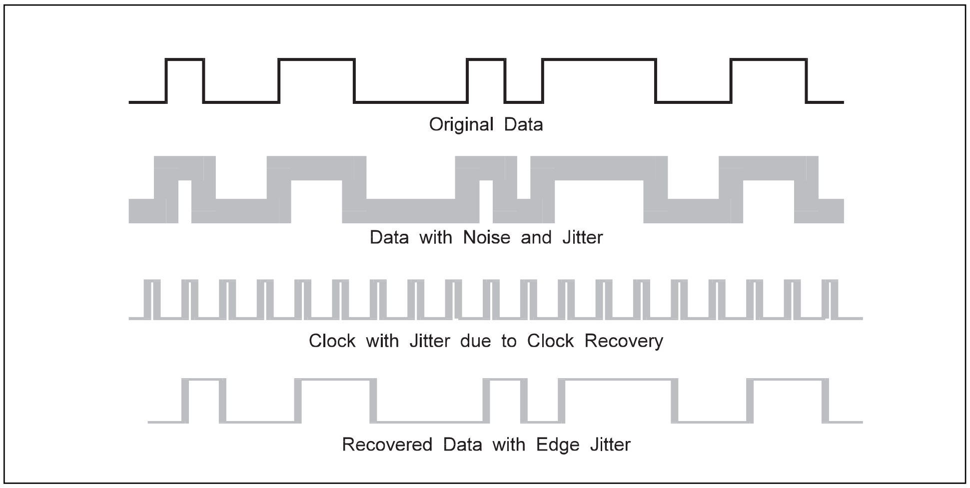

Figure 8-3. Data recovery with extracted clock.

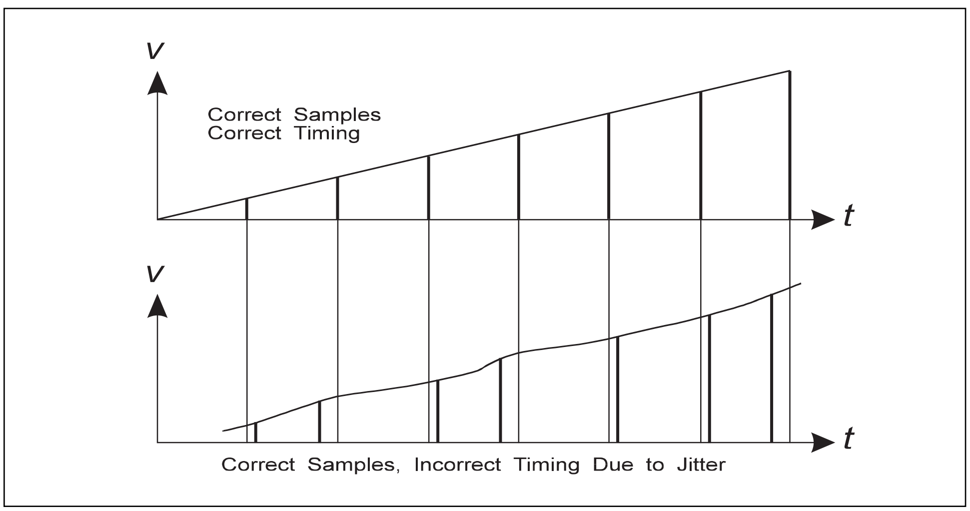

Figure 8-4. Signal errors due to clock jitter.

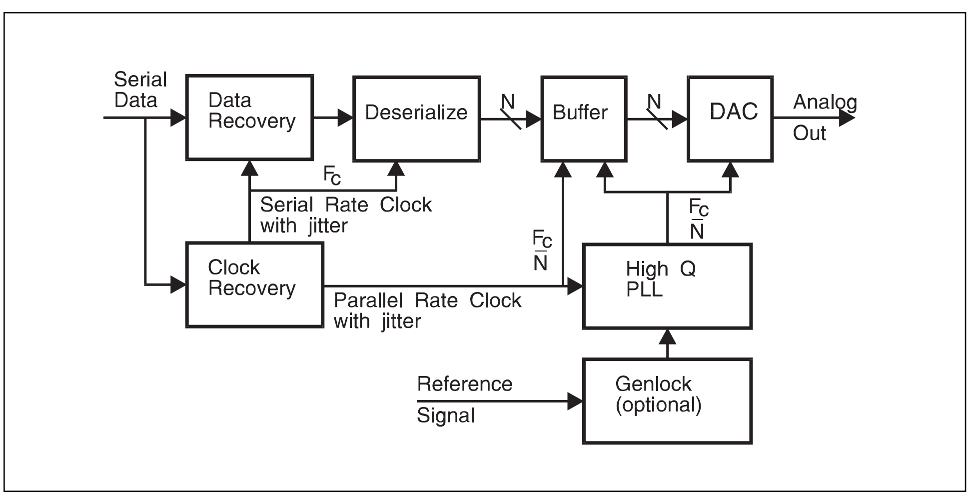

Figure 8-5. DAC with a low-jitter clock.

Based on the acceptance of some jitter in the extracted clock used to recover the data, two types of jitter are defined:

1. Timing Jitter is defined as the variation in time of the significant instants (such as, zero crossings) of a digital signal relative to a clock with no jitter above some low frequency (about 10 Hz).
2. Alignment Jitter (or relative jitter) is defined as the variation in time of the significant instants (such as, zero crossings) of a digital signal relative to a clock recovered from the signal itself. (This clock will have jitter components above 10 Hz but none above a higher frequency in the 1 kHz to 10 kHz range.)

Since the data will generally be recovered using a clock with jitter, the resulting digital information may have jitter on its transition edges as shown in Figure 8-3. This is still completely valid information since digital signal processing will only consider the high/low value in the middle of the clock period. However, if the same clock with jitter (or simple submultiple thereof) is used to convert the digital information to analog, errors may occur as shown in Figure 8-4. Sample values converted to analog at exactly the correct times produce a straight line whereas use of a clock with jitter produces an incorrect analog waveform. The relative effect on the analog waveform produced using the same clock for digital-to-analog conversion and for data extraction from the serial signal depends on the amount of jitter (in unit intervals) and the number of quantizing levels in the digitized signal. Where reduced jitter is required for the digital-to-analog converter (DAC) clock, a circuit such as shown in Figure 8-5 may be used where a high Q (crystal) phase-locked loop produces a clock with much lower jitter.

Typical jitter specifications for serial audio and serial/parallel video signals are shown in Table 8-1. The AES/EBU standard for serial digital audio allows +20 ns of jitter, which is appropriate as the peak-to-peak value of 40 ns is about 1/4 of a unit interval. As explained above, the DAC clock jitter requirements are considerably tighter. A draft AES/EBU standard specifies the DAC clock at 1 ns jitter. However, a theoretical value for 16-bit audio could be as small as 0.1 ns. The important point is that receivers for AES/EBU serial audio should be designed to handle up to 40 ns of jitter, whereas the internal DAC circuitry of such receivers must reduce the clock jitter to an acceptable level.

The situation for parallel digital video is similar; however, practical implementations have generally avoided the whole issue. That is, most or all parallel digital video clocks have jitter in the 1 ns or smaller region, and most or all parallel receivers do not allow or account for clocks with more than that small amount of jitter. For 525/625 video signals, input to a 10-bit full-range D/A converter, the allowable clock jitter is based on a burst frequency, 25% amplitude signal (approximately burst amplitude) causing less than one LSB of error. For high-definition signals, a 20 MHz signal is used. In the author's opinion, designers of new equipment would do well to follow this tradition and not produce parallel clock jitter with values in the region of the published specification. For serial digital systems, the engineering community is still considering whether low-frequency jitter greater than the nominal 0.5 ns should be allowed. That is, should clocks extracted from all serial-digital signals be appropriate as inputs to D/A converters without using jitter reduction or should D/A converters be expected to provide the jitter reduction circuitry.

Table 8-1. Typical Jitter Specifications

Standard	Clock Period	Jitter Spec	% of Clock	DAC Requirement
AES/EBU Audio	163 ns	40.0 ns	25%	1.0 ns spec 0.1 ns 16-bit
Serial NTSC	7 ns	0.5 ns	7%	0.5 ns
Serial PAL	6 ns	0.5 ns	9%	0.5 ns
Serial Component	4 ns	0.5 ns	14%	0.5 ns
Parallel NTSC	70 ns	10.0 ns	15%	0.5 ns
Parallel PAL	56 ns	10.0 ns	20%	0.5 ns
Parallel Component	37 ns	6.0 ns	16%	0.5 ns
Parallel HD	7 ns	1.0 ns	14%	0.1 ns

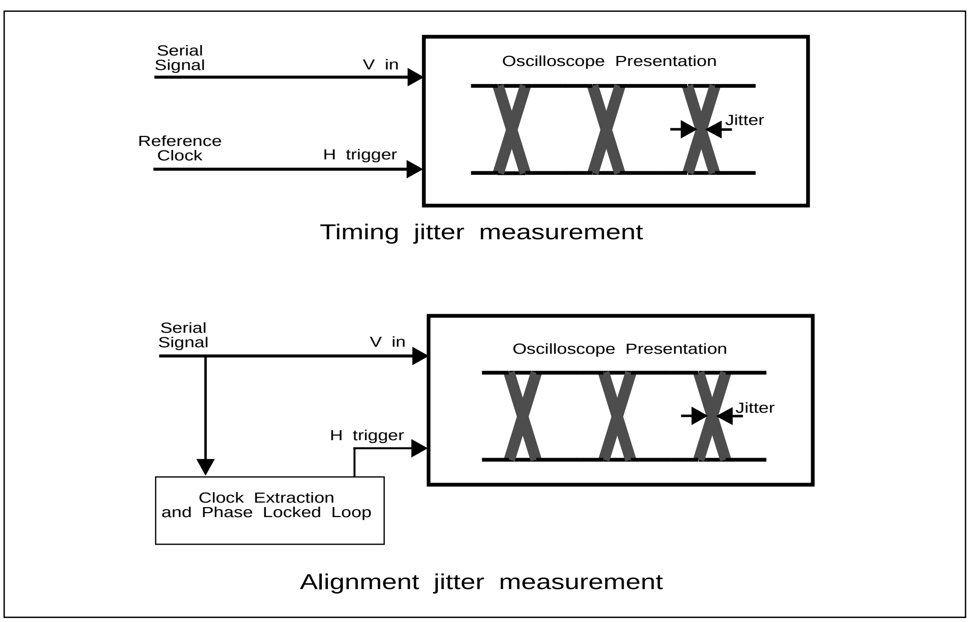

Figure 8-6. Measurements using a clock reference.

Measuring Jitter

There are three oscilloscope-related methods of measuring jitter in a serial digital signal. Measurement of timing jitter requires a jitter-free reference clock as shown in the top part of Figure 8-6. Alignment (relative) jitter is measured using a clock extracted from the serial signal being evaluated as shown in the bottom part of this figure. Timing jitter measurements include essentially all frequency components of the jitter as indicated by the dashed line leading into the solid line in Figure 8-7. The jitter frequency components measured using the alignment-jitter method will depend on the bandwidth the clock extraction circuit. Low-frequency components of jitter will not be included because the extracted clock follows the serial signal jitter. These low-frequency jitter components are not significant for data recovery provided the measurement clock extraction system has the same bandwidth as the data recovery clock extraction system. All jitter frequencies above a certain value will be measured with break points between the two areas at frequencies appropriate for the bandwidth of the clock extraction system used. When deriving the scope trigger it's important to consider the frequency division that usually takes place in the clock extractor. If it's a divide-by-ten, then word-synchronized jitter will not be observed depending on whether the sweep time of the display covers exactly 10 zero crossings. Division by a number other than the word length will ensure that all jitter is measured.

The third method of observing, but not measuring, jitter is the simple but deceptive self-triggered scope measurement depicted in Figure 8-8. By internally triggering the scope on the rising (or falling) edge of the waveform it's possible to display an eye pattern at a delayed time after the trigger point. Using modern digital sampling scopes, long delays may be obtained with extremely small amounts of jitter attributable to the delay circuits in the scope. At first glance, this looks like a straight forward way to evaluate the eye pattern since a clock extraction circuit is not required. The problem is that the components of jitter frequency that are being measured are a function of the sweep delay used and that function is a comb filter. As a result, there are some jitter frequencies that will not be measured and others that will indicate twice as high a value as would be expected. Although there's a low-pass effect with this type of measurement, the shape of the filter is much different than that obtained with an extracted clock measurement; hence, the results may not be a good indication of the ability of a receiver to recover the data from the serial bit stream.

As an example, component serial digital video has a unit interval of 3.7 ns. If the sweep delay is 37 ns the tenth zero crossing will be displayed. In many of today's serializers there's a strong jitter component at one tenth of the clock frequency (due in part to the times 10 multiplier in the serializer). A self-triggered measurement at the tenth zero crossing will not show that jitter. Alternately, a measurement at other zero crossings may show as much as two times too high a value of jitter. To further complicate the matter, if the one-tenth frequency jitter were exactly symmetrical (such as, a sine wave) the fifth zero crossing would also have no jitter. In practice, the fifth crossing also shows a lot of jitter; only the tenth, twentieth, thirtieth, etc., crossings appear to be nearly jitter free.

In troubleshooting a system, the self-triggered method can be useful in tracking down a source of jitter. However, proper jitter measurements to meet system specifications should be made with a clock extraction method. A SMPTE ad-hoc group is developing methods to measure and specify jitter for serial digital video, most likely incorporating a defined-clock extraction system.

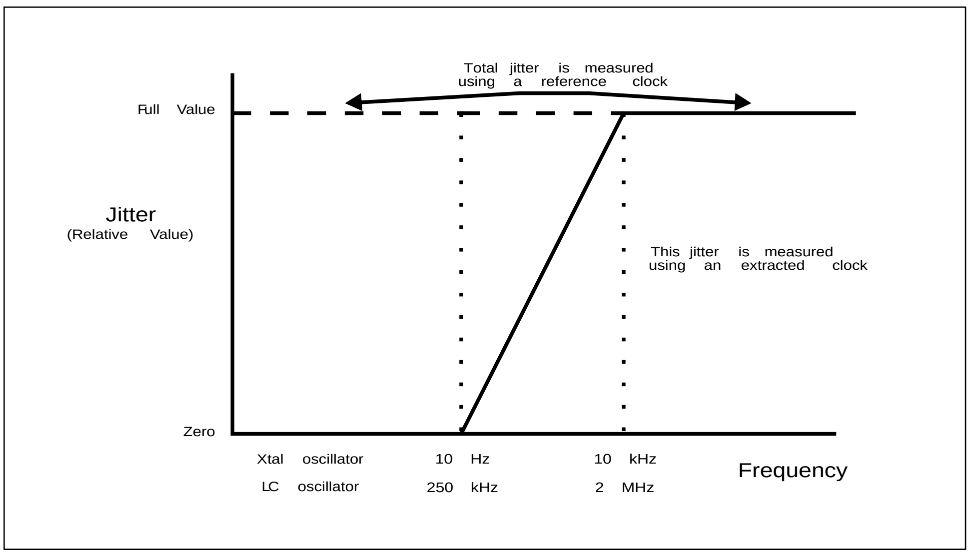

Figure 8-7. Measured jitter components.

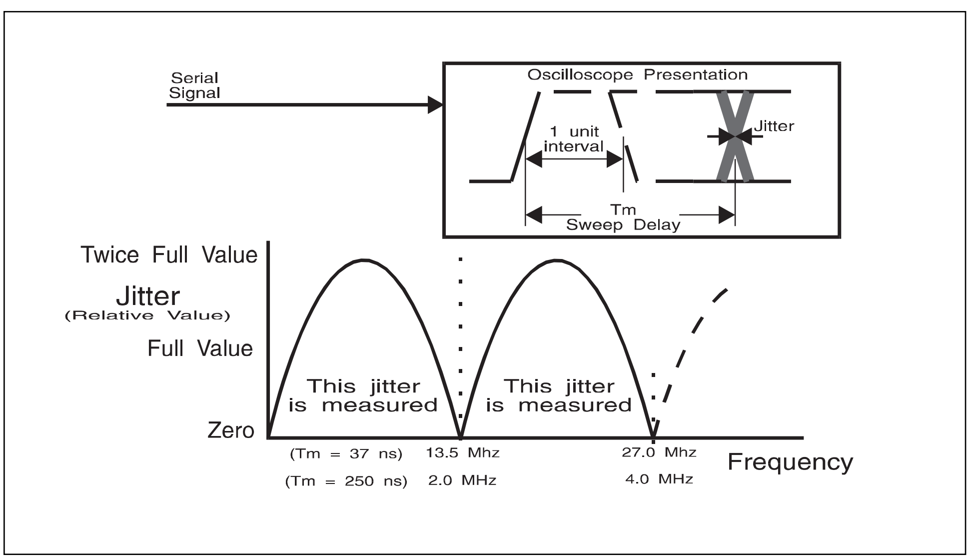

Figure 8-8. Self-triggered jitter observations.

9. System Testing

Stress Testing

Unlike analog systems that tend to degrade gracefully, digital systems tend to work without fault until they crash. This characteristic is most clearly illustrated by the discussion of error detection in Section 7. However, other stressing functions will produce much the same result. When operating a digital system, it's desirable to know how much headroom is available; that is, how far the system is away from the crash point. To date, there are no in-service tests that will measure the headroom; but research is being pursued in this area. Therefore, out-of-service stress tests are required to evaluate system operation.

Stress testing consists of changing one or more parameters of the digital signal until failure occurs. The amount of change required to produce a failure is a measure of the headroom. Starting with the specifications in the serial digital video standard (SMPTE 259M), the most intuitive way to stress the system is to add cable until the onset of errors. Other tests would be to change amplitude or risetime, or add noise and/or jitter to the signal. Each of these tests are evaluating one or more aspect of the receiver performance, specifically automatic equalizer range and accuracy and receiver noise characteristics. Experimental results indicate that cable-length testing is the most meaningful stress test because it represents real operation. Stress testing receiver ability to handle amplitude changes and added jitter are useful in evaluating and accepting equipment, but not too meaningful in system operation. (Measuring the signal amplitude at the transmitter and measuring jitter at various points in the system is important in operational testing as discussed in Sections 6 and 8, but not as stress testing.) Addition of noise or change in risetime (within reasonable bounds) has little effect on digital systems and is not important in stress tests.

Cable-length stress testing can be done using actual coax or a cable simulator. Coax is the real world and most accurate method. The key parameter to be measured is onset of errors because that defines the crash point as described in Section 7. With an error measurement method in place, the quality of the measurement will be determined by the sharpness of the knee of the error curve. As an example, using 8281 coax, a five-meter change in length will go from no errors in one minute to more than one error-per-second.

To evaluate a cable simulator, the natural approach is to compare its loss curve with coax using a network analyzer. Certainly that's an appropriate criteria; but the most important is still sharpness of the error curve at various simulated lengths. Experiments have shown that good cable simulators require a 10- to 15-meter change in added 8281 coax to produce the no-errors to more than one error-per-second change. Simulators that require a longer change in coax (less sharp knee) are still useful for comparison testing but should be avoided in equipment evaluation.

SDI Check Field

The SDI (serial digital interface) Check Field (also known as "pathological signal") is not a stress test; but because it's a full-field test signal, it must be an out-of-service test. It's a difficult signal for the serial digital system to handle and is a very important test to perform. The characteristic of the SDI Check Field is that it has a maximum amount of low-frequency energy in two separate signals. One signal tests equalizer operation and the other tests phase-locked loop operation. It's a fully valid signal for component digital and was originally developed to test D-1 recorders. In the composite domain, it's not a valid signal (hence may be considered a stress test); however, it's also a good test to perform. The SDI Check Field is defined in proposed SMPTE Recommended Practice RP 178.

Scrambled NRZI. It's the mathematics of the scrambling process that produces the pathological signal. Although the quantizing levels 3FF and 000 are excluded from the active picture and ancillary data, the ratio of "1"s to "0"s may be quite uneven for some values of a flat color field. This is one of the main reasons for scrambling the signal. Conversion to NRZI eliminates the polarity sensitivity of the signal but has little effect on the ratio of "1"s to "0"s. Scrambling does indeed break up long runs of "1"s or "0"s and produces the desired spectrum with minimum low-frequency content, while providing maximum zero crossings needed for clock extraction at the receiver. However, there are certain combinations of data input and scrambler state that will, occasionally, cause long runs (20 to 40) of "0"s. These long runs of "0"s create no zero crossings and are the source of low-frequency content in the signal. (All "1"s is not a problem because this produces an NRZI signal at one-half the clock frequency, which is right in the middle of the frequency band occupied by the overall signal.)

The scrambler is a nine-cell shift register as shown in Figure 9-1. For each input bit, there is one output bit sent to the one-cell NRZI encoder. An output bit is determined by the state of the 10 cells and the input bit. A typical distribution of runs of “1”s and “0”s is shown in Table 9-1 for 100 frames of component black. (For the purpose of this analysis the “1”s are high and the “0”s are low as opposed to the transition, no-transition data definition.) The longer runs (16, 19, 22, 32, 33, 34, and 39) occur as single events that the transmission system amplifiers and receivers can handle with relative ease because of their very short duration.

Multiple long run events can be encouraged with certain input signals, which is the basis for the generation of pathological signals. When the input words alternate between certain digital values the multi-event pathological signal will occur for one specific scrambler state, generally all “0”s, hence the sequence usually starts at an SAV. The multi-event sequence ends at the next EAV, which breaks the input sequence. Since the scrambler state is more or less a random number, the multi-event sequence of long runs of “0”s happens about once per frame (9-bits, 512 possible states, 525 or 625 lines per frame).

Pathological Signals, the SDI Check Field. There are two forms of the pathological signal as shown in Figure 9-2. For testing the automatic equalizer, a run of 19 “0”s followed by two “1”s produces a signal with large DC content. Remember a “0” is no transition and a “1” is represented by a transition in the NRZI domain. In addition, the switching on and off of this high DC content signal will stress the linearity of the analog capabilities of the serial transmission system and equipment. Poor analog amplifier linearity results in errors at the point of transition where peak signal amplitude is the greatest. It’s important to produce both polarities of this signal for full system testing. Phase-locked loop testing is accomplished with a signal consisting of 20 NRZI “0”s followed by a single NRZI “1”. This provides the minimum number of zero crossings for clock extraction.

The SDI Check Field consists of one-half field of each signal as shown in Figure 9-3. Equalizer testing is based on the values 300h and 198h

Table 9-1. 100 Frames of Component Black

Length	“1”s	“0”s	Length	“1”s	“0”s
1	112615438	112610564	21		
2	56303576	56305219	22	73759	13579
3	28155012	28154963	23		
4	14076558	14077338	24		
5	7037923	7038595	25		
6	3518756	3518899	26		
7	1759428	1760240	27		
8	1059392	699666	28		
9	610241	970950	29		
10	1169	1060	30		
11	14	156	31		
12	40		32	55	51
13	156	114	33	49	
14			34		3
16	13713	73865	36		
19	101	94	39		50

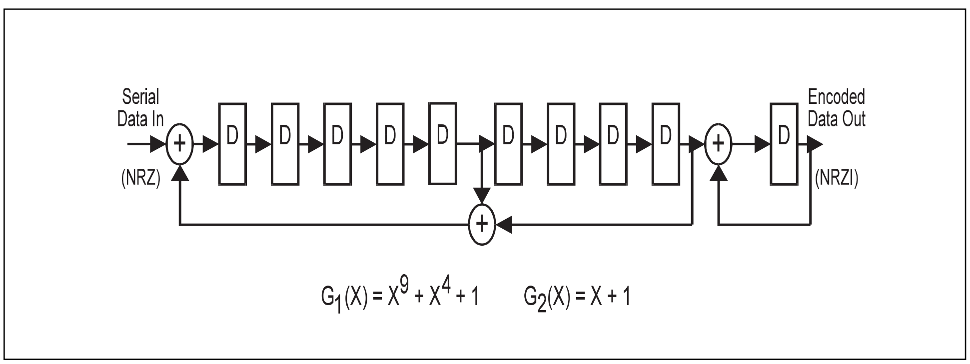

Figure 9-1. Serial scrambler and encoder.

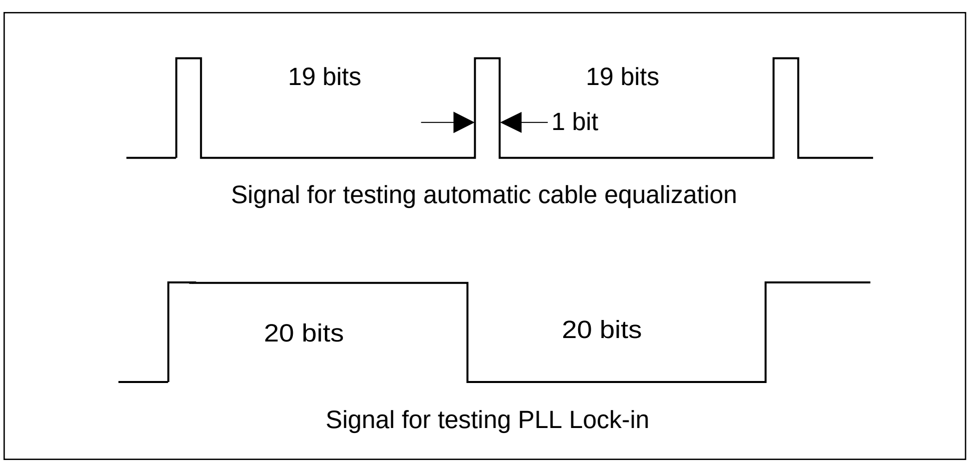

Figure 9-2. Signal forms.

Table 9-2. 100 Frames of SDI Check Field

Length	"1"s	"0"s	Length	"1"s	"0"s
1	112595598	112595271	21		
2	56291947	56292583	22	9344	9425
3	28146431	28147229	23		
4	14073813	14072656	24		
5	7035240	7035841	25		
6	3518502	3518856	26		
7	1758583	1758969	27		
8	879203	879359	28		
9	706058	706029	29		
10	51596	51598	30		
11	146	154	31		
12	16858	16817	32	49	49
13	34405	34374	33	35	32
14	17445	17442	34	3	4
16	9558	9588	36		
18	29	27	38		
19	20944	19503	39	13	14
20	19413	19413	40		

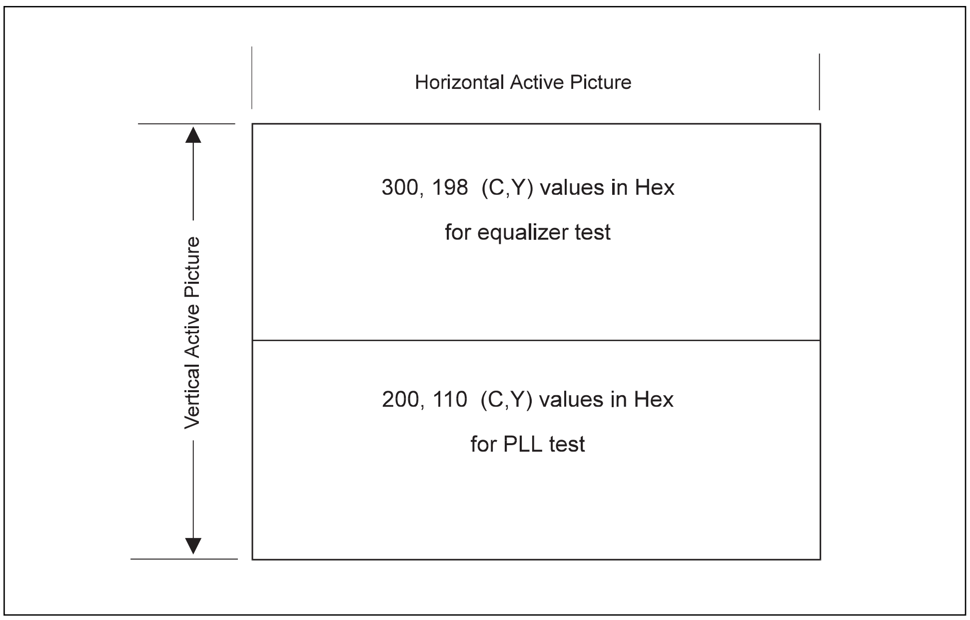

Figure 9-3. SDI check field.

while phase-locked loop testing is based on the values 200h and 110h. In the C-Y order shown, the top half of the field will be a purple here and the bottom half a shade of gray. Some test signal generators use the other order, Y-C, giving two different shades of green. Either one works, but the C-Y order is specified in the proposed recommended practice. One word with a single bit of "1" is placed in each frame to ensure both polarities of equalizer stress signal for the defined SDI Check Field. The single "1" inverts the polarity of the NRZ to NRZI function because the field (for component video) would otherwise have an even number of "1"s and the phase would never change.

Statistics for 100 frames of SDI Check Field are shown in Table 9-2. It's the runs of 19 and 20 that represent the low-frequency stressing of the system. When they happen (about once per frame) they occur for a full active television line and cause a significant low-frequency disturbance. Since the PLL stress signal is a 20 "1"s, 20 "0"s squarewave, you would expect an even number of each type of run. The equalizer stress signal is based on runs of 19 "1"s followed by a "0" or 19 "0"s followed by a "1". As described above, an extra "1" in each frame forces the two polarities to happen and both are represented in the data.

10. Bibliography

Articles and Books

Ainsworth, Kenneth. "Monitoring the Serial Digital Signal," ***Broadcast Engineering,*** Nov. 1991. Tektronix literature number 25W-7156.

Ainsworth, Kenneth and Andrews, Gary. "Evaluating Serial Digital Video Systems," ***IEEE 4th International Montreux Symposium,*** Montreux, Switzerland, June 1991.

Ainsworth, Kenneth and Andrews, Gary. "Comprehending the Video Digital Serial," ***Broadcast Engineering,*** Spanish language edition, Sept. 1992, p. 34-44. Tektronix literature number 25W-7153.

Avis, Richard, Sony Broadcast & Pro-Audio Division, Basingstoke, U.K. "Serial Digital Interface," ***Booklet. Sony Broadcast & Communications***, Basingstoke, England, June 1991.

Elkind, Bob. Ainsworth, Kenneth. Fibush, David. Lane, Richard. Overton, Michael. "Error Measurements and Analysis of Serial CCIR-601 Equipment," ***International Broadcaster's Convention,*** Amsterdam, Holland, June 1992.

Elkind, Bob and Fibush, David. "Proposal for Error Detection and Handling in Studio Equipment," 25th SMPTE Television Conference, Detroit, MI, Feb. 1991. ***SMPTE Journal,*** Dec. 1991. Tektronix literature number 25W-7128-1.

Eguchi, Takeo. Sony Corporation, Japan. "Pathological Check Codes for Serial Digital Interface Systems," 133rd SMPTE Technical Conference, Los Angeles, CA, Oct. 1991. ***SMPTE Journal,*** Aug. 1992, p. 553-558.

Fibush, David and Elkind, Bob. "Test and Measurement of Serial Digital Television Systems," 133rd SMPTE Technical Conference, Los Angeles, CA, Oct. 1991. ***SMPTE Journal,*** Sept. 1992, p. 622-631. Tektronix literature number 25W-7154.

Fibush, David. "Understanding the Serial Digital Video Interface," ***In View,*** Midwest Publishing, Winter/Spring 1991, p. 16-18.

Fibush, David. "Error Detection in Serial Digital Systems," NAB, Las Vegas, NV, Apr. 1993. ***SMPTE Journal,*** Aug. 1993, p. 688-692. Tektronix literature number 25W-7198.

Fibush, David. "Jitter Effects and Measurements in Serial Digital Television Signals," 26th SMPTE Conference, Feb. 1993. ***SMPTE Journal,*** Aug. 1993, p. 693-698. Tektronix literature number 25W-7179.

Fibush, David. "Error Measurements in Studio Digital Video Systems," 134th SMPTE Technical Conference, Toronto, Canada, Nov. 1992. ***SMPTE Journal,*** Aug. 1993, p. 688-692. Tektronix literature number 25W-7185.

Fibush, David. "Test and Measurement of Serial Digital Signals error detection and jitter," ***SBE Convention,*** San Jose, CA, Oct. 1992.

Grass Valley Group, "Digital Television: The Transition," ***Grass Valley Group, Inc.,*** Grass Valley, CA. 1992

Huckfield, Dave. Sony Broadcast & Communications Ltd. "Serial Digital Interfacing," ***International Broadcast Engineer,*** Sept. 1990.

Mendelson, Philip. Digital Magic, Santa Monica, CA. "Building a Serial Component Facility," ***Broadcast Engineering,*** Mar. 1992, p. 38, 42, 44, 48, 50.

Martin, Bruno. Advanced Audio Visual Systems, Montreuil, France. "A Comprehensive Method of Digital Video Signal Analysis," ***135th SMPTE Technical Conference,*** Los Angeles, California, Nov. 1993.

Reynolds, Keith. Grass Valley Group, Grass Valley, CA. "The Transition Process: Getting from A to D," ***Broadcast Engineering,*** Mar. 1992, p. 86-90.

Robin, Michael and Poulin, Michel. CBC, Montreal, Canada. "Performance Evaluation and Acceptance Testing of Bit-Serial Digital Video Equipment and Systems at the CBC," ***134th SMPTE Technical Conference,*** Toronto, Canada, Nov. 1992.

Robin, Michael. CBC, Montreal, Canada. "Bit-serial Digital Signal Jitter Causes, Effects, Remedies and Measurement," ***135th SMPTE Technical Conference,*** Los Angeles, California, Nov. 1993.

Stremler, Ferrel G. "Introduction to Communications Systems," ***Addison-Wesley Series in Electrical Engineering,*** December 1982.

Smith, David R. ***Digital Transmission Systems,*** Van Nostrand Reinhold Company, New York, 1985.

Stone, David. Pro-Bel Ltd. "Digital Signal Analysis," ***International Broadcast Engineer,*** Nov. 1992, p. 28-31.

Tancock, M.B., Sony Broadcast & Communications, Basingstoke, U.K. "The Digital Evolution: The Serial Digital Interface," ***Sony, Basingstoke, U.K. Booklet released at IBC Convention***, Brighton, England, Sept. 1990.

Walker, Marc S. BTS, Salt Lake City, UT. "Distributing Serial Digital Video," ***Broadcast Engineering,*** Feb. 1992, p. 46-50.

Ward, Ron. Utah Scientific, Salt Lake City, UT. "Avoiding the Pitfalls in Serial Digital Signal Distribution," ***SMPTE Journal,*** Jan. 1993, p. 14-23.

Watkinson, John. ***The Art of Digital Video,*** Focal Press, London & Boston.

Watkinson, John and Rumsey, Francis. ***The Digital Interface Handbook,*** Focal Press, London & Boston.

Wonsowicz, John. CBC, Toronto, Canada. "Design Considerations for Television Facilities in the CBC Toronto Broadcast Center," ***134th SMPTE Technical Conference,*** Toronto, Canada, Nov. 1992.

Standards

AES 3 (ANSI S4.40), "AES Recommended Practice for Digital Audio Engineering, Serial Transmission Format for Linearly Represented Digital Audio Data"

AES 11 (Draft ANSI S4.44), "AES Recommended Practice for Digital Audio Engineering, Synchronization of Digital Audio Equipment in Studio Operations"

Recommendation ITU-R BT.601, "Encoding Parameters of Digital Television for Studios"

Recommendation ITU-R BT.656, "Interfaces for Digital Component Video Signals in 525-line and 625-line Television Systems Operating at the 4:2:2 level of Recommendation 601"

Recommendation ITU-R BT.711, "Synchronizing Reference Signals for the Component Digital Studio"

Report ITU-R BT.624, "Characteristics of Television Systems"

Report ITU-R BT.1212, "Measurements and Test Signals for Digitally Encoded Color Television Signals"

EBU Tech 3246, "EBU Parallel Interface for 625-line Digital Video Signals"

EBU N-10, "Parallel Component Video Interface for Non-composite ENG Signals"

SMPTE 125M, "Bit-Parallel Digital Interface, Component Video Signal 4:2:2"

SMPTE 170M, "Composite Analog Video Signal, NTSC for Studio Applications"

SMPTE 244M, "Digital Representation of NTSC Encoded Video Signal"

SMPTE 259M, "Serial Digital Interface for 10-bit 4:2:2 Component and 4*ƒ*sc NTSC Composite Digital Signals"

SMPTE 267M, "Bit Parallel Digital Interface, Component Video Signal 4:2:2, 16x9 Aspect Ratio"

SMPTE 269M, "Fault Reporting in Television Systems"

SMPTE RP 154, "Reference Signals for the Synchronization of 525-line Video Equipment"

SMPTE RP 168, "Definition of Vertical Interval Switching Point for Synchronous Video Switching"

SMPTE RP 165, "Error Detection Checkwords and Status Flags for Use in Bit-serial Digital Television Interfaces"

SMPTE RP 178, "Serial Digital Interface Check Field for 10-bit 4:2:2 Component and 4*ƒ*sc Composite Digital Signals"

Note: *Former CCIR recommendations and reports are now ITU-R documents. The International Telecommunication Union, Radio Communication Sector has replaced the CCIR.*

11. Glossary

125M – See SMPTE 125M.

4:2:2 – A commonly-used term for a component digital video format. The details of the format are specified in the CCIR-601 standard document. The numerals 4:2:2 denote the ratio of the sampling frequencies of the single luminance channel to the two color-difference channels. For every four luminance samples, there are two samples of each color difference channel. See CCIR-601.

4*f*sc – Four times subcarrier sampling rate used in composite digital systems. In NTSC, this is 14.3 MHz. In PAL, this is 17.7 MHz.

AES/EBU – Informal name for a digital audio standard established jointly by the Audio Engineering Society and European Broadcasting Union organizations.

algorithm – A set of rules or processes for solving a problem in a finite number of steps.

aliasing – Defects in the picture typically caused by insufficient sampling or poor filtering of digital video. Defects are typically seen as jaggies on diagonal lines and twinkling or brightening in picture detail.

analog – An adjective describing any signal that varies continuously as opposed to a digital signal that contains discrete levels representing the binary digits 0 and 1.

asynchronous – A transmission procedure that is not synchronized by a clock.

A-to-D Converter (analog-to-digital) – A circuit that uses digital sampling to convert an analog signal into a digital representation of that signal.

bandwidth – 1. The difference between the upper and lower limits of a frequency, often measured in megahertz (MHz). 2. The complete range of frequencies over which a circuit or electronic system can function with less than a 3 dB signal loss. 3. The information carrying capability of a particular television channel.

baseline shift – A form of low-frequency distortion resulting in a shift in the DC level of the signal.

bit – A binary representation of 1 or 0. One of the quantized levels of a pixel.

bit parallel – Byte-wise transmission of digital video down a multi-conductor cable where each pair of wires carries a single bit. This standard is covered under SMPTE 125M, EBU 3267-E and CCIR 656.

bit serial – Bit-wise transmission of digital video down a single conductor such as coaxial cable. May also be sent through fiber optics. This standard is covered under CCIR 656.

bit slippage – 1. Occurs when word framing is lost in a serial signal so the relative value of a bit is incorrect. This is generally reset at the next serial signal, TRS-ID for composite and EAV/SAV for component. 2. The erroneous reading of a serial bit stream when the recovered clock phase drifts enough to miss a bit. 3. A phenomenon which occurs in parallel digital data buses when one or more bits gets out of time in relation to the rest. The result is erroneous data. Differing cable lengths is the most common cause.

bit stream – A continuous series of bits transmitted on a line.

blocking – Occurs in a multi-stage routing system when a destination requests a source and finds that source unavailable. In a tie line system, this means that a destination requests a tie line and receives a "tie line busy" message, indicating that all tie lines are in use.

BNC – Abbreviation of "baby N connector." A cable connector used extensively in television.

byte – A complete set of quantized levels containing all of the bits. Bytes consisting of 8 to 10 bits per sample are typical.

cable equalization – The process of altering the frequency response of a video amplifier to compensate for high-frequency losses in coaxial cable.

CCIR – International Radio Consultative Committee, an international standards committee, now replaced by ITU-R.

CCIR-601 – (See ITU-R BT.601.)

CCIR-656 – The physical parallel and serial interconnect scheme for CCIR-601. CCIR 656 defines the parallel connector pinouts as well as the blanking, sync, and multiplexing schemes used in both parallel and serial interfaces. Reflects definitions in EBU Tech 3267 (for 625-line signals) and in SMPTE 125M (parallel 525) and SMPTE 259M (serial 525).

channel coding – Describes the way in which the 1s and 0s of the data stream are represented on the transmission path.

character generator (CG) – A computer used to generate text and sometimes graphics for video titles or captions.

clock jitter – Timing uncertainty of the data cell edges in a digital signal.

clock recovery – The reconstruction of timing information from digital data.

coaxial cable – A transmission line with a concentric pair of signal carrying conductors. There's an inner conductor and an outer conductive metallic sheath. The sheath aids in preventing external radiation from affecting the signal on the inner conductor and minimizes signal radiation from the transmission line.

coding – Representing each level of a video signal as a number, usually in binary form.

coefficients – A number (often a constant) that expresses some property of a physical system in a quantitative way.

component analog – The unencoded output of a camera, videotape recorder, etc., consisting of three primary color signals: green, blue, and red (GBR) that together convey all necessary picture information. In some component video formats, these three components have been translated into a luminance signal and two color-difference signals, for example, Y, B-Y, R-Y.

component digital – A digital representation of a component analog signal set, most often Y, B-Y, R-Y. The encoding parameters are specified by CCIR 601. The parallel interface is specified by CCIR 656 and SMPTE 125M (1991).

composite analog – An encoded video signal, such as NTSC or PAL video, that includes horizontal and vertical synchronizing information.

compression artifacts – Compacting of a digital signal, particularly when a high compression ratio is used, may result in small errors in the decompressed signal. These errors are known as "artifacts," or unwanted defects. The artifacts may resemble noise (or edge "busyness") or may cause parts of the picture, particularly fast moving portions, to be displayed with the movement distorted or missing.

composite digital – A digitally encoded video signal, such as NTSC or PAL video, that includes horizontal and vertical synchronizing information.

contouring – Video picture defect due to quantizing at too coarse a level.

D1 – A component digital video recording format that uses data conforming to the CCIR-601 standard. Records on 19mm magnetic tape. (Often used incorrectly to indicate component digital video.)

D2 – A composite digital video recording format that uses data conforming to SMPTE 244M. Records on 19mm magnetic tape. (Often used incorrectly to indicate composite digital video.)

D3 – A composite digital video recording format that uses data conforming to SMPTE 244M. Records on 1/2" magnetic tape.

delay – The time required for a signal to pass through a device or conductor.

demultiplexer (demux) – A device used to separate two or more signals that were previously combined by a compatible multiplexer and transmitted over a single channel.

deserializer – A device that converts serial digital information to parallel.

differential gain – A change in chrominance amplitude of a video signal caused by a change in luminance level of the signal.

differential phase – A change in chrominance phase of a video signal caused by a change in luminance level of the signal.

digital word – The number of bits treated as a single entity by the system.

discrete – Having an individual identity. An individual circuit component.

dither – Typically a random, low-level signal (oscillation) which may be added to an analog signal prior to sampling. Often consists of white noise of one quantizing level peak-to-peak amplitude.

dither component encoding – A slight expansion of the analog signal levels so that the signal comes in contact with more quantizing levels. The results are smoother transitions. This is done by adding white noise (which is at the amplitude of one quantizing level) to the analog signal prior to sampling.

drift – Gradual shift or change in the output over a period of time due to change or aging of circuit components. Change is often caused by thermal instability of components.

D-to-A converter (digital-to-analog) – A device that converts digital signals to analog signals.

DVTR – Abbreviation of digital videotape recorder.

EAV – End of active video in component digital systems.

EBU – European Broadcasting Union. An organization of European broadcasters that, among other activities, produces technical statements and recommendations for the 625/50 line television system.

EBU TECH.3267-E – The EBU recommendation for the parallel interface of 625-line digital video signal. A revision of the earlier EBU Tech.3246-E, which in turn was derived from CCIR-601 and contributed to CCIR-656 standards.

EDH (error detection and handling) – Proposed SMPTE RP 165 for recognizing inaccuracies in the serial digital signal. It may be incorporated into serial digital equipment and employ a simple LED error indicator.

Equalization (EQ) – Process of altering the frequency response of a video amplifier to compensate for high-frequency losses in coaxial cable.

embedded audio – Digital audio is multiplexed onto a serial digital data stream.

encoder – In video, a device that forms a single, composite color signal from a set of component signals.

error concealment – A technique used when error correction fails (see error correction). Erroneous data is replaced by data synthesized from surrounding pixels.

error correction – A scheme that adds overhead to the data to permit a certain level of errors to be detected and corrected.

eye pattern – A waveform used to evaluate channel performance.

field-time (linear) distortion – An unwarranted change in video signal amplitude that occurs in a time frame of 16 ms.

format conversion – The process of both encoding/decoding and resampling of digital rates.

frequency modulation – Modulation of a sinewave or "carrier" by varying its frequency in accordance with amplitude variations of the modulating signal.

frequency response rolloff – A distortion in a transmission system where the higher frequency components are not conveyed at their original full amplitude and possible loss of color saturation.

gain – Any increase or decrease in strength of an electrical signal. Gain is measured in terms of decibels or number of times of magnification.

group delay – A signal defect caused by different frequencies having differing propagation delays (delay at 1 MHz is different from delay at 5 MHz).

horizontal interval (horizontal blanking interval) – The time period between lines of active video.

interpolation – In digital video, the creation of new pixels in the image by some method of mathematically manipulating the values of neighboring pixels.

I/O – Abbreviation of input/output. Typically refers to sending information or data signals to and from devices.

ITU-R – The International Telecommunication Union, Radio Communication Sector (replaces the CCIR).

ITU-R BT.601 – An international standard for component digital television from which was derived SMPTE 125M (was RP-125) and EBU 3246E standards. CCIR defines the sampling systems, matrix values, and filter characteristics for both Y, B-Y, R-Y and GBR component digital television.

jaggies – Slang for the stair-step aliasing that appears on diagonal lines. Caused by insufficient filtering, violation of the Nyquist Theory, and/or poor interpolation.

jitter – An undesirable random signal variation with respect to time.

MAC – Multiplexed Analog Component video. This is a means of time multiplexing component analog video down a single transmission channel such as coax, fiber, or a satellite channel. Usually involves digital processes to achieve the time compression.

microsecond (μ) – One millionth of a second: 1×10^{-6} or 0.000001 second.

Miller squared coding – A DC-free channel coding scheme used in D-2 VTRs.

MPEG-2 – Motion pictures expert group. An international group of industry experts set up to standardize compressed moving pictures and audio.

multi-layer effects – A generic term for a mix/effects system that allows multiple video images to be combined into a composite image.

multiplexer (mux) – Device for combining two or more electrical signals into a single, composite signal.

nanosecond (ns) – One billionth of a second: 1×10^{-9} or 0.000000001 second.

NICAM (near instantaneous companded audio multiplex) – A digital audio coding system originally developed by the BBC for point-to-point links. A later development, NICAM 728 is used in several European countries to provide stereo digital audio to home television receivers.

nonlinear encoding – Relatively more levels of quantization are assigned to small amplitude signals, relatively fewer to the large signal peaks.

nonlinearity – Having gain vary as a function of signal amplitude.

NRZ – Non return to zero. A coding scheme that is polarity sensitive. 0 = logic low; 1 = logic high.

NRZI – Non return to zero inverse. A video data scrambling scheme that is polarity insensitive. 0 = no change in logic; 1 = a transition from one logic level to the other.

NTSC (National Television Systems Committee) – Organization that formulated standards for the NTSC television system. Now describes the American system of color telecasting which is used mainly in North America, Japan, and parts of South America.

Nyquist sampling theorem – Intervals between successive samples must be equal to or less than one-half the period of highest frequency.

orthogonal sampling – Sampling of a line of repetitive video signal in such a way that samples in each line are in the same horizontal position.

PAL (Phase Alternate Line) – The name of the color television system in which the V component of burst is inverted in phase from one line to the next in order to minimize hue errors that may occur in color transmission.

parallel cable – A multi-conductor cable carrying simultaneous transmission of data bits. Analogous to the rows of a marching band passing a review point.

patch panel – A manual method of routing signals using a panel of receptacles for sources and destinations and wire jumpers to interconnect them.

peak to peak – The amplitude (voltage) difference between the most positive and the most negative excursions (peaks) of an electrical signal.

phase distortion – A picture defect caused by unequal delay (phase shifting) of different frequency components within the signal as they pass through different impedance elements – filters, amplifiers, ionospheric variations, etc. The defect in the picture is "fringing," like diffraction rings, at edges where the contrast changes abruptly.

phase error – A picture defect caused by the incorrect relative timing of a signal in relation to another signal.

phase shift – The movement in relative timing of a signal in relation to another signal.

pixel – The smallest distinguishable and resolvable area in a video image. A single point on the screen. In digital video, a single sample of the picture. Derived from the words picture element.

PRBS – Pseudo random binary sequence.

production switcher (vision mixer) – A device that allows transitions between different video pictures. Also allows keying and matting (compositing).

propagation delay (path length) – The time it takes for a signal to travel through a circuit, piece of equipment, or a length of cable.

quantization – The process of converting a continuous analog input into a set of discrete output levels.

quantizing noise – The noise (deviation of a signal from its original or correct value) which results from the quantization process. In serial digital, a granular type of noise only present in the presence of a signal.

rate conversion – 1. Technically, the process of converting from one sample rate to another. The digital sample rate for the component format is 13.5 MHz; for the composite format it's either 14.3 MHz for NTSC or 17.7 MHz for PAL. 2. Often used incorrectly to indicate both resampling of digital rates and encoding/decoding.

Rec. 601 – See CCIR-601.

reclocking – The process of clocking the data with a regenerated clock.

resolution – The number of bits (four, eight, ten, etc.) determines the resolution of the digital signal.

4-bits = a resolution of 1 in 16

8-bits = a resolution of 1 in 256

10-bits = a resolution of 1 in 1024

Eight bits is the minimum acceptable for broadcast TV.

RP 125 – See SMPTE 125M.

routing switcher – An electronic device that routes a user-supplied signal (audio, video, etc.) from any input to any user-selected output(s).

sampling – Process where analog signals are measured, often millions of times per second for video.

sampling frequency – The number of discrete sample measurements made in a given period of time. Often expressed in megahertz for video.

SAV – Start of active video in component digital systems

scope – Short for oscilloscope (waveform monitor) or vectorscope, devices used to measure the television signal.

scrambling – 1. To transpose or invert digital data according to a prearranged scheme in order to break up the low-frequency patterns associated with serial digital signals. 2. The digital signal is shuffled to produce a better spectral distribution.

serial digital – Digital information that is transmitted in serial form. Often used informally to refer to serial digital television signals.

serializer – A device that converts parallel digital information to serial digital.

SMPTE (Society of Motion Picture and Television Engineers) – A professional organization that recommends standards for the television and film industries.

SMPTE 125M (was RP 125) – The SMPTE recommended practice for bit-parallel digital interface for component video signals. SMPTE 125M defines the parameters required to generate and distribute component video signals on a parallel interface.

SMPTE 244M – The SMPTE recommended practice for bit-parallel digital interface for composite video signals. SMPTE 244M defines the parameters required to generate and distribute composite video signals on a parallel interface.

SMPTE 259M – The SMPTE recommended practice for 525-line serial digital component and composite interfaces.

still store – Device for storage of specific frames of video.

synchronous – A transmission procedure by which the bit and character stream are slaved to accurately synchronized clocks, both at the receiving and sending end.

sync word – A synchronizing bit pattern, differentiated from the normal data bit patterns, used to identify reference points in the television signal; also to facilitate word framing in a serial receiver.

telecine – A device for capturing movie film as a video signal.

temporal aliasing – A visual defect that occurs when the image being sampled moves too fast for the sampling rate. A common example is wagon wheels that appear to rotate backwards.

time base corrector – Device used to correct for time base errors and stabilize the timing of the video output from a tape machine.

TDM (time division multiplex) – The management of multiple signals on one channel by alternately sending portions of each signal and assigning each portion to particular blocks of time.

time-multiplex – In the case of CCIR-601, a technique for transmitting three signals at the same time on a group of parallel wires (parallel cable).

TRS – Timing reference signals in composite digital systems (four words long).

TRS-ID (timing reference signal identification) – A reference signal used to maintain timing in composite digital systems. It's four words long.

truncation – Deletion of lower significant bits on a digital system. Usually results in digital noise.

VTR (video tape recorder) – A device which permits audio and video signals to be recorded on magnetic tape.

waveform – The shape of an electromagnetic wave. A graphical representation of the relationship between voltage or current and time.

word – See byte.

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