Designing Video Circuits

Part Three

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Part One of ESD's *three-part article on video circuit design discussed underlying theory, video preprocessing, filter design, and video A/D conversion. Part Two, published last month, dealt with sampling standards, video synchronization, and genlock. This month,* ESD *concludes this video design trilogy with a look at digital-to-analog conversion, time-base correctors, and frame synchronizers.*

Digital techniques are well suited for video processing. However the world, as we all know, is analog. Most digital video systems contain an output processor, a subsystem that converts the digital data to an analog format and reinserts burst and synchronization information.

Figure 1 shows an output processor for a line-locked component system. Data rate of the luminance channel (Y) is 13.5 MHz. Color information has a lower bandwidth, and each of the baseband color-difference signals (R-Y, B-Y) is output from the digital system at 6.75 MHz.

After D/A conversion, each signal is low-pass-filtered to remove artifacts from the A/D-D/A conversion process. Color difference signals are quadrature amplitude modulated by the subcarrier (3.58 MHz) and combined to form the chrominance signal. This signal is added to the luminance signal along with timing information. Final output is composite (NTSC) video.

Zero-Order Hold Distortion

A/D and D/A conversion has a distortion associated with it that acts as a low-pass filter. So-called sin[x]/x distortion becomes important for sampled systems in which the sampling frequency is not much greater than the signal being processed.

DACs can be modeled with the equivalent block diagram of **Figure 2a.** Digital data for the DAC arrives at the sampling frequency. The DAC converts the signal to one that is represented as a scaled impulse. The zero-order hold (with impulse response shown in **Figure 2b**) squares up the impulse, producing an analog voltage.

Given the impulse response, the Fourier integral can be evaluated to derive the frequency response of the zero-order hold. For simplicity, the impulse response was assumed to be noncausal. This simply adds a phase shift equal to $-\omega T$ where T is the sampling period.

By taking the Fourier integral and converting from complex to sinusoidal notation, the magnitude function is:

$$F(\omega) = \frac{\sin\left[\frac{\omega T}{2}\right]}{\frac{\omega T}{2}} \qquad \text{Equation 1}$$

In terms of the sampling frequency [f_s]:

$$f(f) = \frac{\sin\left[\frac{\pi f}{f_s}\right]}{\frac{\pi f}{f_s}} \qquad \text{Equation 2}$$

In this example, the sampling frequency for the chrominance components R-Y and B-Y is 6.75 MHz. Bandwidth of the R-Y and

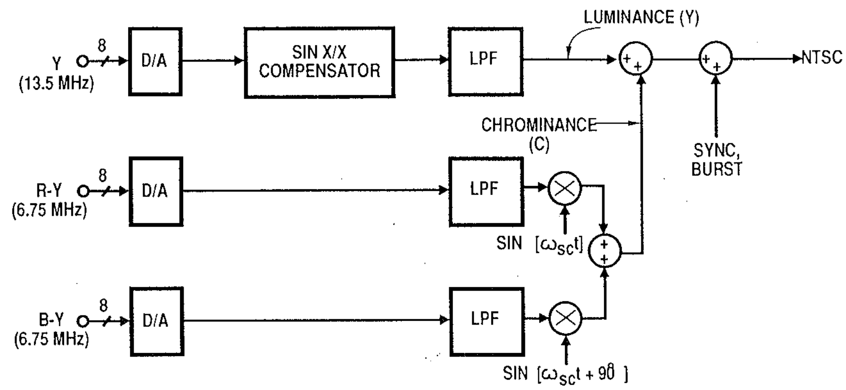


Figure 1: Output processor for a line-locked Y, R-Y, B-Y component system. After D/A conversion, each signal is filtered to remove conversion artifacts. Color-difference signals are then quadrature amplitude modulated by the 3.58-MHz subcarrier and combined to form a chrominance signal, which is then added to the luminance information and the sync is reinserted to obtain the NTSC output.

B-Y components is around 1 MHz. For these values, the zero-order hold frequency response evaluated at 1 MHz is approximately 0.96. This factor can normally be ignored and no compensation is needed.

Distortion becomes important when the analog signal is not greatly oversampled. For the luminance (Y) information, the nominal bandwidth is 4 MHz or more and the sampling frequency, f_s, is 13.5 MHz. Evaluated at 4 MHz, the sin[x]/x response is approximately 0.85, or −1.4 dB. High-frequency luminance will be attenuated, with a loss of picture detail.

Figure 3a shows a possible first-order solution to the problem. If $R_L = R_I$, the circuit's low-frequency gain will be approximately 1. The inclusion of a capacitor across R_I attenuates the impedance at the collector of the transistor. This adds a zero to the amplifier's transfer function, and the 3-dB point can be adjusted to offset the attenuation of the zero-order hold.

Alternatively, the signal can be processed in digital form. By pre-emphasizing the higher frequencies of the Y signal, the overall transfer function of the cascaded system can be made to approach the ideal. The simple finite impulse response (FIR) filter in **Figure 3b** has the transfer function:

$$H(z) = -\tfrac{1}{16} + \tfrac{9}{8}z^{-1} - \tfrac{1}{16}z^{-2} \qquad \text{Equation 3}$$

where z^{-1} corresponds to one digital sample delay of 74 nsec (1/13.5 MHz). By making the substitution $z = e^{-j\omega T}$, conversion is made from discrete time to continuous time. T = 1/sampling frequency.

$$H(j\omega) = e^{-j\omega T}\left[-\tfrac{1}{16}e^{-j\omega T} + \tfrac{9}{8} - \tfrac{1}{16}e^{j\omega T}\right] \qquad \text{Equation 4}$$

A useful identity is $\cos[\omega T] = \dfrac{[e^{-j\omega T} + e^{j\omega T}]}{2}$

$$H(j\omega) = e^{-j\omega T}\left[\tfrac{9}{8} - \tfrac{1}{8}\cos\omega T\right] \qquad \text{Equation 5}$$

The first term, $e^{-j\omega T}$, has a magnitude of unity, and simply adds a linear phase shift. By cascading the magnitude function, [9/8— 1/8 cosωT], with the zero-order hold, the magnitude response of the A/D and D/A holds to within 0.2 dB of unity.

Decoding

Decoders break up composite video into three baseband components. In general, the three components can be R,G, and B, but other standards are possible. Decoding should be as transparent as possible, with minimal losses in video resolution and no introduction of luminance/chrominance crosstalk. After decoding, further processing can reap the advantages of component processing. Before transmission, the components are re-encoded into a composite format.

Originally, television standards were monochrome. When color television was invented, compatibility with existing monochrome standards needed to be retained. By apportioning the NTSC spectrum as shown in **Figure 4a,** color and monochrome information could be sent in the same bandwidth as the original standard.

Luminance is modulated by horizontal sync. Bunches of black-and-white information are spaced at intervals of horizontal frequency, f_H. These signals are further modulated by vertical sync. In the frequency domain, luminance components occur at:

$$Nf_H \pm Mf_V$$

With a luminance bandwidth of 4.2 MHz, the maximum value of N is 277. Luminance information is band-limited to areas around integral harmonics of line frequency, with additional spectral lines offset from Nf_H by the 30-Hz vertical rate. Luminance spectral energy at odd multiples of one-half line frequency is minimal. This empty area in the spectrum is used for transmission of chrominance information.

Distortion becomes important when the analog signal is not greatly oversampled.

By the choice of subcarrier frequency:

$$f_{sc} = \frac{455\, f_H}{2} \qquad \text{Equation 6}$$

the chrominance spectral energy is placed between luminance bunches. This process, called frequency interleaving, allows the spectrum to be shared between two signals with minimal crosstalk.

In the decoding process, luminance and chrominance must be separated from the composite video waveform. After chrominance is separated, baseband color-difference components are produced by multiplying the chroma information with two properly phased subcarrier waveforms. The result is low-pass filtered to eliminate spectral components at twice the subcarrier frequency.

The process of luminance and chrominance separation is the most difficult part of decoding. A filter having amplitude response like the teeth of a comb—i.e., 100% transmission for desired frequencies, and zero transmission for interleaved frequencies—can, in principle, separate luminance and chrominance. Such a system is shown in **Figure 4b.**

Comb filters rely on the 180° shift of chroma phase over one horizontal line. By adding properly scaled and delayed video information, chroma can be emphasized or suppressed. For the system in Figure 4b,

$$H(\omega) = A\,(1+e^{-2j\omega T}) + Be^{-j\omega T} = e^{-2j\omega T}\,(B+2A\cos\omega T)$$

Equation 7

where T is a delay of one line period (63.4 μsec).

If A is set to $-\frac{1}{2}$ B, the transfer function shown in **Figure 4c** results. There are nulls at multiples of horizontal frequency; if the input to this system were composite video, the luminance components would be filtered out, leaving only color information.

For A = ½ B, the situation is reversed and the transfer function is shifted in frequency by $f_H/2$. Thus, the filter will pass luminance and attenuate chrominance.

An inherent disadvantage of comb filters is that they depend on the color being constant from line to line. Errors will occur when there are differences in chrominance and high-frequency luminance on successive lines.

Time-Base Correction

Time-base correctors (TBCs) correct timing variations of a video signal from a video tape recorder (VTR). VTRs are electromechanical devices, and the tolerances of their servomechanisms are not sufficient for broadcast-quality television.

The amount of timing error that can be tolerated in video processing is on the order of nanoseconds. For a broadcast VTR, one television field is recorded on approximately 6.5″ of tape. This translates to a mechanical stability of 40 millionths of an inch per 100 nsec. VTR servomechanisms cannot maintain this accuracy.

Worst-case errors are much worse. The velocity servomechanism can be expected to have an error of 1H or more, which translates to a timing error of $\pm$ 63.4 μsec. Tape stretching can also cause errors on the order of 1H or more.

Digital time-base correctors (**Figure 5**) are characterized by sampling an input video source at a frequency f_1. This sampling clock is locked to the off-tape signal, and thus has all the unstable characteristics of the VTR timebase. The jittery input video is written into digital memory at the f_1 rate.

Another clock (f_2) is derived from a stable (usually crystal-controlled) local reference. This clock controls data recovery from digital memory. Data is read from the memory at f_2 and is processed by the output processor. This signal is D/A-converted and a stable sync and color burst are added.

If the average values of the frequencies f_1 and f_2 are equal, the memory will provide a rubberbanding function, and the jittery timing variations of the input signal will not appear at the output. The size of the digital memory determines the maximum instantaneous time-base error the TBC will correct. For many postproduction editing systems, a TBC window of 16 H-lines is sufficient.

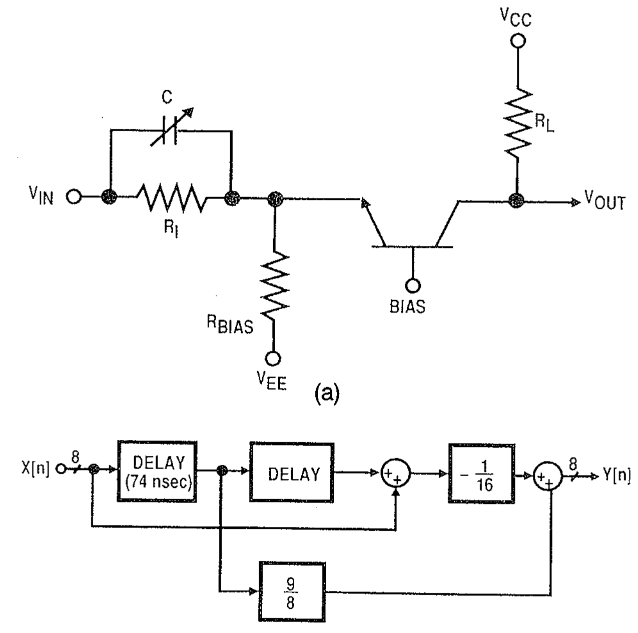


Figure 3: Distortion becomes important when the analog signal is not greatly oversampled. (a) To compensate for this, analog implementations of sin [x]/x filters can be used. (b) Alternatively, the signal can be processed in digital form. Here, the higher frequencies of the luminance signal (Y) can be pre-emphasized, so that the system's overall transfer function better approaches the ideal.

Frame Synchronization

A frame synchronizer can be thought of as a TBC with an infinitely long correction window. TBCs store a number of video lines to eliminate elastic variations of video signals. Frame synchronizers store an entire frame. The block diagram is the same as that of the TBC, except for the fact that the input source is usually stable.

Again referring to Figure 5, consider the two clocks f_1 and f_2. Imagine that f_2 is the local studio reference and f_1 is a clock derived from a remote satellite feed. Even if both clocks are stable, there will be residual timing variation between the two. For instance, the frequency of NTSC subcarrier is specified to be 3,579,545 Hz $\pm$10 Hz. If line-locked sampling at $3f_{sc}$ is used, the differential timing errors between input and output may be as great as 60 Hz. The input picture (clocked at f_1) will move slowly in relation to the reference video. Thus, the two video sources could not be mixed directly.

Frequency deviations of the incoming video will be continuous in one direction. Input to output delay between the two video sources at the output of the frame synchronizer will vary from zero to one full TV frame. At the point where the delay is one full frame, a frame will be

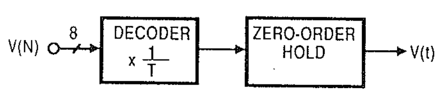
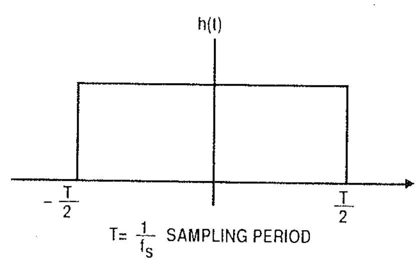


Figure 2: Modeling DACs. (a) Equivalent circuit of a DAC shows the digital input arriving at the sampling frequency. This is then converted to a signal with a scaled impulse. A zero-order hold with impulse response of (b) squares up the impulse, producing an analog voltage.



deleted. This reduces the system delay back to zero, and the timing error begins to accumulate immediately.

If f_1 to f_2 differential timing is in the other direction, a frame will be repeated every once in a while, instead of deleted. In any case, the discontinuity will be only a frame every hour or so, and the viewer will never notice. **ESD:**

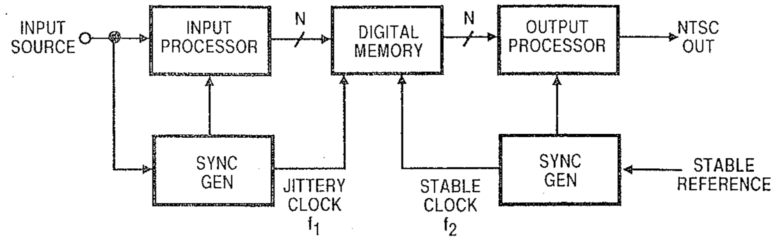


Figure 5: Digital time-base correctors (TBCs) are used to correct timing variations of video signals from video tape recorders. They can be characterized by two clocks. The first clock is locked to the "jittery" input signal. Data is clocked out of memory and is D/A converted at the stable frequency f_2. The clock f_2 is usually crystal controlled.

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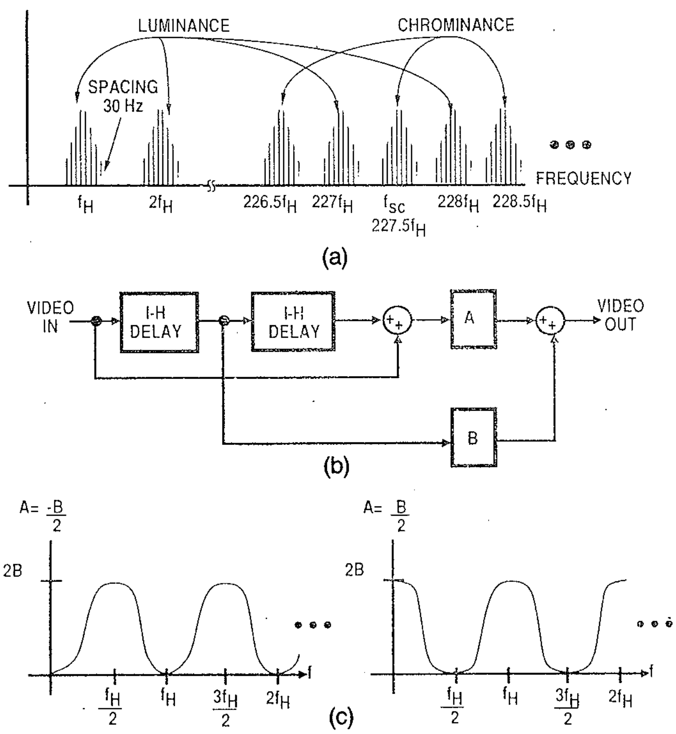

Figure 4: Color TV standards, such as NTSC, were originally monochrome. (a) By apportioning the NTSC video spectrum as shown, both black-and-white and color information can be transmitted with no bandwidth increase. (b) The process of luminance and chrominance separation is the most difficult part of decoding. For the separation, comb filters having amplitude responses like the teeth of a comb are used. These rely on the 180° phase shift of the chroma phase over one horizontal line. By adding scaled and delayed video information, chroma can be emphasized or suppressed. (c) Comb-filter responses for luminance and chrominance filters.