Principles and problems in loudspeaker design

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This month's article concludes with a discussion on cross-overs. The first of the Series 4000 speaker designs you'll find in the project section this issue!

LAST MONTH'S article dealt with the characteristics of a typical moving-coil direct-radiating loudspeaker and the interactions that are likely to occur with the loudspeaker enclosure. Once these problems are understood and the bass performance has been optimised, we are in a position to finish the design. I have discussed the bass end of the audio spectrum first not because it is the most important, but simply because it is the most difficult to optimise. The mid-range is by far the most critical part of the audio spectrum since it is midrange distortion that the ear objects to more than any other.

It was shown last month that drivers have limited frequency responses and that it is therefore necessary to use several drivers, each covering its own frequency range. By far the most common arrangement is the three-way, so called because it uses three drivers to cover the audio range. A woofer covers the bass end, crossing over to a midrange driver somewhere between 400 Hz and 1 kHz. The midrange driver, sometimes called a 'squawker', carries the frequency range from this crossover point up to where the tweeter takes over, usually around 3 kHz to 5 kHz. The tweeter covers the remainder of the audio spectrum up to around 18 kHz, about the limit of human hearing. A crossover is used to separate the input signal from the output of the power amplifier into the three frequency bands.

Passive loudspeaker crossovers

The design of the crossover for any particular group of drivers must be done only after a thorough investigation into the characteristics of the drivers has been carried out. It is essential to choose drivers with an adequate overlap in their frequency responses, or a 'hole' will result in the response of the final loudspeaker. The amount of overlap needed depends on the slope of the filters used in the crossover. If a fast slope is used a smaller overlap is required, but filters with very fast slopes are complicated and expensive.

The basic crossover filter consists of a low pass and a high pass section. In a two way loudspeaker only one of these sections would be used, while in a three way loudspeaker two sections are used, one for the bass-mid crossover and the other for the mid-treble crossover. The simplest crossover is called a first-order crossover and has a slope in its attenuating region of 6 dB/octave. An octave is a range of frequency such that the highest frequency in the band is double the lowest frequency; for instance, an octave above 50 Hz is the frequency range 50 Hz to 100 Hz, while an octave above 5 kHz is 5 kHz to 10 kHz. This is not a precise definition of an octave but is essentially correct and is adequate for loudspeaker analysis.

Figures 2 and 3 show circuit diagrams for series and parallel first-order crossovers. The series configuration is less commonly used since it is only applicable to two-way loudspeakers and has no advantages over the parallel type. If the first-order crossover is terminated with ideal resistive loads, the two slopes will add to give a linear response with the phase response perfectly preserved. In this respect it is fairly unusual since no other simple passive crossover will give a response that is linear in both frequency and phase. Unfortunately 6 dB/octave slopes require a choice of drivers with very broad overlapping frequency responses. Generally it is necessary to have a usable response from a driver to a frequency where the crossover is giving about 12 dB of attenuation. For a 6 dB/octave low pass crossover this would be two octaves above the crossover point. A woofer crossing out at 500 Hz would be very unlikely to have a response to 2 kHz, so a 6 dB/octave filter could not be used.

The most common crossover is the

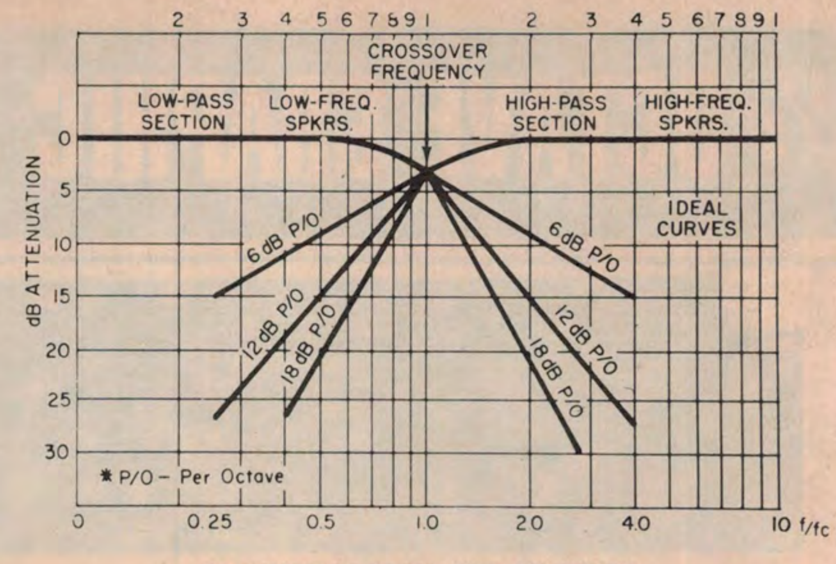


Figure 1. **Typical frequency response curves of a two-way crossover network. The graph shows three different pairs of filters, each having a different rate of attenuation.**

CONSTANT K CROSSOVER NETWORKS

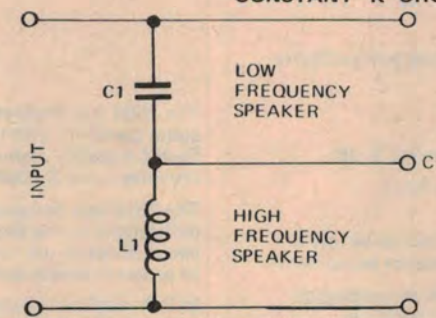


Figure 2 – **series type, 6 dB/octave.**

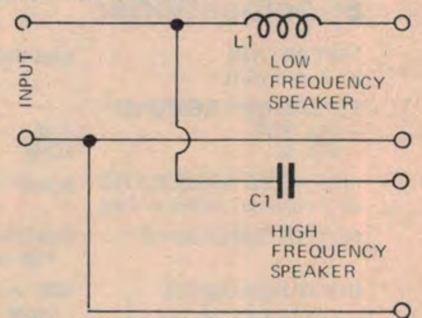


Figure 3 – **parallel type, 6 dB/octave.**

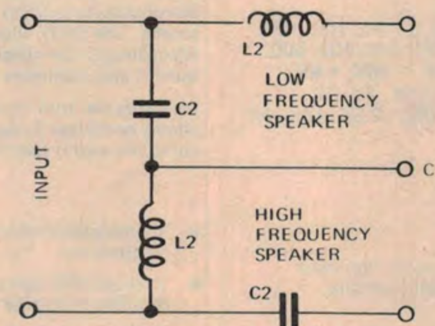


Figure 4 – **series type, 12 dB/octave.**

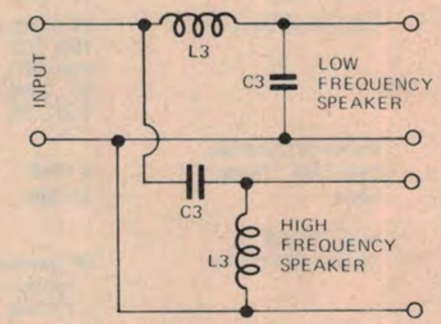


Figure 5 – **parallel type, 12 dB/octave.**

second order crossover, having a slope of 12 dB/octave. Figures 4 and 5 show the circuit diagrams for series and parallel second-order crossovers. Once again the series configuration is less commonly used. When terminated with ideal resistive loads the second-order crossover does not give an overall flat response. The phase characteristic causes the outputs of each half of the crossover to approach a 180 degree phase difference at the crossover point. The two outputs cancel each other, leaving a massive hole in the frequency response of the system. The 'cure' is to invert one of the drivers so that it is driven out of phase normally. The phase inversion around the crossover point brings the two drivers in phase again and the two outputs add, instead of cancelling. Unfortunately they still don't add perfectly and the result is an overall response that has a slight hump in the frequency response of around 2 dB. This is not really noticable, as few drivers have responses that are flat to this degree.

At the present time there is a great deal of discussion as to whether this non-idealistic phase response is audible. Some manufacturers insist that it is audible and design their loudspeakers accordingly, while others are most emphatic that it is not audible. The first work that I know of that was done on the subject was by Helmholtz, in his "Sensations of Tone". The quality of any sound was said to be a result only of the relative intensities of the component sine waves and not their phase relationships. The waveshape could therefore be totally different but they would sound the same.

There is another source of phase error caused by the misalignment of the acoustic centres of the drivers. The conventional way to mount the drivers ▶

FREQ.	C1	C2	C3	L1	L2	L3
100	199	281	141	12·7	9	18
150	133	188	93·8	8·49	6	12
200	99·5	141	70·3	6·37	4·5	9
250	79·6	113	56·3	5·09	3·6	7·2
300	66·3	93·8	46·9	4·24	3	6
350	56·8	80·4	40·2	3·64	2·57	5·14
400	49·7	70·3	35·2	3·18	2·25	4·5
500	39·8	56·3	28·1	2·55	1·8	3·6
600	33·2	46·9	23·4	2·12	1·5	3
750	26·5	37·5	18·8	1·7	1·2	2·4
1000	19·9	28·1	14·1	1·27	·9	1·8
1250	15·9	22·5	11·3	1·02	·72	1·44
1500	13·3	18·8	9·38	·849	·6	1·2
2000	9·95	14·1	7·03	·637	·45	·9
2500	7·96	11·3	5·63	·509	·36	·72
3000	6·63	9·38	4·69	·424	·3	·6
3500	5·68	8·04	4·02	·364	·257	·514
4000	4·97	7·03	3·52	·318	·225	·45
5000	3·98	5·63	2·81	·255	·18	·36
6000	3·32	4·69	2·34	·212	·15	·3
7500	2·65	3·75	1·88	·17	·12	·24
10000	1·99	2·81	1·41	·127	·09	·18

Table 1: **Component values for constant K loudspeaker crossover networks. Inductance in millihenries, capacitance in microfarads, speaker impedance = 8 ohms.**

is to simply bolt them to the front panel. This lines up the chassis of the drivers, but since different drivers have different depths, the voice coils of the drivers are all at different distance from the listener. If two notes are sent simultaneously to both the woofer and the tweeter for example, the note sent to the tweeter will get to the listener momentarily before the note from the woofer. Furthermore, the woofer cone is heavier than the tweeter or midrange cones and this combined with the effect of the air load on the drivers moves their actual acoustic centres even further away from the chassis. Manufacturers concerned with this effect mount the drivers on a multi-level front panel so that the tweeter is further away from the listener than the midrange. Similarly, the midrange is mounted on a plane that is further away from the listener than the woofer. This gives the sound from the midrange and woofers a head start over the tweeter, and attempts to correct for the differences in their acoustic centres.

Both types of phase errors need to be recognised and dealt with independently if a meaningful analysis of the audibility of phase errors in loudspeakers is to be carried out. Even if phase errors of this magnitude are audible (and only experiment can tell us), an extremely good loudspeaker can still be constructed along the more conventional techniques using second-order crossovers with drivers mounted on a plane baffle.

The 4000 Series of loudspeaker projects commencing in this issue use the more conventional approach to driver mounting and crossover design to simplify construction and decrease cost. If you choose to experiment with the audibility of phase in loudspeakers, and construct a Series 4000 loudspeaker with a stepped front panel, the best way to establish the correct distance between the panels is by experiment. The drivers should be connected to the crossover and mounted in separate enclosures. The size of these enclosures is not critical.

Supply the power amp driving the loudspeaker with a source of low repetition pulses (or a low frequency square wave around 20 Hz). If the loudspeaker is now monitored with a microphone and the output of the mic amplifier fed to an oscilloscope, the transient performance of the loudspeaker can be determined.

When the front baffles of the enclosures are aligned, as would be the case in a conventional loudspeaker, the input pulses will be seen to be converted into a series of pulses. Each pulse corresponds to one of the drivers. If the enclosures are moved slowly back with respect to the woofer enclosure these pulses will merge into a single pulse. This is the correct position for the baffles, and using these measurements the final enclosure can be built. If you have the necessary equipment to do this experiment we would be interested in hearing about your results.

The crossovers described so far belong to a class of filters called constant-K filters. These filters are designed on the assumption that the product of the impedances of the capacitor and the inductor in the stage is equal to the square of its characteristic resistance, i.e.

$$Z_C \times Z_L = R_o^2$$

The characteristic resistance of a filter is that resistance into which there is maximum power transfer. Originally, 'K' was used instead of the now more common symbol R_o for the characteristic resistance, hence the name constant K. Table 1 gives values for constant-K filters for a variety of crossover points assuming an 8 ohm resistive load.

M-derived filter sections

The assumptions made to simplify the design of the constant-K filters lead to some non-ideal characteristics. It is sometimes mistakenly thought that constant-K implies constant impedance. This is not the case and the impedance is a variable with the effect occuring ▶

mostly around the crossover point. The other problem with constant-K filters is that the slopes are slowest near the crossover point. The solution to these problems has been known since 1923, when Zobel proposed that other sections could be used to flatten the response within the passband and sharpen the roll-off point. These stages are called M-derived sections, since the values of inductance and capacitance used in the filter are obtained by first deriving them for a constant-K type filter and then converting these values into M-derived values with the use of a mathematical equation that contains the term M. M is simply a number between 0 and 1, usually around 0.6 for crossover applications. Either the phase or frequency characteristics may be optimised but not both at once; 0.6 is a good comprimise. Table 2 and Figures 6 to 9 give values for inductors and capacitors for M-derived crossovers with M = 0.6.

The other major advantage of this filter is that it allows a third-order or 18 dB/octave filter to be built. Third-order filters can be made to have a linear frequency characterisitic when the outputs of the two channels are summed, but like the second-order filters described before, suffer from a very non-linear phase response. Each filter shifts the phase at the crossover point by 180 degrees, so there is a 360 degree phase shift between the two outputs.

Loudspeaker impedance

So far I have assumed that the loudspeakers connected to the crossover are fixed 8 ohm loads, but as was seen in last month's article, this is most definitely not the case. Most drivers have an impedance characteristic that presents maximum impedance at their resonant frequency, dropping to the nominal dc resistance of the driver at a frequency above this, followed by a generally increasing impedance as frequency rises (see Figure 2 in last month's issue). Provided the driver is not being used near its resonant frequency, which should always be the case with midranges and tweeters, this impedance variation can be corrected by a series capacitor-resistor network placed in parallel with the loudspeaker Figure 10 shows a typical circuit. This network has an impedance that decreases with increasing frequency, tending to cancel the increasing impedance of the driver. The component values shown are applicable to an average woofer, although the actual values in any specific application are best established by experiment. This works very well and it is not difficult to obtain an impedance response that is flat within one ohm over most of the driver's operating range.

Matching sensitivities

Once the crossover points have been established from an analysis of the driver's best operating regions, the final step is to equalise the various sensitivites of the different drivers. This is done by a resistor divider network as shown in Figure 11. A simple resistor placed in series with the driver would of course decrease the power in the loudspeaker for a given signal voltage, but this increases the impedance seen by the rest of the crossover altering the crossover frequency point. The resistive divider network shown in Figure 11 can be set to represent a fairly constant 8 ohm ▶

M-DERIVED CROSSOVER NETWORKS

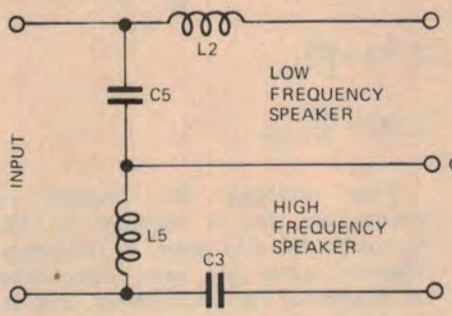


Figure 6 — series type, 12 dB/octave.

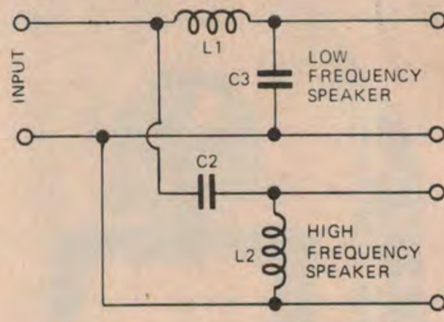


Figure 7 – parallel type, 12 dB/octave.

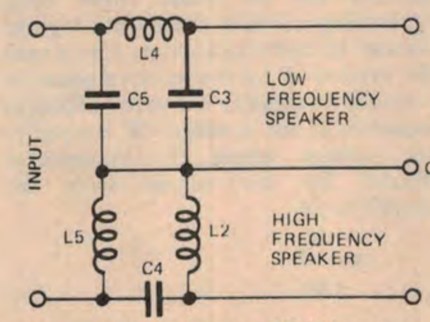


Figure 8 — series type, 18 dB/octave.

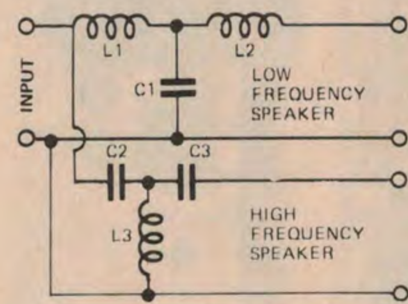
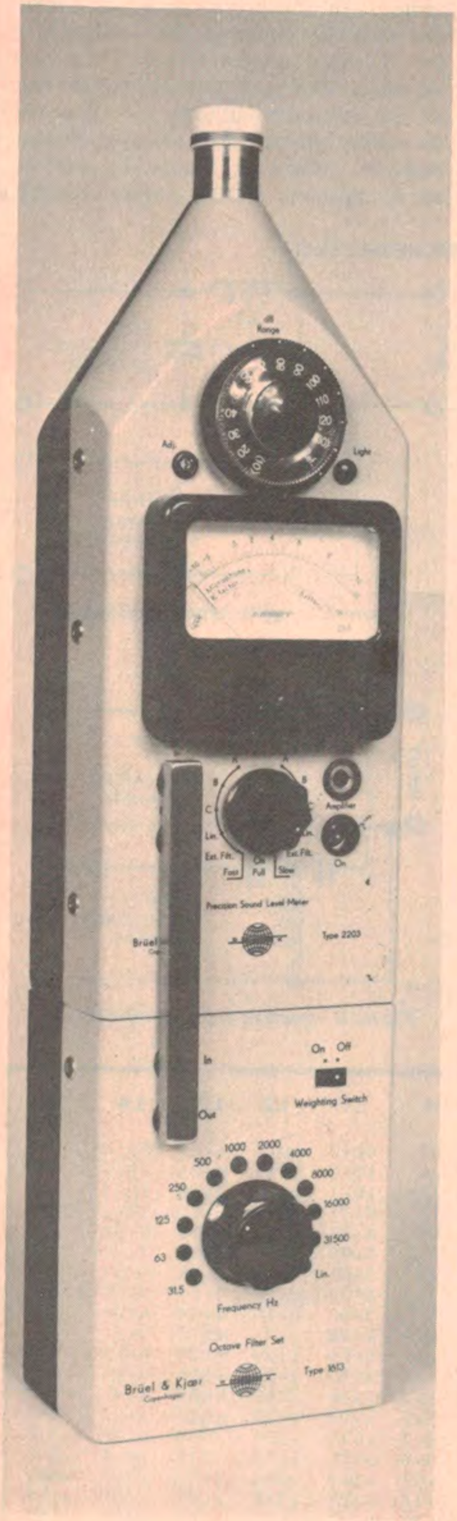


Figure 9 – parallel type, 18 dB/octave.

FREQ.	C1	C2	C3	C4	C5	L1	L2	L3	L4	L5
100	398	124	199	99.4	318	20.3	12.7	6.37	25.4	7.96
150	265	82.9	133	66.3	212	13.5	8.49	4.24	16.9	5.31
200	199	62.1	99.4	49.7	159	10.1	6.37	3.18	12.7	3.98
250	159	49.7	79.5	39.7	127	8.15	5.09	2.55	10.1	3.18
300	133	41.4	66.3	33.1	106	6.79	4.24	2.12	8.49	2.65
350	114	35.5	56.8	28.4	90.9	5.82	3.64	1.82	7.28	2.27
400	99.4	31	49.7	24.8	79.5	5.09	3.18	1.59	6.37	1.99
500	79.5	24.8	39.7	19.9	63.6	4.07	2.55	1.27	5.09	1.59
600	66.3	20.7	33.1	16.5	53	3.4	2.12	1.06	4.24	1.33
750	53	16.5	26.5	13.2	42.4	2.72	1.7	.849	3.4	1.06
1000	39.7	12.4	19.9	9.95	31.8	2.04	1.27	.637	2.55	.796
1250	31.8	9.95	15.9	7.96	25.4	1.63	1.02	.509	2.04	.637
1500	26.5	8.29	13.2	6.63	21.2	1.36	.849	.424	1.7	.531
2000	19.9	6.22	9.95	4.97	15.9	1.02	.637	.318	1.27	.398
2500	15.9	4.97	7.96	3.98	12.7	.815	.509	.255	1.02	.318
3000	13.2	4.14	6.63	3.32	10.6	.679	.424	.212	.849	.265
3500	11.3	3.55	5.68	2.84	9.09	.582	.364	.182	.728	.227
4000	9.95	3.11	4.97	2.49	7.96	.509	.318	.159	.637	.199
5000	7.96	2.49	3.98	1.99	6.37	.407	.255	.127	.509	.159
6000	6.63	2.07	3.32	1.66	5.31	.34	.212	.106	.424	.133
7500	5.31	1.66	2.65	1.33	4.24	.272	.17	.085	.34	.106
10000	3.98	1.24	1.99	.995	3.18	.204	.127	.064	.255	.08

Table 2: Component values for M-derived loudspeaker crossover networks. Inductance in millihenries, capacitance in microfarads, M = 0.6, speaker impedance = 8 ohms.

...loudspeaker design



A Bruel and Kjaer sound level meter. Instruments like this are used to determine the frequency response of a loudspeaker. This instrument has several weighting curves built in as well as a one octave filter set to allow pink noise analysis.

load. Resistor R2 is placed in parallel with the driver, resulting in a decreased total impedance. This impedance is then brought back up to the desired impedance by placing R1 in series. The correct values for R1 and R2, assuming an 8 ohm loudspeaker system, are given by the following three simple equations.

1. $d = \text{antilog} \dfrac{-\text{signal drop in dB}}{20}$

2. $R2 = \dfrac{8d}{1-d}$

3. $R1 = \dfrac{64}{8 + R2}$

First establish the amount of attenuation that is required in dB. Normally by this stage a frequency response curve has been established by measuring the loudspeaker, and an estimate of the required attenuation can be obtained from this. Now use equation 1 above. The antilog of a number can be found either using log/antilog tables or the inverse log key on any scientific calculator. I have used the symbol 'd' for the result of equation 1 mainly to simplify the written form of equation 2, but in reality 'd' is equal to the voltage across the loudspeaker divided by the voltage from the amplifier. i.e.

$$d = \frac{V_s}{V_i}$$

where 'V ' is the signal voltage across the loudspeaker and 'V_i' is the applied signal voltage from the amplifier.

The result of equation 1 is plugged into equation 2, which yields the correct value for R2. The value of R2 is then used in equation 3 to obtain the value of R1. For example, if a midrange is to be decreased in sensitivity by 3 dB, equation 1 becomes:

$$d = \text{antilog} \frac{-3}{20}$$

$$d = \text{antilog} -0.15 = 0.7079$$

So a 3 dB drop in output signal level is equivalent to decreasing the signal voltage across the loudspeaker to $0.7079V_i$. Plugging this into equation 2 gives

$$R2 = \frac{8 \times 0.7079}{1 - 0.7079} = 19.4 \text{ ohms.}$$

Using this result in equation 2 gives

$$R1 = \frac{64}{8 + 19.4} = 2.3 \text{ ohms.}$$

The nearest value resistors to these would be 18R and 2.2R, and with these resistors the impedance presented to the remainder of the crossover would be approximately 7.7 ohms which is close enough in practice.

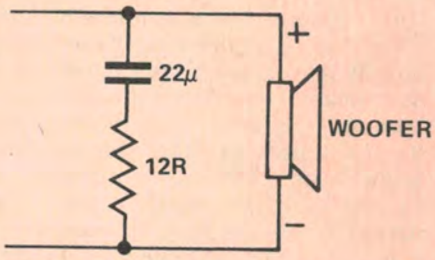


Figure 10. **Circuit to improve the apparent impedance of a loudspeaker.**

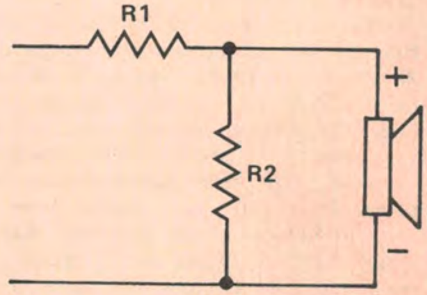


Figure 11. **Potential divider used to compensate for different loudspeaker sensitivities.**

The final test

Now that the bass performance has been optimised through a suitable choice of enclosure size and damping, the crossover points and slopes have been established, and the impedance and sensitivities of the various drivers corrected, the final subjective tests can begin. The importance of good subjective testing in loudspeaker design cannot be overestimated. The most common form of subjective analysis, other than simply listening to some good records, is an A-B test with other loudspeakers. Although this method can give some meaningful results, its validity is generally overestimated in my opinion. The best form of subjective testing is comparison to the original live performance. Simply recording a voice onto high quality recording equipment with a good microphone will tell you more about a loudspeaker than any amount of A-B testing. ●