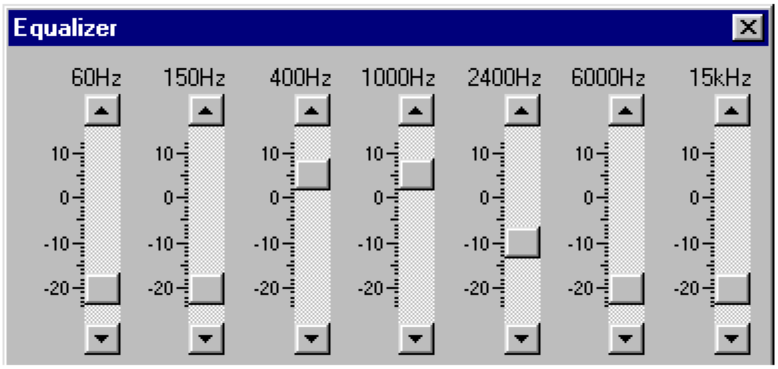
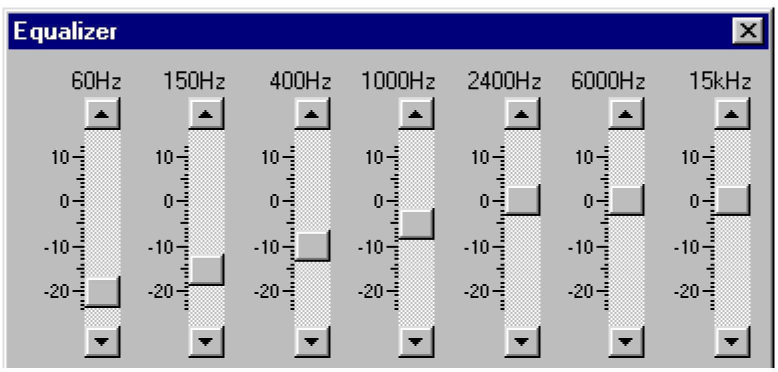
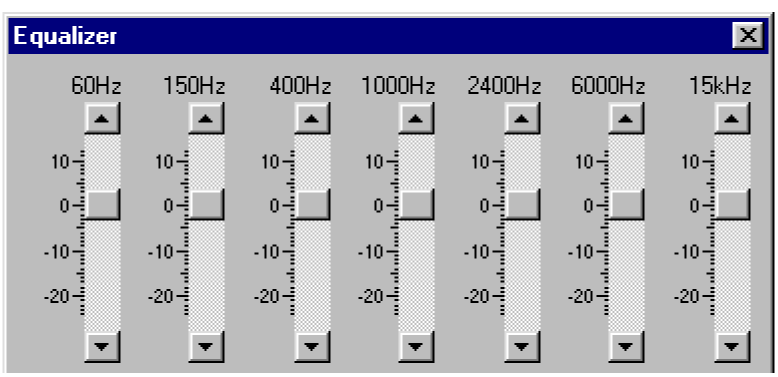
O. Signal Processing







The author's father once told the author around the late 1950s that no tape or recording could be better than the original. Something is lost in the recording. Nothing is perfect. There is no such thing as perfect high-fidelity reproduction. However, the author's father did not live to see the 1980s and beyond, where signal processing challenges this assertion. Here is a specific example. Suppose the original lacks reverb and is too weak in the bass. Then we correct these deficiencies in the processing stage before making the master. The recording is enhanced! You process the signal at home when you play around with bass and treble controls. If you have an equalizer (to be discussed), you process your signal even more.

Philosophically, the original is not reproduced exactly. However, we can get "garbage" along with the good stuff. If we can eliminate some of the "garbage" during the intermediate processing stage, then can we say that the final product is better? The debate over whether a recording is "better" than the original may remind you about our earlier discussion concerning the existence of sound when a tree falls in a forest.

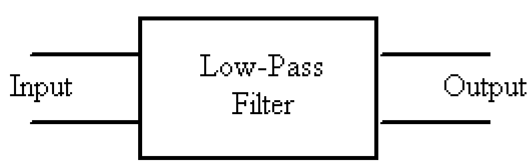
But suppose the source is a digital synthesizer or computer. The information is encoded as 1s and 0s on a diskette. If we copy the diskette, the copy is indeed an exact reproduction of the original! And suppose a digital recording of an artist who hits 5 wrong notes is made and then the mistakes are corrected during the digital-processing stage? The author is sure that his dad did not have these tricks in mind when he made his statement during the early days of stereo and tape recording.

Filters

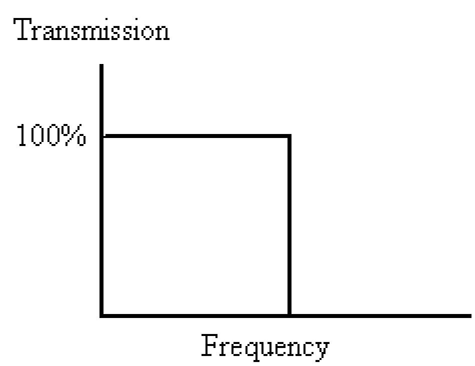
One of the essential features of signal processing is the use of filters. Filters can reduce frequencies that are not as important as others. An *active filter* boosts frequencies transmitted by combining a filter with a dedicated amplifier for the filter. A filter without combined amplifier assistance is called a *passive filter*.

Fig. O-1 illustrates an ideal filter that filters out high frequencies beyond a certain point. The graph at the right gives the percentage of the amplitude that gets transmitted for each frequency. Real filters do not cut off so abruptly.

Fig. O-1. Ideal Low-Pass Filter.



Input sine waves of different frequencies.
See what comes out.



We send in sine waves of different frequencies and see how much of the original wave comes out. We find that 100% of the wave gets through if its frequency is low. If the frequency is higher than a certain frequency, we get nothing (0% transmitted). The filter is ideal. Since actual filters are not "cliffs" that drop off so perfectly, we would find that some intermediate sine waves get partially transmitted. However, when we get to high enough frequencies, they would not get through at all.

Fig. O-2 presents us with a specific ideal filter (labeled with numbers) that transmits low frequencies. We say the filter is a *low-pass filter* because it "passes" low frequencies. Note that sine waves less than 275 Hz get transmitted and sine waves beyond 275 Hz cannot pass through the filter. This boundary value of 275 Hz is called the *cutoff* frequency. Sine waves with frequencies beyond 275 Hz get "cut off." Fig. O-2 analyzes what happens to an incoming 50-Hz ramp wave. Note that the filter rule applies to sine waves. So we need to decompose the ramp wave into sine waves for analysis (lower left diagram). The first harmonic is 50 Hz, the second 100 Hz, and so on. The relative amplitudes we know from our earlier chapter on Fourier analysis. All harmonics below 275 Hz pass through the filter (see lower right diagram).

Fig. O-2. Sending a Ramp Wave Through An Ideal Low-Pass Filter.

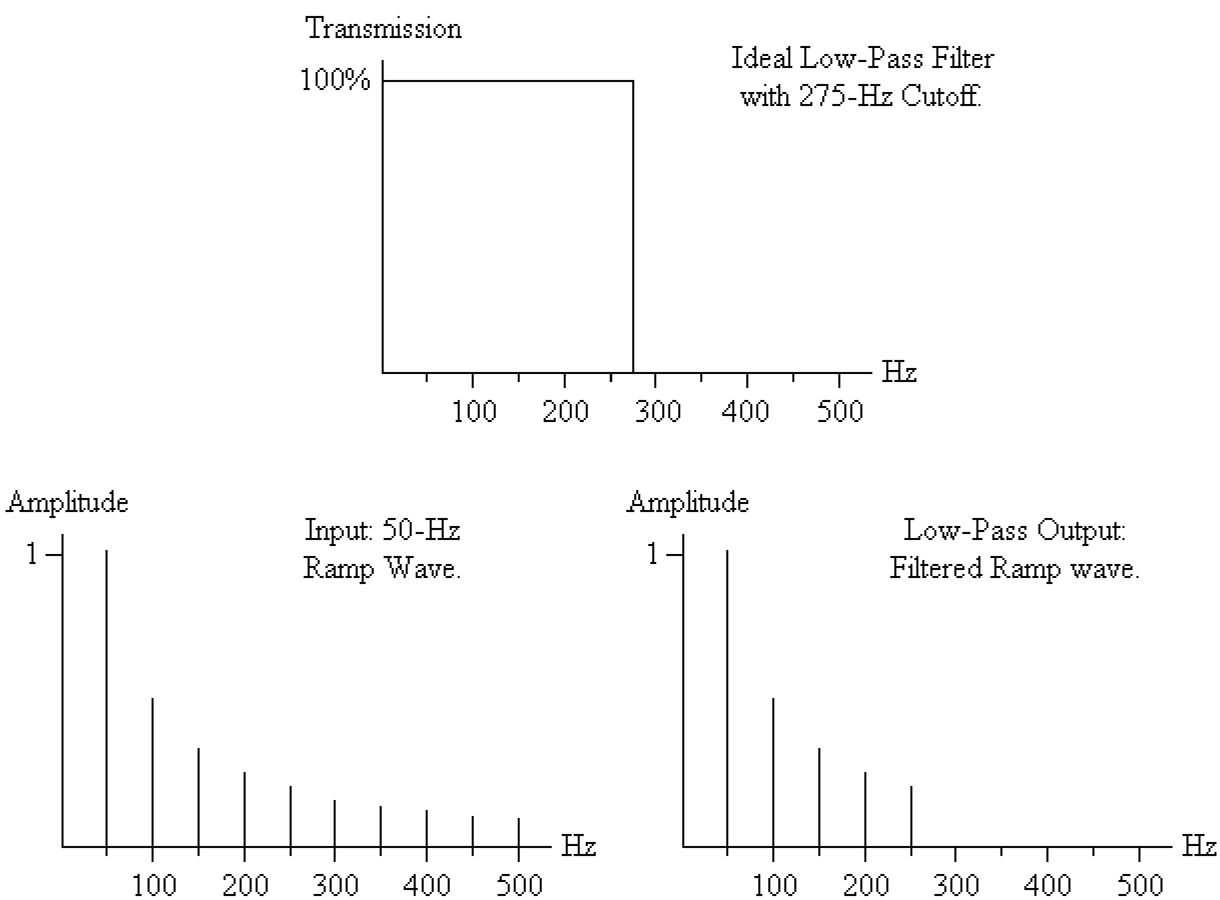
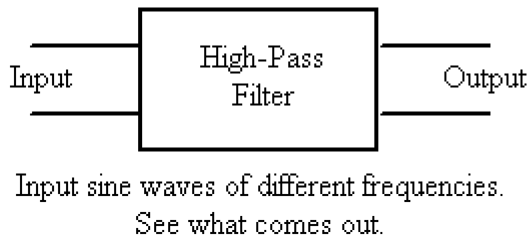


Fig. O-3 illustrates an ideal *high-pass filter*. This filter passes high frequencies instead of low frequencies as we saw earlier. Once again, we send in sine waves to determine the transmission rule for the filter.

Fig. O-3. Ideal High-Pass Filter.



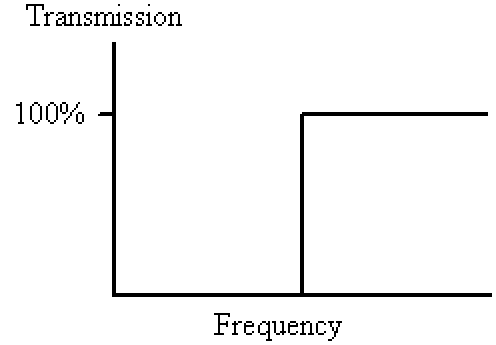


Fig. O-4 gives a specific example of an ideal high-pass filter using our ramp wave again. This time, the higher harmonics of the ramp wave pass through.

Fig. O-4. Sending a Ramp Wave Through An Ideal High-Pass Filter.

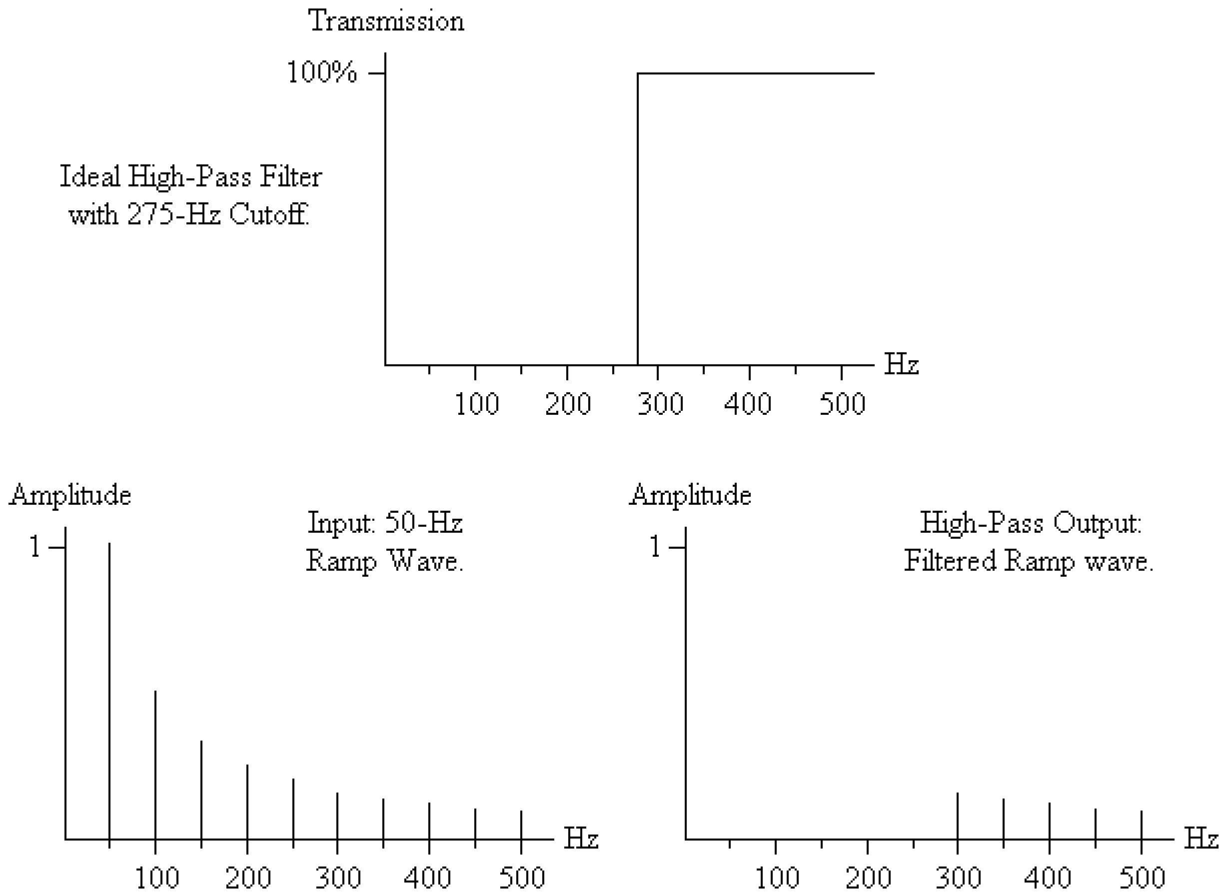
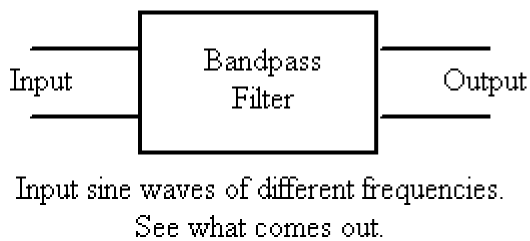


Fig. O-5 illustrates an ideal *bandpass filter*. This filter passes middle frequencies. We have a low cutoff and a high cutoff.

Fig. O-5. Ideal Bandpass Filter.



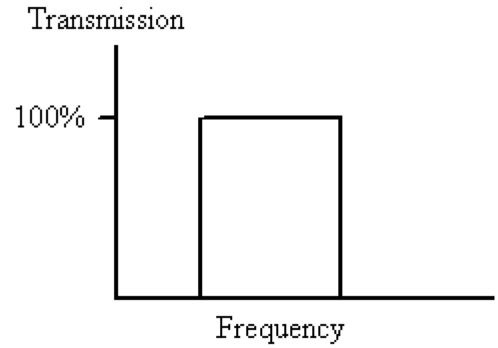
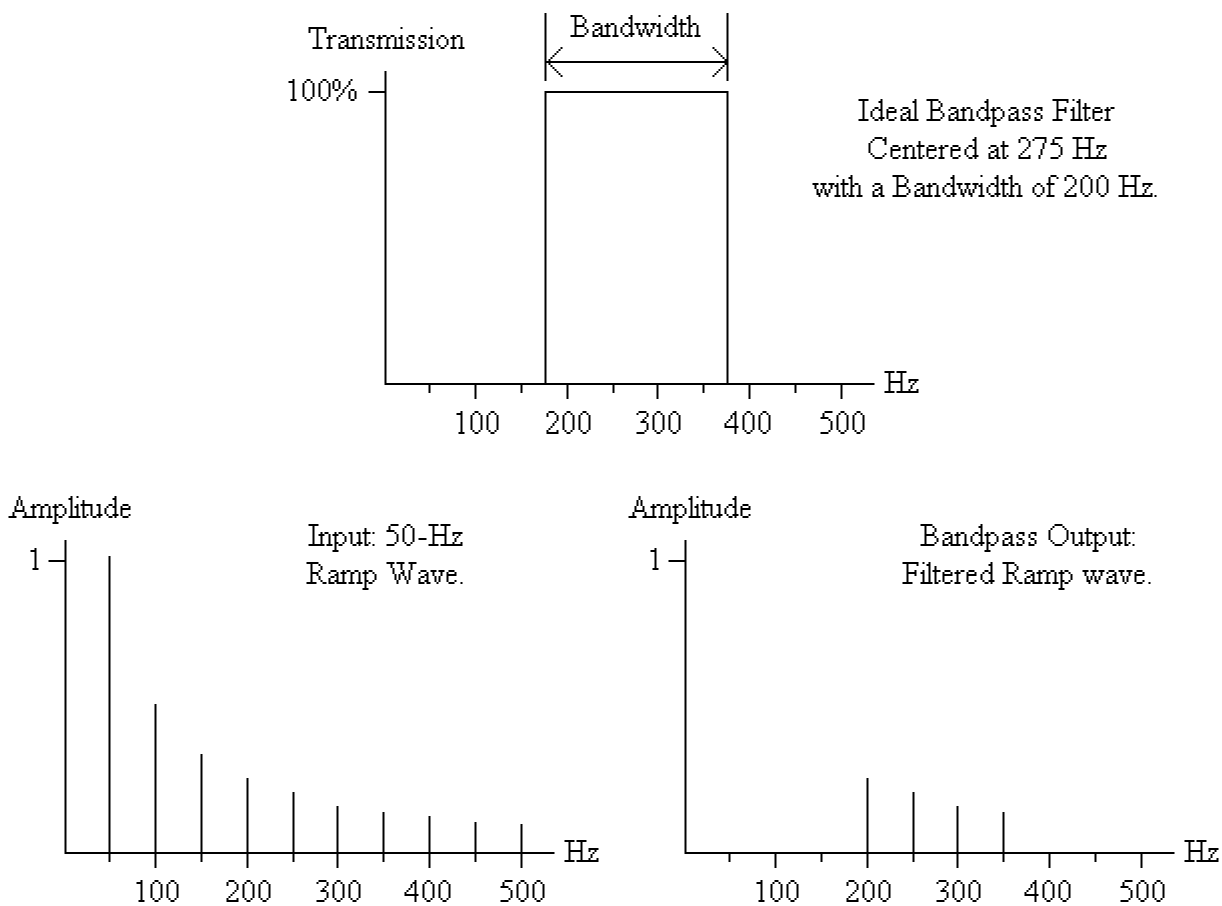


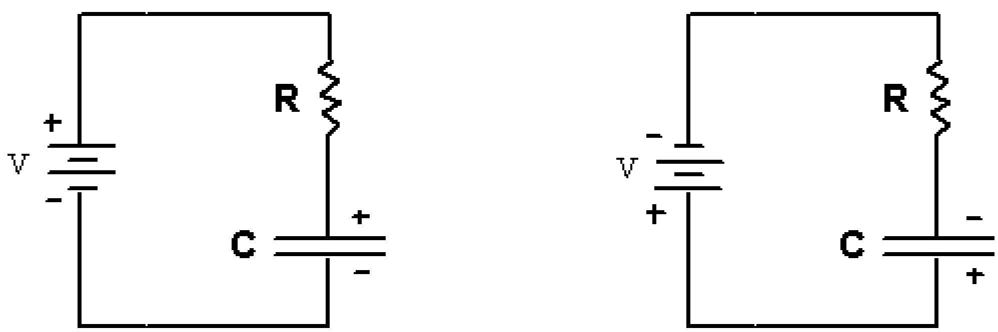
Fig. O-6 gives a specific filter. It is convenient to refer to the bandpass filter by its central frequency and then give the width of the filter (the *bandwidth*). The filter in Fig. O-6 is centered on 275 Hz and it has a 200-Hz bandwidth (175 to 375). It passes frequencies within this band.

Fig. O-6. Sending a Ramp Wave Through An Ideal Bandpass Filter.



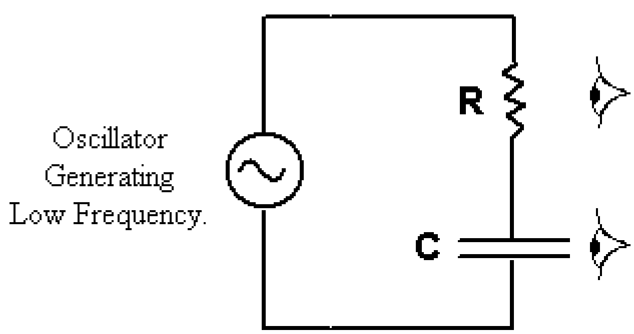
A simple filter can be made with one resistor and one capacitor (RC circuit). To understand why, we review the concept of charging a capacitor. In Fig. O-7 we charge a capacitor two ways. In the left case, electrons (negative) gather on the bottom plate of the capacitor as they try to get to the positive side. The absence of electrons at the top makes the upper plate positive. The battery is reversed in the right case so the charge on the capacitor is reversed. A key idea is that it takes time to charge the capacitor in either case because of the resistor. Now imagine flipping the battery orientation back and forth. This is basically what the oscillator does in Fig. O-8.

Fig. O-7. Charging a Capacitor.



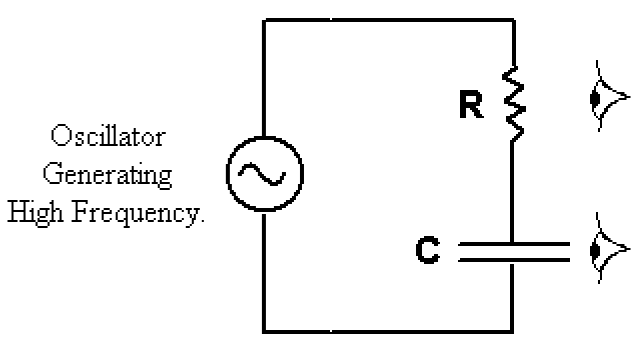
Plus on Top Capacitor Plate. Plus on Bottom Capacitor Plate.

Fig. O-8. Sending an Oscillating Signal into an RC Circuit.



Changes are so slow that very little current ever flows through R. This small I means small voltage signal ($V = IR$) across R.

The low frequency gives the capacitor time to charge one way and then the other. The signal is seen at the capacitor.



The rapid surges in current are seen at R but charge never gets a chance to collect on the capacitor.

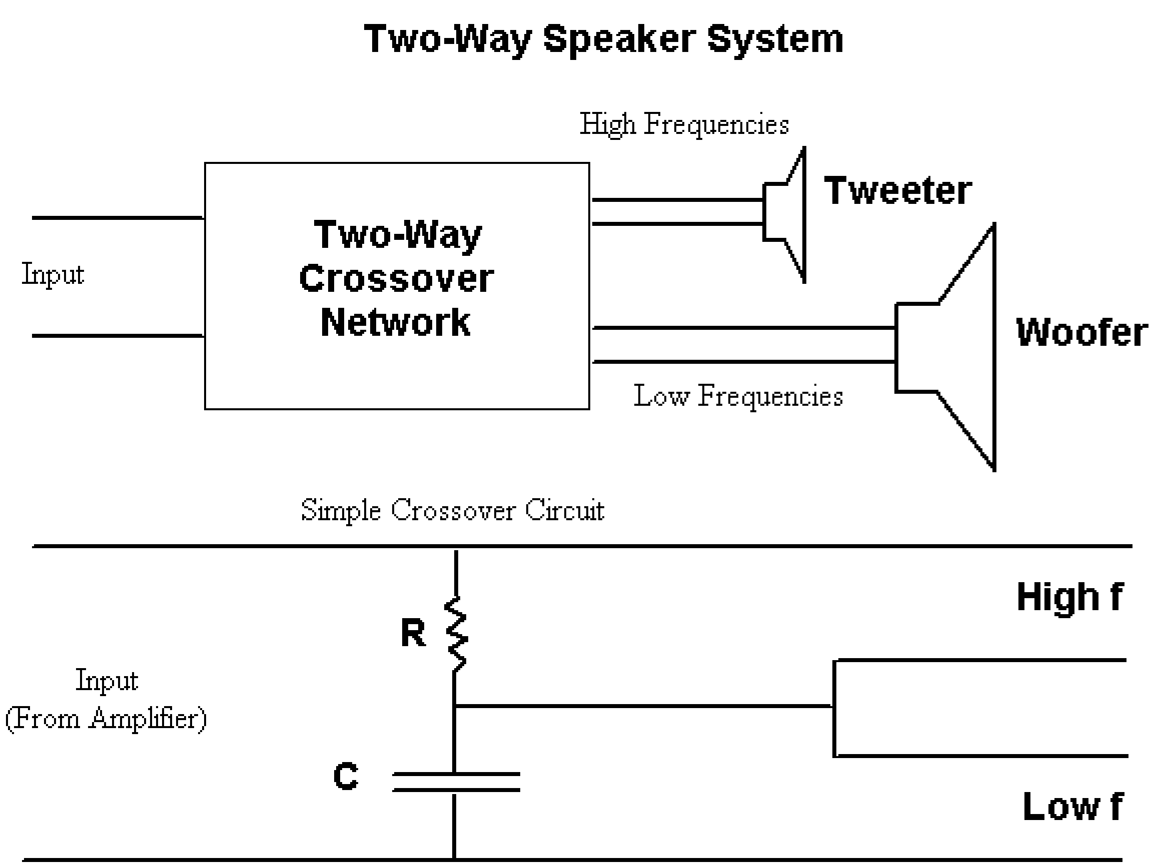
Rapid changes do not allow the capacitor to charge. The signal is not seen at the capacitor.

Carefully study the comments in Fig. O-8. From these, we see that the low-frequency oscillations are directed to the capacitor, while the high-frequency oscillations are picked up at the resistor. The RC circuit splits the frequencies. By choosing specific numerical values for the resistance and the capacitor, the frequency where the division occurs can be chosen.

In a two-speaker unit, this simple RC circuit can be used to direct low frequencies to the larger bass speaker and send the high frequencies to the smaller speaker. This circuit is called a *crossover circuit*. The low frequencies "cross over" to the bass speaker, called a *woofer*. Remember this by thinking that "woof-woof" is a low-frequency sound. The woofer is large and can support longer wavelengths. The high frequencies "cross over" to the smaller speaker, which is called a *tweeter*. Remember this by thinking that "tweet-tweet" is a high-frequency sound. The tweeter better supports short wavelengths. The size of a vibrating system is related to the wavelength.

Long strings produce bass, short ones high pitches. Similarly, large speaker membranes can support long wavelengths very well. Small speakers handle the short-wavelength high tones better. Fig. O-9 illustrates a crossover network which routes the low and high frequencies to the appropriate speakers. Compare the lower diagram in Fig. O-9 with Fig. O-8.

Fig. O-9. Crossover Network for Two-Speaker System.



We are now ready to give RC circuit diagrams for the low-pass and high-pass filters. Our *crossover network* can serve as a guide. Fig. O-10, reproduces our *crossover network*. To make a low-pass filter, focus on the lower section with the capacitor. Throw out the wire at the upper right of the resistor. Then bend the resistor down to the left and straighten out the wire to its left. The result is the low-pass filter in Fig. O-11. For the high-pass filter, discard the wire to the lower right of the capacitor. Bend the capacitor up and straighten out the wire to its left. Then flip the whole thing from top to bottom to obtain the circuit in Fig. O-11. For alternating signals, it doesn't matter that the circuit is flipped from top to bottom.

Fig. O-10. Basis For Low-Pass and High-Pass Filters.

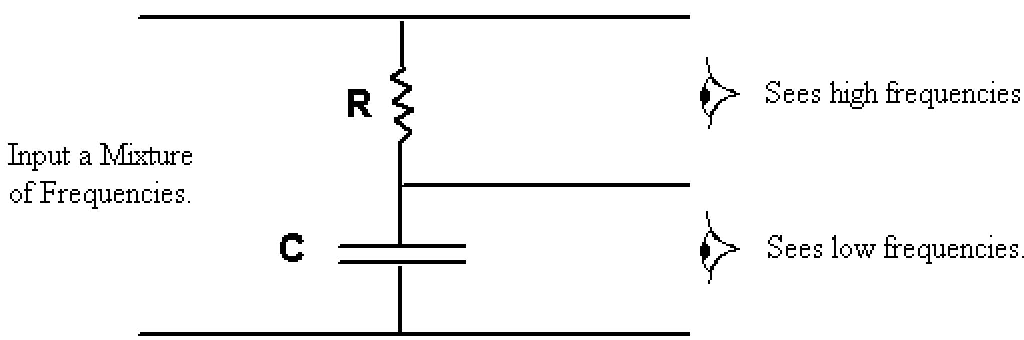
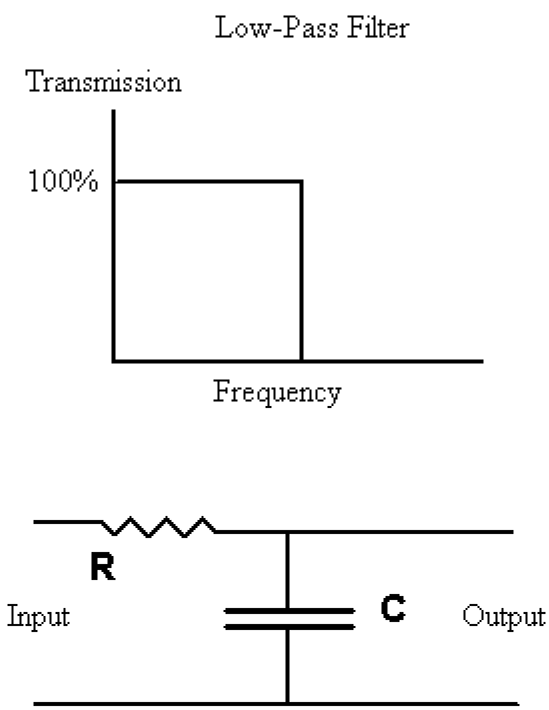
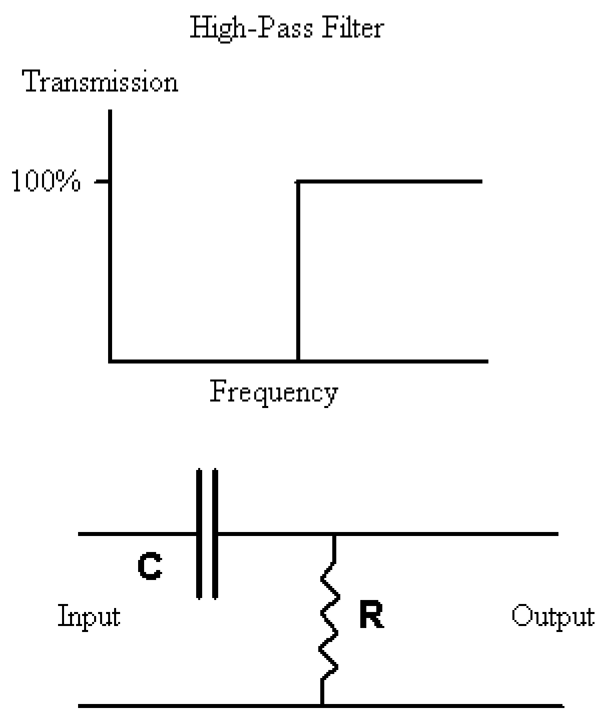


Fig. O-11. Low-Pass and High-Pass Filters.





We can obtain a bandpass filter by combining a low-pass filter with a high-pass filter provided that there is common overlap between the two. See Figs. O-12 and O-13.

Fig. O-12. Low-Pass and High-Pass Filter to be Combined to Form a Bandpass Filter.

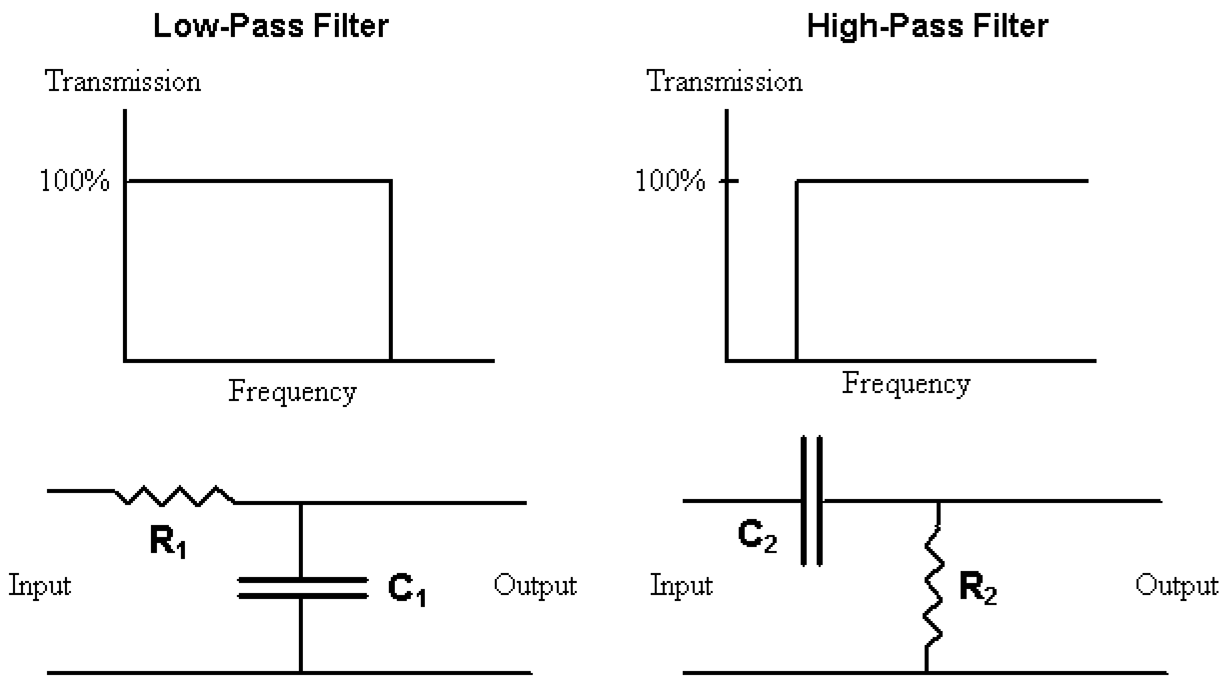
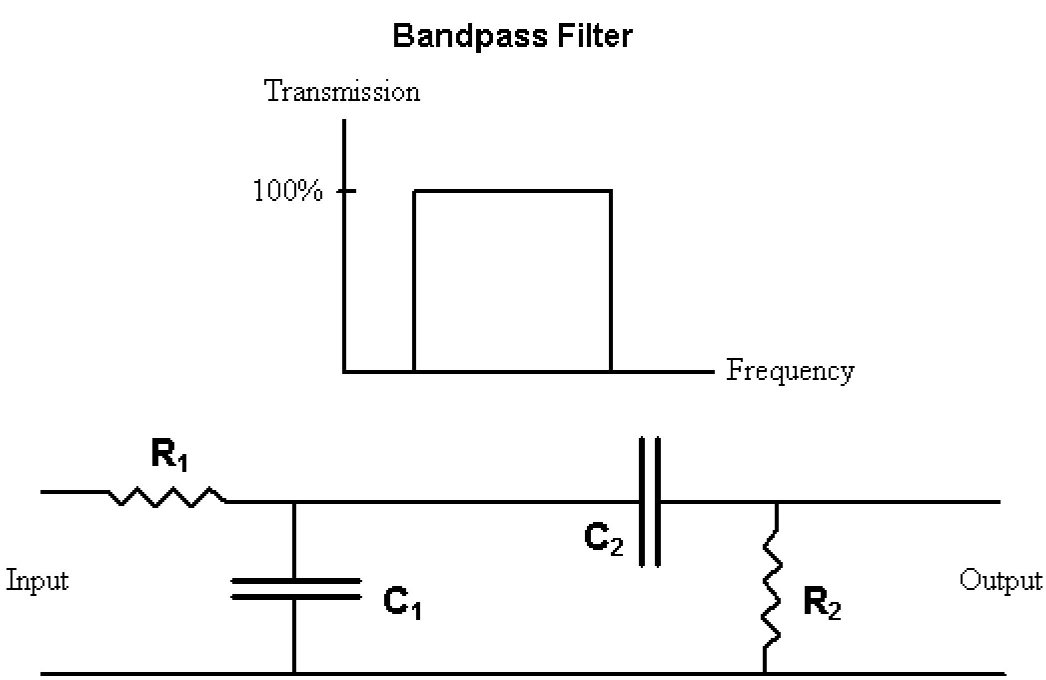


Fig. O-13. Bandpass Filter Made by Combining Above Filters.



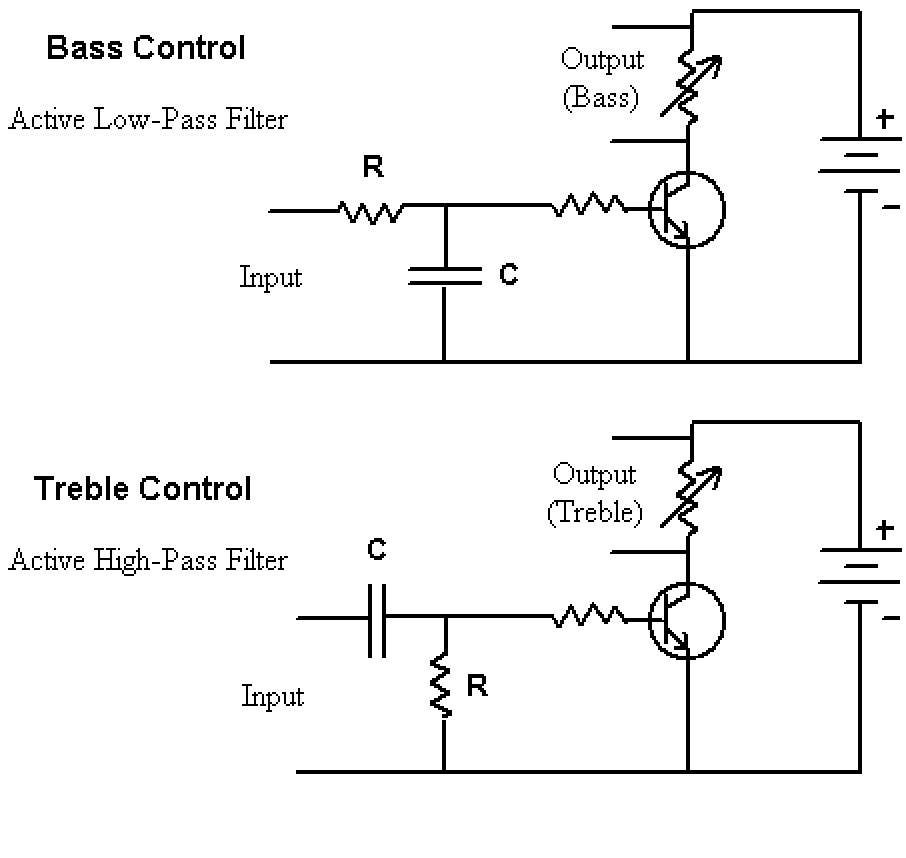
Equalizers

Equalizers are filter circuits that enable us to achieve a more "equal" sound in our homes or elsewhere. The equalizer can help to correct for deficiencies in the original sound or correct for less-than-ideal room acoustics in our homes or cars. The simplest filter circuits that offer some control are the bass and treble controls. The *bass* control enables us to modify the lower-frequency half of the audio spectrum while the *treble* control allows us to adjust the higher-frequency half.

A bass control and treble control are found in Fig. O-14. The same input audio signal is sent through each of these filters. The output of each filter is sent to an amplifier to boost the filtered output. This makes the filter an active filter. The RC-circuit alone is a passive filter, one that filters without any subsequent amplification as part of the filtering process. Active filters amplify the filtered output as this is more desirable. It gives us control of the volume (loudness).

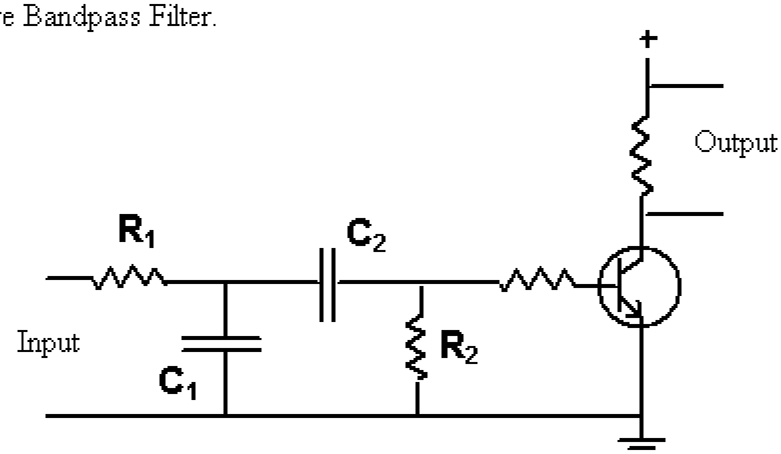
The amplifier unit has a control that adjusts the volume overall. Amplifiers usually have bass and treble controls. These respectively provide control over the lower and higher frequency regions of the spectrum. We can boost the bass or treble independently.

Fig. O-14. Bass and Treble Controls.



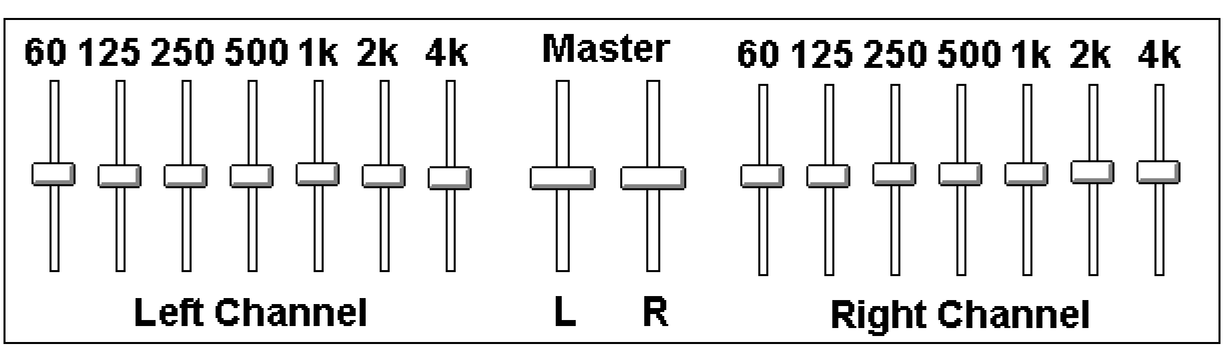
The bass and treble controls let us adjust the lower and upper parts of the audio spectrum. Think of the range of audio frequencies split into two parts, the lower from 20 - 500 Hz, and the upper from 500 - 20,000 Hz. Now break the audio spectrum into many small bandwidths and let an active bandpass filter control each section. We then have far more control of the sound. An active bandpass filter is given in Fig. O-15. Note that a power source (battery) is indicated by the "+" symbol and ground.

Fig. O-15. Active Bandpass Filter.



An equalizer is an audio component that gives control over more than two regions of the spectrum. Active bandpass filters are employed to boost specific frequency bands. A seven-band equalizer is illustrated in Fig. O-16. The audio spectrum is broken up into 7 regions. A better equalizer is one that breaks the spectrum into 12 regions or more. Note how the central frequency for each region progresses by doubling the previous frequency. We can adjust the amplified levels of each of these bandwidths by sliding levers up and down. Adjustments can be different for left and right channels. Specific settings depend on the music played, room acoustics, and our personal preferences. Some equalizers include a reverb control.

Fig. O-16. Seven-Band Stereo Equalizer.



Dolby

We consider a special kind of signal processing now, one aimed mainly at reducing noise on playing tapes. Noise is heard on playing a blank tape. The noise is particularly dominant above 5 kHz. This is due to room temperature, which gives energy to the magnetic dipoles ("baby magnets") on the tape. They do not align correctly due to this energy.

Fig. O-17. Noise on Tape Playback.

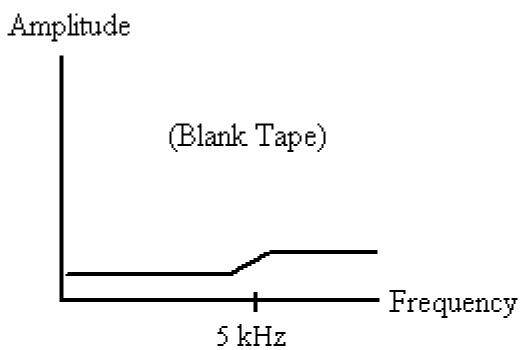


Fig. O-17 indicates the amplitude of noise on playback. *Noise* is the term used when virtually all frequencies are present. Make a "shhh" sound. This is noise. A release of steam is another example. In Fig. O-17 we find some presence of all frequencies. However, there is a greater amount for frequencies over 5 kHz. Due to the larger amount of high frequencies, the tape noise sounds more like a "hiss." In fact, it is called *tape hiss*. The dipoles do not truly get randomized when we erase the tape. Similarly, they do not align 100% correctly when we record.

You can cool the system since thermal agitation due to heat is the culprit. But you probably do not want to get into cryogenics. Temperatures have to get really low. Dolby thought of a very simple way to approach the problem in the 1960s. The first method, called *Dolby-A*, was developed for professional use. A commercial version for the average consumer, *Dolby-B*, became widely available in the 1970s. The basic idea is to simply use an active filter to boost the high frequencies when you are recording, then filter them on playback. Remember, hiss is relevant only on playback. You can't tape hiss. So boosting on taping just raises the level of the music at high frequencies. Then on playback, use a filter that reverses your boost at high frequencies to bring the music back to where it should be.

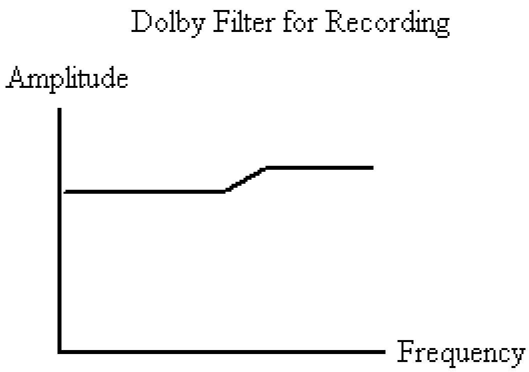
The key here is that you are lowering all high frequencies on playback. This includes the playback hiss. The high-frequency noise is lowered along with the taped high pitches on playback. The high pitches move down to levels where they should be and the hiss moves down also. The result is less hiss.

The Dolby processing circuit for playback matches in reverse what the recording circuit does. You boost high frequency on taping and then you lower high frequency on playback. In order to use Dolby during recording, you must tape with Dolby activated on your tape recorder. There is a switch to set Dolby to on. Then you must play back the tape with Dolby on.

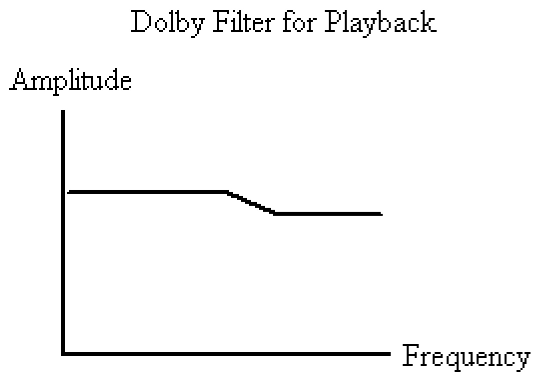
Playback with Dolby on activates the playback circuit which applies the proper correction for the high frequencies. Leave the Dolby switch to the on position and automatically the right Dolby processing circuit is used. Some people like playing Dolby tapes with Dolby off. You then have the hiss but the high frequencies are enhanced from the original taping. The enhancement helps mask the hiss (cover it up). However, you are then hearing the high frequencies louder than that intended by the musicians.

Fig. O-18 puts into graphical form many of things we have been discussing. Dolby boosts high frequency on recording and compensates for this on playback. Note how the music and noise get equalized in the final diagram which indicates what we hear.

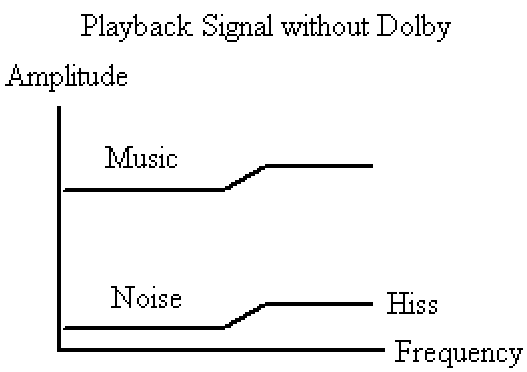
Fig. O-18. Dolby Filters for Recording and Playback.

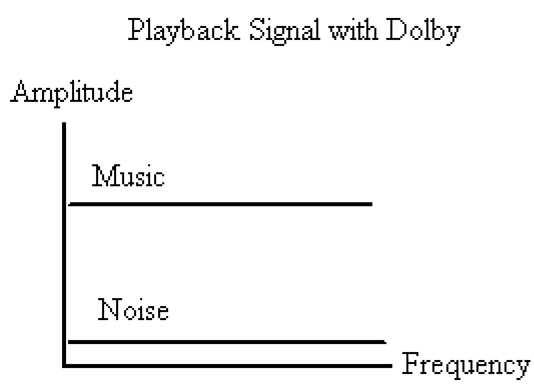


Playback tape noise is not relevant during recording. So we boost the high end in preparation for suppressing high frequencies later. Lowering the intensity of the high end to the proper level during playback will drive the noise that arises during playback down.



During playback, the high frequencies are suppressed somewhat. This is okay since the high frequencies were boosted during taping. The noise that appears on playback gets "wasted" as the high frequencies (music) are reduced to their proper levels (see immediately below).





In the diagram immediately above at the left, the tape was made with Dolby but is incorrectly being played back without the Dolby playback filter. High frequencies were taped higher so they are louder on playback without the Dolby filter. Note that noise now creeps in during the playback, especially over 5 kHz (the high end). With proper playback (diagram to the right), the hiss is lowered.

There are a few Dolby Noise Reduction Systems. The common commercial form of Dolby is Dolby B. Dolby A is a more elaborate form, used in professional recording. In Dolby B, the high-frequency end above 5 kHz is enhanced ten-fold. On playback, the high frequencies are suppressed ten-fold to compensate. This takes the noise down to one-tenth its usual value. We refer to such a drop as 10 decibels (10 dB).

Dolby proposed in 1983 using two Dolby B systems back to back to get a hundred-fold suppression of noise. On the decibel scale which we will learn about later, a hundred-fold drop is a 20 dB decrease.

Back in the old days when such sophisticated processing circuits didn't exist, faster tape speeds had to be used. Cassettes play at 1 and 7/8 ips (inches per second). There is much hiss at this speed. Tape decks in the 1950s typically had the higher speeds: 3 and 3/4 ips (twice the cassette speed) and 7 and 1/2 ips (four times the cassette speed). Playing tape hiss twice as fast doubles the frequency (pushes it up an octave). So 5 kHz gets pushed to 10 kHz by doubling the speed. Double again and you get 20 kHz. Fine recordings could be made in the early days with tape speeds at 7 and 1/2 ips. However, you need much tape, a big reel compared to a small cassette.

If you suggested taping music at cassette speed in 1960, people would think you were joking. One would only record speech at such a low speed. In fact, the professionals used 15 ips for higher-quality musical recordings. However, 25 years later, the improvements in signal processing made it possible to have excellent recordings at cassette speeds. You need little tape because it moves across the tape head so slowly. Now, the *Rachmaninoff Third Piano Concerto* (40 minutes) needs 400 ft instead of 1600 ft of tape (at the 7.5 ips we discussed earlier). Therefore, the cassette can be small. However, cassettes are no longer as popular as they once were due to compact discs.

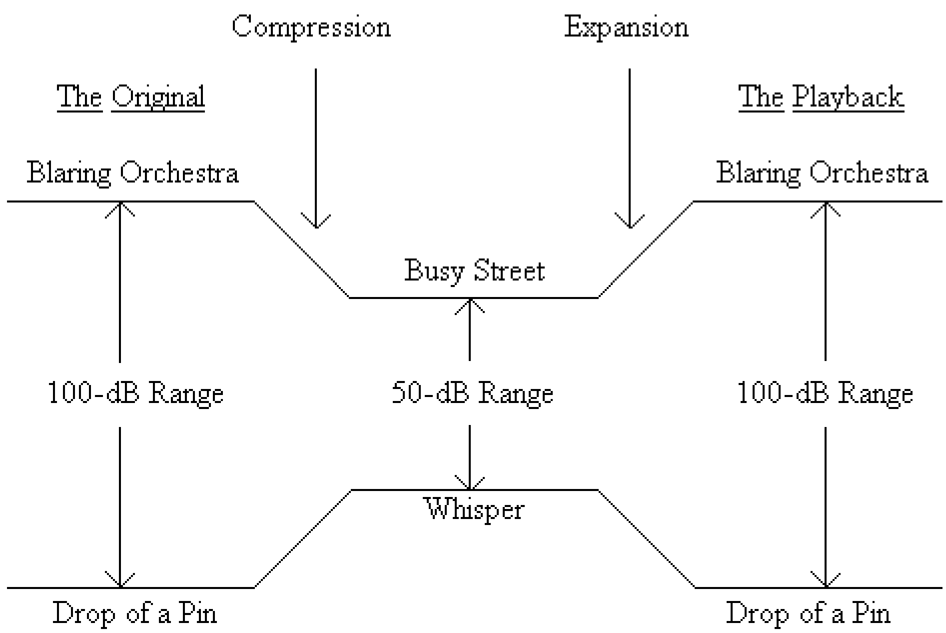
The dbx Compander

Tape is not able to handle the range of loudness levels we find in real life. From an extremely soft violin playing solo (barely audible) to the blare of the full orchestra, the decibel change is about 100 dB. A tape can't record this difference. The tape just gets saturated. Tapes can successfully record a range of 50 dB. The decibel scale is tricky. This doesn't mean half the orchestra. Think of a range of 50 dB as going from a whisper to a busy street.

There is an ingenious way to capture a sound as quiet as a drop of a pin and also the full orchestra. A processing circuit compresses the 100 dB range to 50 dB as the sound is recorded on the tape. The tape can faithfully store the compressed range of 50 dB. The various loudness levels are "pushed together" for recording. Then on playback, another processing circuit expands the signal from the tape. Fig. O-19 illustrates the idea. The original sound is compressed and the compressed version is recorded. The recording is expanded on playback to recover the full range of the original sound.

The system described in Fig. O-19 is called the *dbx* system. It does wonders for curing tape hiss. The tape hiss appears at the "whisper level" before expansion. When tape hiss appears on a blank tape without dbx or Dolby, you hear it like a whisper at normal volume. Expansion takes the "whisper" down to the level of a "drop of a pin." The hiss is essentially gone! Can you hear the drop of a pin with almost any other sound going on?

Fig. O-19. The dbx System.

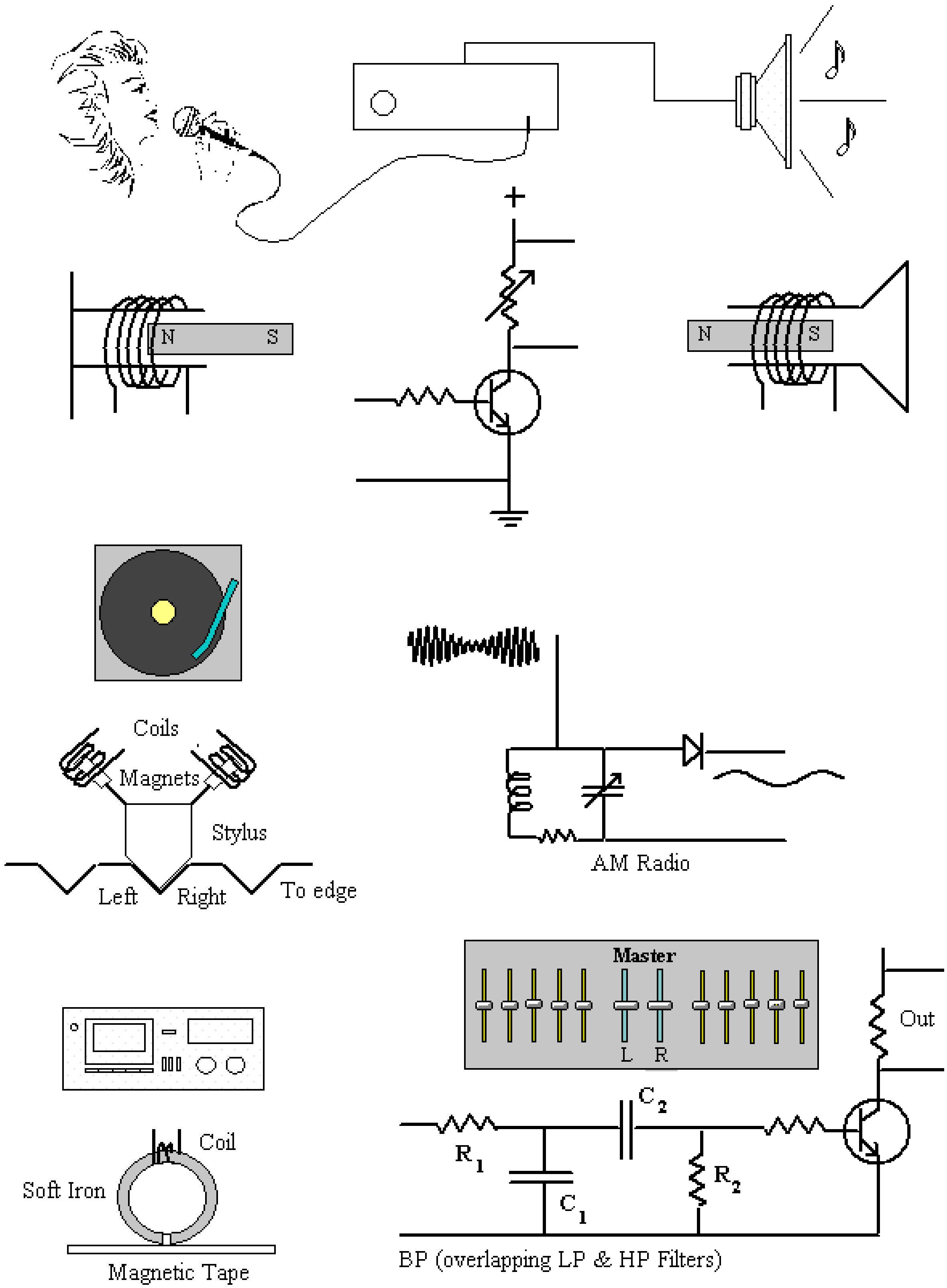


Like Dolby, dbx requires the dbx switch to be on during recording and during playback. It is interesting to take a regular recording without dbx and try playing it back with dbx. The soft sounds are made too soft and the loud sounds too loud.

When the music goes from soft to loud, this change is expanded by the dbx processing circuit. It sounds as if the music is moving quickly right at you due to the sudden increase in sound. Then the reverse is perceived when the music drops. It appears as if the source of the music quickly retreats to a great distance.

Finally, Fig. O-20 summarizes most of the sound components we have covered in this text. Connect the output of the microphone schematic to the input of the amplifier circuit. Then connect the amplifier output to the speaker. Filter abbreviations are used: LP for low-pass, BP for bandpass, and HP for high-pass. Refer to the appropriate places in the text for detailed drawings and explanations.

Fig. O-20. Summary of Basic Sound Components.



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