The Evolution of an EQ Design

By Fred Forssell, Forssell Technologies Inc.

1st draft

This discussion covers the steps used to design a functional multi-band equalizer for use in professional audio applications such as recording, mastering, and sound reinforcement. The aim of this design is to provide EQ adjustments with minimal sonic coloration other than the change in frequency. We are <u>not</u> trying to design an effects device in this discussion.

Given the goal of a sonically neutral device, we need to design this EQ with a minimum number of series connected stages. Most commonly encountered EQ designs use series connected filter sections. Sometimes there is an input buffer amplifier also used in the signal path of such designs. Such a design would result in the number of series connected amplifiers equal to 1+N, where N is the number of independent filter sections. For example, a 4 band EQ using such a design would have 5 series connected amplifiers. It may be obvious to the reader that connecting 5 amplifiers in series will have a greater sonic impact than using a single amplifier in the entire signal path. So the next question becomes, "Is it possible to build a 4 band EQ using one series connected amplifier?". Fortunately for us minimalist designers, the answer is a definite yes!

If you have done much circuit design, you are probably saying to yourself right about now, "OK, what are the trade-off's if I use a one amplifier design." That's a fair question. If the answer to that question is "There are no significant trade-offs to a one amplifier EQ design" then why would anyone design an EQ with 5 series connected amplifiers? While there are trade-offs to the single amplifier design which I consider to be minor, I personally cannot imagine why anyone would use the multiple series amplifier approach. It simply makes no sense to me.

If we ignore the topic of why one would use an input buffer amp for the moment (we will get into this later), the only reason that I could see to use a topology using series connected EQ sections is to make their transfer functions additive. What does that mean? It means that if one were to make all the sections of an EQ design have the same frequency range, and then set each section to the same frequency and gain setting, they would add. If you boosted 1 KHz by 6 dB in each of 4 sections, the result would be 24 dB of gain boost at 1 KHz. While I am open to being shown the error of my thinking, I do not see this as a reason to use this topology. It simply has too great an effect on the sonic quality of the signal passing through the EQ and that is not the style with which I design. It must be said however, that the EQ design presented here will not boost the signal by 24 dB if all 4 of the filter sections with set as described above. It will boost it by 6 dB.

Single amp EQ designs have been around for quite a long time. Many graphic equalizers with octave frequency centers use a single amplifier topology. An excellent example of such a design can be found in National Semiconductor's Audio/Radio Handbook[1] published some 23 years ago. An example of this topology is shown in Figure 1.

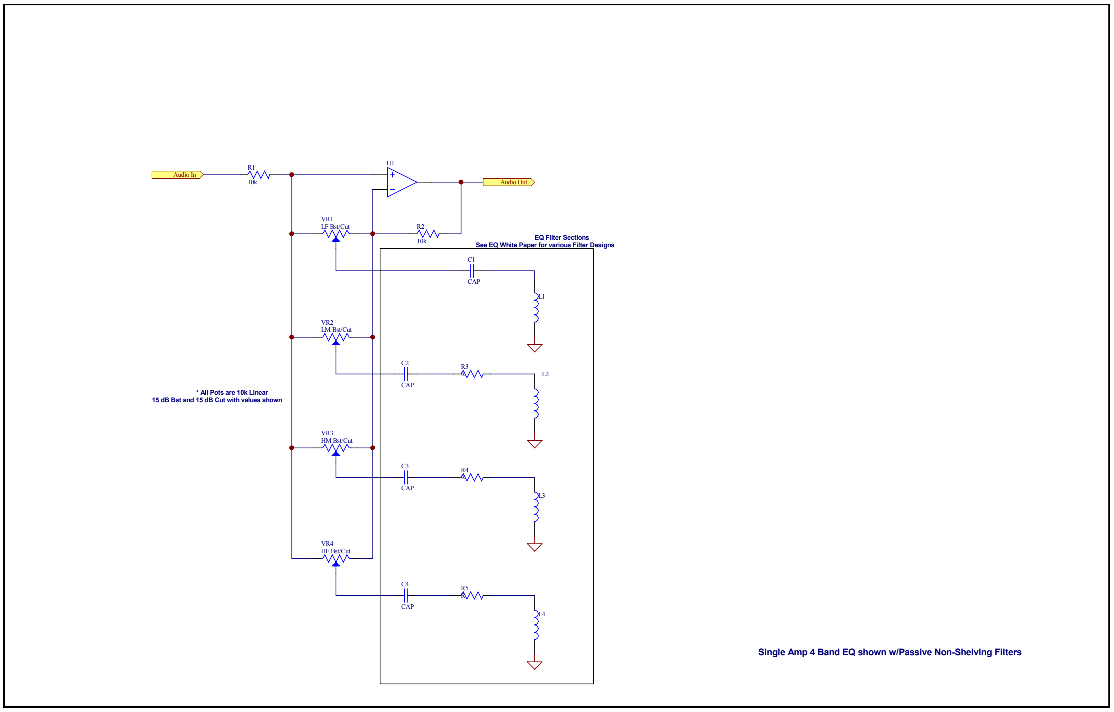

Figure 1
A single amplifier EQ topology

This is the basic circuit that can be used with up to 10 filter sections in parallel. The example shown in Figure 1 uses only 4 filter sections. While it may appear that unlimited parallel filter sections could be added, practical limits to the number of sections exist because of the limited open loop gain of the single amplifier. Ten sections seem to be a good number of sections if you only need low amounts of boost/cut (gain). If you wish to have higher amounts of gain like +/- 15 dB, then 4 sections is a more practical limit.

Without resorting to detailed math, it can be said this topology operates in the following way. When in a "cut" mode, the filter sections form the shunt element in a frequency dependent input attenuator. We all know that it is possible to build an input attenuator by placing a resistor in series with the signal path (Rseries), and then placing a shunt resistor from the downstream side of this resistor to ground (Rshunt). Making these two resistors equal in value will result in a 6 dB loss across Rseries. If we could make Rshunt only have its resistance value at a specific frequency instead of the entire audio passband, then we would have a "dip" filter with 6 dB of cut at whatever frequency it was set for. This, in essence, if what happens with the LC filter sections shown above.

So think of this circuit as being a frequency dependent passive attenuator with an output buffer amplifier. The buffer amplifier provides low output impedance for the EQ to drive the outside world properly.

We now know how to accomplish the "cut" mode of the EQ, but how do we get the "boost" mode? Think for a second about the effects of a negative feedback network around a non-inverting amplifier. This network has a series connected resistor from the output to the inverting input, and a shunt resistor from the inverting input to ground. This creates the exact same attenuator configuration that we discussed above, but with the opposite results. The more signal we drop across the Rseries in the feedback loop, the higher the gain of the amplifier (because of less negative feedback). If we make the feedback network's Rshunt be a frequency dependent resistor, as we did in the "cut" mode example, then we have less feedback at the frequency to which we have set the network resulting in a boost at that frequency. Pretty simple isn't it. Simple, and in my opinion, quite elegant.

To summarize the operation of the topology shown in Figure 1; In "cut" mode the filters act as frequency dependent shunt resistors forming a passive input attenuator. In boost mode, the filters act as frequency dependent shunt resistors in the negative feedback network of a non-inverting amplifier, varying the amount of negative feedback at specific frequencies. In either mode, the opamp effectively acts as a buffer amplifier providing low output impedance to drive the outside world. The input impedance is effectively equal to the value of the series resistor at the input plus the parallel values of the gain pots used. If we use a 10k series resistor and 4 10k pots (as Figure 1 does), the input impedance is a bit greater than 10k. The actual input impedance will vary depending on the setting of the gain pots, but it will never fall below 10k ohms. Making the input and feedback series resistors have the same value results in equal amounts of boost and cut gain. The input configuration circuit in Figure 1 is single-ended, not balanced.

If the designer is concerned about having balanced inputs, or wished to have either an input impedance greater than10k, or wanted the input impedance held constant, then a buffer amplifier could be used in front of the circuit shown in Figure 1. The trade-off is, of course, this places an additional amplifier in the series connected signal path. This trade-off is often justified because it creates a more hum-free interface in studio environments where lots of different patching options occur.

Filter Designs

The filter design shown in Figure 1 uses inductors "L" and capacitors "C" to achieve a bandpass filter response. Removing either the L or the C result in a shelving response. At the resonate frequency of an LC filter network, the impedance approaches zero. The frequency of resonance is equal to….

$$Fo = \frac{1}{2\pi(LC)^{1/2}}$$

If one were to use an ideal L and C (which do not exist), then the Q of the filter would be infinite, creating a filter with infinitely steep sides. Said another way, the filter would only have an effect at the frequency of resonance and do nothing at any other frequency. That's simply not possible in the real world. Even if it were, it would not result in a very practical EQ design for use on audio signals. Most of the time we want our audio EQ to have a fairly broadband. The bandwidth of a bandpass filter such as those shown in Figure 1 is usually expressed as a numeric value called "Q". Q is also known as the

"Quality factor". This is an old term referring to the "quality" of the L and C. The better the quality, the higher the Q factor.

In bandpass filter usage Q is a number equal to resonate frequency divided by the bandwidth of the filter or…

$$Q = \frac{Fo}{BW}$$

The bandwidth of the filter is measured at the – 3 dB points on either side of the resonate frequency. So, a 1 KHz filter with – 3 dB points at 500 Hz and 1.5 KHz has a bandwidth of 1 KHz and a Q of 1 or…

$$Q = \frac{1000}{1000} = 1$$

A 1 KHz filter with – 3 dB points at 300 Hz and 1.7 KHz has a Q of 0.714 or …

$$Q = \frac{1000}{1400} = 0.714$$

By selecting the resonate point of the C and L, we can obtain a desired Q within the limits of the quality of the actual L and C used. The highest Q obtainable is usually limited by the quality of the inductor because it is much more difficult to build an inductor that approaches ideal than it is a capacitor. The actual Q of a given LC filter can be adjusted by inserting a resistor in series between the C and the L. This resistor also serves to create the same Q for all of the filter sections, something that should be addressed if you build a passive EQ such as that shown in Figure 1. As the filter frequency drops, the inductor size increases. Larger inductors have higher DC resistance than smaller inductors, everything else remaining the same. So compared to the highest frequency filter, the lowest frequency filter will have a lower Q (wider bandwidth) due to the higher DC resistance of the inductor. Adjusting the value of each filters series R will allow the designer to match the Qs for all of the sections. Adding this resistor will also affect the amount of boost/cut that is possible. While it is possible to have a switched R (or a pot) for the series resistor, using such a variable R as an variable Q adjustment (like a Pultec EQ) will result in changes in boost/cut as well as Q. In my opinion, a good EQ design with a variable Q will not allow much (or any) gain change as the Q is varied. The GML 8200 EQ design is a good example of an EQ design that allow the Q to be varied while the amount of boost/cut remains constant.

By far the biggest battle facing a designer of the EQ shown in Figure 1 will be in obtaining good inductors, particularly for the low frequency sections. All inductors will have a self-resonate point. This is a point at which the inductance and inter-winding capacitance create a resonate frequency within the inductor itself. It can be difficult to find high value inductors to be used in low frequency filters that have self-resonate points that are significantly higher that the 20 KHz upper limit of the audio passband. If the self-resonate point falls within the audio passband, boost/cut gain settings for the LF section will also occur at the self-resonate frequency. This is obviously undesirable. Other limitations of real-world inductors relate to the core characteristics such as core saturation (a real problem at LF), other core induced non-linearity, and shielding problems. While it is possible to find people who still know how to wind high quality high inductance coils, it is becoming a lost art and at any rate, very expensive to do correctly. Most designers opt for a way to simulate large inductors instead of having them wound. It is possible, cost effective, and easy to obtain the smaller inductors required for the higher frequencies of an audio EQ design.

So we can assume from the last paragraph that it is possible to design and build circuits that simulate inductors, thereby eliminating the need for wound inductors. There are two basic circuit topologies that do mimic inductors— the negative impedance converter (NIC) and gyrators. NIC are ruled out of our EQ design because they do not exhibit all the correct properties of an inductor (they have an inverse frequency to impedance curves) while the gyrator does exhibit the correct properties of an inductor.

There are three different gyrator topologies that I know of, and possible more. One is too complicated (it uses two opamps and 5 resistors), while the other two circuits use a single opamp. Feeling that simpler is almost always better for audio circuits, I prefer the single opamp approach. As I said there are two topologies that use a single opamp. In experiments that I have done with both these circuits, I have found that one has a distinct advantage over the other. I therefore only use one topology of single opamp gyrator and it is shown in Figure 2.

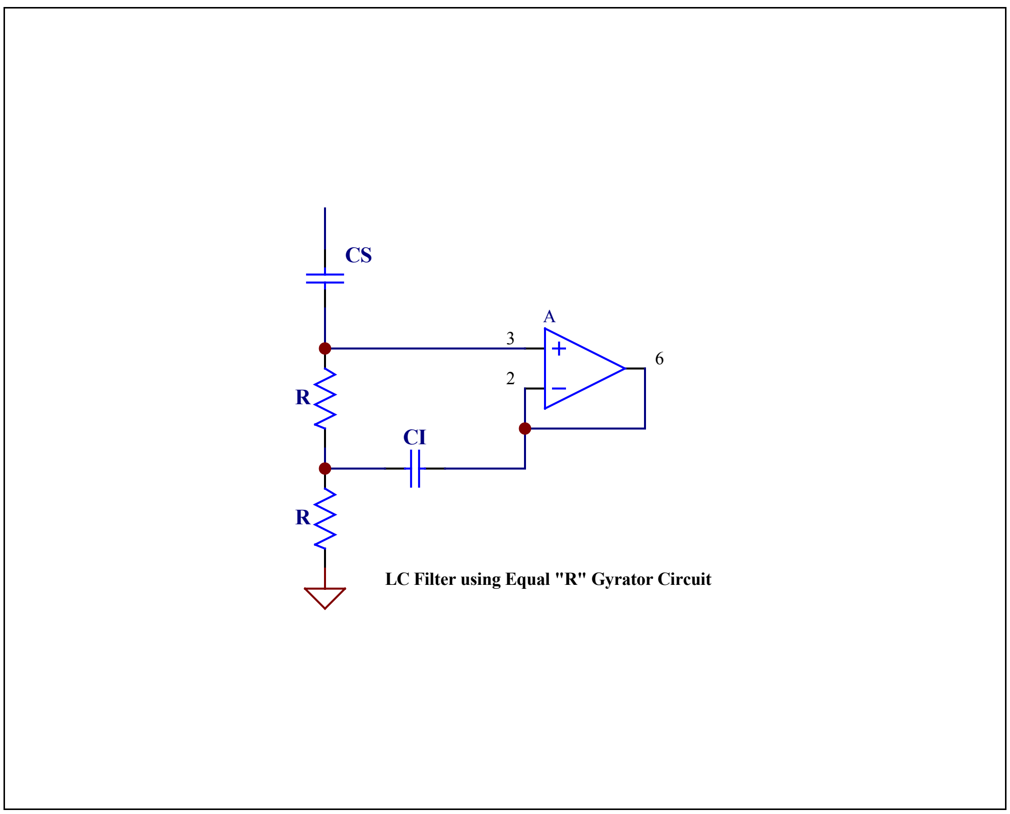

Figure 2
A single opamp gyrator

The following equations are used to obtain the circuit values for this gyrator.

$$Fo = \frac{1}{2\pi R\sqrt{CsCi}}$$

$$Q = \frac{1}{2}\sqrt{\frac{Ci}{Cs}}$$

$$Loss = \frac{2R}{Rseries + 2R}$$

Using 10k Rseries resistors on both the Input Rseries and the feedback Rseries, 1k ohm resistors for R (there are two of them) in the gyrators, and 10k linear pots, you will get approximately 18 dB of boost/cut for your EQ design.

If you look closely at the opamp shown in Figure 2, you will see that it is connected as a voltage follower with a gain of 1 (unity). You can therefore use any circuit that is non-inverting unity gain with a high input impedance (JFET's work well), has low output impedance, and little or no DC offset. Discrete JFET source followers, vacuum tube cathode followers (with a large output coupling cap) can all be used. High linearity, wide bandwidth, good stability, low distortion, and high headroom are all desirable features for a gyrator's active circuitry.

Eliminating CS (shorting it) will result in a low frequency shelving circuit. Eliminating CI (lifting it at one end) results in a high frequency shelving circuit. Adding an R in series after CS results in a lower Q (getting lower as this R increases) and lower gain (boost/cut). To change frequency of a filter you can use a double pole switch to switch in different values of CS and CI. Make sure that you use a shorting switch (rotary) and include 1 meg ohm resistors from each caps switch contact to ground so it doesn't pop when you change frequencies.

At this point, it my be obvious that the LC bandpass circuit is good for making peak/shelf filter sections, but not so good for making a parametric filter section. While there are many good state-variable and biquad filter topologies that can be used to build parametric filters, few are useable with the single EQ amplifier circuit that we are discussing. Those of you paying attention will not that I said few, not none. There are a circuit topologies that are useable as parametric filter section with our single amplifier topology. The one that I'd like to discuss here is called the "Constant Amplitude Phase Shift" or CAPS circuit. Figure 3 shows the caps circuit.

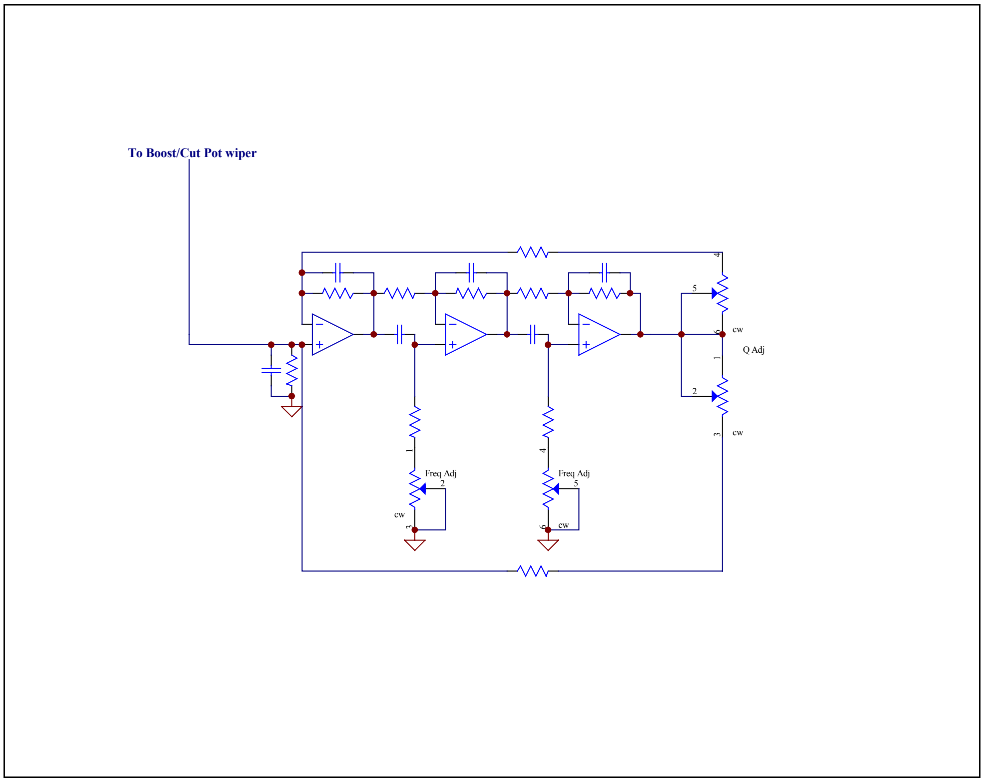

Figure 3
Constant Amplitude Phase Shift (CAPS) circuit

The CAPS topology was first presented (to my knowledge) by Mr. Steve Dove is a series of article printed in the UK's Studio Sound magazine[2], which covered the design of a large format console. Mr. Dove's excellent article went into great detail on the EQ section of console design. He discussed several different filter topology as well as the advantages and disadvantages of each. He then presents the design of the CAPS circuit in modest detail. The design presented in the article uses more than one EQ amplifier, but each amp is used in the same configuration as our Figure 1 design. Therefore, we can replace one or more of the passive filter sections (or gyrator filter sections) with Mr. Dove's CAPS design.

Mr. Dove's CAPS design uses two series connected all-pass filters with a buffer amp in front of them. Positive and negative feedback are wrapped around the entire series connected signal path. By manipulating the feedback values the Q of the circuit can be changed without altering the gain of the filter. The amount by which the gain changes or doesn't change is a function of selecting the value of positive and negative feedback resistor values. If a dual pot is used as a Q control, then the matching of the two sections and their inverse curves will dictate how constant the gain is while varying the Q over its desired range.

There are other all-pass circuit configurations, some of which can be configured a bandpass filters and/or shelving filters, but we'll have to save that discussion of another time.

An analytical look at the CAPS circuit does not immediately reveal how it works. With our LC bandpass filter it was fairly intuitive that the LC circuit presented a low impedance at filter's resonate point, and that this low impedance working in conjunction with series connected resistors (either at the input or in the negative feedback loop) to produce the desired effect. How the CAPS circuit works is less intuitive. In a nutshell, what the CAPS circuit does is produce a phase shift at the desired frequency (and with the desired bandwidth or Q) that is summed with either the input signal or the feedback signal. Think about what would happen if the phase is 180 from that of the input or feedback signal; it cancels it out, effectively attenuating it. That is (very basically) how the CAPS circuit works. However with the CAPS circuit, the two all-pass filters introduce a 90 degree phase shift at a frequency set by the RC value network connected to the non-inverting amplifier in each all-pass network. Because there are two these filters, the second all-pass filter inverts the phase shift of the previous stage. The results in a +90 and – 90 degree phase shift at the frequency to which the filters are set. That is exactly what we need for a bandpass filter. The first opamp in the series of three "closes the loop" around the filter, allowing both positive and negative feedback. This allows the damping (Q) and the gain to be adjusted in a manner that allows low gain change as the Q is adjusted. The rest of the details of the CAPS circuit can be obtained from Mr. Dove's excellent article series.

Opamps

All opamps used in the filter section and the EQ amplifier itself should be unity gain stable designs. Further, wide bandwidth, JFET input stages, high headroom (high voltage supply rails) and low output impedance are desirable traits for all of the amplifiers used in such a design**.**

That said, if you are making gyrator-based filters, I encourage you to explore voltage follower circuits other than those using IC opamps as active elements. If space and/or time is a consideration, then IC opamp based gyrators can be used with good results if you are careful to use really good quality IC amps such as Burr-Brown OPA604 or Analog Device AD825 devices. The 604 series opamps offer an advantage in that they can use bipolar voltages up to 24 VDC. Discrete Class A opamps and discrete follower circuit will also work well if you are careful to address DC offset issues.

High open loop gain opamps of either discrete components or IC types can be used for the EQ amplifier. If the open loop gain of the EQ amplifier only moderately high (less than 100) you may see asymmetrical boost/cut curves, with the boost curve not matching the maximum gain and slope of the cut curves. Remember from our discussion above that the cut curves in this EQ are passive cut (an input attenuator) with a unity gain buffer amp. The boost curve relies on changes to the feedback factor of the buffering opamp. Because the total gain of a non-inverting amplifier is a function of the open loop gain of the amplifier and the feedback factor, the actual amount of maximum boost is also a function of the open loop gain of the amplifier. Further, high frequency distortion will generally be higher than lower frequency in the audio passband if a wideband amplifier is not used. The effect will get worse when high frequencies are boosted because negative feedback around the amplifier is being reduced.

References

1. Audio/Radio Handbook National Semiconductor Corp, Section 2.17 Octave Equalizers pp 2.56-2.61
2. Steve Dove, "Designing a professional mixing console, parts 7,8,9" Studio Sound Magazine, April, May, June, 1981