TEK TEXT

Graphic Equalizer Design Provides Flat Unity Gain Response

Audio systems typically include filter circuits tuned to a user specified frequency band.

Johnny Molina

Gain or attenuation of these bands can enhance the "timbre" of the audio sound once the signal source becomes audible through a pair of speakers. One such filter circuit is a graphic equalizer.

Graphic equalizers for audio equipment commonly use a group of bandpass filters to separate the audio spectrum into different frequency bands (*Figure 1*). The amplitude of each band can then be individually adjusted to balance the response of the various ranges. The outputs are then summed to create a single output. However, it is difficult to build filters that will sum to a signal that is equal to the input. There should be no audible difference even when the amplitude of each band is made identical. The output has ripples in the frequency response.

The circuit approach shown in *Figure 2* achieves this desired response. State variable filters are used to implement the required transfer functions of each stage. Each state variable filter has simultaneous, bandpass, lowpass and highpass outputs. The bandpass signal is routed to the gain and/or signal processing stage while the low-pass and highpass outputs are summed and then routed to the next-higher frequency bandpass filter. This recursive filtering technique assures that the roll-off charac-

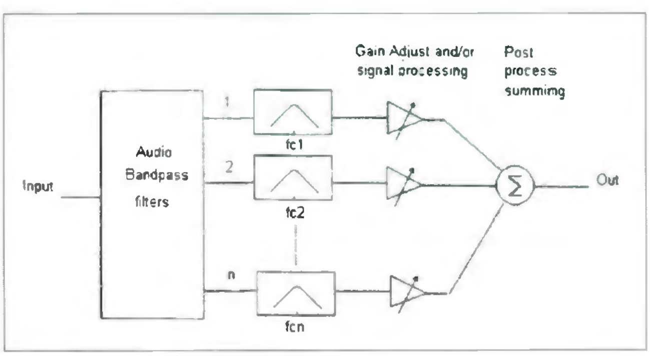

Figure 1. The basic filter cuircuit.

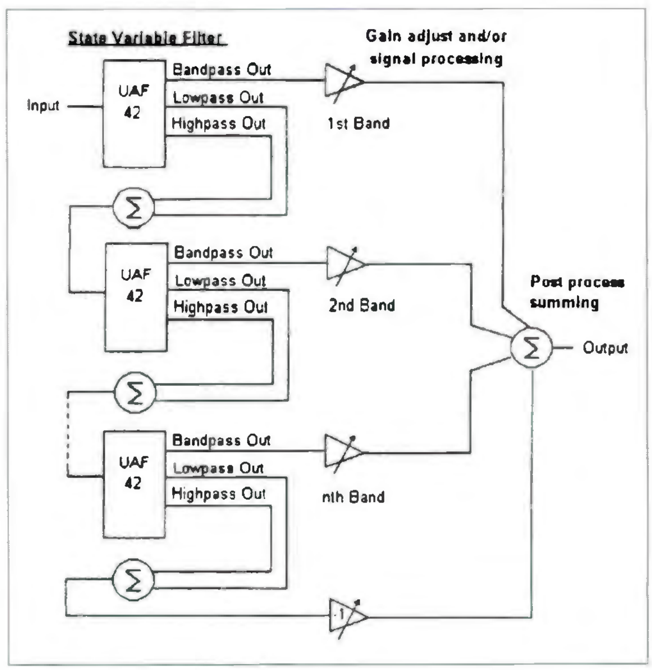

Figure 2. The better filtler circuit.

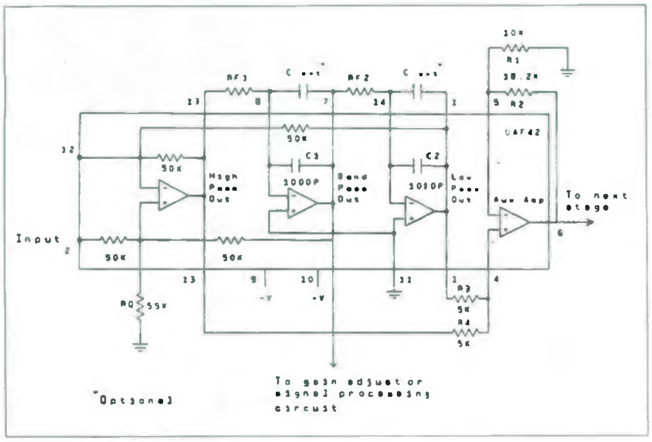

Figure 3. The UAF with externalcomponents.

teristics of one filter ideally "mesh" with the next to produce a smooth frequency response when summed at equal amplitudes.

State variable filter chips like Burr Brown's UAF42 , provide lowpass, bandpass and highpass outputs. Added features like an auxiliary FET input op amp and two on board 0.5 percent 1000 pF integrator capacitors lower the required component list. *Figure* 3 shows a diagram of the UAF42 along with the external components required to realize each stage of this filter.

IMPLEMENTING A DESIGN

A typical design would have the audio frequency range broken up into ten frequency bands starting with an fc of 31.25 Hz and ending at fc=16 kHz (fc=center frequency). Each band is centered one octave beyond the previous one and has a half power bandwidth of fc/1.4. The bandpass out nodes of each UAF42 provides the frequency range of interest which can then be attenuated, gained up, compressed, noise gated or subjected to any desired signal processing. All the bands are then summed back together using a post processing summing circuit.

The lowpass and highpass out nodes are summed and gain adjusted using the internally supplied auxiliary op amp. This amplifier then drives the subsequent stage. The signal from the final summing amplifier is inverted and then routed to the post processing summing amplifier as shown in *Figures* 2 and 5.

CALCULATING COMPONENT VALUES

Figures 3 and 4 show the circuit layout for each individual stage. The center frequency setting resistors RF1 and RF2 can be set using equation 1,

$$(1)\quad RF1 = RF2 = \frac{1}{2\pi\,(1000pF + C_{ext})}\,fc$$

where C_{ext} is an optional externally supplied capacitor. For center frequency designs below 100 Hz, it's recommended that a good NPO ceramic or mica external capacitor be used in order to avoid large RF1, 2 values. The table below shows RF1, 2 values for this circuit. All resistor values are standard 1 percent tolerance available "off the shelf".

fc (Hz)	RF1,2	Cext
31.2	511K	0.01uF
62.5	255K	0.01uF
125	1.27Meg	—
250	634K	—
500	316K	—
1K	158K	—
2K	78.7K	—
4K	40.2K	—
8K	20K	—
16K	10K	—

Note that a seven band version of this circuit can easily be implemented by simply omitting the 31.2, 62.5 and 125 Hz bands.

SETTING THE BANDPASS -3DB FREQUENCY RANGE

The –3dB bandwidth of each stage and the filter Q are related by the following:

$$(2)\quad Q = \frac{fc}{BW\ {-3bd}}$$

For audio graphic equalizer designs where each filter stage center frequency is an octave above that of the previous stage, a Q of 1.4 is recommended. Q setting resistor Rq can be set using equation 3.

$$(3)\quad RQ = \frac{25K}{Q - 1} = 59K \qquad (Q{=}1.4)$$

SUMMING AND GAIN ADJUSTMENTS

The gain and phase of all the signals summed into the post processing summing amplifier should be as close to unity and as free of externally induced phase distortion

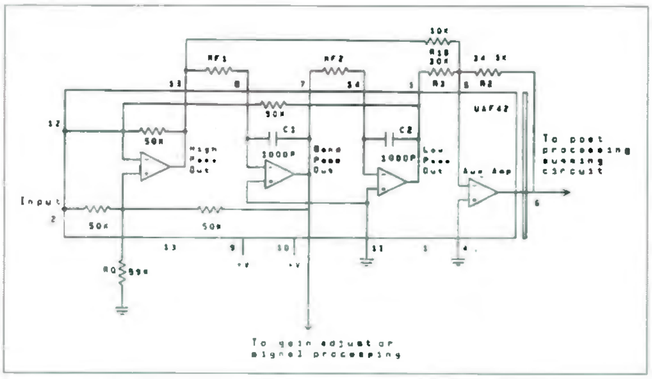

Figure 4. How the Figure 2 circuits can be achieved.

as possible. This is essential for maintaining inaudible differences between the input and output signals when the gain of all frequency bands are set the same. The UAF42 does not provide unity gain response at all three outputs. Gain adjustments must be made. The pass band gain of the highpass and lowpass transfer functions is inversely proportional to the filter Q. That is,

$$(4) \quad A_{lp} = A_{hp} = \frac{1}{Q}$$

where A_{lp} and A_{hp} are the lowpass and highpass gains respectively. Thus for a Q of 1.4, the passband gain at both the high and lowpass outputs is 0.71 V/V. *Figure 3* shows how resistors R3 and R4 sum and couple the lowpass and highpass outputs to the input of the summing and gain adjust (auxiliary) amplifier. This also attenuates the signal by a factor of two. The gain loss due to these effects is adjusted back to unity with R1 and R2.

Note that *Figure 2* shows that the summing and gain adjust circuit of the final stage inverts the signal prior to passing it on to the post processing summing amplifier. *Figure 4* shows how this can be achieved. The auxiliary amplifier, R1, R2 and R1b are used to create an inverting summing amplifier. This is required to compensate for the 180 degree phase inversion that's inherent from input to bandpass out of each UAF42 stage. *Figure 5* shows a more detailed schematic that includes a post processing summing circuit.

CIRCUIT PERFORMANCE

A slightly modified version of this circuit has been implemented by Joseph Brennan, audio engineer at Sky Walker Sound studios in Santa Monica, CA. Brennan designs and implements audio signal processing circuits for major film producers such as Oliver Stone. A seven band version of the above mentioned circuit was recently used on the sound track of Stone's soon-to-be-released *Natural Born Killers*.

Brennan reports that THD+N for the circuit, which included signal processing circuitry along with the filter, was below 0.015 percent. Other tests show the gain ripple from 10 Hz to 20 kHz to be within 0.4 dB. Phase ripple is between ±5 degrees. db

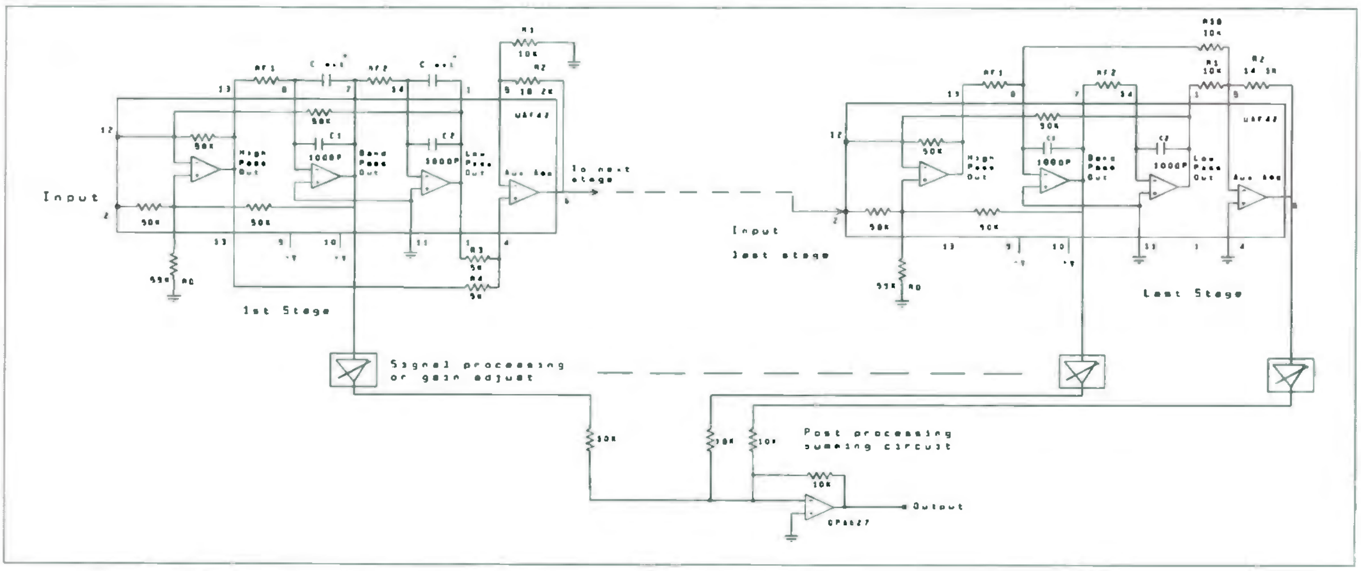

Figure 5. Final UAF42 stage with auxiliary amp used as an inverting summing amplifier.