Crossovers

By Mike MacLeod.

A practical guide to the loudspeaker Crossover.

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A design and construction manual.

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Chapter 1

Crossover Networks.

The most important component in a high fidelity system is the **crossover network** and it is the make or break of a system, being as important as the amplifier. Not only does it split the frequencies between the drivers, but it also determines the power going to them. Without a network, you would simply blow the tweeter. If you look inside a so-called “hi-fi” speaker, you would be very surprised, as sometimes there is only a capacitor connected to the tweeter. This not only applies to hi-fi speakers, but to car systems as well. A good quality low loss network is essential no matter where it is used, as the crossover network completely influences the final sound.

Type of network

So what exactly are these networks and what do they do? They are filters to give them their proper name, actually a family of filters, and this can be further broken down into specific types. For our discussion we are going to concentrate on filters used in audio and to only discuss those that apply to high fidelity loudspeakers. The types of filter that that we’re going to be dealing with are called resonant **LCR filter networks**, more about this later.

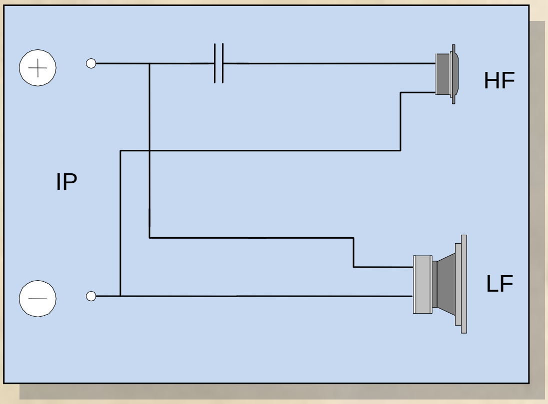


***Fig 1**. A simple 1st order crossover*

The two main filter groups are: active filters and passive filters.

- **Active filters** are electronic filters utilizing resistors, capacitors and opamps. This implies that they require their own power source which makes them ideal for integrating into or using them with separate power amplifiers for each loudspeaker.

- **Passive filters** on the other hand utilize inductors, capacitors and resistors and are housed inside the speaker enclosure. They don’t require power and are usually mounted as one unit supplying all the drivers and are the ones that we are going to discuss.

Passive filters come in two distinct topologies, namely **series** networks and **parallel** networks. There are three reasons why a crossover is needed:

1. A single loudspeaker cannot produce the whole audio range without causing distortion, especially at high power, so the frequency range is split between two or three drivers.

2. To protect the tweeters, which are only rated at about 5watts and without a suitable crossover, the voice coils would burn out very quickly indeed.

3. To improve the frequency response as much as possible. All loudspeakers have very irregular frequency responses, sometimes with very large peaks outside of their desired operating range. A woofer may have a large peak at around 1300 to 1500 Hz. A crossover can limit the frequency to below this peak by delivering more power to a midrange driver or a tweeter.

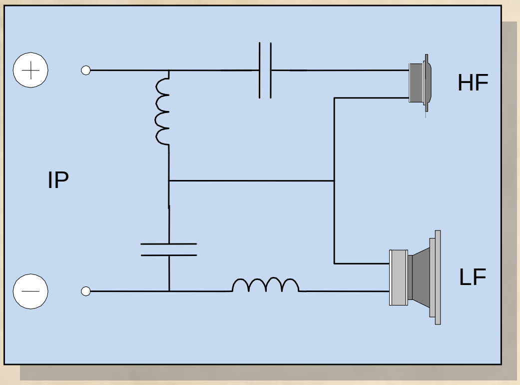


***Fig 2**. Series network*

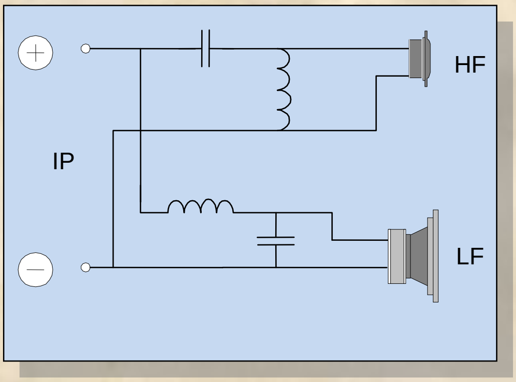


***Fig 3**. Parallel network*

***Fig 2& 3.** Series and parallel networks. Note the series connection between the two drivers in the first network, hence the name.*

Kinds of network

Crossover networks are named by the number of drivers or speakers in the system and by the steepness of the cut-off action.

- **Two-way systems** are comprised of two drivers, namely the woofer and tweeter. The woofer is fed by a low pass filter while the tweeter is fed from a high pass filter.

- **Three-way systems** have a midrange driver added, totalling three drivers, but these filters pass only a certain band of frequencies that fall between those of the woofer and tweeter and are thus called band-pass filters.

Crossover Order and Cut-off Rate

The simplest network is called a **1st order network** because it has only one element in each filter. A woofer has a coil to block high frequencies while a capacitor blocks the low frequencies to the tweeter. The effectiveness of these elements to attenuate or cut the signal is known as the cut-off rate or slope and is 6dB per octave. This means that for every halving of the cut-off frequency the power drops by 6dB. The crossover point between the woofer and tweeter occurs at the -3dB point – see Fig 4. From this you can see that the drivers still carry a fair amount of signal below the cut-off frequency and this will still damage the tweeters. At a constant power, the diaphragm excursion of the drivers, without a crossover, will increase by a factor of four as the frequency is halved. Another problem with 1st order networks is that whenever the two speakers are reproducing the same frequency at roughly the same levels, there will be interference between them, reinforcing some frequencies or cancelling others.

By adding additional components - called cascading, to the basic 6dB filter, a coil in the tweeter filter and a capacitor in the woofer filter, creates a **2nd order** network with a 12dB slope and this has become the standard, as going any steeper starts creating problems like transient and other distortion in the woofer and tweeter, as well as adding extra cost for very little extra gain in quality. The crossover point occurs at -6dB.

Parallel networks have become the choice of just about everyone and this is the network that we will describe from here onwards. Second-order filters can have a Bessel, Linkwitz-Riley or Butterworth characteristic depending on design choices and the components used.

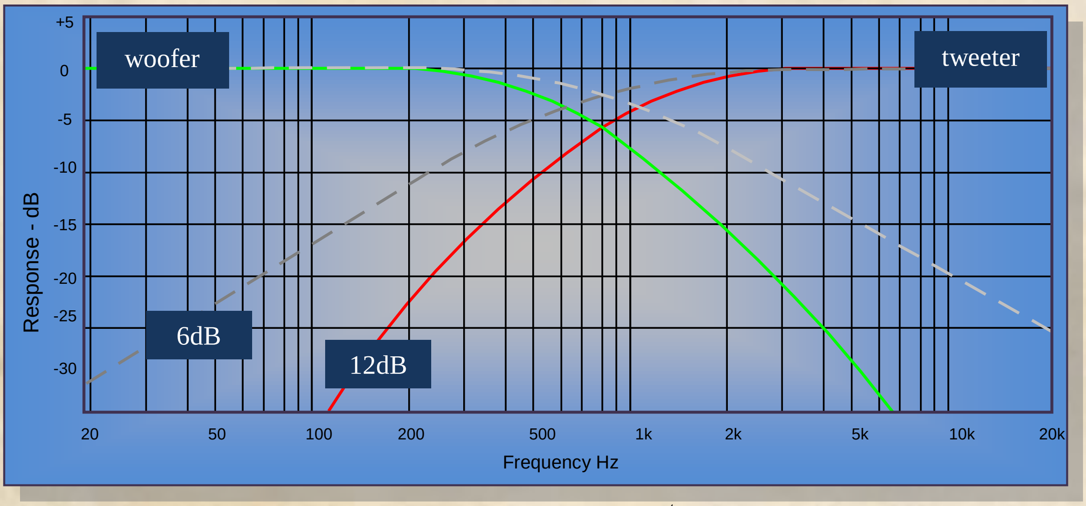


Fig 4. *These are the typical response curves of: 1st order, the grey dotted lines, and 2nd order networks, green and red lines, crossing over at 800Hz . Note the slow decline in response beyond the crossover frequency of the 6dB slope.*

Crossover points.

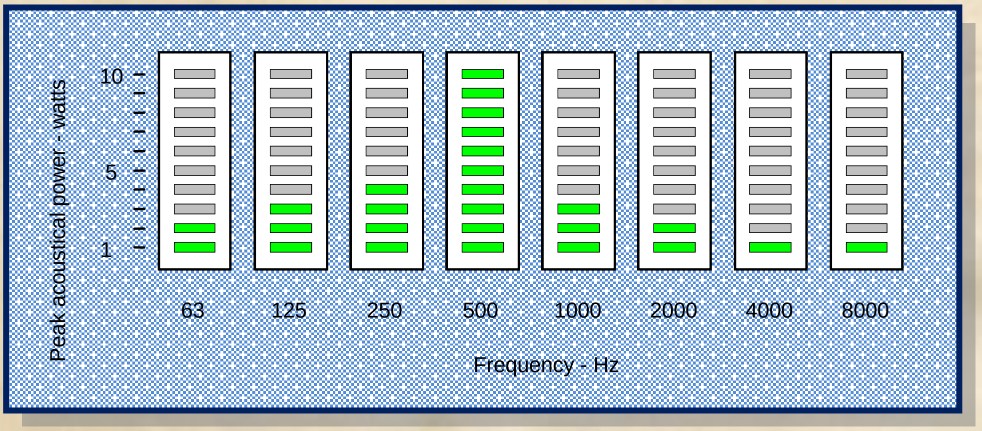


Fig 5.

A spectral analysis of a symphony orchestra shows that the bulk of the power is concentrated in the 500 Hz region, see **Fig 5 above**. A typical voice ranges from 500 Hz to 2000 Hz and a typical hi-fi system from 20 Hz to 20 480 Hz (10 octaves). If this is split between the woofer and tweeter, then the crossover point would be about 784 Hz. A single driver cannot cope with the full range of sound at high powers, without causing Doppler and other distortion. The choice then would be to keep the woofer below the voice range, which has a lot of dynamic power of its own, to less than 500 Hz. This is the professional choice. Top C (4186 Hz) on the piano, still has a fair amount of energy in it, so 5000 Hz would be the upper or tweeter crossover point which is about the limit, maybe dropping to 4500 Hz.

Now, due to the fall off of power as the crossover point is reached, the beauty of the three-way network comes to the fore and that is to let the midrange operate between the two crossover points of 500 Hz and 5000 Hz. In a two-way system, the crossover point is usually 2000 Hz and as a result the woofer has to cover a greater frequency range and the tweeter must handle more power. I personally prefer 2500 Hz as the crossover point in my Camm DynaTone loudspeakers.

Octaves

The audio spectrum is broken up into 10 bands or octaves as follows:

1^{st} octave	20 – 40 Hz
2^{nd} octave	40 – 80 Hz
3^{rd} octave	80 – 160 Hz
4^{th} octave	160 – 320 Hz
5^{th} octave	320 – 640 Hz
6^{th} octave	640 – 1280 Hz
7^{th} octave	1280 – 2560 Hz
8^{th} octave	2560 – 5120 Hz
9^{th} octave	5120 – 10 240 Hz
10^{th} octave	10 240 – 20 480 Hz

Summary

1. Passive LCR crossover networks utilize (***L***)inductors, ***C***apacitors and ***R***esistors.
2. Butterworth networks have 6dB slopes and crossover at -3dB.
3. Linkwitz/Riley networks have 12dB slopes and crossover at -6dB.
4. Most of the acoustic power in a piece of music is concentrated at 500Hz and this is the limit for a woofer crossover point.
5. Tweeters should cut off below 5000Hz.
6. A midrange crossover operates between the two points of the woofer and tweeter in a 3-way system, usually 500 and 5000Hz.
7. A 2-way system crosses over at about 2500Hz.

Getting down to the nitty gritty

A loudspeaker voice coil has a resistance that can be measured with a digital multimeter. For hi-fi use we stick to 8 ohm drivers (car audio uses 4 ohm drivers) and if we measure the voice coil with our DMM, the resistance is found to be about 6 ohms, even though manufacturers rate them as 8 ohm speakers. That's all well and done and this is where most people make the mistake by using the quoted figure of 8 ohms in the formula to calculate the coil and capacitor values. The voice coil resistance is denoted as $\mathbf{R_e}$ in ohms, symbol Ω.

Fig 6. A typical 12 inch woofer

A loudspeaker has a second property relating to voice coil resistance and that is its **impedance**. This is a property whereby the speakers apparent resistance changes due to varying AC frequencies and is the combined effect of its resistance, inductance and capacitance that opposes the current applied to it and can be as high as 40 Ω at the speakers' resonant frequency. It is also measured in ohms and is denoted as $\mathbf{Z_x}$, the x subscript refers to T=tweeter W=woofer M=midrange in the formula and charts. More on measuring it later.

A loudspeaker driver cone has a tendency to resonate or vibrate; it is suspended on a spring after all – the surround, which can be made of rubber, foam or paper. Much like a tuning fork which when tapped on a hard surface resonates at its designed frequency, a loudspeaker will resonate at its natural frequency when driven by an AC signal. Various factors affect this frequency – stiffness of the surround, mass of the cone, the stiffness of the 'spider'(the yellow springy locating support below the cone), the atmospheric pressure and even objects placed in front of the speaker. Of course it goes without saying that the compliance of the box is a major factor too. The **resonant frequency** is denoted by the term $\mathbf{F_s}$ and is measured in Hertz.

A second phenomenon that occurs is the change in impedance as the frequency varies and this is where people make the mistake when determining the values of crossover networks. More on measuring this later on.

So far everything has related to standard text book formulas but there is always a fly in the ointment and that is the work done by people that affect these simple formulas. Most of the pioneering work into audio filters was done by Stephen Butterworth and lends his name to the networks that we are discussing. The fine tuning of these formulae are called **alignments** and they manipulate the magnitude response of the output at the cut-off point.

Filter Characteristics

There are three primary filter alignments that can be used, and they differ only in the damping factor, Q, or "quality factor". This is an abstract term that is applied to many passive components in many applications, and is effectively the inverse of damping. Thus, Q=1/d or d=1/Q.

- ✓ The classic 2nd order filter network is the **Butterworth alignment** that has been the mainstay of nearly all crossover systems in common use. Butterworth networks have 6dB slopes and crossover at -3dB. The disadvantage of this alignment is that the tweeter is 180° out of phase producing a dip in the response curve at the crossover point. Reversing the connections to the tweeter brings the signal back into phase but

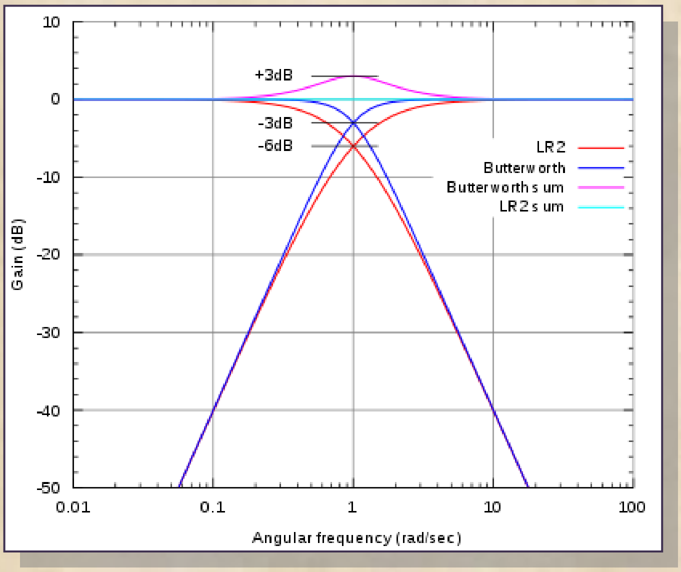


Fig 7.

because the drivers are now summed the output is now +3dB, producing a hump in the response curve, which is more tolerable than the dip.

- ✓ The **Linkwitz-Riley** alignment is a re-arrangement of two cascaded 2nd order Butterworth filters. The resulting Linkwitz–Riley filter has a 6 dB gain at the cutoff frequency. This means that summing the low-pass and high-pass outputs causes the gain at the crossover frequency to add to 0 dB, so the crossover behaves like an all-pass filter, having a flat amplitude response with a smoothly changing phase response.

- ✓ A third 2nd order network alignment is the **Bessel** and it is characterized by a slight peak in the summed magnitude response at the crossover frequency of 1.25 dB and a moderate dip in power response of -1.756 dB.

To alter from one alignment to another is just a matter of multiplying the values of the capacitor and inductor as calculated using the basic text book formulas and multiplying them with the 'nth' multipliers from the following table:

	1st order	**2nd order**		
	As calculated with the text book formulas. For higher orders multiply the standard values with the appropriate multiplier.	Butterworth	L/R	Bessel
C1		0.707	0.50	0.57
C2		0.707	0.50	0.57
L1		1.414	2.00	1.74
L2		1.414	2.00	1.74
		nth multipliers		

Table 1

1st Order Networks

A 1st order network is one that has only one element per leg, a capacitor for the tweeter and a coil for the woofer.

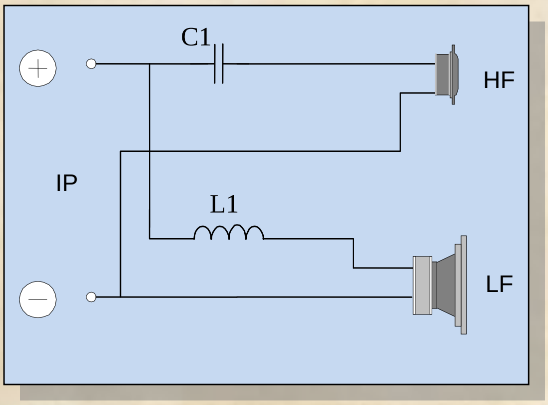


***Fig 8**. A 2-way 1st order filter.*

A capacitor reacts to an alternating frequency by limiting the current through it and acts like a variable resistor, blocking low frequencies. This property is called the capacitors reactance and is inversely proportional to the frequency applied to it. The current through a capacitor leads the voltage whereas it lags through an inductor and is 45° through the cap and 45° through the coil. They therefore cancel, producing an in phase signal from the drivers at the crossover frequency. The two half power signals sum to produce full power and so the response through the crossover region is flat.

Mathematically the formula is written as: $X_c = 1/2\,\pi\, f\, C$, where f is the frequency and C is the capacitance in Farads. The so called 'crossover frequency' is the frequency at which the **capacitive reactance** equals the tweeters impedance, which we measure at our ***chosen crossover frequency*** i.e. 4500hz. Substituting the tweeters impedance, Z_T, in the formula and solving for C we get:

$$C = \frac{1}{2\,\pi\, f\, Z_T} * 10^6\ \mu F$$

Because a Farad is such a large number we usually work with micro Farads which is 10^{-6}, and is denoted μF.

A coil or inductor has similar properties in that it blocks high frequencies while allowing the low frequencies to pass through and its reactance is proportional to the frequency it handles. Mathematically the formula is written as: $X_L = 2\,\pi\,f\,L$, where L is the inductance in Henry's. The woofer gets progressively less power as the frequency increases. Again the so called 'crossover frequency' is the frequency at which the **inductive reactance** equals the woofers impedance at our desired crossover point i.e. 500hz. Substituting the woofers impedance, Z_W, in the formula and solving for L we get:

$$L = \frac{Z_W}{2\,\pi\,f} * 10^3 \text{ mH}$$

Because a Henry is such a large number we usually work with milli Henry's which is 10^{-3} and is denoted mH.

2nd Order Networks

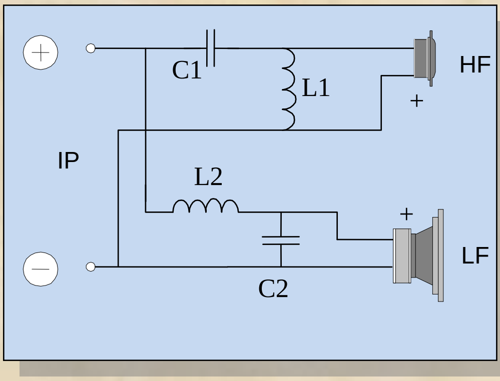


Fig 9. *A 2-way 2nd order filter.*

A 2nd order network or **L-network**, has an extra capacitor and inductor – see **Fig 9** above. The capacitor C2 in the woofer circuit shunts further high frequencies to ground giving a sharper or steeper cut off. Similarly the inductor L1 shunts the low frequencies to ground, giving the tweeter more protection. But now we choose an alignment 'n' from Table 1 and multiply the values calculated for 1st order networks:

$$C\,1 = \frac{1}{2\,\pi\,f\,Z_T} * n \quad * 10^6 \ \mu\text{F}$$

$$C\,2 = \frac{1}{2\,\pi\,f\,Z_w} * n \quad * 10^6 \ \mu\text{F}$$

And the same for the inductors:

$$L1 = \frac{Z_T}{2\,\pi\, f_h} * n * 10^3 \text{ mH}$$

$$L2 = \frac{Z_W}{2\pi\, f_L} * n * 10^3 \text{ mH}$$

Adding a midrange speaker

We could add a midrange speaker to our system creating a 3-way network. Now, as the power falls off as the crossover point is reached, we let the midrange operate between the two crossover points of, for example, 500 Hz and 5000 Hz, thereby reinforcing the sound at the crossover points.

To do this we add a **bandpass filter** to the 2-way network. A capacitor and an inductor in series forms a 1st order bandpass filter and by adding another capacitor and inductor both in parallel creates a 2nd order bandpass filter to go with that of our 2-way network to handle the power to the midrange driver. A point to note here is the fact that this filter produces a 2dB gain in its stop band, so when choosing a midrange driver, ensure that its sensitivity is 2dB ***lower*** than the woofer and tweeter, both of which must both have the exact same sensitivity. Another point to note is that it is the midrange driver that now becomes 180° out of phase while the tweeter is back in phase, so the leads to the midrange driver are reversed and the tweeter is connected with the correct polarity.

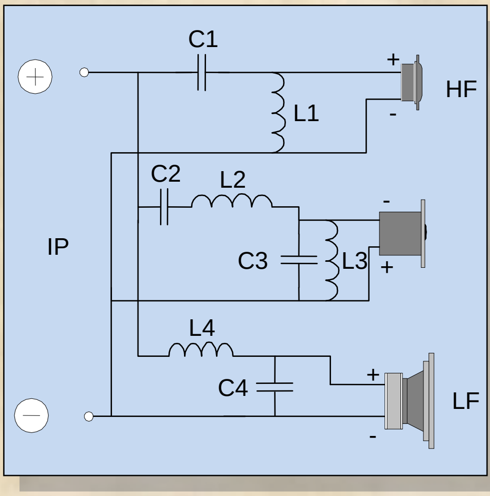


Fig 10. *A 3-way, 2nd order crossover - note the midrange polarity change.*

The formula for our 2nd order midrange filter is:

$$C2 = \frac{f_h - f_L}{\sqrt{2}\ \pi\ Z_{ml}\ f_m{}^2} * n * 10^6\ \mu F$$

$$C3 = \frac{1}{\sqrt{2}\ \pi\ Z_{mh}\ (f_h - f_L)} * n * 10^6\ \mu F$$

$$L2 = \frac{Z_{mh} * \sqrt{2}}{4\ \pi\ (f_h - f_L)} * n * 10^3\ mH$$

$$L3 = \frac{Z_{mL}\ (f_h - f_L)}{2\sqrt{2}\ \pi\ f_m{}^2} * n * 10^3\ mH$$

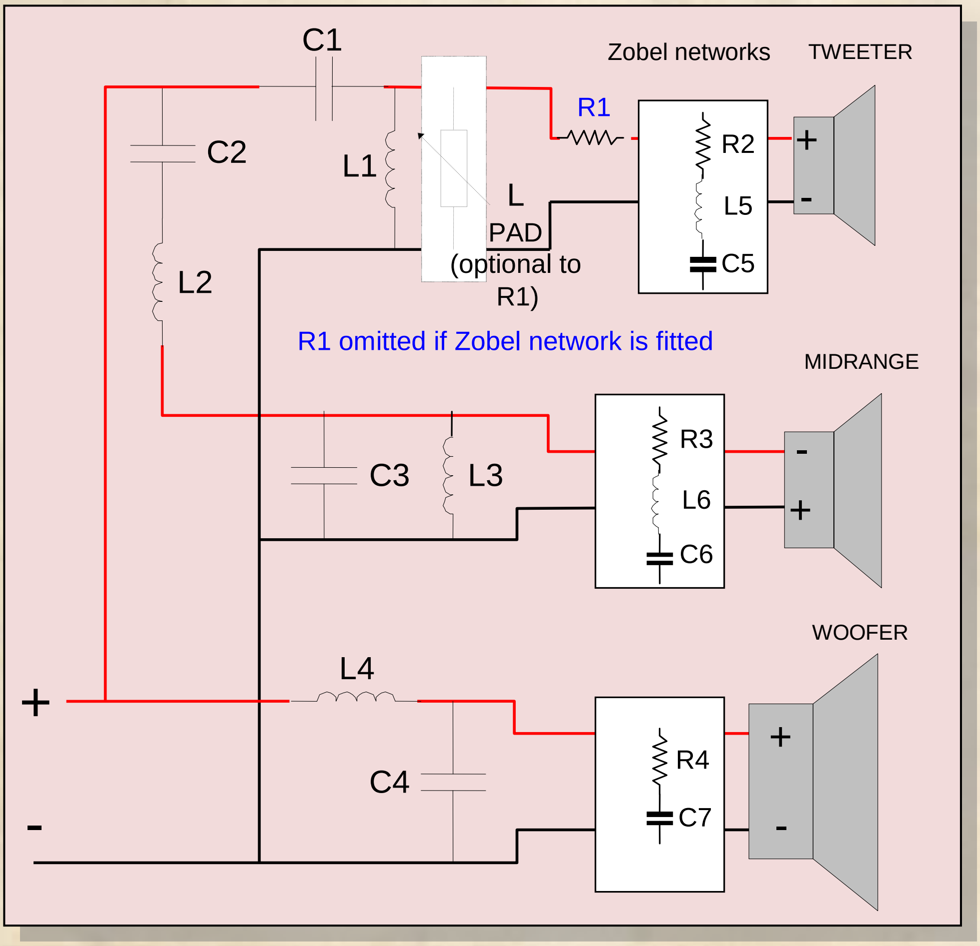
$$Fm = \sqrt{(f_h * f_L)}$$

The term F_m is the geometric mean centre frequency between the high and the low cutoff frequencies. *In the work sheets you will see that C2 and L3 have the term Fm instead of* Fm^2 *as in the box above, as I have cancelled the square root sign in the geometric formula and used* $Fm = F_h * F_L$ *instead , it just made it easier for me.*

Sometimes a resistor, R1, is inserted in series with the capacitor going to the tweeter to attenuate its output when the tweeter sounds too bright. You could also fit an **L-pad** instead of the resistor giving you a variable control. An L-pad is a potentiometer with a constant 8Ω resistance on its output no matter where you set it to.

The following diagram, Fig 11, sums up the 3-way 2nd order topology. Also all the crossover topologies we are talking about are the same and are not named Butterworth, Linkwitz/Riley Bessel etc, they are from a guy by the name of **Wilhelm Cauer** and some others before him. It is the alignments that are named Butterworth, Linkwitz/Riley, and Bessel etcetera.

Fig 11.



Note the midrange driver polarity change.

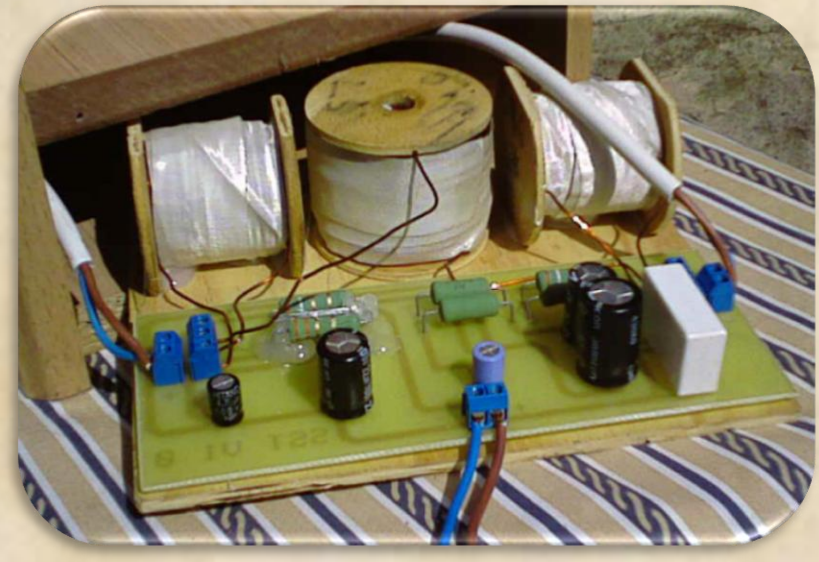


***Fig 12**. The finished crossover in a 2-way system.*

Summary:

1. The frequency response of the Butterworth filter is maximally flat (i.e. has no ripples) in the passband.
2. The Linkwitz/Riley has 0 dB gain at the crossover frequency, so the crossover behaves like an all-pass filter, having a flat amplitude response with a smoothly changing phase response.
3. The Bessel alignment is characterized by a slight peak in the summed magnitude response at the crossover frequency and a moderate dip in power response.
4. A 1st order filter always has a -3dB response.
5. The two-way 2nd order filter has the disadvantage of having the tweeter 180° out of phase.
6. In the 3-way system the midrange is out of phase and the tweeter in phase.
7. A resistor or an L-pad tames the peaks of tweeters.
8. For a two-way system just omit the midrange portion of the circuit.

Test Setup

So far everything is hunky dory regarding the theory of crossover networks but in the real world things don't quite operate correctly and loudspeakers are no different. As I alluded to earlier, a loudspeakers impedance changes with frequency and that is the problem facing the constructor, what figure does one use to calculate the values of the caps and coils? To make things clearer I am going to leave the theory a bit and explain how to test the speaker as this will allow you to see this behaviour for yourself.

Now that you've purchased your speakers!

Okay, now that you have obtained your prized speakers, the fun begins. Each speaker cone has a natural tendency to resonate, that is, it vibrates at a maximum rate at a specific frequency when it has a signal applied to it.

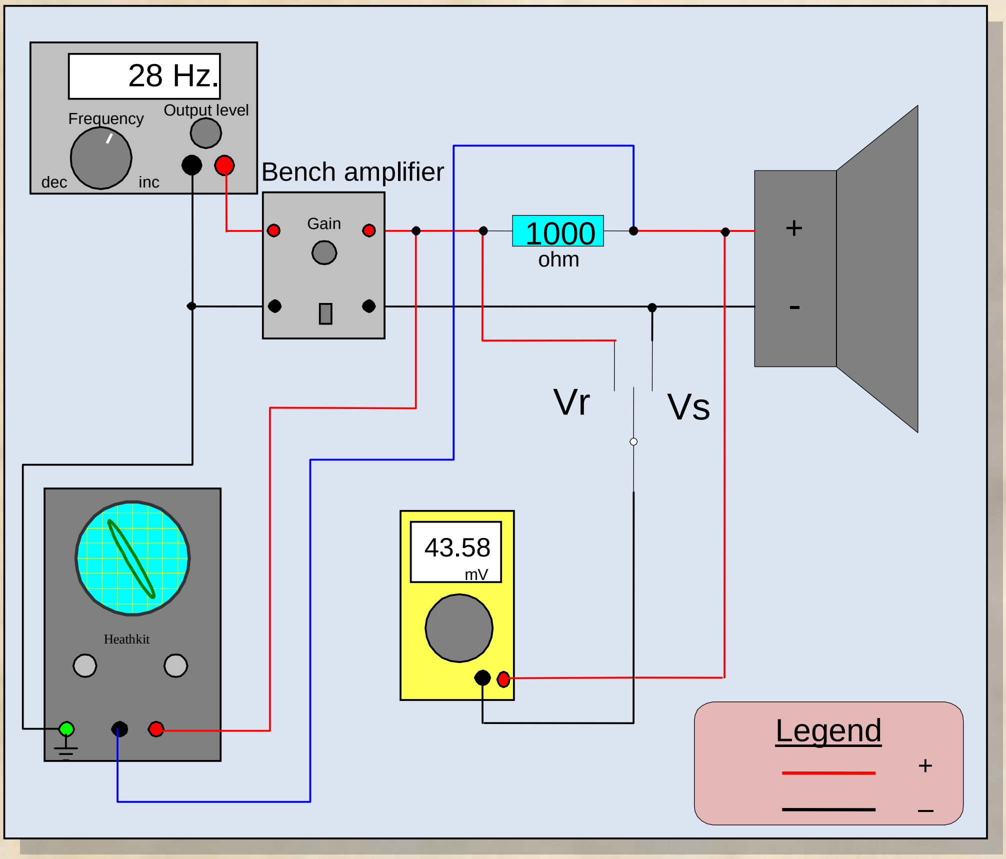
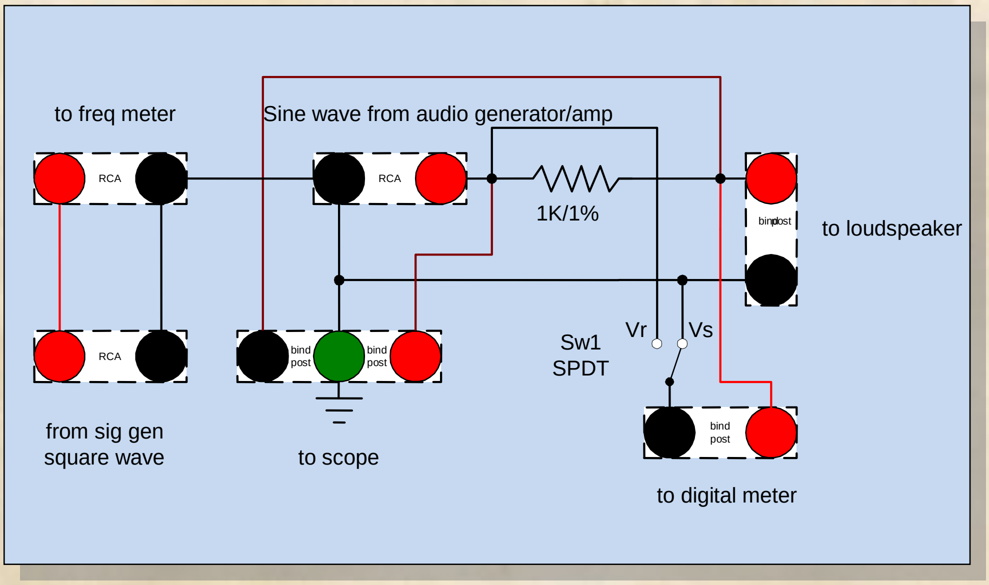


Fig 13. *Test setup.*

The frequency where this maximum resonance takes place is known as the speaker's **resonant frequency, F_s**. To do all your measurements, you will need a voltmeter, sine wave audio generator with variable voltage output and a frequency meter. I add a bench amp to help boost the power. Connect your speaker up as per Fig 13. You could make up a box with phono sockets and the switch fitted, and then wire in the 1000-ohm resistor directly to the tabs, to make life easier – see **Fig 14 below.**

Fig 14.



With everything switched on, sweep the audio generator up and down to ensure that everything is working properly and that there is a reading on the voltmeter - you can also feel the loudspeaker cone vibrating. I prefer a DMM here because it's easier to read. Note the switch positions marked $\mathbf{V_R}$ and $\mathbf{V_S}$ in the diagram, one measures the voltage across the resistor and is measured in Volts and the other measures the voltage across the speaker and is measured in milliVolts. To determine the impedance you divide V_S by V_R :

$$\text{Speaker impedance in ohms} = \frac{V_S \text{ mVolts}}{V_R \text{ Volts}}$$

The 1000-ohm resistor sets the current level and if you ensure that V_R is equal to 1 Volt at each measurement, this is important because the voltage varies with the change in frequency, then your multimeter will read the impedance directly. If your audio generator output level cannot be adjusted then you will have to manually work out the impedance with the above formula. To measure the resonant frequency, don't believe the manufacturers figures, you test the speaker in free air, that is, on its own and not in a box. The following measurements are all done outside on a clear day so that obstructions don't affect the readings. Set the audio generator to its lowest reading, say 10 Hz and place the test switch in the V_S position, also make sure your DMM is set to ***AC milliVolts***. Now slowly increase the frequency, watching the multimeter. It will start increasing to a maximum point and then start to drop back down again. With the reading at its highest, switch the test switch to V_r and set output level to 1 volt, then go back to V_s and record the voltage and frequency displayed on the frequency counter. If you have an oscilloscope, connect the vertical and horizontal inputs across the resistor and watch for the collapse of the Lissajous circle to a flat line, showing zero phase shift. You have now found the speakers resonant frequency, **F_s**. Look at the "Impedance data chart", **Fig 15** below and

Impedance data chart

FREQUENCY HZ.	V_S mVolts.	V_R Volts.	IMPEDANCE Ohms.
10	13.3	1.256	***10.58***
15	19.7	1.147	***17.17***
20	26.2	1.038	***25.24***
25	39.1	1.030	***37.96***
30	41.7	0.965	***43.21***
35	31.0	0.963	***32.19***
40	22.2	0.933	***23.79***
50	13.5	0.895	***15.08***
60	10.5	0.847	***12.39***
70	8.3	0.802	***10.35***
80	7.2	0.776	***9.27***
90	6.2	0.730	***8.49***
100	9.1	1.136	***8.27***
200	9.1	1.272	***7.15***
300	7.1	0.930	***7.63***
400	6.1	0.751	***8.12***
500	5.6	0.648	***8.64***
600	5.5	0.580	***9.48***
Resonant Frequency - F_s			
28	***42.01***	***0.964***	***43.58***

Fig 15.

set each frequency, making sure that V_R is 1 Volt and record the impedance reading on the multimeter with the switch in the V_S position or record both V_R and V_S as required, if you are doing it manually. When you have finished tabulating all the data, you can plot your impedance curve on the "Impedance Curve chart", **Fig 16** below. The Appendix A has all the blank forms for you to

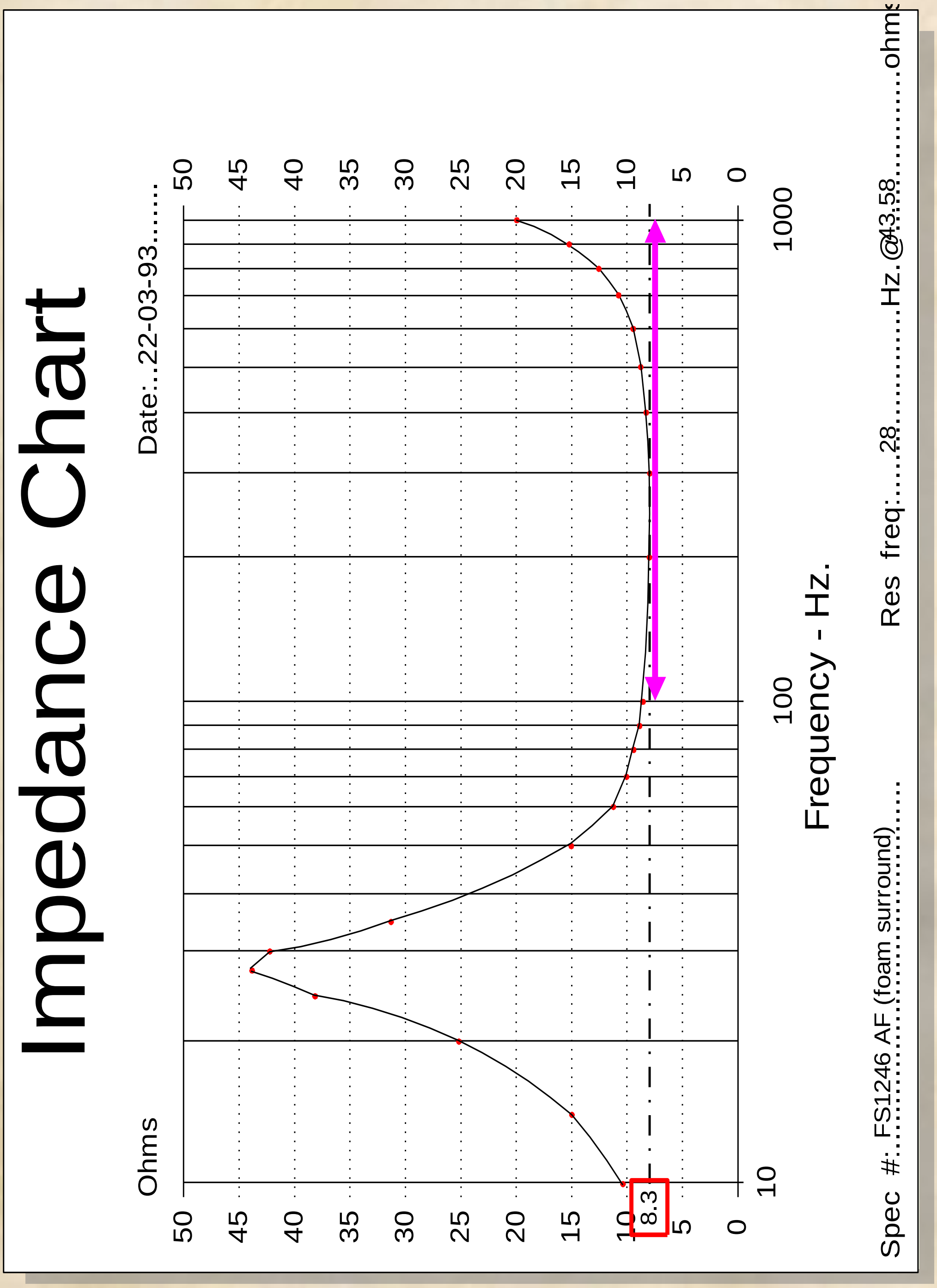


Fig 16.

copy and fill in with your own test figures. Try holding your hand or a book just above the speaker and watch the meter readings, you will see why one does these tests outside where there are no obstructions and make sure there is *no* wind blowing. To see what I mean, gently push the cone in and out and watch the meter readings. *Note that these sample figures are from my speaker, yours will be different.*

Measurements galore.

If you now look the example sheet at Fig 17, you will see that it makes provision for working out the **impedance** for the crossover networks, as well as data for tuning the cabinets to. The page is divided into two, L/H and R/H, which refers to the left hand and right hand speakers in your system. The second row is used to work out the impedance of the speaker at the desired **crossover points**, usually at 500 HZ and 5000 HZ in three-way systems. You will see each block has place for two calculations and this is for the midrange driver (with some sample figures in the R/H speaker column), which operates between the two frequencies of the crossover points. To avoid confusion, the subscript to Z is left open for you to fill in and allows you to identify the chart quickly. X represents the following subscripts: w = woofer, m = midrange, t = tweeter.

The sample chart refers to our woofer that was tested and tabulated in Fig 15. There is also place in the side column to detail at which frequencies the figures were taken.

To measure $\mathbf{R_e}$ accurately, set your digital multimeter to its lowest setting and then short your test leads together and note their resistance, mine are about 0.2Ω. Subtract this from your final readings for best accuracy, especially on readings below 10Ω. The following terms are used:

- R_e = Is the D.C. resistance of the coil measured with a multimeter.
- F_s = Speaker resonant frequency.
- Z_{fs} = Impedance at resonance.
- Z_{f3} = Is 0.707 of the value of Z_{fs} – where the signal power is 3dB down.
- F_1 = The frequency above F_s where the impedance equals Z_{f3}.
- F_2 = The frequency below F_s where the impedance equals Z_{f3}.
- Q_{ts} = Total Q of the speaker.

TYPE : …Akayo…. P/N : …….FS1246 AF……… DATE : ……22-3-93…….		
	LEFT HAND	**RIGHT HAND**
Woofer parameters : R_e F_s Z_{fs} Z_{f3} F_1 F_2 Q_{ts}	……6.8….. Ohms. …..28.0…. Hz. …..43.58… Ohms. …..30.81… Ohms. …..36…… Hz. …..22….. Hz. ……0.79..	………………… Ohms. ………………… HZ. ………………… Ohms. …………… Ohms. …………… Hz. …………… Hz. ……………
Z_x = $\frac{V_s}{V_r}$ $\frac{mV}{V}$	$\frac{43.58}{1}$ = Z_{fs} = …43.58..Ohms.	______ = Z_{fs} = ……...Ohms.
Crossover points: F_l = …..500......Hz. F_l= F_h = …4500…..Hz. **x** = w-woofer m-midrange t-tweeter F_h=	Z_w = $\frac{5.6}{0.648}$ = Z_w = ..8.64...Ohms. $Z\_$ = ______ = $Z\_$ = ………..Ohms.	Z_m = $\frac{5.9}{0.762}$ = Z_m = …7.8...Ohms. Z_m = $\frac{7.8}{0.762}$ = Z_m = .11.18...Ohms.
Speaker Q: $r_0 = \frac{Z_{fs}}{R_e}$ **Q_{ts}** = $\frac{F_s}{F_1 - F_2}$ x $\frac{R_e}{Z_{fs}}$ x $\sqrt{ro}$	$r_0 = \frac{43.58}{6.8}$ = 6.4 $\frac{28}{14}$ x $\frac{6.8}{43.58}$ x $\sqrt{6.4}$ Q_{ts} = .0.79..	r_0 = ……….. = ……….. …….. x …….x $\sqrt{\ldots}$…….. Q_{ts} = …………..
Check **F_s** : (±1 %) **F_s** = $\sqrt{F_1 \times F_2}$ Hz	$\sqrt{..36…x…22..}$ = ..28.14..Hz	$\sqrt{………..x………}$ = ……….Hz
Max Output. SPL =10LOG(P) + S P= rms watts. s=sensitivity.dB@1m/1w	=10LOG…100….+…92……. =………112………… dB SPL.	=10LOG…………+………….. =………………………dB SPL.
Calculate **F_5**: L_{ref} = Re x √2 Find L_{ref} above F_s and record F_5.(300-1000 Hz.)	= ..6.8.. x √2. = .9.61.. Ω F_5 = ……615….. Hz.	= …… x √2. = ……… Ω F_5 = ……………. Hz.

Z_{f3} = .707 x Zf_s F_1 and F_2 is the frequency of Z' above and below F_s.
L_{ref} is for Zobel impedance equalisers for some woofers whose impedance curve rises sharply above +-600Hz. See Fig 16.

***Fig 17.** Sample Data Calculation Chart*

The resonant frequency, $\mathbf{F_s}$ and its impedance $\mathbf{Z_{fs}}$, is taken from the chart in Fig 15. Calculate $\mathbf{Z_{f3}}$, which is 43.58 x .707 = 30.81 ohms, this is where the power is down by 3dB. Set the resonant frequency on the audio generator and then slowly increase it until the DMM reads 30.81 and record this frequency as F_1. Now slowly decrease the frequency below resonance until again you get 30.81 on the DMM and record this as F_2. At the bottom of the chart is a formula for checking $\mathbf{F_s}$ using F_1 and F_2 and the result should be within ±1 % of each other. Notice that I write all the important figures that I use to either make the crossovers or for use in other formulas in **red**, this makes them stand out and makes them easier to see on a page full of data.

The last row on the Woofer Data chart, Fig 17, has a calculation for a reference impedance $\mathbf{L_{ref}}$. This is for determining the frequency $\mathbf{F_5}$ used in the Woofer impedance equalizer filter calculations, Fig 18, and requires patience to determine. It is usually quite high above F_s at about 1000 Hz but differs between loudspeakers, so take your time and sweep slowly above F_s until you get the reference impedance. Mine was located at 615 Hz, with Lref = 9.61Ω.

Summary:

1. Do all speaker measurements outside in still air, clear of any obstacles.
2. Make a jig to hold the speaker vertically when doing the tests so that the sound wave isn't reflected back at the cone as would happen if laid flat on a table.
3. $\mathbf{V_R}$ measures the voltage across the resistor and is measured in Volts
4. $\mathbf{V_S}$ is measured in milliVolts and measures the voltage across the speaker.

Back to class

Okay, time to sit and relax with your favourite beverage in hand and let the excitement and adrenalin rush of your first loudspeaker test fade to more manageable levels.

Hopefully you will now appreciate the conundrum one faces when choosing the correct impedance for the crossover calc’s. Make no mistake, the preceding crossover networks work extremely well, my first three hi-fi’s that I sold had these Linkwitz/Riley crossovers fitted and ten years later the customers still swear by them.

The difficulty lies in the new technology in loudspeaker manufacture. Mid’s and tweeters now employ dome technology like soft textile domes or titanium/aluminium diaphragms and because of this they sound harsh at their fundamental frequencies, as they act more like woofers due to their very low resonance, which disturbs the crossover network. This is because these frequencies fall within the crossover region and upset the networks' balance, causing annoying resonance. Mids and tweeters that are ferro-fluid cooled don't suffer as much from this phenomenon because the fluid acts like a mechanical damper giving them smoother responses, but unfortunately not all of them are cooled this way. The days of paper cones are numbered.

Zobel Filters

The second problem is the rise in impedance of the woofer and if you look at **Fig 16**, say from 100 Hz you can see how steeply it rises above 300Hz. Bugger, right through a 3-way system lower crossover point of 500Hz and leaving nothing at 2500Hz of a 2-way system.

Now luckily for us Otto Zobel working out of the Bell Laboratories came up with several filters for transmission lines and we can adapt them for our use. For the woofer we can use a **Zobel impedance equaliser** that flattens the rise in impedance. It is a simple network and consists of a resistor and capacitor in series placed in parallel to the driver, see Fig 18 for details. In the following discussions the mathematics gets very involved and to keep this book simple I will not go into detail but rather make use of charts with the relevant formula in them which we just fill in and follow the laid out procedure. To make things even simpler, I will include software to do all the tedious calculations for you☺.

The formula for calculating the component values is as follows:

$$\mathbf{C7} = \frac{L_e}{R_e^{2}} \times 10^{3}\ \mu F$$

$$\text{Where:}\quad L_e = \frac{R_e}{2 \times \pi \times F_5} \times 10^{3}\ mH$$

R_e is the voice coil dc resistance as measured with a DMM and **F_5** you get from the data chart. Calculate the speakers' **inductance, L_e** first before calculating the capacitance. Don't scale these values, as the inductance L_e is a constant value at its own frequency.

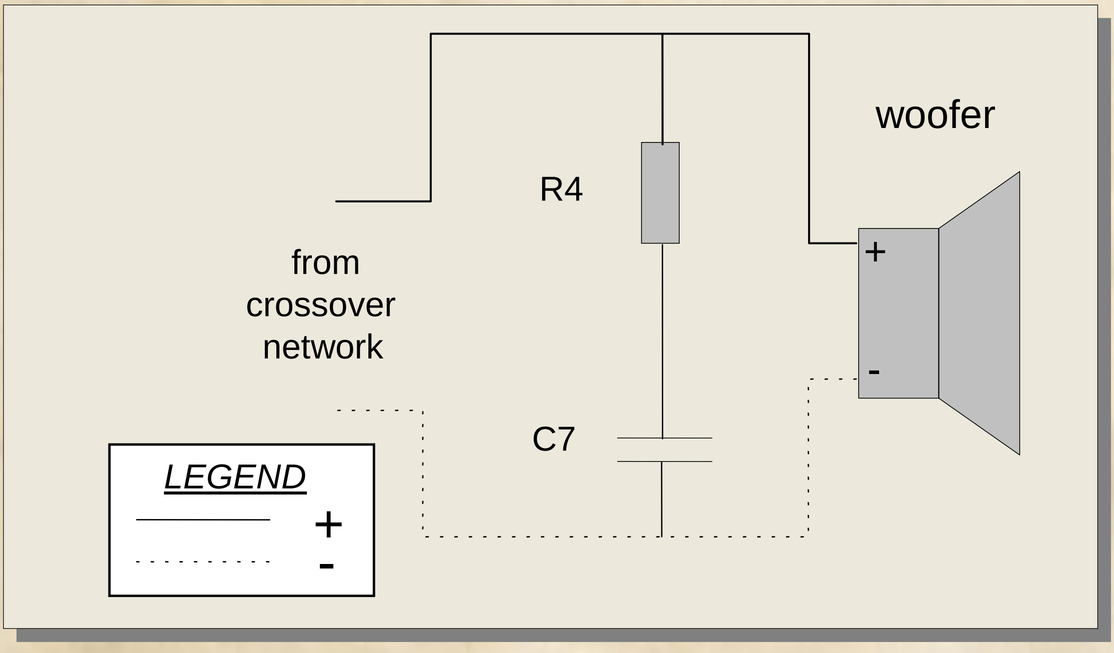


Fig 18.

Woofer impedance equaliser equalizer

For the tweeter resonance at the crossover frequency, the problem is improved with a **Zobel notch filter**, see Fig 19 below, that you insert across the driver leads. R_n is the speaker's DC resistance, R_e, and F_n you get from the Impedance Data Chart for the applicable speaker (usually it's resonant frequency),

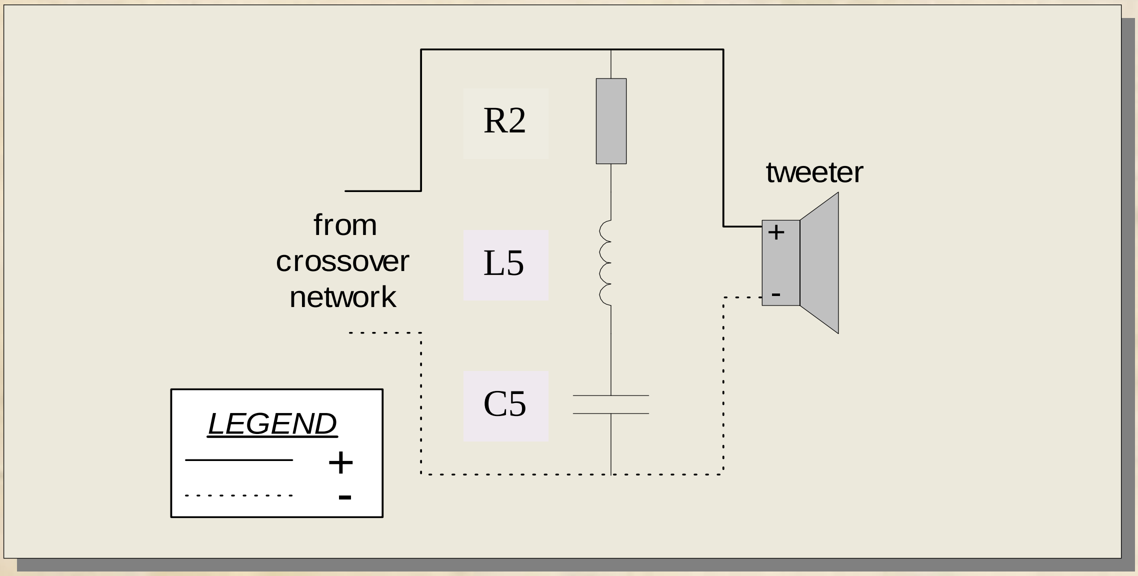


Fig 19.

Tweeter notch filter

$$C_{ideal} = \frac{1}{2 \times \pi \times R_e \times F_n} \times 10^6 \quad \mu F$$

$$L5 = \frac{1}{C5 \times (2 \times \pi \times Fn)^2} \times 10^3 \quad mH.$$

or $\mathbf{F_n}$ may be a frequency that you want to control.
Calculate $\mathbf{C_{ideal}}$, then choose the nearest standard value capacitor for C5 to enable a small enough coil to be wound (< 6mH). See Fig 17 for the relevant data for the above filters and proceed to the next chapter.

Summary:

1. Fit an impedance equalizer to woofers that have a steep rise to their impedance curve.
2. Fit a notch filter to tweeters that have annoying resonance at the crossover frequency.

Doing the math

Using the impedance data from Fig 17, calculate the capacitor and inductor values on "Crossover Network Calculation Chart" in Fig 20. With modern drivers, Zobel networks are a necessity and their aim is to keep the voice coil impedance constant. Therefore check your impedance curve chart and use an impedance that falls on the flattest part of the curve. On my chart this is at about 100 Hz and is equal to 8.3 Ω. This then does not upset the crossover and actually makes it work more efficiently. If you already have loudspeakers and they don't sound good, you can always use this info to tweak them.

Capacitors and inductors.

A crossover network consists of **capacitors** (non-polarised) and **inductors** (coils) to control the power and frequency to the drivers. Basically a capacitor blocks low frequencies and an inductor blocks high ones. Using one of each, that is, one capacitor on the tweeter and one coil on the woofer, is called a **first order network** and has a 6-dB per octave cut-off slope. Adding an inductor to the tweeter and a capacitor to the woofer, makes it a **second order network** with a 12-dB slope. This is the preferred choice, as going any steeper is not only more expensive, but introduces transient distortion when the speaker encounters sudden changes in signal. Audio grade **polypropylene** capacitors are the best ones to use, but unfortunately do not come in very high values, usually to about 4.7 µF. You can either connect them in parallel to get higher values or use special non-polarised electrolytic capacitors that are much cheaper. Polyester is a good second choice, with higher values. Whatever you use, make sure that the **voltage rating** is high enough for the power output of your amplifier. The following table refers :

Power - Watts rms.	50	100	150	250	500
Voltage – 4 ohm load	20	28	35	45	63
- 8 ohm load	28	40	50	63	90

Doubling the Voltage in the table above gives a good safety margin.

Table 2.

Crossover points : .....500.........&.....4500........ Date : ......22-3-93..........

Z_w * = ...8.64......... Z_{mh} = ...11.18......... Z_{ml} = ...7.8....... Z_t = ...7.38........

Alignment: …L/R…… **n** = C x …0.5… & L x…2.0….

Woofer :

$$L4 = \frac{Z_w}{2\pi \times F_L \times 10^{-3}} * \mathbf{n} = \frac{8.64 \times 2}{6.28 \times 500 \times 10^{-3}} = \text{.........}\mathbf{\mathit{5.50}}\text{.......mH}$$

$$C4 = \frac{1}{2\pi \times Z_w \times F_L \times 10^{-6}} * \mathbf{n} = \frac{1 \times .5}{6.28 \times 8.64 \times 500 \times 10^{-6}} = \text{........}\mathbf{\mathit{18.42}}\text{........uF}$$

Midrange :

$$L2 = \frac{Z_{mh} \times \sqrt{2}}{4 \times \pi \times (F_h - F_L) \times 10^{-3}} \times \mathbf{n} = \frac{11.18 \times \sqrt{2}}{12.566 \times 4} \times 2 = \text{......}\mathbf{\mathit{0.629}}\text{......mH}$$

$$L3 = \frac{Z_{ml} \times (F_h - F_L)}{2 \times \sqrt{2} \times \pi \times F_m \times 10^{-3}} \times \mathbf{n} = \frac{7.8 \times 4000}{8.8857 \times 2250} \times 2 =$$

(Where $F_m = F_h \times F_L$

....4500...x....500.......

= ......2250000.........)

$$= \frac{31200}{19992.825} = \text{........}\mathbf{\mathit{3.12}}\text{......mH}$$

$$C2 = \frac{F_h - F_L}{\sqrt{2} \times \pi \times Z_{mL} \times f_m \times 10^{-6}} \times \mathbf{n} = \frac{4000}{4.442 \times 7.8 \times 2.25} \times .5 =$$

$$= \frac{2000}{156.00195} = \text{........}\mathbf{\mathit{25.65}}\text{.......uF}$$

$$C3 = \frac{1}{\sqrt{2} \times \pi \times Z_{mh} \times (f_h - F_L) \times 10^{-6}} \times \mathbf{n} = \frac{1}{4.442 \times 11.18 \times 0.004} \times .5 =$$

$$= \frac{1}{0.19864} \times .5 = \text{...........}\mathbf{\mathit{2.51}}\text{.......uF}$$

Tweeter :

$$L1 = \frac{Z_t}{2\pi \times F_h \times 10^{-3}} \times \mathbf{n} = \frac{7.38}{14.14} \times 2 = \text{.......}\mathbf{\mathit{0.522}}\text{......mH}$$

$$C1 = \frac{1}{2 \times \pi \times Z_t \times F_h \times 10^{-6}} \times \mathbf{n} = \frac{1}{0.05657 \times 7.38} \times .5 = \text{........}\mathbf{\mathit{2.39}}\text{........uF}$$

* Notes: 1. Use value of R4 if using an impedance equaliser – see page 30.
2. '**n**' is the multiplier for the different alignments – see Table 1 page 16.

Fig 20.

Inductors.

Inductors are coils of copper wire wound onto **bobbins**, or coil formers. See **Fig 21** below for details. The inductance of a coil depends on the number of turns, as well as its cross sectional area and ranges from about 0.001 to 6.5 mH. This is one of the more difficult areas of speaker construction and that is the winding of the coils. To prevent insertion losses and to be able to withstand high powers, 1mm enamelled **copper wire** is used and all data and tables refer to this diameter. The best place to get copper wire is from your local motor re-winding works and about 4kg is about enough.

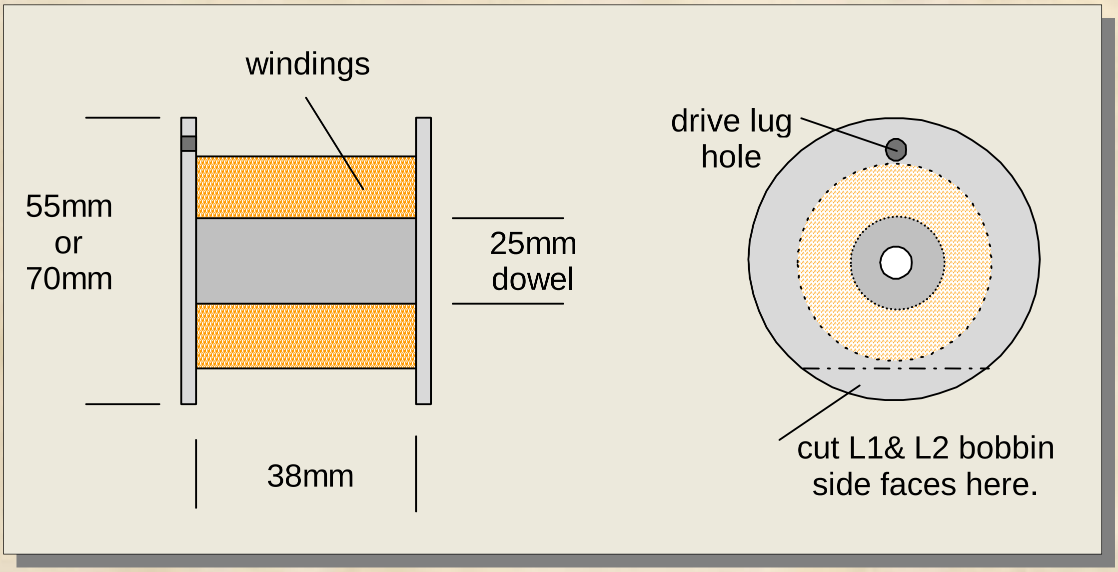


Fig 21.

Winding coils.

I have not found a source for the plastic **bobbins**, so I make mine out of wood. Cut the discs from 4mm plywood, about 55mm for L1 and L2 and 70mm for L3 and L4, using a hole saw fitted to your drilling machine. The centre part of the bobbin is 25mm dowel stick or broom handle, with a hole through its centre the same size as that of the hole saw's drill and cut to a length of 38mm. At this stage I must point out that the bobbin must have ***no metal parts*** to it, as this affects the inductance.

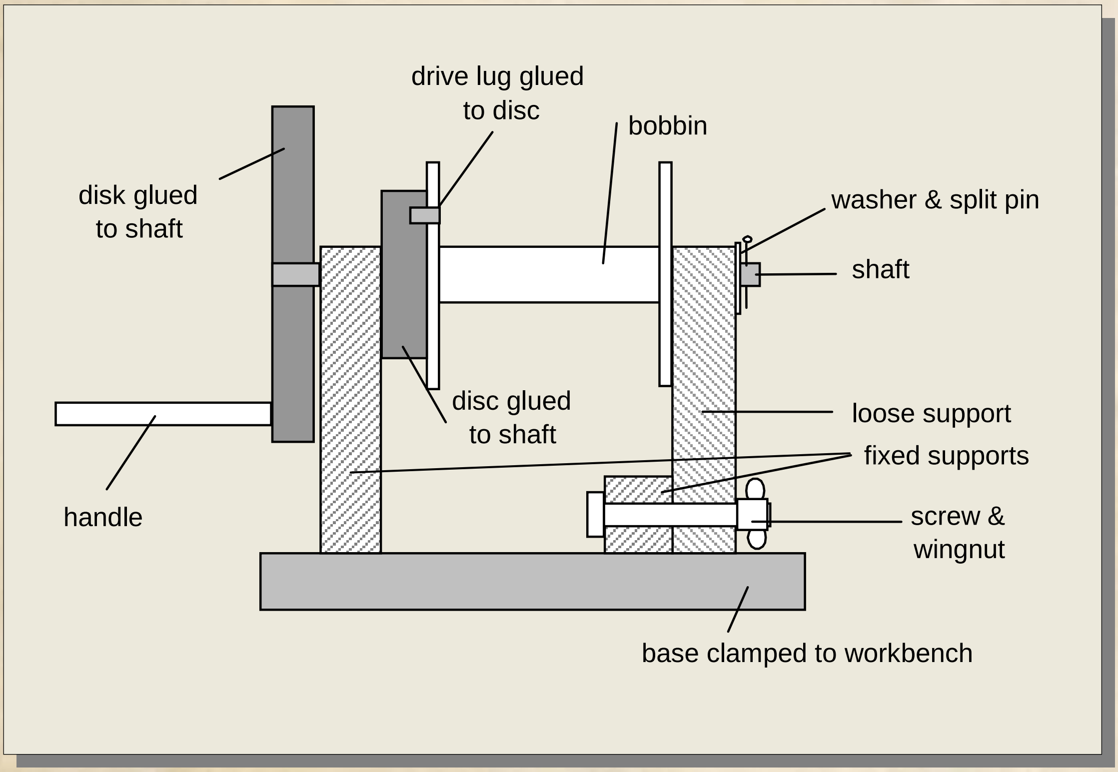


Fig 22.

Assemble the bobbin by gluing the two discs to the centre former and temporally bolting it together until it is dry. Drill a locating hole on one cheek or side that will engage with a driving lug (a dowel) fitted on a winding jig. I had to make a winding jig to help me wind coils and you can easily make one yourself from scrap wood, see **Fig 22** above. When you start winding, make sure that **all** coils are wound in the same direction and from the *same side*, that is, decide if you are going to wind them *clockwise* or *anti-clockwise* or start from the *left* or *right*. Before starting to wind the wire on, score a groove on the inner face of the bobbin at a tangent from the centre, outwards to the edge, so that the first winding starts as close to the side as possible. The wire actually lies in the groove, and make the start wire long enough, ± 120mm, as it is has to reach the **terminal block** where all the connections are made. Also remove about 10mm of the enamel from the end of the wire and then bend it over the edge of the bobbin side, so that it doesn't get in the way when the winding process starts.

The following formulae are used to determine the inductance and number of turns:

$$N = \sqrt{L \times \left(\frac{3m + 9w + 10h}{8.87 \times m^2}\right) \times 10^6} \quad \text{turns.}$$

m = **mean diameter.** **W = width of bobbin.** **H = height of windings.**
N = number of turns **L = inductance in milliHenries.**

All dimensions in mm.

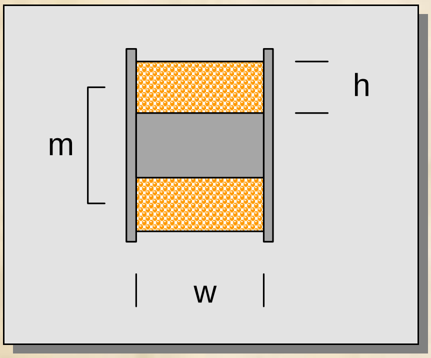


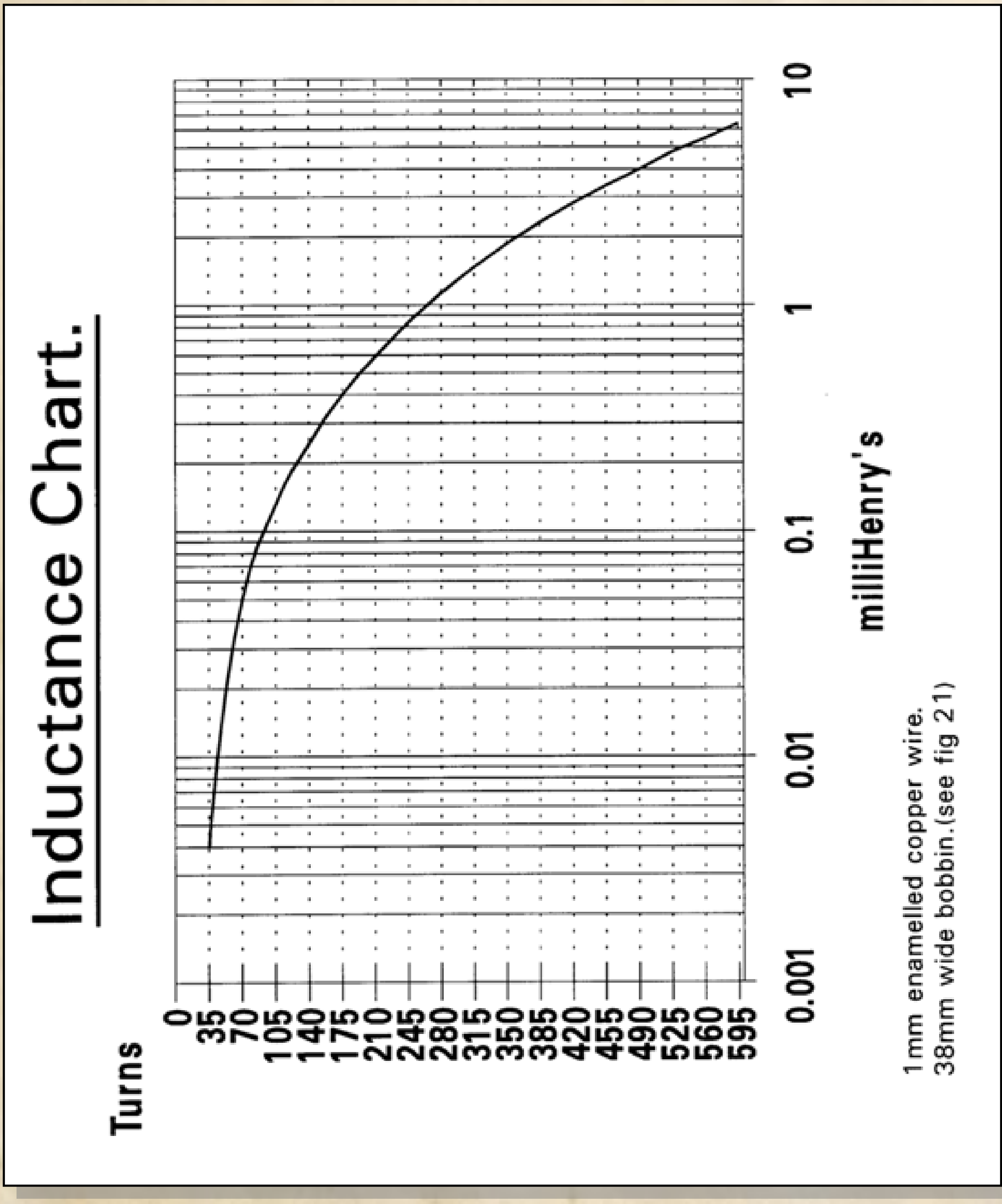
Fig 23.

Note:
In practice, the actual number of windings is about 5 % less than the ideal, as determined in the formula, due to the difficulty in keeping the windings neat and close wound, so be prepared to add extra windings.

$$L = \frac{7.87 \times m^2 \times n^2}{3m + 9w + 10h} \times 10^{-6} \text{ mH.}$$

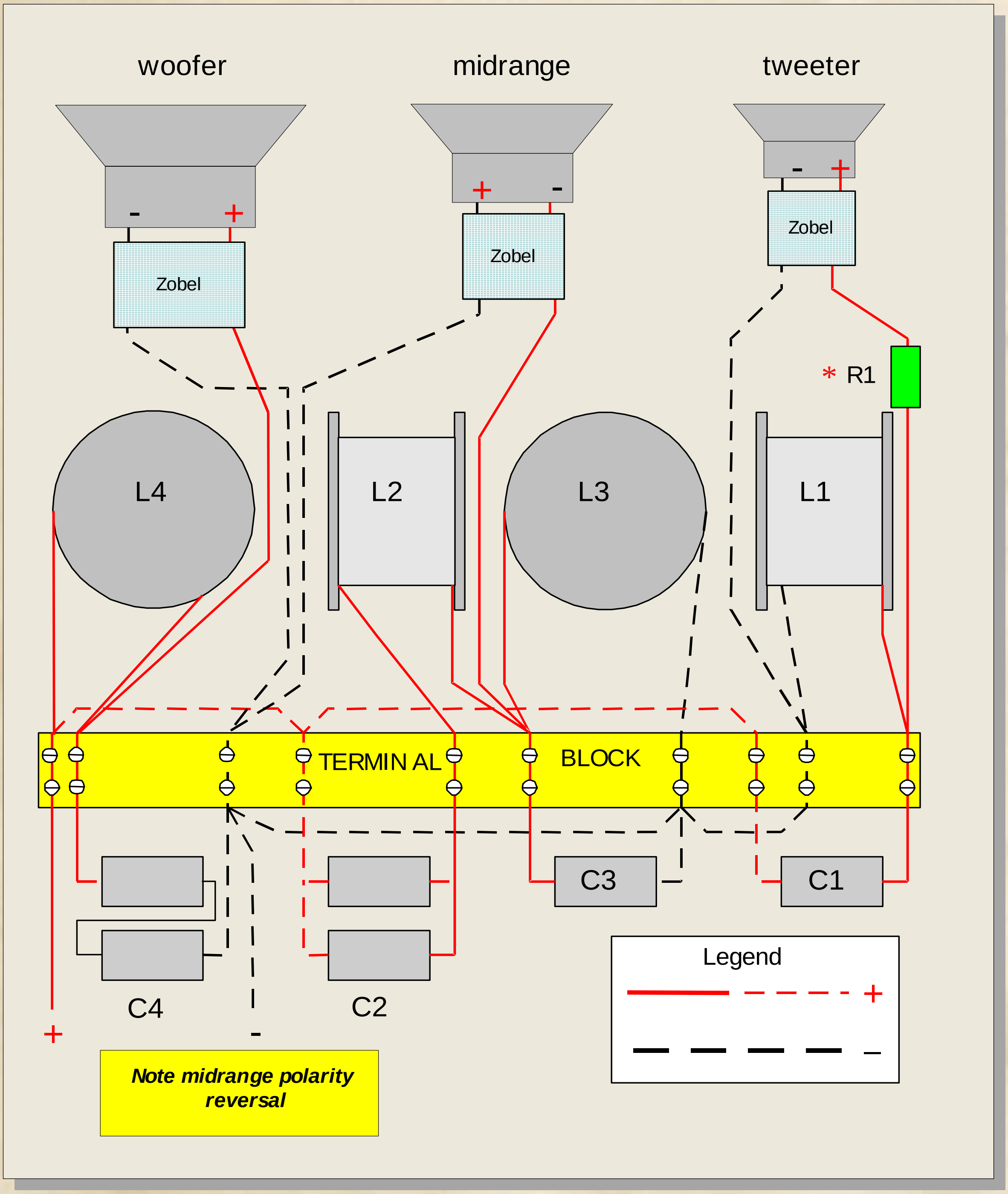
Refer to the "Inductance chart", Fig 24, for the number of turns required. When winding the coils, make sure that the windings are tight and close wound, that is, there are no gaps between the windings, do not wind them on randomly. My DMM has the facility for measuring inductance, so I wind on the number of turns, as determined from the chart. Then, connecting one lead of my DMM to the cleaned start winding and using a sharp object like a pin or knife blade with the other lead connected to it, I pierce the enamel to get a reading. I then add or remove wire as required. Don't worry if you do not have a DMM that measures inductance, the chart is accurate enough. To keep the windings from coming undone, wrap fibreglass insulation tape, or any similar high temperature tape, around the coils on the bobbin. Also mark each coil with its position and value, to avoid confusion when assembling the network later on.

Tip: have a pad and pencil handy with the number of rows you're going to wind written down in a column. As you finish winding each row, cross off its number on the sheet of paper, believe me things get hectic and it's easy to lose count. Hopefully you'll have an assistant to help out. Now where's my wife gone!



Inductance Chart.
Turns
0
35
70
105
140
175
210
245
280
315
350
385
420
455
490
525
560
595
0.001
0.01
0.1
1
10
milliHenry's
1mm enamelled copper wire.
38mm wide bobbin.(see fig 21)

Fig 24.



* R1 omitted if a Zobel filter is fitted.

Fig 25.

Assembling the crossover networks.

The easiest way that I found in assembling the network, was to glue the coils onto a piece of wood and to use a chocolate block, that is, one of those long plastic terminal blocks with a double row of screws, to connect the wires to. See Fig 25 and Fig 12 for a general layout. To prevent **magnetic interaction** between adjacent coils, mount them at right angles to each other about 25mm apart. Start by hot gluing L4 face down, then L2 standing upright, (if you look at Fig 21 you will see that bobbins L1 and L2 get segments cut off the side discs, so that they may stand upright) and then L3 face down, with L1 standing upright. The capacitors are mounted directly to the terminal block, with jumper wires connecting the components where necessary. The whole assembly then gets screwed down on top of the damping material, which acts as a shock absorber, at the bottom of the speaker cabinet. You can also use point to point wiring if you wish, but I try and avoid having to solder the joints, as they tend to become brittle and come loose after a while.

You will notice that C2 consists of two capacitors in parallel, a 15 uF and a 10 uF giving a total of 25 uF. I solder the one to the other before connecting to the terminal block. R1 is a damping resistor sometimes used for controlling the rise in impedance of the tweeter as the frequency rises and is normally 2R2 ohms @5w, but rather use a **Zobel impedance filter**.

Capacitor and resistor combinations.

Sometimes it is the value of available capacitors that will determine the crossover points for the crossover network. I juggle the figures around so as to obtain a capacitor value as close to that of a standard value. For example you may only be able to buy 2µf2, 2µf7, 3µf3, 4µf7 in polyester or polypropylene but we need up to 48µf and I couldn't bear to parallel ten 4µf7 caps to get this value. Bipolar (non-polarised) caps come in values ranging from 2µf2, 4µf7, 22µf, 47µf, 100µf and 220µf and using my calculator, I try to get as near to the value required with as few parts as possible, by putting them in series or parallel. Remember capacitors in parallel add in value, as do resistors connected in series. Resistors in parallel decrease in value as do capacitors connected in series. With capacitors in series, the total value will be less than the lowest capacitor, which also applies to resistors connected in parallel☺.

To work out the values required, use the following formulae:

For parallel capacitors: $C_T = C_1 + C_2 \ldots\ldots + C_6$

For series capacitors: $C_T = \dfrac{C_1 \times C_2}{(C_1 + C_2)}$ or $\dfrac{1}{C_T} = \dfrac{1}{C_1} + \dfrac{1}{C_2} + \dfrac{1}{C_3} \ldots + \dfrac{1}{C_6}$

For parallel resistors: $R_T = \dfrac{R_1 \times R_2}{(R_1 + R_2)}$ or $\dfrac{1}{R_T} = \dfrac{1}{R_1} + \dfrac{1}{R_2} + \dfrac{1}{R_3} \ldots + \dfrac{1}{R_6}$

For series resistors: $R_T = R_1 + R_2 \ldots + R_6$

For example, I needed C5 to be 39µf, so reworking the formula and solving for C_x gave me the following values:

(When selecting values, try the nearest value *above* the one you require, which in my example was 47µf.)

$$C_x = \frac{39 \times 47}{(47 - 39)} = 229\mu f$$

taking 220µf as the closest to this and reworking in the formula gives:

$$C_T = \frac{220 \times 47}{220 + 47} = 38.72\mu f.$$

So, a 47µf cap in series with a 220µf cap gives 38,72µf which is close enough to 39, considering that the tolerance of the caps is between 10% and 20%. Follow the same procedure for calculating the resistors and you usually need only two connected in either series or parallel. For example, R_a = 7.2Ω, so use a 3.9Ω and 3.3Ω in series or for R_b = 2.23Ω, use a 6.8Ω resistor in parallel with one of 3.3Ω giving a total of 2.22Ω, also close enough considering the tolerance of the devices. Non polarised electrolytic capacitors have high internal resistance, called high **ESR**, which supposedly affects the frequency response and is why some people prefer polypropylene types, which are very expensive. To overcome this, connect a 0.1µF/100v polyester capacitor in parallel with the bipolar cap, usually where the signal goes through the cap i.e. at C1 and C2.

Scaling Factor.

An easy way to change values in a crossover network is to use a scaling factor on each element in the filter:

$$\text{Scaling factor, } Sf = \frac{\text{n1 (old value)}}{\text{N1 (new value)}} \times \text{Element2}$$

For instance, say you plug your data into the crossover filter formula and get a capacitor value of 18.42 μF (element1) with a coil of 5.50mH (element2) at 500Hz and you have a 22μf capacitor (a standard value), then:

$$Sf = \frac{18.42}{22} = 0.8373$$

We now scale the inductor:

$$Sf \times \text{Element2} = 0.8373 \times 5.5 = 4.6\text{mH}$$

To check that the values are correct, use the frequency formula:

$$F = \frac{1}{2\pi \times \sqrt{(L \times C)}} \text{ Hz.}$$

$$= \frac{1}{6.28 \times \sqrt{(4.6 \times 10^{-3} \times 22 \times 10^{-6})}} \text{ Hz.} = 500\text{Hz.}$$

Zobel Filters

If you find that the woofers impedance curve is not flat and rises steeply with an increase of frequency on the impedance curve chart, Fig 16, then you should consider fitting a **Zobel impedance equaliser**, consisting of a resistor R4 in series with a capacitor C7, both in parallel with the speaker. See Fig 18. Calculate the speakers' inductance, L_e first before calculating the capacitance. Don't scale these values, as the inductance L_e is a constant value at its own frequency.

Remember to watch the voltage rating of the capacitors and should you have ones with a low voltage rating, putting them in series allows them to be used at a higher voltage. For example, suppose you need 50μF at 100v and have a couple of 100μF at 50v in your spares box, then putting two in series will give you 50μF at 100v.

Summary.

1. Use non-polarised capacitors with a voltage rating high enough for your amplifier.
2. The charts are based on 1mm diameter enamelled copper wire.
3. Coils are all wound in same direction starting at the same side of coil former – see phase dots on schematic diagram. Phase dot equals the start of a winding.
4. Mount adjacent coils at right angles to each other, with sufficient space between them to prevent magnetic interaction.
5. Fit an impedance equalizer to woofers that have a steep rise to their impedance curve.
6. Fit a 100n polyester capacitor in parallel with bi-polar types to lower the ESR (equivalent series resistance), although I've never tried this.
7. You can spray your finished coils with clear lacquer to hold them together and prevent magnetic resonance between windings which can cause buzzing in the coil.

Shelving networks

Sometimes it may be difficult in matching the sensitivities of the woofer and tweeter or it may be that you already have a tweeter with a sensitivity higher than that of your woofer. It may also be a case of your tweeter sounding too bright or loud. The choice then would be to fit an L-pad or make your own voltage divider that you insert between the crossover and Zobel network or the speaker itself if no Zobel network is fitted, see Fig 27.

I have seen on certain websites, the mistaken impression that ‘The woofer should have the lowest efficiency, since it will require the most power,’- wrong, wrong, wrong: the lower the woofer efficiency the lower its output will be and conversely the higher its sensitivity the more output it will give for less input power, so get the highest efficiency one, not the lowest one. Just remember that to get a 3dB increase from the speaker requires doubling the amplifier power! Tweeters, especially dome ones will, by the nature of their construction have higher sensitivity ratings than woofers that have springy surrounds and heavier cones, so choose woofers with the highest sensitivity and match the tweeter to the woofer.

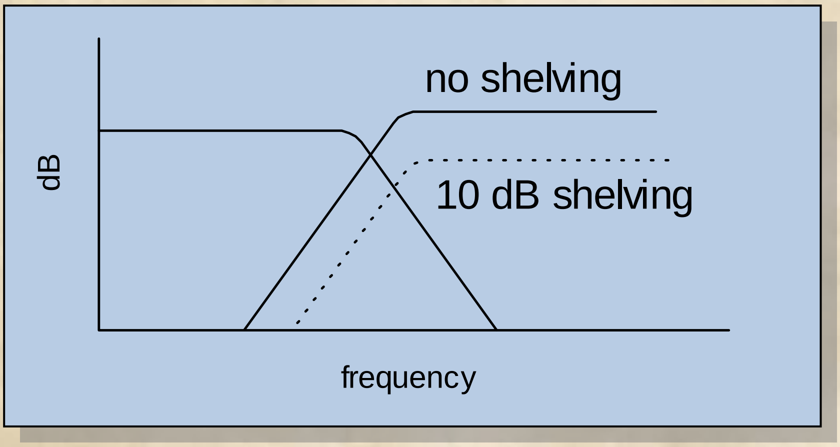


***Fig 26**. The effect of a shelving network on a 12dB crossover*

Looking at Fig. 26, notice how the crossover point has shifted due to the sensitivity mismatch and is also the reason why you never mix 4Ω and 8Ω speakers.

The mathematics for calculating the resistor values in the shelving network is far too involved for the style of this book but as I mentioned previously, the free spreadsheet will calculate this for you and to my mind it would be better to match the sensitivities of your drivers than to mess around fitting a shelving network, or just easier fitting a custom L-pad potentiometer.

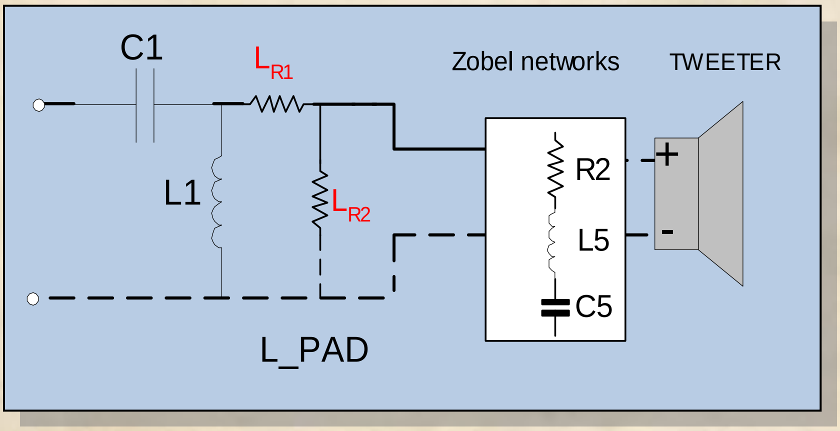


Fig 27. *Shelving resistors in a crossover network*

Insertion losses

The one disadvantage of passive crossover networks, are the insertion losses due to the resistance of the inductors, especially that of L4 in the woofer branch of the crossover. For example, an insertion loss in a 100w system of just 1dB means that the amp will deliver only 80w of power. Why is this you may ask, well it is to do with the resistance of the inductor or coil. As an example an inductor of 4.6 mH has a resistance of 1.2 ohms.

$$dB = 20 \log 10(R1/R2) + 1)$$

This gives an attenuation of -1.13 dB or a loss of 23w. As an aside, L4 in a 2-way network has a coil inductance of 0.86 mH and a resistance of approximately 0.52 ohms. This gives us an attenuation of -0.58 dB or a loss of only 12 w.

Summary.

1. If you cannot match the sensitivities of your drivers then you will need to fit an L-pad or attenuator network if the tweeter is too bright.
2. Inductors in passive crossover networks introduce insertion losses in the system, especially in the woofer branch where maximum power is required.

In some ways, this is the most important test of all. Learn to listen critically and you will get more out of all your listening.

David B. Weems

Testing the crossover points

The first test is a visual one, making doubly sure that everything is assembled correctly on the crossover pcb, if you're using one that is, or connected as per Fig 25 if using a terminal block, refer to Fig 11 for details. Before connecting the drivers, measure each output terminal of the crossover network with a DMM set to measure DC resistance and make sure that the reading is not less than 6 ohms which would indicate a short circuit somewhere that will blow your amplifier.

The following test assumes that you have finished your cabinets and the drivers are fitted and secured. It's best to make your connection leads long enough so that the speakers can be connected to the crossover outside the box. Use the test setup in Fig 13 by connecting your crossover to the test board you hopefully made as per Fig 14. This is why I prefer to have a bench amplifier as part of the setup so that should anything go wrong, you don't blow your expensive amplifier.

You can either test the crossover as a whole by connecting everything at once or you can test the individual branches of the crossover. I just connect everything up and do a listening test with music and listen closely to each speaker. If you take the second option, start by connecting the woofer but instead of connecting the tweeter or midrange fit a 5watt resistor that is of equal resistance to their impedance.

Switch everything on and listen for any funny noise or smoke escaping from the components☺. If all is well sweep the signal generator up and down to verify that there is sound coming from the speaker. Now slowly vary the signal from half to two times the crossover frequency, say 250 Hz to 1 kHz, and measure the AC voltage across the woofer terminals. The voltage should start dropping just below the crossover point and drop rapidly above the crossover point.

Do the same with the midrange if your system is a 3-way system, but now disconnect the woofer and fit a 5w resistor of equal impedance. It should have a low voltage at half the lower crossover frequency, rising as the frequency increases, then levelling out above the crossover point. The tweeter is tested the same as the woofer (disconnect the midrange and fit a 5w resistor) but this time by varying the frequency from half to two times the upper crossover point, say 2500Hz and 10 KHz. For a 2-way system vary the frequency from 1 KHz to about 5 KHz. At this stage you can connect all the drivers and vary the signal generator over the full audio spectrum with the volume of the amp turned up. If there are any problems, sort them out and try again. Also check your boxes for leaks and excessive vibration while doing these tests.

Connect everything up and ensure that your speakers are installed airtight even if you have to apply some silicone compound to the sealing gasket. Once you have got everything finished just make sure that the speakers are connected in phase. In other words make sure that the positive wires go to the positive terminals. If there seems to be a drop in performance check the phasing by playing a constant 100 Hz tone through the system while standing dead centre between the speakers, about 2 metres in front. Get an assistant to change the wires around on one speaker and see if the volume increases or decreases. If it decreases then your system was correct and the wires should then be changed back again, they should be connected with the volume loudest which indicates that the system is in phase.

Then sit back and enjoy your masterpiece. My system has been running for over 15 years and I still get a kick out of listening to it, especially when people pop by and are amazed at the sound quality.

In conclusion, does one build a 2-way system or a 3-way one? My personal choice is the 2-way system with smaller woofers as in my Camm DynaTone speakers, discussed in The Ultimate Loudspeaker Design book. For the larger 12 inch speaker discussed in The Lost Art Of Loudspeaker Design, I would use a 3-way system. A thing to note is that in a 2-way system because of the higher crossover point of 2500 Hz, the woofer coil, **L4,** in the crossover network is lower in value to that of the coil in the 3-way system and thus has a lower resistance due to less windings. What this means is that the insertion loss is lower in the 2-way system or looking at it another way it means that a 2-way system is more efficient.

I don’t see the need for any fancy components or tweaking with oscilloscopes, I mean you’re not going to hear any squiggles that you see with some fancy

Fig 28. *A typical crossover with capacitors costing an arm and a leg.*

software plot. Besides it depends on the music you are going to listen to. Any loud rock will drown out any inconsistencies in the sound anyway. One thing I did find though was that a good system really shows up bad recordings and they sound terrible so don't be alarmed. If you want to spend bags of money on some fancy looking capacitors, that's your indaba. I built my system with stock over the counter components, okay I used polyester caps where the signal goes through the cap, C1and C2, Fig 11, and ordinary bi-polar caps where the signals are shunted to ground. The loudspeaker drivers were cheap Far East jobs and they still sound fantastic. I would love to buy a set of Seas or PHL Audio or Vifa drivers but the exchange rate is way too high for me here in South Africa. As for loudspeaker cables, the only thing to worry about is that they are thick enough to handle the current over the distance, so don't fall for the hype and bullshit claims for any superior components.

Summary:

1. Before testing the crossover, make sure that there are no short circuits.
2. Test the crossover with the speakers fitted to the cabinets.
3. Test the cabinets for rattles, buzzes and air leaks.
4. Test the crossover points with the test setup on page 23.
3. Make sure that both speaker cabinets are connected with the correct phase, i.e. make sure the positive wires go to the positive terminals or else there will be a drop in sound level.

References:

1. The Lost Art Of Loudspeaker Design, M. MacLeod
2. The Ultimate Loudspeaker Design, M. MacLeod
3. Great Sound Stereo Manual, D. B. Weems
4. Design Your Own Speakers - Practical Electronics, R. A. Honeycutt
5. An Introduction To Loudspeakers and Enclosure Design, V. Capel

Glossary.

- A -

AB test
A direct comparison between two pieces of equipment, or two design alternatives. Everything in the system is unchanged except the item under test, which is switched between the two alternatives. Also see ABX test.

Absorption coefficient
The sound absorption coefficient defines the fraction of sound energy absorbed by, for example, one reflection from a wall. Energy is proportional to the square of sound pressure, so for an absorption coefficient of 0.2, 20% percent of the energy is absorbed, reducing the sound pressure by 10.6%.

Acoustics
The effect of a room on music. Rooms are typically described as "dead" when they have a lot of sound absorbing stuff like drapes and heavy furniture in the room, which reduce the number of audible reflections (echoes). A room with bare concrete walls has tons of audible reflections and is extremely "live".

Acoustic Suspension
A type of loudspeaker enclosure design, made popular by the Acoustic Research AR3 among others. The air in a sealed enclosure acts like a spring, and is used as such in this design. The result is a lower resonant frequency, and lower bass, than would otherwise be obtainable. These designs also typically use an enclosure filling material that makes the enclosure act larger than it really is.

Active
Containing tubes or transistors that require an external source of power, and typically provide amplification

Aliasing
An effect that occurs when an analogue signal is digitally sampled. . Samples of a 10,000 Hz tone are identical to samples of a 34,100 tone. In general, samples of a tone of frequency f are identical (except for a phase shift) to samples of a tone of frequency Fs - f, where Fs is the sampling rate. Thus if f equals, say Fs/2 + f0, the digital recording will produce a false tone of Fs/2 - f0. Anti-aliasing filters are used

to eliminate tones above Fs/2 to avoid this effect.

Amp
Short for amplifier, and also the abbreviation of Ampere, the unit for electrical current. Electrical current is sent through the speaker input terminals to make the speaker produce sound. A good 120 watt amplifier can momentarily send a peak current of 80 amps through a speaker, which will really rattle your windows; maximum sustained average current at 120 watts, which is also very loud, is around 4 amps.

Amplifier
Same as power amplifier. Converts a low-power signal from a pre-amplifier into a high-power signal (tens to hundreds of watts) needed to drive a loudspeaker.

Anechoic chamber
A room where the walls, ceiling, and floor have been covered with acoustic absorbing material to (nearly) totally eliminate reflections. Useful for engineering development of loudspeakers since it isolates the performance of the speaker. But the response in a real room will be very different.

- B -

Bass-reflex
See vented enclosure.

Bi-amplified
Using two stereo power amplifiers. For a typical system with two tweeters, midranges and woofers, one amplifier drives two woofers, and the second amplifier drives the midranges and tweeters.

- C -

c
The speed of sound, 344.4 meters per second at 1 atmosphere pressure, 20^0 C; 347.3 m/s @ 25^0 C; very little change with pressure.

Capacitor
A component in crossover networks. Available at electronic supply stores. Capacitance is denominated in Farads. A cousin of a resistor, but presents a high resistance to low frequencies, and low resistance to high frequencies. Perversely, more capacitance means less resistance.

Cascade
To add a second element to a filter or connect the output of one network to the input of a second network.

Circuit
A conglomeration of electronic components. More or less interchangeable with "network", but networks are usually small, tidy and passive, and circuits tend to be big, messy and active.

Class A
A type of amplifier design that generally produces less distortion than class AB, or B amplifiers, but with lower efficiency lower power output.

CLIO
A PC based audio measurement system made by Audiomatica. Includes a calibrated microphone and software. This system can measure impedance, component values of inductors and capacitors. Most important, the time-domain and frequency response, both amplitude and phase, of drivers and systems can be measured. It also measures room reverberation time, SPL sound levels, and sundry other stuff. This was an absolutely essential tool in developing my system, and I highly recommend it to anyone serious about building sound systems

Coloration
An alteration of the character of music due to imperfect frequency response. Can make music sound shrill, dull, boomy, etc.

Comb filtering
A filter that reduces the frequency response at a fundamental tone and all of its odd harmonics. If a signal is combined with its duplicate, except that the duplicate is delayed by a path-length difference d (a time-delay of d divided by the speed of sound) the two signals will cancel at all frequencies where d is an odd multiple of half of the wavelength. For example, if d=12 inches, cancellation will occur at wavelengths of 24, 8, 4.8, etc. inches, corresponding to frequencies of 565, 1695, 2825, etc. Hz. If these frequencies are plotted as vertical lines on a linear frequency scale, it will look something like the teeth of a comb. There is a doubling of the sound pressure at the frequencies between these teeth. The overall result is a roller-coaster frequency response.

Cone
The cone-shaped part of a loudspeaker driver that moves the air. Made of paper,

polypropylene, or recently out of high-tech materials such as Kevlar.

Convolution
A mathematical process. To convolve a 128-point HRTF with a sound wave, both functions must be sampled at the same rate, e.g. 44,100 samples per second. The 128 HRTF samples then cover a time period of 2.9 milliseconds. The HRTF samples are first flipped in time. They are then aligned with the first 128 sound wave samples. Paired samples are multiplied together, and the 128 products are added together to create one new HRTF-processed time sample. The HRTF is then shifted one notch to align with sound samples 2-129, and another multiply and add is performed. This shift, multiply, and add is repeated until you run out of sound samples. Since a 5-second sound segment has 220,500 samples, we are talking about 28 million multiplications!

Critical Band
The ear behaves as if it contains a bank of filters, each filter passing frequencies within the critical band. Above 1 kHz the critical bandwidth is close to 1/6 octave. Below 100 Hz it is approximately 30 Hz. Two tones separated by more than a critical bandwidth are received by the cochlea more-or-less independently. Within a critical band a louder tone can make a softer tone inaudible, which is called "masking"

Crossover
An electronic network for dividing sound into high and low frequencies for reproduction by woofers, tweeters, etc. A crossover for a two-way system would consist of a high-pass filter that sends the high frequencies to the tweeter, and a low-pass filter that sends the low frequencies to the woofer. For a three-way system a band-pass filter is added to send the middle frequencies to the midrange speaker. The response beyond the crossover frequency is reduced by an amount that depends on the order of the crossover filters. A 1st order filter reduces the response 6 dB per octave, a 2nd order 12 dB per octave, a 3rd order 18 dB per octave, and a 4th order 24 dB per octave.

Crossover frequency
The nominal dividing line between frequencies sent to two different speaker drivers and occurs at the -3 dB point.

- D -

Damping factor
A characteristic of amplifiers; 8 Ohms (the nominal impedance of common

loudspeakers) divided by the output impedance. A large damping factor is allegedly good because it gives an amp tighter control over the loudspeaker. This is not true, and a damping factor of 1000 is really not better than a factor of 200.

dB
A logarithmic scale. A 3 dB increase means twice as much power; a 6 dB increase means 4 times as much. A 60 dB increase is one-million times as much, and a 120 dB increase is one-million times one-million as much power.

Diffraction
When a sound wave from a loudspeaker reaches an edge of the speaker cabinet, it creates a new source of sound that appears to emanate from the edge. This is called edge diffraction. Diffraction will occur any time a wave encounters an obstacle, or abrupt change in a surface. Rounding the corners of a speaker cabinet will reduce, but not eliminate, diffraction.

Diffuse-field Equalization
A manner of equalizing data, such as head-related transfer functions, such that the power spectrum average over all directions is uniform. This eliminates measurement effects which are a function of frequency, but not a function of the direction of the incoming sound.

DMM
Is a digital multimeter

Dome
Same function as a cone, but shaped as it is named. Used for many tweeters and some midranges. Made of fabric, aluminium, titanium, or other high-tech material.

Doppler distortion
Harmonic and intermodulation distortion caused by the motion of a loudspeaker cone. This distortion is a result of the change in pitch of the high frequencies as the woofer cone moves in and out.

Doppler shift
A change in pitch caused by a relative motion between the sound source and an observer. The pitch is higher when the source and observer are getting closer, and

lower when they are moving apart.

Driver
"Driver" always refers to a single naked speaker (e.g. a woofer or a tweeter) rather than an ensemble of drivers, enclosure, and crossover network, which is a loudspeaker. The primary parts of a driver are (1) the cone, which moves the air, and (2) the voice coil and magnet, which moves the cone.

e
A magical mathematical number equal to 2.71828......... It is an irrational number, so unlike rationals, the decimals run on forever, never forming a repeating pattern. One form of Euler's equation, $e^{i\pi}+1=0$, relates 5 special numbers.

- E -

Electrostatic Speaker
A loudspeaker that uses a large, flat (sometimes curved), thin membrane instead of a cone. The membrane is situated between two screens that are charged with a high electrical voltage to move the membrane. Ideally the entire membrane moves back and forth as a unit. The large area and low moving mass are advantages. The limited membrane travel seriously limits the low frequency response. Also presents a difficult load impedance to an amplifier.

Indaba
An African council meeting where important matters are discussed. Your choice or answer to a decision related to a question or problem.

- F -

Filter
An electronic circuit designed to selectively pass certain sound frequencies. Also see crossover.

Feedback
Positive feedback in an amplification loop causes the output signal to grow uncontrollably, and results in the annoying squeal when a microphone gets too close to a loudspeaker. Negative feedback is a technique intentionally employed in typical amplifier designs to reduce distortion. Hence, positive feedback is a negative quality , and negative feedback is a positive quality (some folks dispute the latter).

If engineers didn't make things hard to understand we wouldn't get paid so much.

Fletcher-Munson Curves
Curves originally obtained by two Bell Lab scientists relating subjective sound levels to objective levels of pressure/power.

Fourier transform
A graph of sound pressure vs. time is an illustration of a time-domain function. Sound can also be described as a frequency-domain function, as the superposition of a group of sinusoidal waves spanning the audio frequency spectrum. Mathematically both functions contain exactly the same information and each function perfectly defines the sound. A Fourier transform, and inverse transform, convert functions back and forth between the time and frequency domains. This is a error-free transformation, but the width of the frequency range and frequency resolution (fine-grain detail) is limited by the time resolution and duration. The longer the duration, the finer the detail, and the finer the time detail, the wider the frequency range. A fast Fourier transform is a technique for performing these mathematical operations very quickly with a computer.

Frequency
The rate at which sound pressure varies. Denoted by f and denominated in Hertz=cycles per second; abbreviated Hz, (or kHz, meaning thousands of Hz). A 1000 Hz (1 kHz) tone pushes your eardrum back and forth 1000 times each second. The lowest bass note on an organ keyboard is a rumbling 16.4 Hz; the highest is a squeak of 18,794 Hz, which is inaudible to most males over the age of 40. These sample tones are audible with good loudspeakers or headphones, but are not audible with many computer speakers: a 100 Hz tone (12 kb wav file);and a 10kHz tone (44 kb wav file). Frequency can also be defined mathematically as the derivative of phase with respect to time, in which case it is usually denoted by ω and denominated in radians per second. $\omega=2\pi f$.

Frequency band
A specific range of frequencies. For example a typical subwoofer band is 20 Hz to 100 Hz.

Frequency response
The response of a system to a fixed input voltage, at different input frequencies. Same as frequency domain response.

- G -

Group delay
Mathematically group delay is the derivative of the phase, in radians, with respect to frequency. Filters cause such a delay. Group delay can also mean an average of this delay over some frequency band.

- H -

Harmonic
A tone whose frequency is an integer times the frequency of the fundamental (lowest) tone. Every note played on a musical instrument consists of a fundamental tone plus many harmonics.

Harmonic distortion
Spurious harmonics produced by the sound system. Usually not quite as objectionable as intermodulation distortion because harmonics occur naturally in music. See the discussion in the section on music and ears.

Head-related transfer function
The sound wave that reaches the eardrum is quite different than the sound wave travelling through the air towards the ear. The incident sound is modified by reflections from the head, pinna, and within the ear canal. These effects are mathematically represented by the head-related transfer function (HRTF). An impulsive sound will reach the eardrum spread out over time, and this function of time is the HRTF.

High-end Audio
Equipment that is always expensive and pretentious, and sometimes excellent.

- I -

i (or j)
The basic "imaginary" number equal to the square root of minus one. Multiplying a quantity by i is equivalent to adding a 90^0 phase shift. Quantities involving i are called "complex" numbers, and if you don't already know about this stuff it is a bit complex.

Impedance
The technically correct term for the resistance produced by inductors, capacitors, and networks. For the technically minded, the real part of an impedance is equal to resistance, and the imaginary part is called reactance.

Impulse response
The response of a system to a "spike" input that abruptly rises from zero and then abruptly decays back to zero. The response of the system to any input can be predicted from the impulse response. The impulse response measured in a room also shows a series of echoes of the direct response, which are reflections from walls, etc.

Incoherent addition
Sound from two loudspeakers can combine coherently or incoherently. This is generally different at different spots in the room, and the following refers to one specific spot. Coherent addition means all of the pressure peaks and lows of the sound wave from one speaker arrive at exactly the same time as the peaks and lows of the other. The pressures then add. Twice the pressure means 4 times the acoustic power, so the sound level increases 6 dB. Incoherent addition means the peaks and lows from one arrive at random times compared to the other. In this case the power adds and the sound level increases 3 dB. For frequencies below 100 Hz, where the sound tends to be monaural, and the distance between pressure peaks and lows is more than 5 feet, addition tends to be coherent. At higher frequencies addition tends to be incoherent.

Inductor
A component in crossover networks. Available at electronic supply stores, or can be made by winding wire in a doughnut shaped coil. Inductance is denominated in Henries. A cousin of a resistor, but presents a high resistance to high frequencies, and low resistance to low frequencies. More inductance means more resistance.

Infinite baffle
A speaker enclosure that is completely sealed except for openings occupied by the driver cones. The enclosure performance is similar to mounting the drivers on a panel of infinite extent, which explains the name. Also called a sealed enclosure, which is actually more accurate.

Inter-modulation distortion
Spurious tones created by the sound system. The frequency of a spurious tone equals the frequency of one tone in the music added to or subtracted from the frequency of a second tone in the music.

- J -

j (or i)
The basic "imaginary" number equal to the square root of minus one. Multiplying a quantity by j is equivalent to adding a 90^0 phase shift. Quantities involving j are called "complex" numbers, and if you don't already know about this stuff, it is a bit complex.

- K -

k
Abbreviation for 1000. 1kHz = 1000 Hz. Alternatively, frequently used as the

algebraic symbol for the propagation constant.

Kinetic energy
Energy due to motion. Proportional to an objects mass and the square of its velocity. As a car accelerates, energy from burning gasoline is converted into kinetic energy. To slow a car down, the kinetic energy must be converted into heat by the brakes.

- L -

Linear
If the input amplitude is increased by a factor of, say 1.2, then the output amplitude increases by a factor of 1.2, no more and no less. This is the ideal for all parts of a sound system, since no distortion is introduced.

Load
Whatever is connected to the output of an amplifier or electronic circuit.

Litz wire
Wire woven in a pattern to reduce the "skin effect." For ordinary stranded wire each strand tends to stay the same distance from the wire centre. The magnetic field produced by the current in the wire effects the strands differently depending on this distance, causing the skin effect. In Litz wire the strands weave in and out, such that all strands experience almost the same magnetic field, and carry nearly equal current.

Lobing
A vertical array of two or more drivers will beam energy in a series of vertical lobes, over a portion of the frequency band.

Loudspeaker
Converts electrical energy to sound energy. This can mean an entire system such as a woofer, midrange, tweeter, crossover network, and enclosure. It can also mean an individual speaker driver such as a woofer.

- M -

Matlab
A software package for engineering analysis that does just about anything an engineer could desire. Lots of graphics capability, full kit of analysis tools. Produced by Mathworks (Cleve Molar, a founder of Mathworks, was a classmate of mine at Caltech - too bad I didn't have his entrepreneurial spirit).
Midrange
A speaker driver 2-5" in diameter for reproducing the middle range of sound frequencies.

Minimum phase filter
Not an easy thing to explain. A given amplitude response can in theory be produced by an infinite set of filters, each of which creates a different phase response. A phase response that is a non-linear function of frequency smears the response in time. The filter that produces the amplitude response with the least time-smearing is the minimum phase filter of the set.

Mode
Characterized by having a specific pattern (e.g. of sound pressure). Some sound features, like wavelength, can have any value. With modes there is no halfway; there are modes 1,2,3, etc., but no mode 1-1/2.

- N -

Network
A bunch of electrical components hooked together.

Non-linear
Anything other than linear. A non-linear relationship between the input and output of an amplifier, loudspeaker, or whatever creates harmonic and intermodulation distortion.

- O -

Octave
A one octave increase in frequency means doubling the frequency. A note one octave higher than another sort of sounds the same, but at a higher pitch. 1/3 octave smoothing means averaging the frequency response over 1/3 octave intervals. This smoothes out (i.e. obscures) abrupt changes in frequency response.

Oscilloscope
A piece of electronic lab equipment with a screen similar to a television screen, which shows an instantaneous picture of the voltage of an electrical signal. When connected to the output of an audio amplifier the picture is an analogue of the sound pressure coming out of the loudspeaker

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- P -

Passive
Not requiring an external source of power; containing resistors, capacitors, inductors, etc., but no tubes or transistors; provides no amplification.

Pentode
See valve.

Phase
A measure of how well two waves are aligned. "In phase" means the crests align perfectly and add together; "180^0 out of phase" means crests align with valleys and the waves cancel each other. A system has a flat (i.e. good) phase response if the relative phase of the input and output signals is the same for all frequencies (actually a linear variation is equally good). This is particularly important for good transient response.

Phon
The unit of subjective sound level. Defined such that the sound level in phones

equals the level in SPL for a 1000 Hz tone. Generally used in tests where listeners are asked when a tone at a different frequency sounds as loud as a 1000 Hz tone, to define the subjective frequency response of the ear.

Pi (π)
Another magical mathematical number, 3.14159....... It is an irrational number, so unlike rationals, the decimals run on forever, never forming a repeating pattern. Entire books have been devoted to Pi, which has roots far deeper than its relationship with circles.

Pink noise
Pink noise is random noise where the power is spread uniformly over a specific spectrum of frequencies, such as 20-20,000 Hz for audio. It sounds like a hiss, much like tuning between FM stations. For a theoretically infinite spectrum it is called white noise.

Pinna
The exterior part of the ear. The ridges are important in determining the direction sound comes from.

Placebo effect
If you believe something will make an improvement, it will make an improvement (usually). A scientific fact so well documented that trials of new drugs are always blind-tested vs. a placebo (something with no "real" effect). The fact that a significant percentage of people taking the placebo are always cured is a rather remarkable proof of the power of the brain. It follows that one is often happier as a stupid believer than as a wiseacre cynic.

Port
A tube inserted in a vented enclosure. Reinforces the sound of the speaker.

Power Spectrum
A power spectrum plot shows the relative levels of power at different frequencies. The power around 4 kHz is about 20 dB below (1% of) the power at the lowest frequencies, and power drops another 20 dB for frequencies above 12 kHz.

Pre-amp
Preamplifier. Converts an electrical signal from a CD player or other music source into a stronger signal to drive a power amplifier. Also controls the source selection, volume, and treble and bass response.

Propagation constant
The mathematical term which defines wave propagation. It is equal to 2π divided by

the wavelength.

Push-pull
A type of amplifier design. A tube or transistor that outputs positive voltage swings is matched with a second tube or transistor that outputs negative voltage swings. This results in a symmetrical response for positive and negative voltages.

- Q -

Q
The "quality factor" of a network or system. The lower the losses, the higher the Q. In electrical circuits losses occur due to resistance. In mechanical systems losses occur due to friction. An example of a high Q mechanical system is a bell, which rings for a long time after it is struck.

- R -

Resistor
A basic component in electronic circuits that resists the flow of electrical current. Resistance is denominated in Ohms. Higher resistance results in less current. All electrical components (except superconductors) have some resistance, wanted or not. Most home speakers are nominally rated at 8 Ohms of resistance (4 Ohms for cars), but in reality the resistance is closer to 6 Ohms in most of the band, has a peak of 20-100 Ohms at resonance, and also rises at high frequencies due to voice coil inductance.

Resonance
A reinforcement of sound due to echoes adding in phase with the original source. In general, a motion or electrical or other response at a frequency where a system responds much more strongly than at other frequencies. The higher the system Q, the narrower the frequency band where the response is strong.

Reverberation
The echoes in a room that one hears after the original sound stops. The classic example is organ music in a cathedral. The usual measure of reverberation time, denoted RT_{60}, is equal to the time it takes the sound to decay 60 dB after the sound source stops, in seconds.

- S -

Scalar
A physical quantity that involves magnitude, but not direction. Examples are speed, temperature, and pressure. Quantities that also involve direction, such as velocity, are called vectors.

Scientific notation
A convenient way of writing very large or very small numbers. $2x10^{27}$ means 2 followed by 27 zeros. (The 27 should be a superscript; some web browsers don't get it right). Basically the decimal point is moved 27 places to the right, filling in any spaces with zeros. So $1.6x10^{27}$ means 16 followed by 26 zeros. $2x10^{-27}$ means moving the decimal point 27 places to the left, or a decimal point followed by 26 zeros and then the number 2.

Sensitivity (of a loudspeaker)
Sensitivity is normally defined as the SPL level produced at a distance of 1 meter from the speaker, with 1 Watt of input power. For example a drivers spec's would be: 89dB at 1w/1m. For a 3dB increase, the power from the amplifier must double, i.e. to 2w. Therefore to get to 120dB, the standard level, would require 1024w!

Single-ended
A type of valve (tube) amplifier design where the output stage is biased such that a DC current exists when the input voltage is zero. A positive (negative) input voltage causes the output current to increase (decrease) relative to the DC bias level. The output transformer does not respond to the DC bias current, so the final output is only the AC part due to the input voltage. This type of design does not respond symmetrically to positive and negative input voltages.

Skin effect
At very high frequencies electrical current tends to flow in a thin layer on the surface of a conductor, a layer of thickness roughly equal to the "skin depth" δ. The skin depth for copper at 20 kHz is about 0.5 millimetres (0.018 inches). The 16 AWG gauge wire I use for speaker connections consists of 19 strands of .28 millimetre (.011 inch) diameter wire. The overall diameter of the bundle is 1.5 millimetres (.060 inches). Assuming that the strand bundle acts like a solid wire of the same diameter (a good assumption), the skin effect for this wire is shown by a plot of the current [40.1 kb] magnitude as a function of wire radius. The blue curves are the exact Bessel function solution for a cylindrical conductor. The red curves are the exponential solution exp(-x/δ) for an infinitely thick rectangular slab. The Bessel solution is very close to the exponential solution for wires greater than 10 δ in diameter. However

when the thickness is of the same order as δ, the Bessel function solution exhibits much less of an effect. The effective resistance of the wire is increased by 3.6% and 13.2% at 10kHz and 20 kHz respectively. Since the total resistance of a 10 meter length is about 0.1 Ohm, this increase of about .01 Ohms is insignificant compared to the typical voice coil resistance of 6 Ohms. The wire inductance is slightly reduced by the skin effect, but is tiny compared to the voice coil inductance in any case. The skin effect can be reduced by using Litz wire.

Slew rate
A characteristic of amplifiers. The maximum rate of change of output voltage in response to a square-wave input, usually measured in volts per microsecond. The higher the better.

Solid-state
See transistor.

Speaker
Same as loudspeaker.

Spectrum
A range of frequencies; e.g. the standard audio spectrum extends from 20 Hz to 20 kHz.

SPL
Sound pressure level. A logarithmic dB scale is used, akin to the Richter scale for earthquakes. A 3-dB increase means doubling the power. Zero dB SPL is the threshold of hearing; a quiet room has a background SPL of about 40-dB; a loud rock concert can go to 120-dB; the threshold of pain is 135 dB. Peak sound pressure is 3 dB higher than root-mean-square (RMS) average pressure. SPL normally refers to RMS pressure. Also see sensitivity.

Subwoofer
Like a woofer but even bigger in diameter and lower in frequency.

- T -

Thiele parameters
A set of parameters that define the behaviour of a driver in an enclosure. Used for designing enclosures and analyzing the sound response

Time domain response
The response of a system measured over an interval of time. One can mathematically transform the time domain response to obtain the frequency domain response (and vice-versa), using a Fourier transform.

Transient distortion
Is the blurring of notes when the speaker encounters sudden changes in signal.

Transistor
The basic component of most amplifiers. A glob of magic stuff (semiconducting solid-state material) that amplifies electrical current. From smaller than a pin head up to an Oreo cookie in size (transistor case included). Reviled by some high-end types as having a harsh inhuman sound.

Triode
See valve.

Tweeter
A speaker driver on the order of 1" diameter for reproducing the highest sound frequencies.

- V -

Valve (tube)
Short for vacuum tube; a small glass bottle with all the air removed, containing metal sheets and screens (three for a triode, five for a pentode) to control electron flow. The basic component of amplifiers built prior to the 60's. Many high-end audio types swear they sound better than transistors. High power output tubes are usually pentodes; some folks think triodes produce a cleaner sound.

Vented enclosure
Same as Bass-reflex. A speaker enclosure with a tube or "port", which is a hole or a cardboard tube connecting the inside of the box with the outside world. The port generates sound in phase with the speaker driver, and generally extends the bass response.

Vector
A physical quantity that involves direction as well as magnitude. Velocity is a vector; it's magnitude is speed, and the vector points in the direction of travel.

Voice coil
The coil of wire in a speaker driver that conducts the electrical current from the amplifier. This current creates a magnetic field that interacts with the permanent magnet of the speaker to move the speaker cone.

Volt
Using an analogy of water flow, electrical voltage is analogous to pressure, and electrical current analogous to the volume of water flow. Voltage pushes current through a circuit. One volt will create a current of 1 amp through a 1-Ohm resistor. In general, current is directly proportional to voltage and inversely proportional to resistance. (Ohms law).

- W -

ω
The commonly used symbol for frequency in radians per second. Equal to 2π times the frequency in Hz.

Watt
A unit of power. A 200 watt amplifier can make the walls shake pretty well.

Wavelength
The distance between two crests (or valleys) of sound wave pressure. Usually denoted by the symbol λ. Varies from 56 feet for a 20 Hz tone to 0.7 inches for a 20 kHz tone. In general the wavelength in inches is equal to 13,543 divided by the frequency in Hz.

Window
A Fourier transform of a segment of a pure 400 Hz tone will produce a spectrum with a sharp peak at 400 Hz. But since the segment is finite, if the tone amplitude is constant, the spectrum will have "side lobes" that extend over a wide range of frequencies. The side lobes can be sharply reduced by contouring the amplitude with a "window."

White noise
A random noise used in measurements, as it has the same amount of energy at each frequency.

- Woofer
A speaker driver 6-15" in diameter for reproducing low sound frequencies.

- Z -

Zobel network
Generally used to mean a network that compensates for the voice-coil impedance. Without a Zobel the driver input impedance rises with frequency, which affects the crossover response. A Zobel flattens the input impedance by making the impedance of the circuit equal the voice coil dc resistance.

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Crossovers Revision History

Rev1.0

Added chapter on shelving networks.

Rev1.1

Fixed Lpad formula in spreadsheet at cell D14 to read: =D12/B21.
Tidied some drawings.

Rev1.12

Fixed spelling mistake at cell D8 ‘comment’ in spreadsheet .
Added some detail to 3-way schematic drawing.
Changed woofer description of sensitivity as seen on certain websites.

Rev1.13

Fixed component numbers in calculation sheets. Updated midrange formula to have ‘n’ factor. Fixed tweeter notch filter formula.