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ORIGINAL VIDEO FILTERS WERE PASSIVE L-C CIRCUITS SURROUNDED BY AMPLIFIERS. TODAY, YOU CAN ACHIEVE SMALLER AND MORE EFFICIENT DESIGNS BY COMBINING THE AMPLIFIER WITH AN R-C FILTER. MOREOVER, SENSITIVITY ANALYSIS AND PREDISTORTION METHODS HAVE OVERCOME THE POOR PERFORMANCE THAT GAVE EARLY VIDEO FILTERS A BAD REPUTATION.

Active filters for video meet antialiasing and reconstruction requirements

HIGH-PERFORMANCE OP AMPS and specialized software for PCs enable the design of wide-bandwidth active filters, but those advantages don't address the requirements of any specific application. For video filters, the application and signal format add nuance to each circuit design. The two major video applications are antialiasing filters and reconstruction filters.

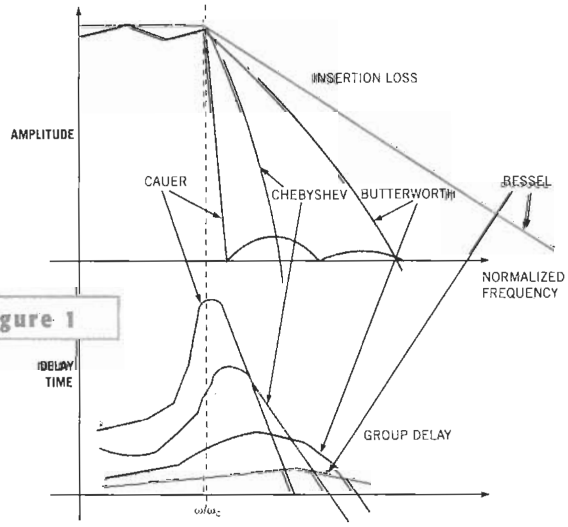
Antialiasing filters, which precede an ADC, attenuate signals above the Nyquist frequency, or one-half the sample rating of the ADC. These filters usually have the steepest possible response to reject everything above the cutoff frequency. (The filter response drops by −3 dB at the cutoff frequency.) For ITU-601 applications and others, analog filters combined with digital filters and an oversampling ADC help to achieve this performance. Little filtering is necessary for applications such as PC graphics.

Reconstruction filters, also called (sinx)/x or zero-order-hold correctors, follow DACs to remove multiple images that the sampling creates but not to remove the DAC clock. Reconstruction filters are seldom as selective as antialiasing filters because the DAC's hold function also acts as a filter—an action that lowers the required selectivity but introduces loss in the response. The available video formats are RGB, component video, composite video, and RGB PC graphics.

All applications and formats require a video filter to be "phase linear," a condition that group delay (delay versus frequency) specifies. The degree of required phase linearity depends on the application and the video format. For example, antialiasing filters and component formats have tighter phase-linearity specifications than reconstruction applications and composite video. NTSC, PAL/DVB, ITU, SMPTE, and VESA specify the requirements for the various applications and formats.

You can compare filters to determine the optimum design for a given application or format. Some examples that follow compare Rauch and Sallen-Key realizations with regard to their gain-bandwidth-to-cutoff ratios, using predistortion and element-sensitivity techniques to achieve accuracy in the design. The examples include an ITU-601 antialiasing filter;

Figure 1



Filter types exhibit different characteristics of amplitude and group delay versus frequency.

a 20-MHz antialiasing and reconstruction filter; and an HDTV-reconstruction filter.

Whether you use a video filter for antialiasing or reconstruction, the filter must have a lowpass characteristic to pass the video-frame rate. You should therefore beware of ac coupling. The industry categorizes lowpass filters by their amplitude characteristic or by the name of the polynomial, such as Bessel, Butterworth, Chebyshev, or Cauer, that describes it. **Figure 1** shows these characteristics normalized to a 1-rad bandwidth. Normally, you'd choose a filter with the best selectivity and the fewest poles to minimize cost, but the additional need for phase linearity limits the available choices.

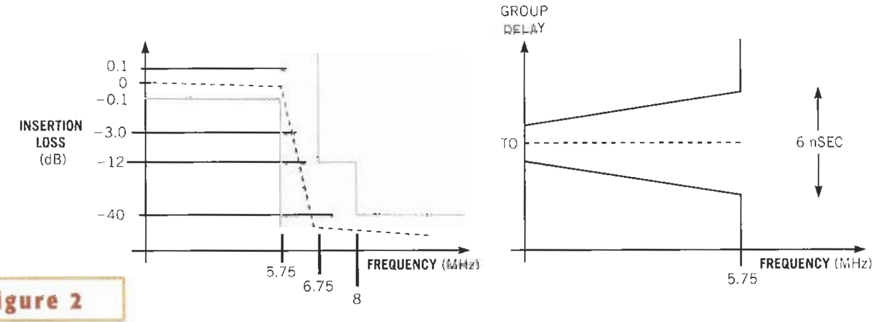


Figure 2

This filter template illustrates antialiasing requirements in accordance with the ITU-R BT.601-5 standard.

PHASE LINEARITY AND GROUP DELAY

A filter's phase linearity is specified as envelope delay or group delay versus frequency. A flat group delay indicates that all frequencies are delayed by the same amount, which preserves the shape of the waveform in the time domain. Thus, absolute group delay is less important than the variation in group delay. (Do not confuse group delay with channel-to-channel variation, which is specified as "time coincidence.")

Group delay is undesirable but not unacceptable for video, so you may ask how much is acceptable and why it is acceptable. The answer depends on the application and the video format. For example, ITU-470 loosely specifies group delay for composite video, but ITU-601 specifies it tightly to ensure "generational stability," both for MPEG-2 compression and to control phase jitter before serialization. So, you may need to know what filter characteristics to look for to ensure phase linearity.

The group-delay curves in **Figure 1** show a peak near the cutoff frequency (ω/ω_C=1). The steep phase change near the cutoff frequency causes this problematic peak. To get an idea of scale, a three-pole, 6-MHz Butterworth filter has a group-delay variation of 20 to 25 nsec over its bandwidth. Adding poles or increasing the filter's selectivity increases that variation. Other, more exotic filters that minimize group-delay variation include Bessel, phase approximation, Thompson-Butterworth, and LeGendre (**Reference 1**). Nevertheless, the Butterworth characteristic is most common for video.

GROUP-DELAY PROBLEMS

All formats and applications are sensitive to group-delay variation. The degree of sensitivity depends on the number of signals and their bandwidths. Composite NTSC/PAL has only one signal, and ITU-470 specifies group delay. Those requirements are easy to meet. RGB and component video both have multiple signals. The RGB signals have equal bandwidths, but component-video signals do not, making group-delay matching easy with RGB but difficult with component video.

Because the Pb and Pr signals have half the bandwidth of the luma (Y) signal, their group delay is double that of the Y signal. One approach is to slow the Y signal by adding delay stages. Another is to equalize bandwidths by doubling the sample rates of Pb and Pr. That approach raises the 4:2:2 sampling rate to 4:4:4, allowing you to treat the signal as RGB. (The 4:2:2 sampling originally indicated the number of times the color subcarrier was oversampled. ITU-601 replaced the subcarrier frequency with 3.375 MHz. The 4:2:2 sampling occurs at 13.5 and 6.75 MHz.) Reconstruction applications discard or average the additional Pb

TABLE 1—COMPONENT SENSITIVITIES, INCLUDING BANDWIDTH AND Q PREDISTORTION FORMULAS (SALLEN-KEY REALIZATION, ω_0=1 RAD/SEC)

	Gain K=3-1/Q	Gain K=1	Gain K=2
Sensitivity S_x^y	$R_1=R_2=C_1=C_2=1$	$R_1=R_2=1$	$R_1=C_1=1$
S_x^ω ($x=R_1, R_2, C_1, C_2$)	−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	NA	−1/2
S_{RB}^K	9/14	NA	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]	

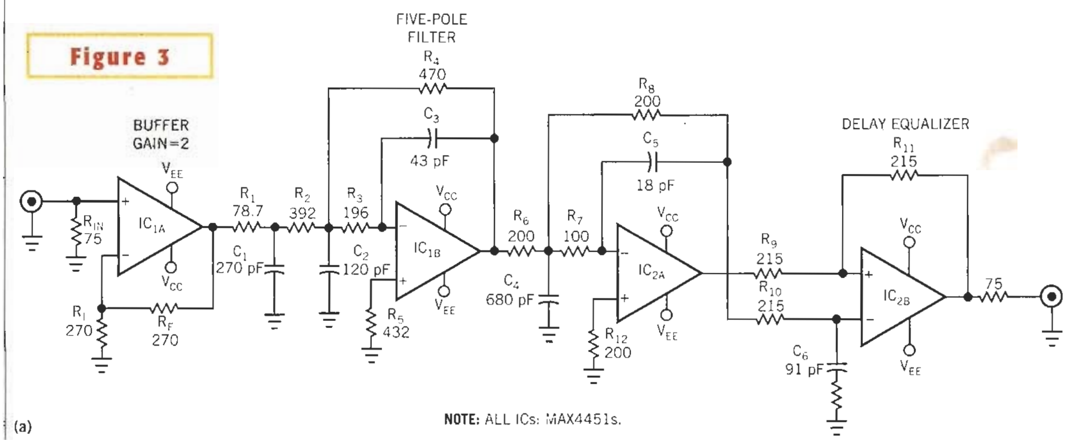
TABLE 2—COMPONENT SENSITIVITIES, INCLUDING BANDWIDTH AND Q PREDISTORTION FORMULAS (RAUCH REALIZATION, ω_0=1 RAD/SEC)

	Gain K=1	Gain K=2	Gain K=2
Sensitivity function S_x^y	$R_1=R_2=R_3=1$	$R_1=1$, $R_3=H_o$, $R_2=(H_o/1+H_o)$	$C_1=1$, $C_2=C_1/100$
S_x^ω ($x=R_2, R_3, C_1, C_2, S_{R1}^\omega=0$)	−1/2	−1/2	−1/2
S_{R1}^Q	11/3	11/3	11/3
S_{R2}^Q	−11/6	0	0
S_{C1}^Q	11/2	11/2	11/2
S_{C2}^Q	−11/2	−11/2	−11/2
S_{R3}^K	1	1	1
S_{R1}^K	−1	−1	−1
S_{R3}^Q	11/6	0	0
ω_c(Actual)	ω_c(Design)[1 − 3ω_cQ/2GBW]		
Q(Actual)	Q(Design)[1+3ω_cQ/2GBW]		

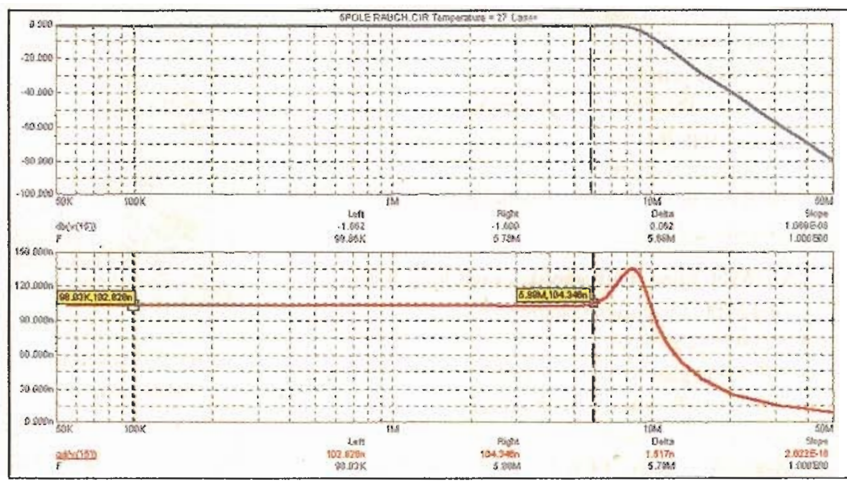
and Pr samples during antialiasing.

The other component-video format (S-VHS) is confusing to some. The Y channel is the same as in YPbPr, but the chroma signal, C, looks like it needs bandpass filtering rather than lowpass filtering. As for YPbPr signals, bandpass filtering causes group-delay and timing problems, so don't do it! Unless you're encoding analog signals, you can lowpass-filter Y and C with the same filter. S-VHS is more forgiving of bandwidth limitations than of problems caused by trying to equalize the delay. You typically see S-VHS in reconstruction applications, for which the main concern is correct timing between Y and C.

Figure 3



(a)



(b)

A five-pole, 5.75-MHz Butterworth filter for ITU-601 antialiasing uses a Rauch circuit with delay equalizer (a) to achieve the output response (b).

CHOOSING AN OP AMP

After choosing a filter characteristic, the next step is to implement it with an actual circuit. The most commonly used single-op-amp circuits are the Sallen-Key configuration in noninverting form and the Rauch configuration in inverting form. An important consideration for op amps operating in the wide bandwidths of video applications is the minimum gain bandwidth. Video signals are typically 2V p-p, so you need to refer to the large-signal gain bandwidth. Don't confuse that parameter with the 2V p-p, 0.1-dB gain bandwidth, which is much lower.

For filter circuits, an important question to answer is how much larger than the filter's cutoff frequency does the op-amp gain bandwidth have to be? For a Rauch filter, the phase argument of the filter's response characteristic is

$$\mathrm{ARG}[K(j_\omega)]\mathrm{NONINV}= -(\omega_C/\mathrm{GBW}_{RAD})(1+R_F/R_I). \qquad (1)$$

For a Sallen-Key filter, the argument is

$$\mathrm{ARG}[K(j_\omega)]\mathrm{NONINV}=\pi -(\omega_C/\mathrm{GBW}_{RAD})(1+R_F/R_I). \qquad (2)$$

In these equations, R_F and R_I are the gain-set resistors in ohms, GBW_{RAD} is the op amp's gain-bandwidth product, and ω_C is the filter's cutoff frequency in radians per second. You set the gain by introducing values for R_F and R_I and solving for $\omega_C/\mathrm{GBW}_{RAD}$. For the noninverting case, $R_FR_I=0$, and, for the inverting case, $R_F/R_I=1$. A unity-gain Rauch circuit has $R_F/R_I=1$, and a Sallen-Key circuit has $R_F/R_I=0$. Thus, for the same phase error, a Sallen-Key circuit requires half the gain bandwidth of a Rauch circuit. As the required gain increases, the phase error of the two circuit types converges, leaving little advantage for the Sallen-Key in gain bandwidth, but you must also consider other issues.

PREDISTORTION, BANDWIDTH, AND Q

Anything less than an infinite $\mathrm{GBW}_{RAD}/\omega_C$ ratio causes the closed-loop poles of a filter to move. For this reason, an actual filter often exhibits a lower bandwidth (ω_C) than does the paper design (**Reference 2**). You can compensate for this effect with predistortion by increasing the design bandwidth. Formulas for the Sallen-Key and Rauch circuits help to calculate a design bandwidth that provides the actual necessary bandwidth (**tables 1** and **2**).

The next design aspect to consider is component tolerance. To determine component tolerance, you need a sensitivity function (**Reference 3**): S_X^Y gives the ratio between a change in the value of part X and the consequent change in parameter Y. For example, **Table 1** shows that the Q in a Sallen-Key circuit has greater sensitivity to variations in C_1 and C_2 than does Rauch circuit. Thus, a Sallen-Key filter is less tolerant of parasitics than a Rauch filter. The sensitivity function, S_X^Y, lets you predict the effect and then design accordingly. Now, consider some typical designs.

DESIGN OF ANTIALIASING FILTERS

For antialiasing filters, a template such as the one in **Figure 2** for ITU-601 determines the sensitivity. The specified bandwidth is 5.75 MHz±0.1 dB with an insertion loss of 12 dB at 6.75 MHz and 40 dB at 8 MHz and with a group-delay variation of ±3 nsec over the 0.1-dB bandwidth. Such performance is too difficult for an analog filter alone, but four-times oversampling modifies the requirements to 12 dB at 27 MHz and 40 dB at 32 MHz.

Using software or normalized curves, you find that a five-pole Butterworth filter with a −3-dB bandwidth of 8.45 MHz satisfies the requirement for selectivity but not for group delay (**Reference 1**). You need a delay stage, for which the important op-amp parameter is the 0.1-dB, 2V p-p bandwidth. You use this bandwidth number in **equations 1** and **2** to get an accurate design. A schematic for this application with curves showing its gain and group-delay characteristics, uses four-times oversampling (**Figure 3**).

Next, consider PC video. VESA does not specify templates for antialiasing or reconstruction filters. The XGA resolution (1024×768 pixels at 85 Hz) has a sampling rate of 94.5 MHz and a Nyquist frequency of 47.25 MHz. A Rauch realization of a 20-MHz, four-pole Butterworth filter achieves greater than 35-dB attenuation at the Nyquist frequency (**Figure 4**). This design uses the MAX4450/4451 for its transient-response characteristic and signal bandwidth of 175 MHz at 2V p-p.

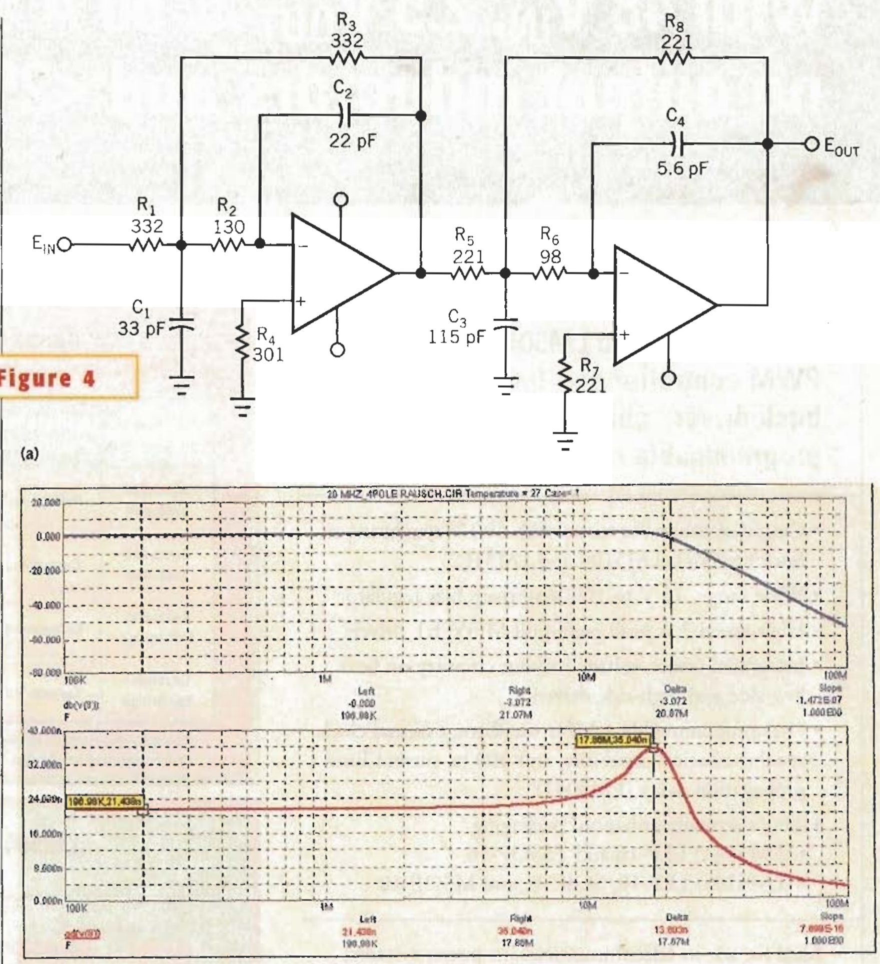


Figure 4

A four-pole, 20-MHz Butterworth filter for XGA graphics antialiasing uses a Rauch circuit (a) to achieve the output response (b).

RECONSTRUCTION FILTERS

Many designers do not fully understand reconstruction filtering after a DAC. Designers often think that the purpose of reconstruction filters is to remove the sample clock, but nothing is further from the truth. When a system samples a signal, the samples comprise multiple recurring images of the signal centered on harmonics of the sample clock. A reconstruction filter removes all but the baseband sample. If the antialiasing filter has done its job, the DAC output looks like Point A in **Figure 5**, and you then want to remove all samples

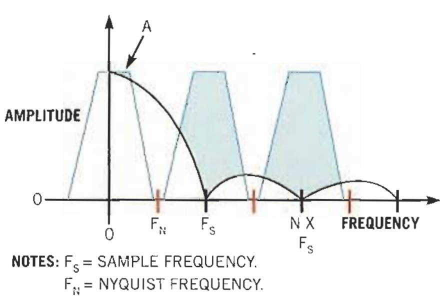


Figure 5 The reconstruction filter's job is to remove all but the baseband sample, or Point A, from a typical DAC's output spectrum, which appears here in terms of the sampling, F_S, and Nyquist frequencies, F_N.

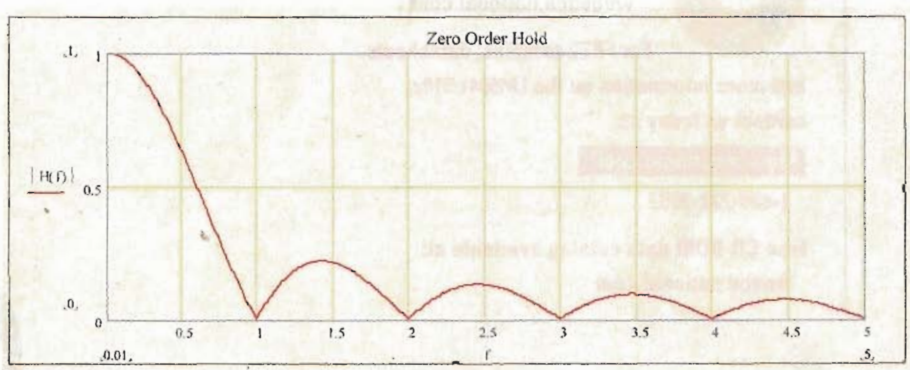
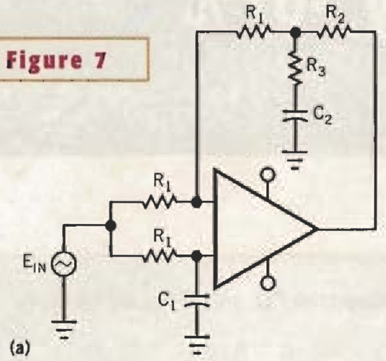


Figure 6 The hold function of a DAC produces a (sinx)/x response with nulls at multiples of the sampling frequency.

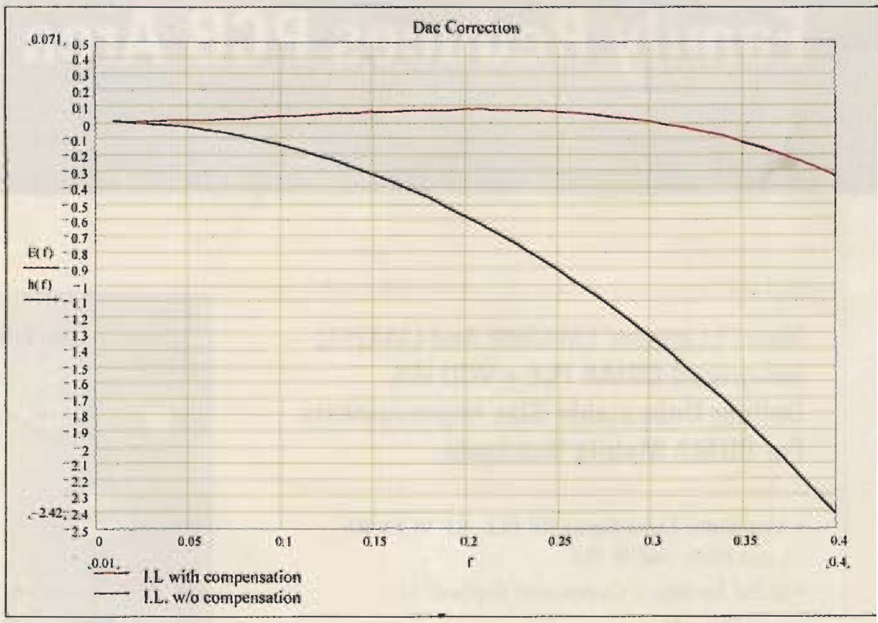
Figure 7



(a)

NOTES:
1. THE COMPONENT VALUES ARE FUNCTIONS OF THE SAMPLING FREQUENCY OF THE ADC.
2. $R_1 \times C_1 = 1/(4 \times f_{SAMPLE})$.
3. $R_2 = R_1/10$.
4. $R_3 = R_1/50$.
5. $C_2 = 12 * C_1$.

(a)



(b)

An amplitude equalizer circuit (a) provides (sinx)/x correction for a DAC output (b).

to the right of it. Thus, reconstruction is similar to antialiasing except, because each sample exists only for an instant, the DAC holds it for one clock period, thereby creating the familiar staircase approximation to a sloping line.

The hold function corresponds to a digital filter whose Sinc-function characteristic, or sinx/x, is similar to that of a Butterworth or a Bessel filter (**Figure 6**). Notice that the response is down 4 dB at half the sample frequency. The second job of a reconstruction filter is to restore that loss, which requires an amplitude equalizer (**Figure 7a**). This equalizer is based on a delay stage and has a response like a Bessel filter. The design of the equalizer is based on the DAC sample rate, F_S. **Figure 7b** shows the DAC's frequency response with and without an amplitude equalizer. Like the delay stage, you can include an equalizer in any reconstruction filter.

The hold response also has a pole centered on the sample clock, which completely removes the clock. Nevertheless, most reconstruction applications refer to clock attenuation as a figure of merit.

The most common requirement for NTSC/PAL reconstruction is an attenuation of greater than 20 dB at 13.5 MHz and greater than 40 dB at 27 MHZ, where ω_C depends on the applicable video standard. A three-pole Butterworth with Sallen-Key configuration is a good choice for two reasons. First, its gain of 2 drives a back-terminated cable. Second, you can adjust the group-delay variation to optimize performance without using a delay equalizer. **Figure 8** shows both NTSC and PAL designs and the corresponding gain and group-delay characteristics. These applications usually include digital amplitude correction for the DAC, but you can easily add that feature if necessary.

To illustrate a circuit for XGA, a 20-MHz, three-pole Butterworth filter in the Sallen-Key configuration includes **Figure 8**'s circuit for amplitude correction. (See **Figure 9** on the Web version of this article at www.edn.com.) Complementing the antialiasing filter of **Figure 4**, this filter has a gain of 2 to drive a back-terminated, 75Ω coaxial cable.

The last application is a reconstruction filter for HDTV. Based on the templates in SMPTE 274 and 296M, it has a center frequency of $\omega_C = 0.4 \times F_S = 29.7$ MHz. The DAC usually includes amplitude correction, but you need to add group-delay compensation. The resulting 30-MHz, five-pole Sallen-Key filter has greater than 40-dB attenuation at 74.25 MHz and a group-delay stage with gain of 2 to drive a back-terminated, 75Ω coaxial cable. (See **Figure 10** on the Web version of this article at www.edn.com.)

ACTIVE VIDEO-FILTER DESIGN

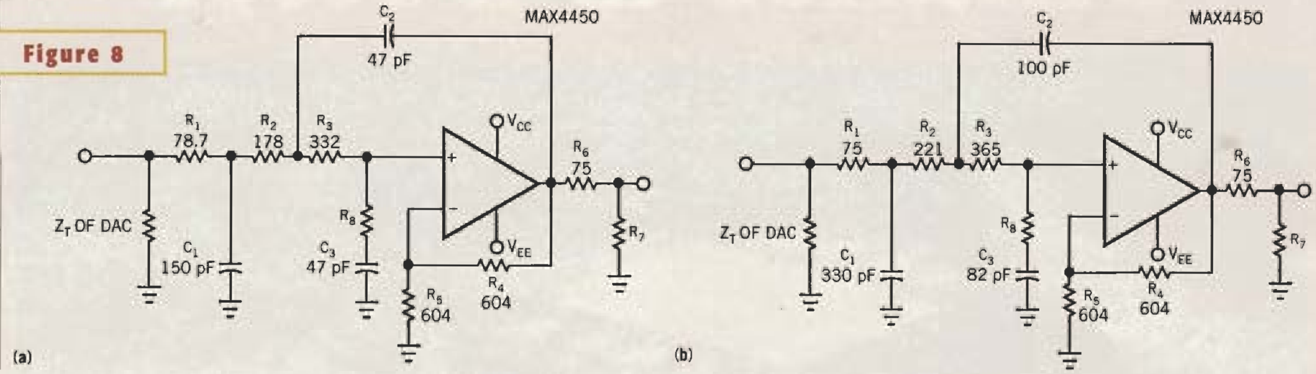
Whether you design filters by hand, with the aid of software, or with a combination of these approaches, the actual response may differ from what you expect. One cause is the discrepancy between a calculated response and the actual response you obtain using standard component values.

You can minimize error by choosing standard 5% capacitor values and deriving the resistor values from them. The reason is practical: Capacitors are available with 1 or 2% tolerance but only at 5% values, whereas resistors are available that combine 1% values with 1% tolerance. Such components give the best approximation and the most precise amplitude response.

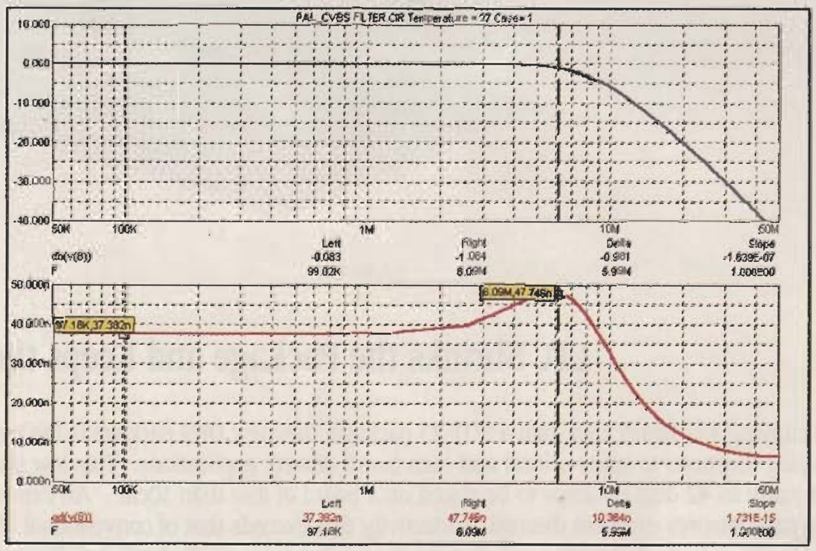
Once you build it, a filter may be unstable and oscillate. In that case, short the input to ground and see whether it continues to oscillate. If it stops, the impedance is too high. Lowering the design impedance should eliminate the oscillation. If it continues, note whether the oscillation is near or just below the filter cutoff frequency. If it is just below the cutoff frequency, the oscillation is probably due to components or parasitics. If the oscillation is above the cutoff frequency, it's probably due to the op amp or the circuit layout.

Good layout seems an art, but it's based on a few simple principles, such as having a clean supply voltage and a solid ground. At minimum, that requirement means filtering with low-ESR capacitors

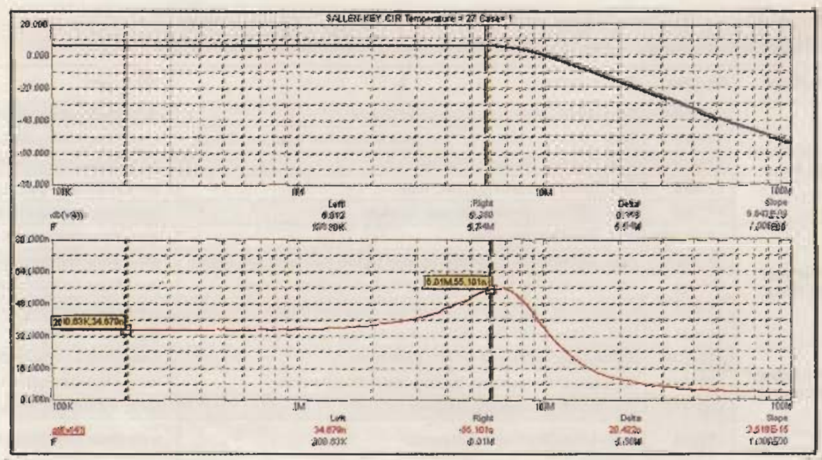
Figure 8



and sometimes with a regulator. The loop that the bypass-capacitor connections form must be small, or the resulting parasitic inductance will resonate with the capacitance. A good ground plane is essential to good analog design, but as bandwidth increases, the ground plane may add parasitic capacitance that can detune the filter. To avoid that problem, remove the ground plane beneath the offending parts and traces.□



(c)



(d)

You can design reconstruction filters with group-delay adjustment in PAL (a) and NTSC (b) versions. The respective amplitude and group-delay responses are in c and d.

References

1. Williams, Arthur B and Fred Taylor, *Electronic Filter Design Handbook, Second Edition*, McGraw Hill, ISBN 0070704341, 1988.

2. Kennedy, Eldredge J and Maxwell F Kennedy, *Operational Amplifier Circuits: Theory and Applications*, International Thomson, 1997.

3. Bode, Hendrik W, *Network Analysis and Feedback Amplifier Design*, RE Krieger Publishing Co, 1945.

Authors' biographies

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DESIGN NOTES

High Frequency Active Antialiasing Filters – Design Note 313

Philip Karantzalis

Introduction

High frequency (1MHz or higher) active lowpass filters are now practical alternatives to passive LC filters mainly due to the availability of very high bandwidth (100MHz or higher) integrated amplifiers. Analog signal filtering applications with bandwidths in the megahertz region can be implemented by a discrete active filter circuit using resistors, capacitors and a 400MHz operational amplifier such as the LT®1819 or the LT6600-10, a fully integrated lowpass filter. The LT6600-10 has a fixed 10MHz frequency response equivalent to a fourth order flat passband Chebyshev function. An LT1819-based active RC lowpass filter can be designed to have a Chebyshev, Butterworth, Bessel or custom frequency response (up to 20MHz).

The LT6600-10 Lowpass Filter

The LT6600-10 is a fully integrated, differential, fourth order lowpass filter in a surface mount SO-8 package (Figure 1). Two external resistors ranging from 1600Ω to 100Ω set the differential gain in the filter's passband from –12dB to 12dB respectively. The LT6600-10 passband gain ripple is a maximum of 0.7dB to –0.3dB up to 10MHz and attenuation is typically 28dB at 30MHz and 44dB at 50MHz. The signal to noise ratio (SNR) at the filter's output is 82dB with a $2V_{P-P}$ signal for a passband gain equal to one (a SNR suitable for up to 14 bits of resolution). In addition to lowpass filtering, the LT6600-10 can level shift the input common mode signal. For example, with a single 3V supply, if the input common mode voltage is 0.25V, then the output common mode voltage can be set to 1.5V. The LT6600-10 operates with single 3V or 5V or dual 5V power supplies.

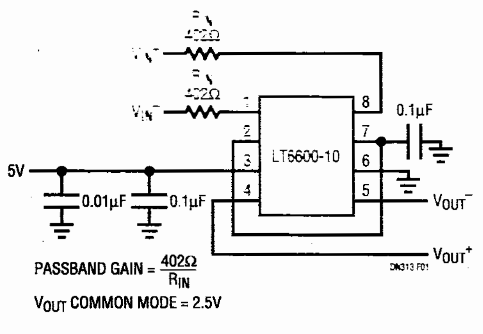

Figure 1. LT6600-10 10MHz, Lowpass Filter Features a Single Supply and Only Two External Resistors

An LT1819-Based RC Lowpass Filter

The LT6600-10 greatly simplifies lowpass filter design because it only requires two external resistors to set the differential gain, but the passband is fixed. For more flexibility, the LT1819 400MHz, high slew rate, low noise and low distortion dual operational amplifier is a good choice. Figure 2 shows a differential, 10MHz, 4th order, lowpass filter using two LT1819s. This approach allows for adjustable passband up to 20MHz but at the expense of a large number of passive and active components and high sensitivity to the variation of component values. For example, a component sensitivity analysis of Figure 2 shows that in order to maintain a passband ripple similar to the LT6600-10 (±0.5dB up to 10MHz), the component tolerance must not exceed ±0.5% for the resistors and 1% for the capacitors. Also, the LT1819 gain-bandwidth product should not be less than 300MHz. If a Butterworth, Bessel or custom filter response is desired, ±1% resistors and ±5% capacitors are adequate. These filters have lower sensitivity than a "flat" passband Chebyshev filter. The LT1819-based filter operates with single 5V or dual 5V power supplies (for a single 3V power supply filter circuit use an LT1807, a dual 325MHz, rail-to-rail operational amplifier).

Antialiasing 10MHz Filters for a Differential 50Msps ADC

An LT6600-10 or an LT1819-based 10MHz lowpass filter provides adequate stopband attenuation for reducing aliasing signals at the input of a high speed analog-to-digital converter (ADC) such as the LT1744, a 50Msps, differential input ADC. Figure 3 shows the gain response

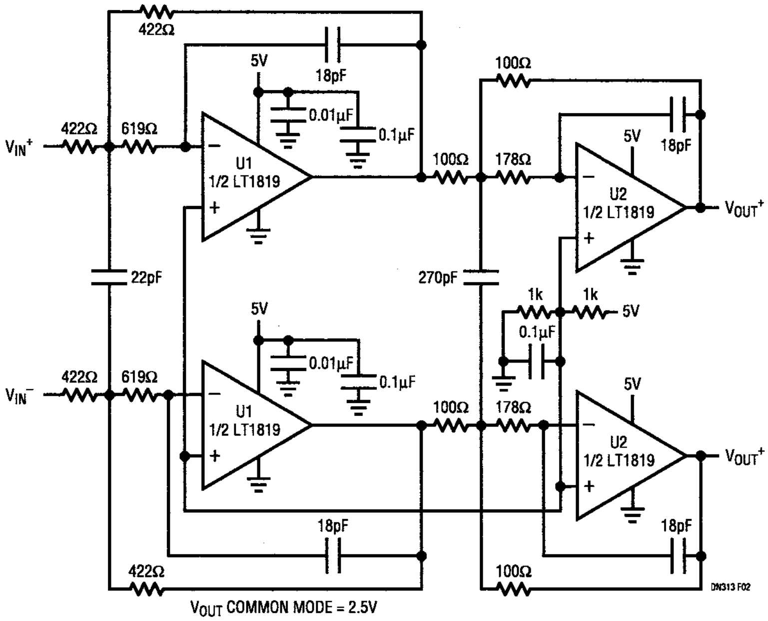

Figure 2. Another 10MHz, Single Supply Lowpass Filter, Similar to Figure 1. This Circuit Features the LT1819 Op Amp and Adjustable Bandwidth Up to 20MHz

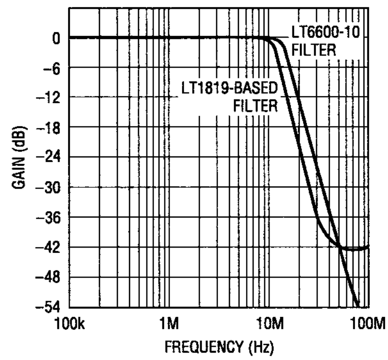

Figure 3. The Frequency Response of the Two 10MHz Antialiasing Filters Shown in Figures 1 and 2

of the LT6600-10 and the LT1819-based 10MHz filters. The LT1819 filter is designed to have higher attenuation at 20MHz than the LT6600-10 filter in order to achieve sufficient stopband attenuation. The stopband attenuation beyond 40MHz of the LT1819 circuit is limited to –42dB by printed circuit stray paths and differential component mismatches that decrease the common mode rejection at very high frequencies. The stopband attenuation of the fully integrated LT6600-10 filter continues increasing beyond 40MHz. Figure 4 shows the DC to 10MHz plot of a 1MHz $2V_{P-P}$ differential sine wave processed by an LT6600-10 plus an LT1744 14-bit ADC. The plots are an averaged 4096-point FFT of a 1MHz sinewave digitized at 50 million samples per second. The 1MHz harmonic distortion of Figure 4 is virtually the same when an LT1819-based filter is used with an LT1744. In a 10MHz bandwidth, the measured signal-to-noise plus distortion for the LT6600-10 plus LT1744 circuit is 74.5dB and essentially equal to the dynamic range of an LT1744 ADC (a minimum of 75.5dB for a $2V_{P-P}$ signal). The LT1819-based filter is slightly noisier, the measured signal-to-noise ratio plus distortion for the LT1819 plus LT1744 circuit is 71.5dB.

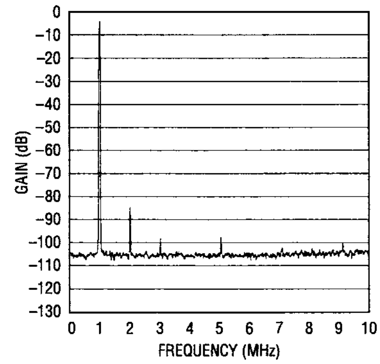

Figure 4. Spectral Plot of a 1MHz, $2V_{P-P}$, Differential Sine Wave Input to an LT6600-10 Filter Plus an LT1744 14-Bit ADC (a DC to 10MHz Plot of a 4096-Point Averaged FFT with a 50MHz Sample Rate)

Conclusion

The LT6600-10 offers a high performance, 10MHz differential filter with gain, in a small package (SO-8) while the LT1819 op amp can be used to create a variety of differential filters up to 20MHz.

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