Active Filters for Video

Modern op amps and specialized design software for the PC make it possible to design high-bandwidth active filters. Unfortunately, this doesn't take into consideration the problems associated with video. For that, you'll need to understand the applications and formats used, as well as the nuances of active filter design. In an effort to do that we'll look at today's major active-filter applications in broadcast and graphic (PC) video.

Originally, video filters were passive L-C circuits surrounded by amplifiers, but the increased gain-bandwidth product (GBW) of modern op-amps makes it possible to combine them with R-C circuits to achieve smaller, more accurate designs. Active filters developed a bad reputation because of problems getting repeatable results until sensitivity analysis methods provided solutions for these problems in the 1960s.

Today, op-amps like the MAX4450/4451 and specialized design software for the PC, make it possible to design high-bandwidth active filters. Unfortunately, this doesn't take into consideration the problems associated with video. For that, you'll need to understand the applications and formats used, as well as the nuances of active filter design. In an effort to do that; we'll look at today's major active-filter applications in broadcast and graphic (PC) video, which are:

- Anti-Aliasing Filters. These are placed before an analog-to-digital converter (ADC), to attenuate signals above the sample rate (Nyquist Frequency) of the ADC.
- Reconstruction Filters. These are placed after a digital-to-analog-converter (DAC), to filter the video and to amplitude-correct for the DAC response. They are also called sin (X)/(X), or zero-order-hold corrector depending on which author you read.

Normally, these filters are designed for the steepest possible response to reject everything above the cutoff frequency[1] . To avoid distorting the complex video waveform, the filter must be phase linear, where the linearity is specified as group-delay variation; and the amount allowed will depend on the video format. The principal video formats in use today are:

- Primaries or Native Video. These are the gamma-corrected primaries (R', G', B'), or linear R, G, B, depending on whether the video is broadcast or graphics. All these signals have the same bandwidth, and the group-delay variation must be sub-pixel to suppress visual artifacts.
- Component Video. There are two component-video formats; color difference (Y, Pb, and

Pr) found in broadcast and luma/chroma (Y-C), or S-VHS, found in VCRs. Both have a luma (Y) bandwidth that's greater than the color (Pb/Pr or C) bandwidth making it difficult to keep the signals time-coincident.

- Composite Video. Composite video (Cvbs) is only found in SDTV, not HDTV. It's a single channel format, and is the most forgiving of group-delay variation; in fact, group delay is seldom specified except for the RF Modulator path.

Different filter characteristics are compared for selectivity and group delay, including exotics, such as the Legendre, Equal-ripple, and Transitional types for use as video filters. The Sallen-Key and Rauch realizations of these filters are evaluated for op-amp GBW-to-cutoff ratio (GBW/ω_C), and element sensitivity. In the process, pre-distortion and the sensitivity function are introduced to accurately design a filter and predict its response. The op-amp's requirements for slew rate, offset voltage, settling time, transient response, differential gain and phase, and the spurious free dynamic range are also determined. Then, using the MAX4450/51, several active video-filter designs are shown. The first is for ITU-601 anti-aliasing, based on using an over-sampling, or FIR digital post filter. Another is a 20MHz design for XGA PC Graphics, and finally a prototype design for HDTV based on SMPTE274M & 296M.

Reconstruction filter requirements are complicated by the DAC "hold" function, which is also a filter of sorts. Although this helps by reducing the selectivity, it introduces loss as the reconstruction bandwidth approaches the Nyquist frequency. Over-sampling reduces the amount of compensation required in SDTV broadcast applications, but at higher sample rates, an amplitude equalizer is used, and a design is shown using the MAX4450, based on the delay equalizer, which can be used for HDTV and graphics.

Finally, we look at the practical side of video filters, tuning, oscillation problems, and the causes and some cures.

Filters and their Characteristics

Whether used for anti-aliasing or reconstruction, video filters have a low-pass characteristic to pass the video frame rate. Such filters are categorized by their amplitude-vs-frequency characteristic, or the name of the person whose polynomial describes it, e.g.; Bessel, Butterworth, Chebyshev, or Cauer. Figure 1 is a collection of these characteristics, normalized to a one-radian bandwidth (ω_C). Normally you would use the most selective one, with the minimum number of poles, but video requires selectivity and phase linearity, which limits the choice.

Phase Linearity and Group Delay

Because it's so important, we need a way to specify the property of phase linearity, and the best indicator is the group or envelope delay (GD), shown as the second set of curves in Figure 1. What's group delay? It's defined as the derivative of phase with respect to frequency, but practically, it's the time delay as a function of frequency. A flat group delay curve indicates that all frequencies experience the same delay, preserving the waveform's shape in the time domain, obviously a desirable trait for video. But how much variation is acceptable, and why?

Visually, group-delay variation first affects the high frequency details in an image. If the group-delay variation over the filter's bandwidth is less than a pixel, it's theoretically invisible. So, why do ITU-601 and EIA-770, have tighter requirements? It's done to insure "generational stability" of the video during editing (MPEG-2), and to control phase jitter prior to serialization. Amplitude variation is controlled for the same reason. Non-broadcast applications relax these specifications, but all the multi-signal video formats (RGB, YPbPr, & Y-C) require less than one-half pixel group-delay variation to preserve their shape. So, what do we look for in a filter characteristic to achieve this?

In Figure 1, you'll notice some filters have a peak in the Group-Delay curves near the cutoff frequency, (ω/ω_c=1). These are called "phase ears" because they're due to the steep phase change at the cutoff frequency. To get an idea of the scale of things, a 3 pole, 6MHz Butterworth filter has a group-delay variation of 20-25nsec, about 1/3rd of a pixel in broadcast video. Increasing the number of poles, or using a more-selective filter, will increase this. Delay equalizers can correct it, but require more op-amps, and increase the complexity. There are exotic types of filters, and we should mention them[2].

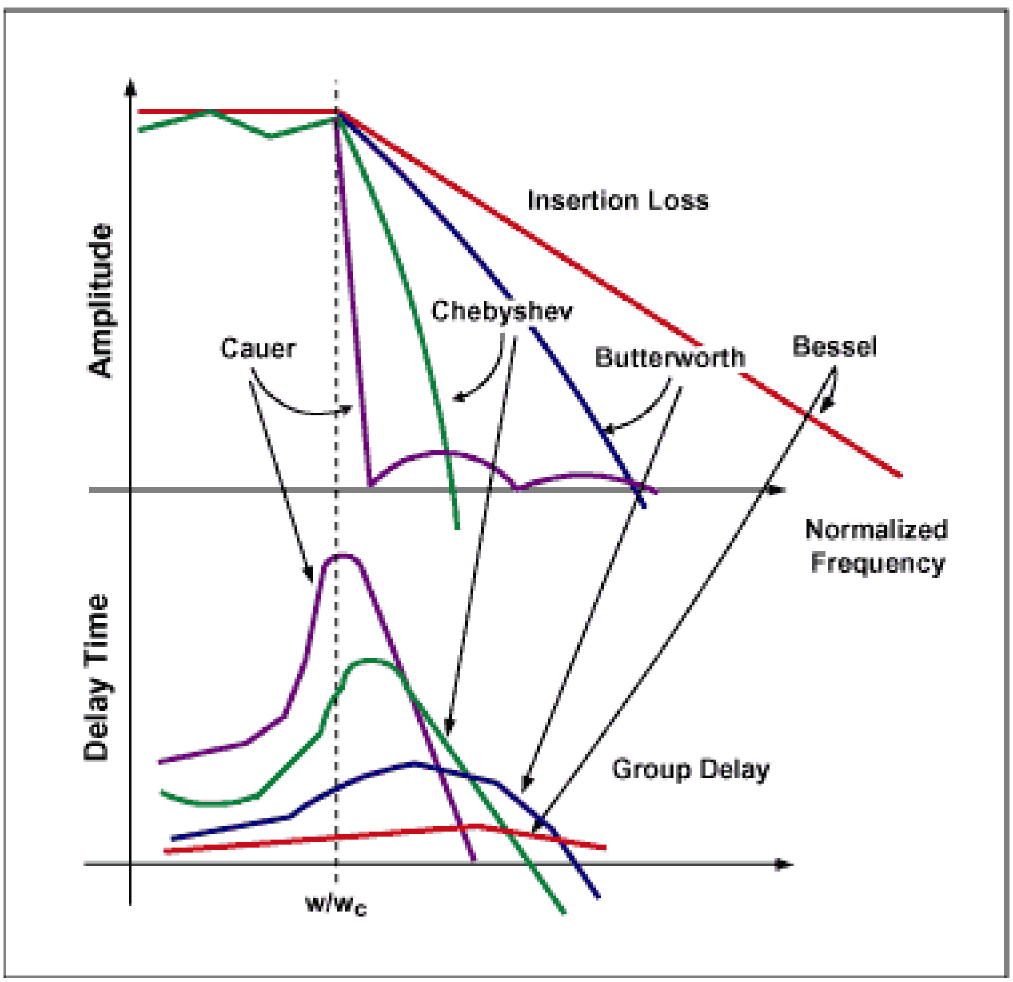

Figure 1. Amplitude and group-delay vs. frequency characteristic of the various filters.

The Transitional types are optimized for minimal group-delay variation in the pass-band. One type is flat until its amplitude response falls 6 or 12db, then transitions to a more selective response. In some literature, these are also known as Thompson, or Thompson-Butterworth filters. Another type, the Equal-Ripple, Phase Approximation, holds the phase change to 0.05 or 0.5 degrees, giving a ripple in the group delay, before transitioning. Both trade selectivity for linearity; their shortcoming is band-edge response. The Legendre filter trades pass-band loss for band-edge response. A 3-pole Legendre has better selectivity than a 4-pole Butterworth, and about the same group delay, but has about ½db loss at the high end of its response. Despite all this, the Butterworth characteristic is still the most commonly used filter in video.

Group Delay Problems with Component Video

The multi-signal formats, RGB and component video, are the most sensitive to group delay variation, both within the channel and between channels. The latter is called time coincidence to differentiate it from group delay, and it controls the registration of the image components when they're displayed. Only the RGB format has the same bandwidths for all channels. The component formats don't, and that causes problems controlling the time-coincidence, and group-delay variation. An example of this is the anti-aliasing of the Pb and Pr signals in ITU-601-5. The filters for Pb and Pr have half the bandwidth of the luma (Y) signal. Using that bandwidth, the group delay of Pb and Pr would be double that of the Y signal. To correct it, a delay equalizer would have to be used to slow down the Y signal[3]. Video encoders were originally built this way, but today the sample rate of Pb and Pr is doubled to equalize the bandwidths. This raises the 4:2:2 sampling to 4:4:4[4], and you treat it like RGB. It isn't as straight forward for the other component format, S-VHS.

The Special Case of S-VHS

S-VHS is a two-channel, (Y-C) videotape format. It has the same luma (Y) signal bandwidth as YPbPr, but with Pb and Pr combined into a single chroma channel (C) that looks like it should be band-pass, rather than low-pass filtered. To make the matter worse, the source is usually a VCR. Tape stretch causes the timing to vary, especially in a "pause" or fast-forward mode[5]. TV sets will adapt to this, but trying to digitize it produces a varying number of samples per line. Thankfully, this format is seldom digitized, so unless you're creating it in the encoding process, low-pass filter both channels (Y&C) with the same bandwidth filter. S-VHS is more forgiving of that this, than it is of the problems caused by trying to equalize the Group Delay. Besides, the Chroma signal has a bias voltage that's lost when it's AC coupled. You're more likely to see this format as an output from a VCR, where the primary concern is preserving the timing between Y and C. Using the same filter on both channels will do this nicely.

Choosing an Op-Amp

After a filter characteristic is chosen, the next step is to choose a circuit to implement it. In filter design, this is called the realization process; and for active filters, it starts with choosing the op-amp. The first parameter needed is the minimum gain-bandwidth (GBW), and it depends on whether an inverting (Rauch) or non-inverting (Sallen-Key) circuit is used.

To prove that, solve for the phase shift as a function of frequency, putting the gain in terms of the feedback ratio (Rf/Ri), and compare the phase of a non-Inverting amplifier:

$$Arg[K(j\omega)]_{non\text{-}inv} = -(\omega_c / GBW_{non\text{-}inv}) \times (1+Rf/Ri) \quad \text{Eq 1}$$

With an inverting amplifier:

$$Arg[K(j\omega)]_{inv} = \pi - (\omega_c / GBW_{inv}) \times (1+Rf/Ri) \quad \text{Eq 2}$$

Where Rf and Ri are the gain-set resistors in Ohms, GBW_{rad} is the op-amp gain-bandwidth in radians, and ω_c is the cutoff frequency of the filter in radians. The π term just indicates phase inversion, discard it, and set the equations equal to each other. Add values for Rf and Ri[6] for the gain, and solve for (ω_c/GBW_{rad}). Doing that, you'll find a unity gain Rauch circuit requires twice the GBW of a Sallen-Key, for the same phase shift.

To determine the requirements for a particular filter, just set ($Arg[K(j\omega)]$) to some excess phase like 0.1 radian (~5.7°), and solve for GBW_{rad}/ω_c. For example, based on 0.1 radians, a unity-gain Sallen-Key requires a ratio of 10, while a Rauch requires 20. Just remember, these filters handle large signals, so the large signal (2 Volt) GBW must be used. Based on this, we would choose the Sallen-Key circuit to realize all our video filters, except for something called "sensitivity".

Pre-Distortion, Bandwidth, Q, and Sensitivity

It's important to understand that anything less than an infinite GBW_{rad}/ω_c ratio will cause the closed loop poles of a filter to move, which is why you get a lower bandwidth, and more peaking (Q), in a filter than it was designed for[7]. To compensate for the finite GBW, the poles of the characteristic are moved to get the correct bandwidth, and Q. In filter design, this is called "pre-distortion". Using the pre-distorted bandwidth for ω_c, we can calculate component values. Once we have them, we need to know the tolerance to maintain the characteristic, and that requires something called a sensitivity function.

The active filter we will design are is a cascade of 1st- and 2nd-order stages. The 1st-order stage has a linear relationship between the parameters and the component values. For example, a ±1% resistor and a ±2% capacitor produce a ±3% total change in ω_c. In the S-K and Rauch circuits, ω_c is the square root of the product of the component values. To see how this affects the tolerance, we first describe the stage in terms of its Q and cutoff frequency, ω_c. Using this form, and the partial derivative, we get the sensitivity function[8], ($S_X{}^Y$), that shows the ratio between a change in part "X" and parameter "Y".

This helps in the design. For example, from Table 1, the Q sensitivity of a Sallen-Key to R1, R2 and C1, C2 is large, ±5:1 to ±10:1 depending on gain, but they're equal and opposite. Using parts with the same temperature coefficient, they will self-compensate each other over temperature. Unfortunately, the Q sensitivity to DC gain is also large, and requires better than 1% tolerance to avoid peaking in the response. The important point is that, using S_X^Y, you can predict the effect of component tolerance, and then design accordingly.

Table 1. Component Sensitivity values, with BW and Q Pre-Distortion formulas for the Sallen-Key Realization, ω_o =1 rad/sec.

Sensitivity S_x^y	Gain K=3-1/Q R1=R2=C1=C2=1	Gain K=1 R1=R2=1	Gain K=2 R1=C1=1
$S_x\omega$ (x=R1,R2,C1,C2)	-1/2	-1/2	-1/2
S_K^Q	14	50	10
S_{R1}^Q	4.5	0	4.5
S_{R2}^Q	-4.5	0	-4.5
S_{C1}^Q	9.5	1/2	5.5
S_{C2}^Q	-9.5	-1/2	-5.5
S_{Ra}^K	-9/14	N/A	-1/2
S_{Rb}^K	9/14	N/A	1/2
ω_c(Actual)	ω_c(Design)[1-1/2(3-1/Q)$^2\omega_c$/GBW]	ω_c(Design)[1-ω_cQ/GBW]	
Q(Actual)	Q(Design)[1+1/2(3-1/Q)$^2\omega_c$/GBW]	Q(Design)[1+ω_cQ/GBW]	

Table 2. Component Sensitivity values, with BW and Q Pre-Distortion formulas for the Rauch Realization, ω_o =1 rad/sec.

Sensitivity Function S_x^y	Gain K=1 R1=R2=R3=1	Gain K=2 R1=1, R3=H_o, R2=(H_o/1+H_o)	Gain K=2 C1=1, C2=C1/100
$S_x\omega$ (x=R2,R3,C1,C2, $S_{R1}\omega$=0)	-1/2	-1/2	-1/2
S_{R1}^Q	1/3	1/3	1/3
S_{R2}^Q	-1/6	0	0
S_{C1}^Q	1/2	1/2	1/2
S_{C2}^Q	-1/2	-1/2	-1/2

S_{R3}^{K}	1	1	1
S_{R1}^{K}	-1	-1	-1
S_{R3}^{Q}	1/6	0	0
ω_c(Actual)	ω_c(Design)[1-3ω_cQ/2GBW]		
Q(Actual)	ω_c(Design)[1+3ω_cQ/2GBW]		

Once the characteristic, realization, and GBW_{rad}/ω_c ratio are known, the other op-amp parameters can be used to narrow the choice. They should be:

- Slew rate; the op-amp should be able to slew from black to white in less than 10% of a pixel, and settle to 0.1%, with <2% overshoot in >>1 pixel.
- The offset voltage shouldn't exceed 1-2% of the peak-to-peak video.
- SFDR should be >50db, including crosstalk, distortion, and noise.

Based on that, we'll look at some applications using the MAX4450/51.

Anti-Aliasing Filter Design

The first application is anti-aliasing, where the selectivity is determined by the quantization, or number of bits (N) in the ADC we're filtering. Selectivity is determined by the insertion loss (IL) required at the Nyquist frequency, which is:

IL (db) = 20 x Log 2^N Eq 3

This defines a template, together with the bandwidth (ω_c), like the one for ITU-601-5 in Figure 2. The video is sampled at 13.5MHz, and the specified bandwidth is 5.75MHz ±0.1db, with an insertion loss of 12db at 6.75MHz (Nyquist) and 40db at 8MHz, and a group-delay variation of ±3nsec over the 0.1db BW. This is too difficult for any analog filter, but 4X over-sampling modifies these requirements to 12db at 27MHz and 40db at 32MHz for the same ω_c.

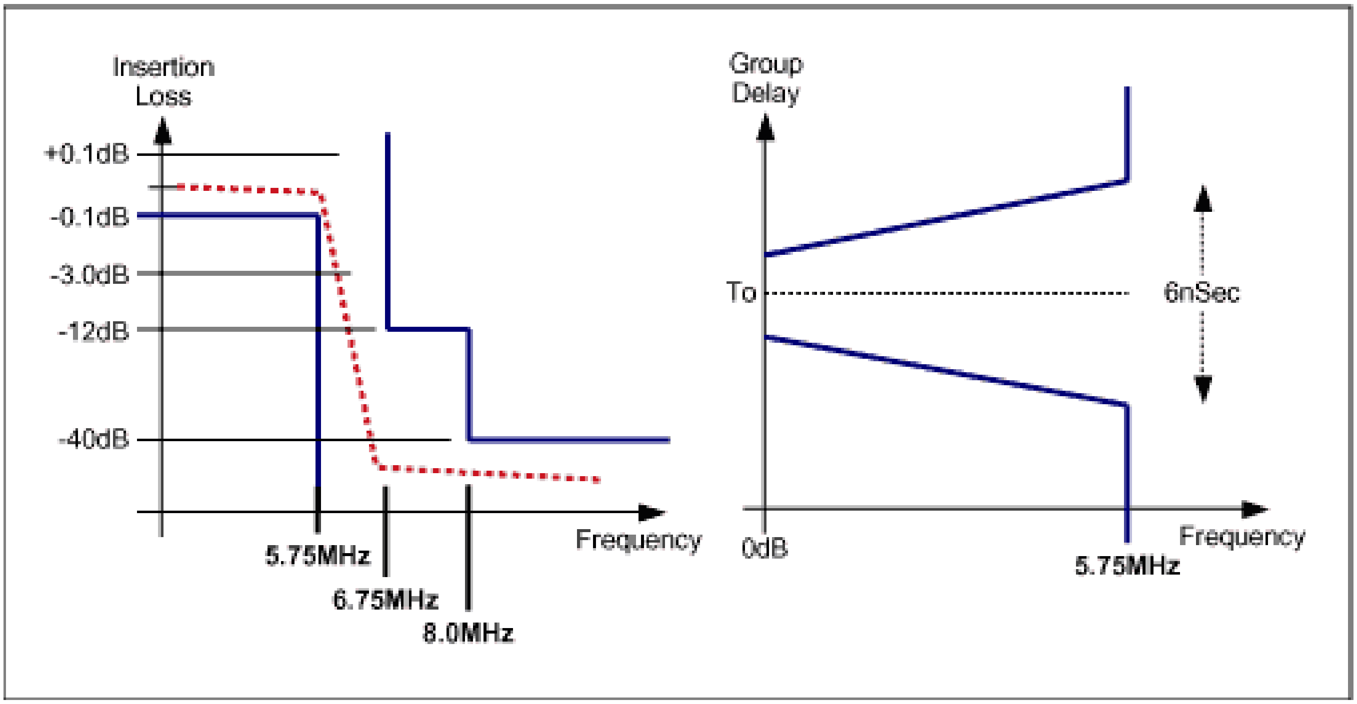

Figure 2. Filter template showing the requirements for anti-aliasing in accordance with ITU-R BT.601-5.

Using the normalized curves[9] to see if this is feasible. First, the -3db BW (ω_c) is calculated by dividing the 0.1db BW (5.75MHz), by the normalized frequency (ω/ω_c) at which the IL is 0.1db. For a 5-pole Butterworth, this is 5.75MHz/0.68 = ω_c = 8.46MHz. Then divide any frequency by the result (8.46MHz) to get a normalized frequency, and re-enter it on the curves (frequency axis) to get the IL at that frequency. For example, divide 27MHz and 32MHz by 8.46MHz to get 3.19 and 3.78 respectively. Enter these on the curves, and you get an IL equal to 50db and 58db respectively. This satisfies the selectivity requirement, but not the group delay; and a delay equalizer is added to correct that. The schematic using the MAX4450/51 with gain and group-delay characteristics is shown in Figures 3a and 3b. The important parameter for this design is the 0.1db, $2V_{P\text{-}P}$ bandwidth of the op-amp, which is specified in the MAX4450/51 data sheet[10].

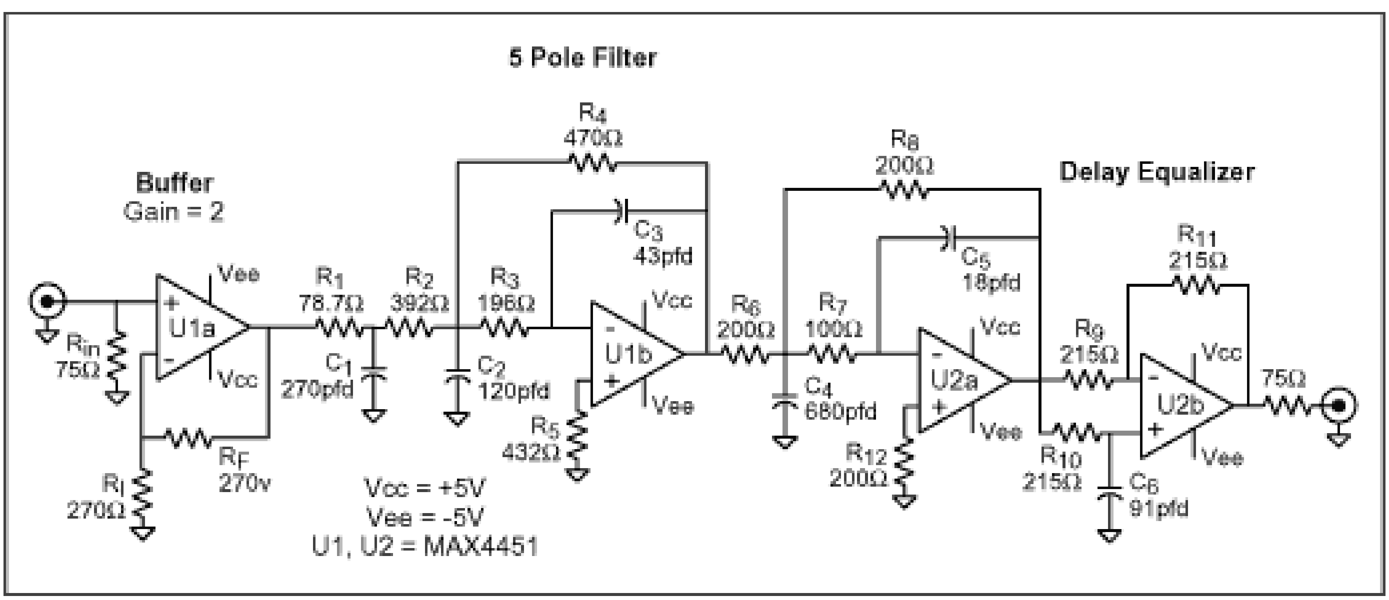

Figure 3a. The schematic of a 5.75MHz, 5-pole, Butterworth filter for ITU-601 anti-aliasing, using the MAX4451 in a Rauch circuit with a delay equalizer.

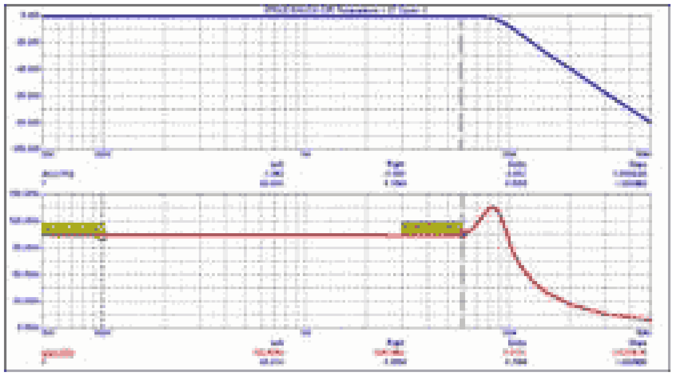

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Figure 3b. The response of a 5.75MHz, 5-pole, Butterworth filter for ITU-601 anti-aliasing, using the MAX4451 in a Rauch circuit with a delay equalizer.

At the other end of video is graphics, and HDTV, where over-sampling isn't an option; we'll look at graphics first. The XGA (1024 x 768) resolution is the most common in PCs today, with a sampling rate between 45 and 86MHz, depending on which VESA format is used. That's a Nyquist of 22.5-43MHz, and assuming 8bit video, an insertion loss of 45db or more is needed. Looking at the Butterworth curves at $\omega/\omega_c = 2$, or twice the -3db BW, a 6pole filter only gives 35-37db loss. This could be a very large filter, depending on how close we want to approach Nyquist attenuation. In reality, most anti-aliasing filters for graphics are only 3- or 4-pole filters chosen for their group delay, not for their Nyquist attenuation. Today, anti-aliasinf filters are a combination of two filters. First is a "roofing filter", placed before the ADC; the second is a highly-selective, digital FIR type, placed after the ADC. The "roofing filter" removes high frequencies that cause ringing in the digital filter. A typical filter for XGA is the 20MHz, 4-pole Butterworth, using a MAX4450/51 in the Rauch realization shown in Figures 4a and 4b. Again, the 175MHz large signal ($2V_{P-P}$) bandwidth of the MAX4450/51 and excellent transient response were the reasons it was chosen. A more selective Characteristic, such as the Legendre or Chebyshev, provides more selectivity, but requires more GBW and Group Delay Compensation, due to higher Q.

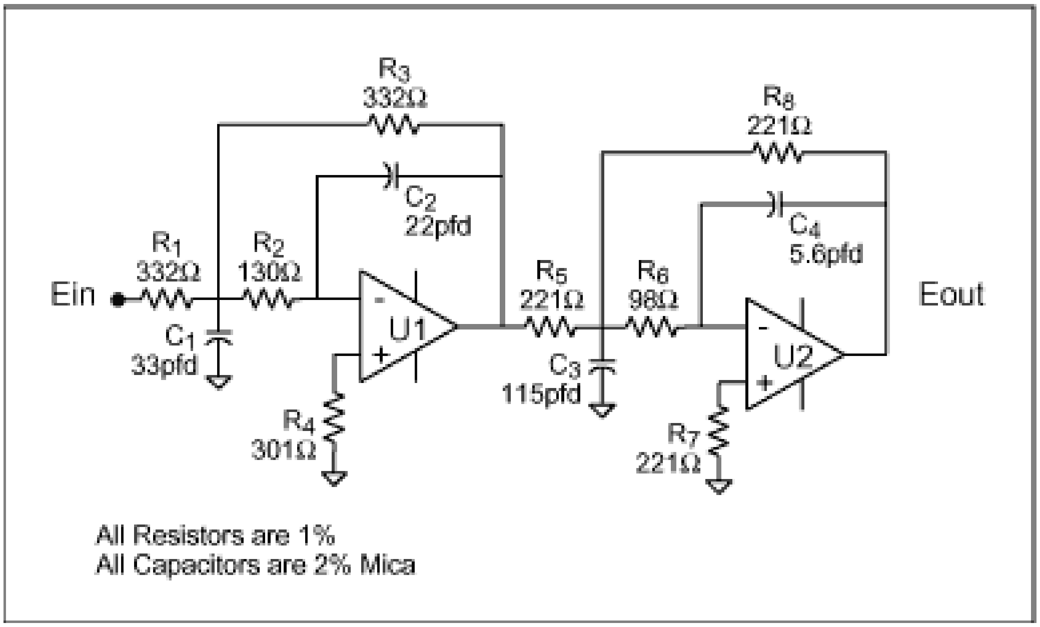

Figure 4a. A 4 pole, 20MHz Butterworth anti-aliasing filter for XGA video using a Rauch realization.

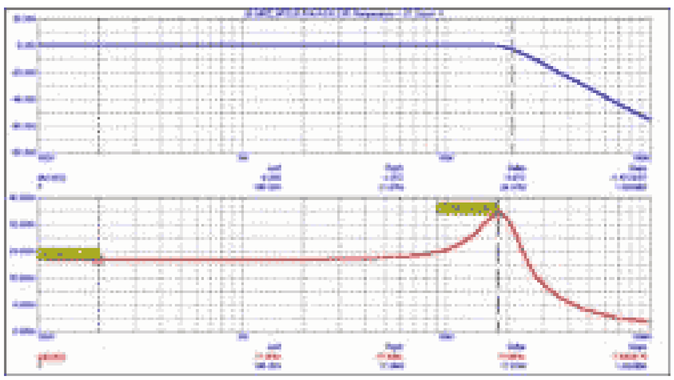

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Figure 4b. Response of a 4 pole, 20MHz Butterworth anti-aliasing filter for XGA video using a Rauch realization.

The last anti-aliasing application is HDTV, which is similar to the XGA filter above. The requirements are based on sampling at 74.25MHz, or 148.5MHz, per SMPTE274M or SMPTE296M, and there is a template for the anti-aliasing filter[11]. Like the template for ITU-601-5, this is too difficult for an analog filter, and the only choice left is a compound filter, like that used in the graphics format. At present, 1080i with 74.25MHz sampling is being used by one manufacturer, which requires a 30MHz, 0.1db bandwidth, with <1nsec group delay variation. The roofing filter suggested here is a 5-pole Butterworth using the 1 in a Sallen-Key realization and a single delay-equalization stage, using a single Quad MAX4393. The schematic with the gain and group delay is shown in Figures 5a and 5b. Although this requires more GBW than a Sallen-Key, the response will not rise back up again, when the GBW of the Op-Amp runs out. Other characteristics, such as LeGendre and Chebyshev, will improve the selectivity at the cost of Group Delay.

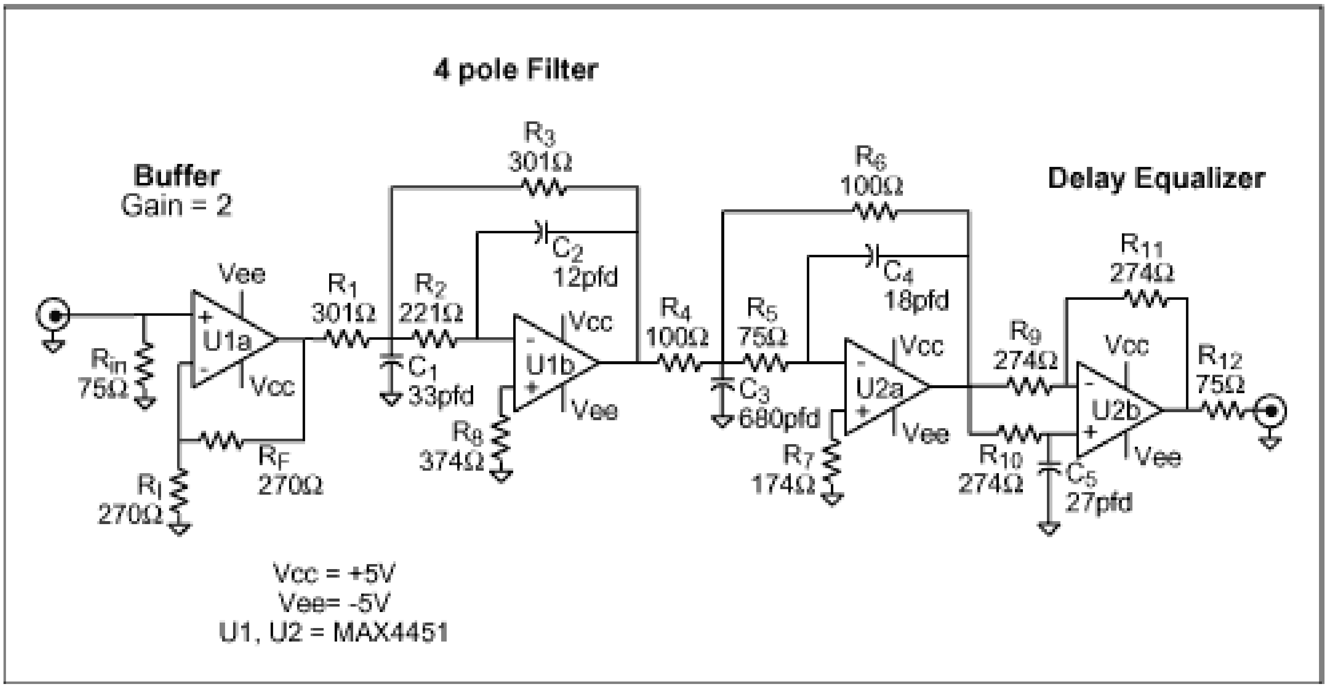

Figure 5a. An HDTV anti-aliasing filter using the MAX4450/51 with Group Delay compensation.

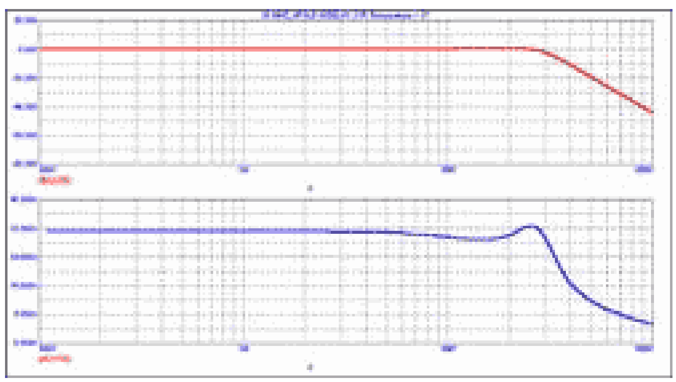

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Figure 5b. Gain and group-delay characteristics of an HDTV anti-aliasing filter using the MAX4450/51.

Reconstruction Filters

The reconstruction filtering after a DAC is one of the most poorly understood of video applications. Unlike anti-aliasing, the requirements aren't clearly specified, and most people think of it as removing the sample clock, but nothing could be further from the truth.

When an ADC samples a signal, it creates multiple, recurring images, centered on the sample-clock's harmonics. The job of the reconstruction filter is to remove all but the baseband component. If the anti-aliasing filter did its job, the output of the DAC looks like image A in Figure 6, and the images to the right are what we want to remove. Simple enough, except for one thing, the DAC samples are impulses and exist for only for an instant in time. To improve this, the DAC "holds" the impulse for a clock period, creating the familiar staircase waveform seen at a DAC output. This "hold" is a digital filter with a sin (x)/(x) characteristic, shown in

Figure 7, from whence it got its name, Sinc[12] corrector. Notice at $0.5F_S$, the Nyquist frequency, the response is down 4db. Overlaying this on Figure 6 shows the insertion loss as a crosshatched area, that includes the signal. The second job of the reconstruction filter is to restore this loss. The good news is the "hold" has a pole of attenuation centered on F_S, so you don't have to filter the sample clock, but applications still use the attenuation at F_S as a reference.

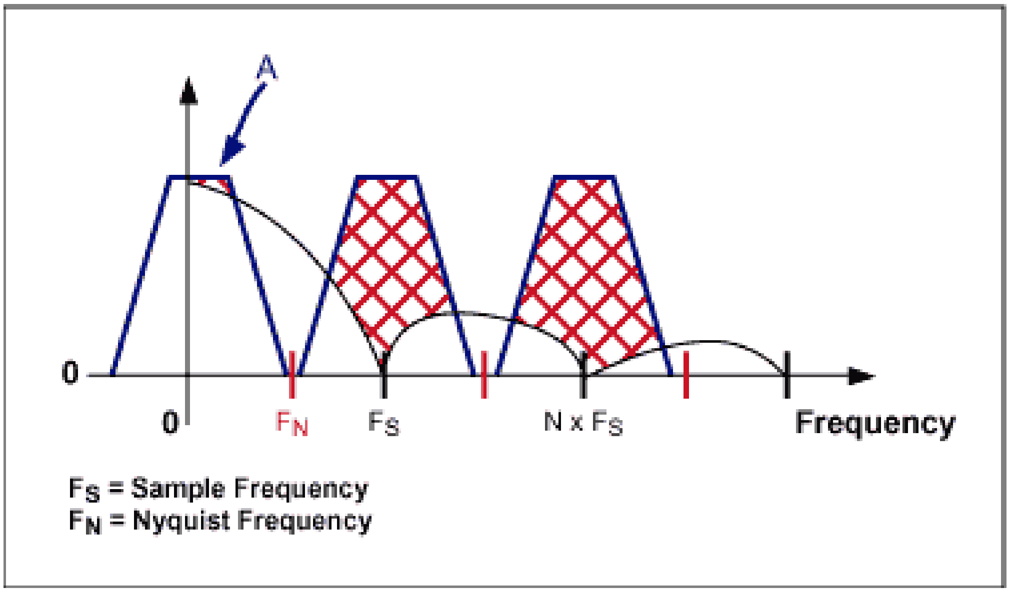

Figure 6. DAC output spectrum in terms of sampling frequency F_S and Nyquist frequency F_N.

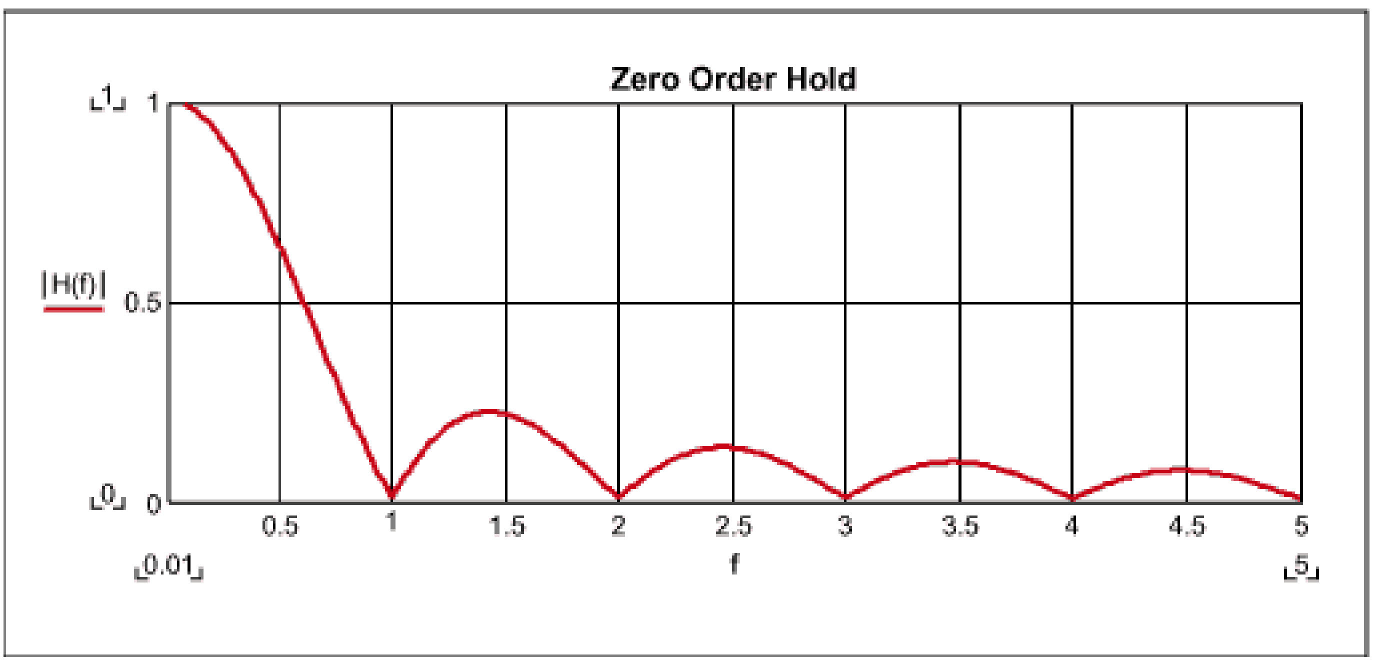

Figure 7. The sin(x)/(x) response of the DAC "hold" versus the sample frequency.

Now that we understand what a Reconstruction Filter does, we can look at some designs. Considering broadcast video first, we can assume 4X over-sampling is used for SDTV, to match the anti-aliasing. Practically, this drops the Nyquist bandwidth to $(0.5\ F_S)/4$, where the DAC loss is less than 0.3db, so little compensation is needed, and the selectivity required becomes a function of the image proximity. From the DAC response, this is about 4 x 1.44 x F_S (~78MHz), and it's attenuation is 13db due to the "hold" alone. The sample clock (54MHz) is down 30db or more, so adding another 18-20db to it, is all that's required. Using the Butterworth curves again, a 3-pole filter with an 8MHz bandwidth has >-50db @ 54MHz,and >-55db @ 78MHz. Here, I've turned to the Sallen-Key realization for two reasons. It has a single-stage gain of +2 to drive a back terminated cable; and, using R12, I can tune the group delay at the expense of the band edge selectivity. This little known attribute of the Sallen-Key variant can be used to save the expense of a separate delay equalizer, and thus optimize performance. The filter's Q is low, being only a 3-pole, so Q sensitivity is addressed by 1% resistors for Rf and Ri. A schematic, with gain and group delay, is shown in Figures 8a, 8b and 8c, using the MAX4450/51 for NTSC and PAL. Another use for R12 is to compensate for GBW in the Op-Amp.

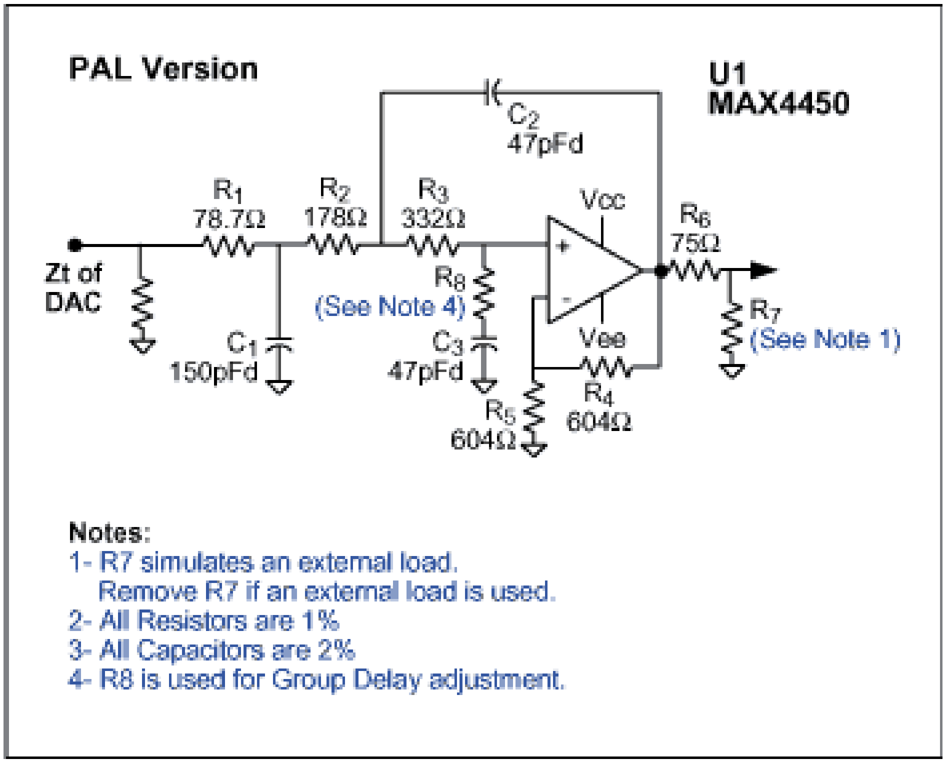

Figure 8a. SDTV Reconstruction filter with group-delay adjustment, PAL Version.

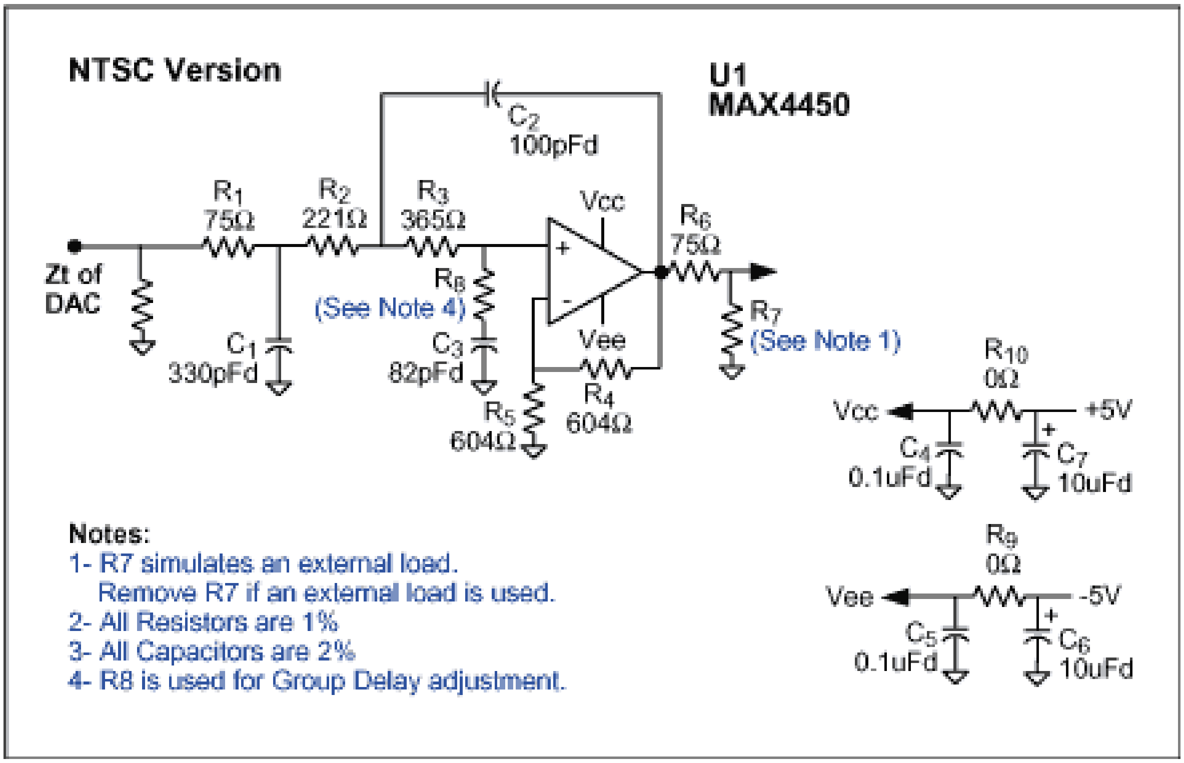

Figure 8b. SDTV Reconstruction filter with group-delay adjustment, NTSC Version.

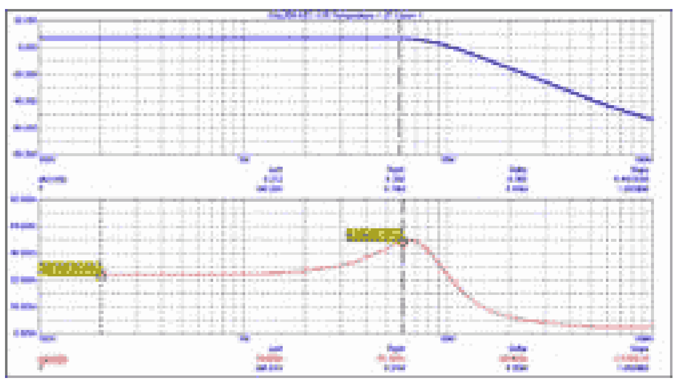

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Figure 8c. Response and group delay for the NTSC reconstruction filter of Figure 8b.

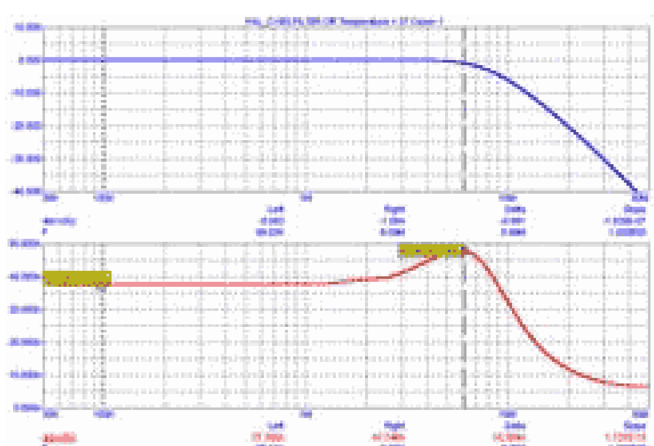

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Figure 8d. Response and group delay for the PAL reconstruction filter of 8a.

Graphics and HDTV are too high in frequency to over-sample, so to compensate for the sin (x)/(x) loss, we'll need an equalizer. The problem with most amplitude equalizers is that they corrupt the group delay. The one shown in Figure 9 is a little different because it is based on a delay-equalization stage, and its group delay is similar to a Bessel filter. The nice part about the circuit is you can design it directly from the DAC sample rate, F_S. An example of this is shown in Figures 10a and 10b, using the MAX4450/51 op-amp in a 20MHz, 3-pole Butterworth, Sallen-Key, to complement the anti-aliasing filter in Figure 4 for XGA graphics.

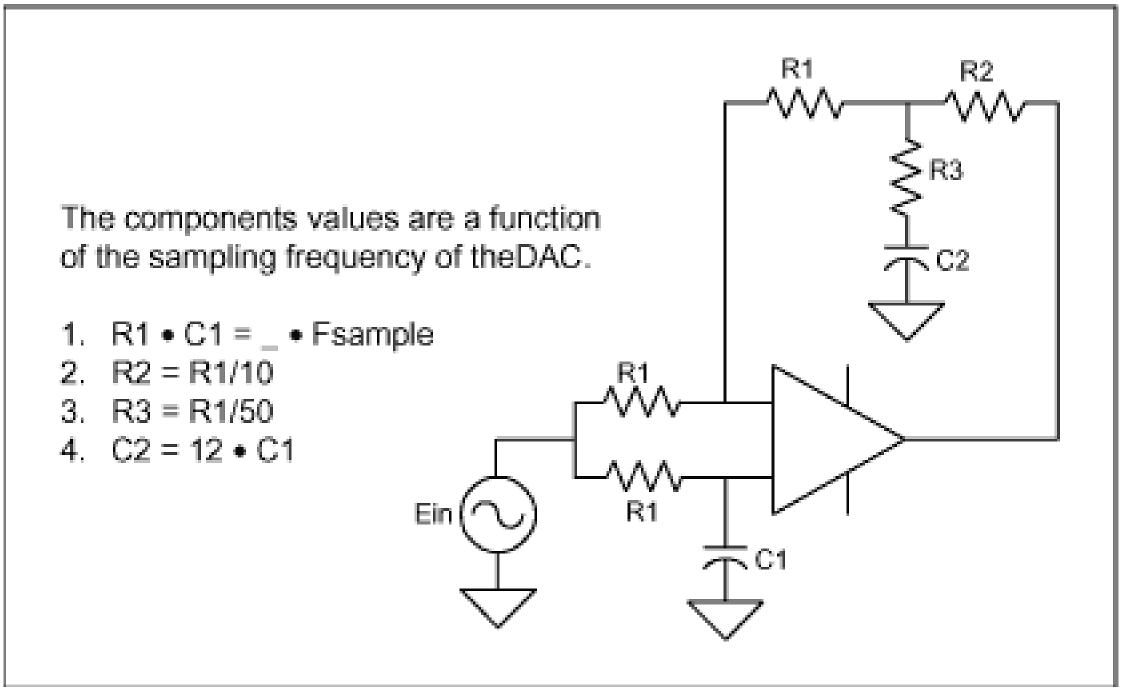


Figure 9a. Amplitude equalizer for DAC.

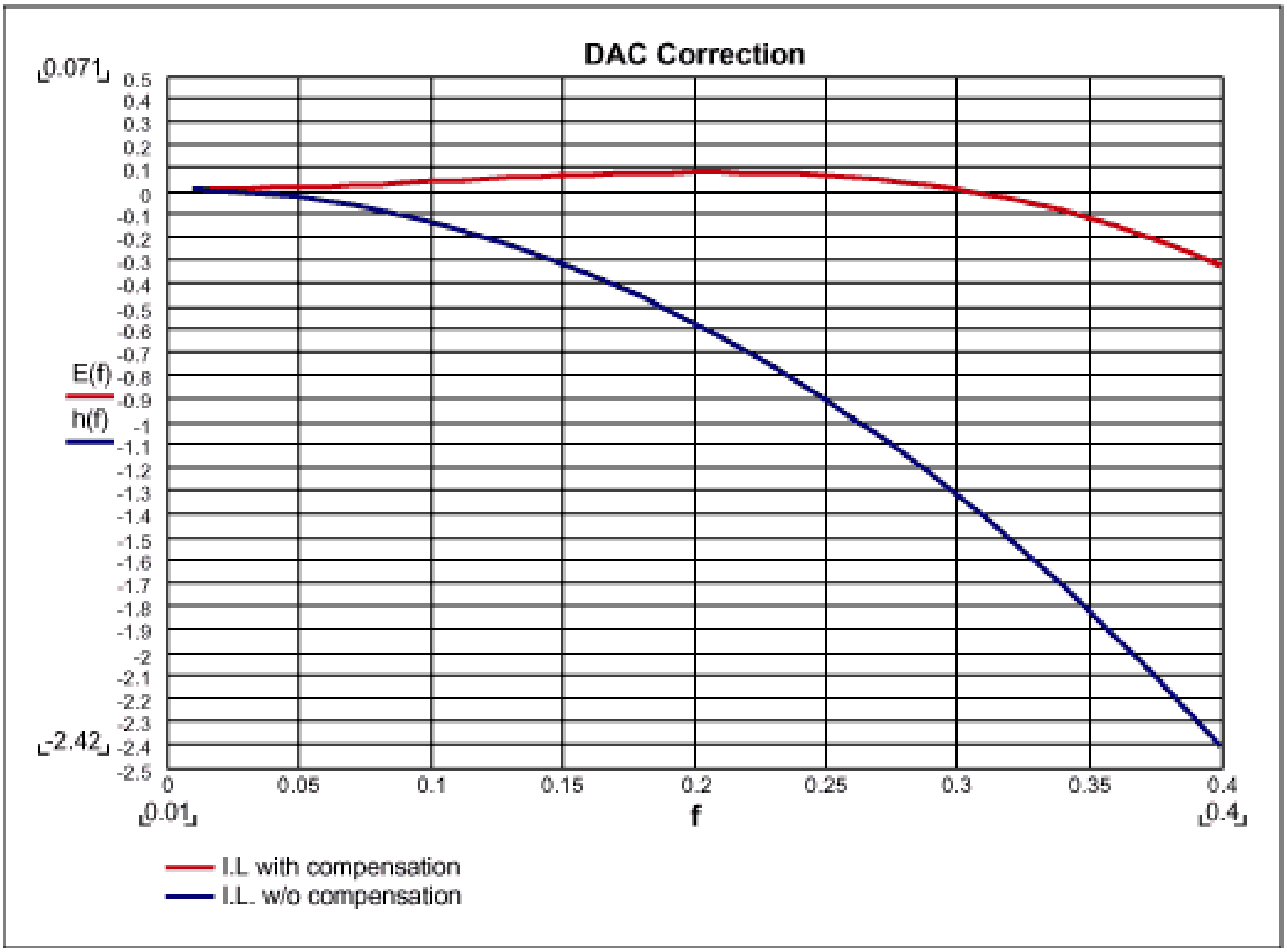


Figure 9b. DAC amplitude-equalizer response, with and without compensation.

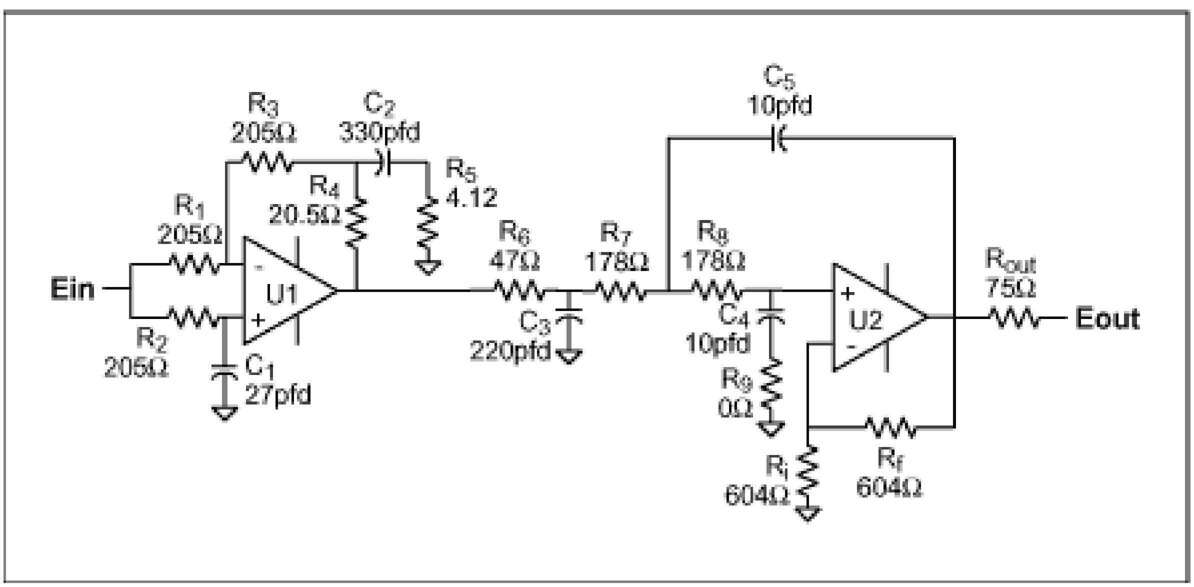
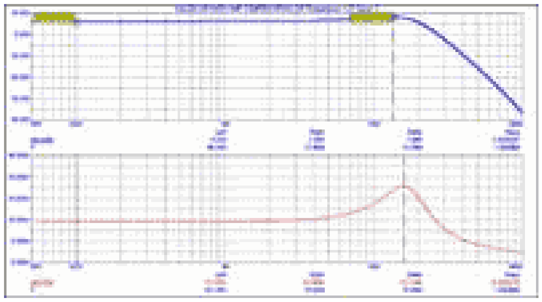


Figure 10a. A 20MHz, 3-pole Butterworth filter with sin(x)/(x) amplitude compensation for the DAC using the MAX4450/51.



[For Larger Image.]

Figure 10b. Response of the 20MHz, 3-pole Butterworth filter with sin(x)/(x) amplitude compensation for the DAC using the MAX4450/51.

The last reconstruction filter is for HDTV. From SMPTE274/296M we know this has a 29.7MHz pass-band (0.4 x F_S), sampled at 74.25MHz. The DAC is usually part of an ASIC, where the sin (x)/(x) correction is done prior to the actual conversion, so we don't require an equalizer. We will need some group delay compensation. Despite the fact that the DAC "hold" filters the clock, most Reconstruction Filters are defined in terms of how much they filter the sample clock frequency. The 30MHz, 5 Pole Sallen Key filter in figure 11 has >-40db @74.25MHz, the most common requirement for this application. Unfortunately the Sallen-Key has a tendency to rise back up when the GBW runs out, and the 4 pole design in Figure 5, does not.

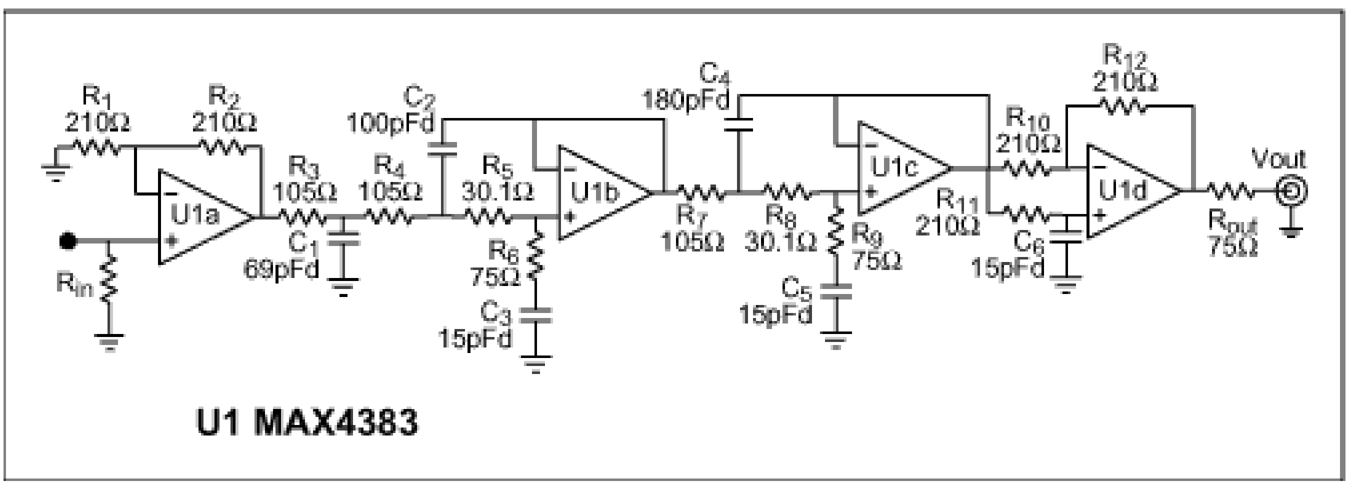


Figure 11. A 5 Pole, 30 MHZ HDTV Reconstruction Filter.

The Practical aspects of Active Video Filter Design

Whether you design your filter by hand, with software, or a combination of the two, chances are the response won't be exactly what you wanted when it's simulated and built. The most common causes of this are round-off error when converting to a standard component value and modeling errors in the simulation.

Round-off errors can be minimized by choosing standard (5%) capacitor values in the design, and deriving the resistor values from them. The reason for this is practical; the capacitors can be had with 1% or 2% tolerance, but only at 5% values, while the resistors are available with 1%

values and tolerance. This gives the best approximation, and the most precise amplitude response.

The group-delay portion of the response may also need correcting, but it usually can't be changed independently of the amplitude response. Often, the solution is a delay equalizer stage; but the Sallen-Key circuit in Figure 8a allows trading off selectivity for group delay. As R12 becomes a larger portion of R2, the circuit becomes less selective; and the group delay phase ears are lower. This is used in Figure 8 to keep the group delay variation under 35nsec in a reconstruction filter, without adding a delay equalization stage.

Modeling errors don't become apparent until after the circuit is built. The SPICE macro model published for most op-amps is based on typical, not worst-case, conditions at room temperature; and doesn't consider second-order or parasitic components. Behavioral simulation is more accurate; but it may be quicker to just breadboard the circuit, and test it.

Once built, the circuit may be unstable and oscillate, particularly if the filter has a high Q. If it does, short the input to ground and see if it continues to oscillate. The op-amp needs to return the input current to ground to operate properly, and the filter wants some (preferably) low impedance at the input. If the circuit continues to oscillate, check the frequency. If the oscillation is around the pole frequency, or just below, it's most likely due to the choice of components. Lowering the normalization impedance in the design will help this. If the oscillation is above the pole, it's probably due to the op-amp, itself, or the circuit layout.

Good layout seems like it's an art, but it yields to science once you adopt a few simple principles. The first is to have a clean supply voltage. At minimum, this requires filtering with low-ESR caps, with some impedance in the supply line, such as a ferrite bead or resistor. In some cases, a separate regulator may be needed. The loop formed when the bypass cap is attached must be small to avoid adding inductance that could resonante with the cap, and the entire circuit should be on a ground-plane PC board. As the bandwidth increases, the ground plane may add parasitic capacity that will de-tune the filter components. To avoid this, remove the ground plane beneath the offending part(s) and traces.

[1] The cutoff frequency is where the filter response drops by -3db.
[2] Electronic Filter Design Handbook, Taylor and Williams, Mc-Graw Hill, ISBN 0-07-070441-4.
[3] ibid.
[4] The 4:2:2 sampling originally indicated the number of times the color sub-carrier was over sampled, ITU-601 replaced the subcarrier frequency with 3.375MHz. 4:2:2 is sampled at 13.5, and 6.75MHz respectively.
[5] Video Demystified, Jack, K. High Text Publications, ISBN 1-878707-23-X.
[6] For the Non-Inverting case Rf/Ri=0, while the Inverting case is Rf/Ri=1.
[7] Operational Amplifier Circuits; Theory and Applications, E.J.Kennedy.
[8] as defined by H.W. Bode in Network Analysis and Feedback Amplifier Design (D. Van Nostrand, Princeton NJ, 1945).
[9] Electronic Filter Design Handbook, Taylor and Williams, Mc-Graw Hill, ISBN 0-07-070441-4.

[10] MAX445/51 Data sheet is available at www.maxim-ic.com.
[11] SMPTE 274M and SMPTE 296M specifications.
[12] The Sinc function in Mathmatics is sin (x)/(x).

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