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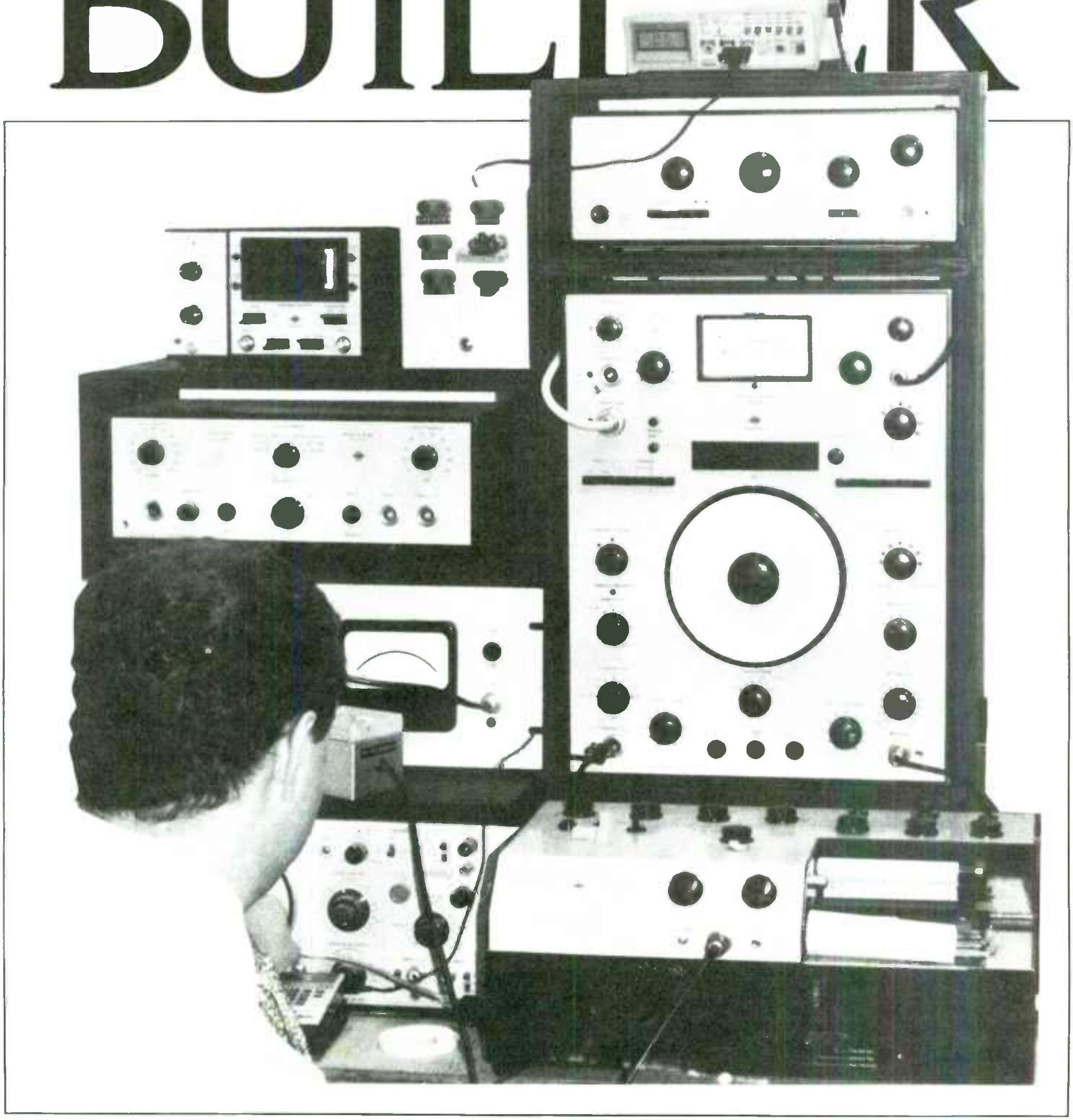
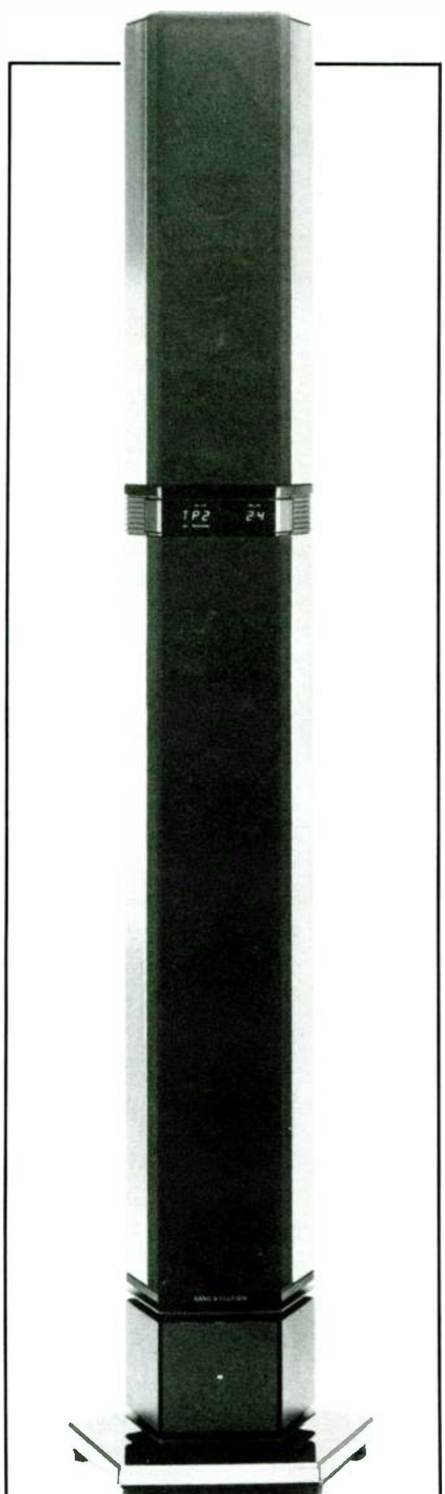
SPEAKER BUILDER



Good News



The Beolab Penta is the latest stereo loudspeaker system developed by **BANG & OLUFSEN**. Each unit is 65" tall, 5" wide, and finished in brushed stainless steel. Nine active drivers and a 150W amplifier are built into each speaker, while a digital display in each unit shows operating status.

According to the manufacturer, the Beolab Penta system's five-sided enclosure reduces the standing waves that cause sound distortion in conventional box speakers.

The nine drivers include four woofers, four midranges, and one dome tweeter, all in line-source configuration. The system automatically turns on when it senses a signal, and turns off when the signal ceases.

Available this fall, the Beolab Penta system will retail for $2,598.

For more information, write Bang & Olufsen of America, Inc., 1150 Feehanville Dr., Mount Prospect, IL 60056.

Fast Reply #IK535

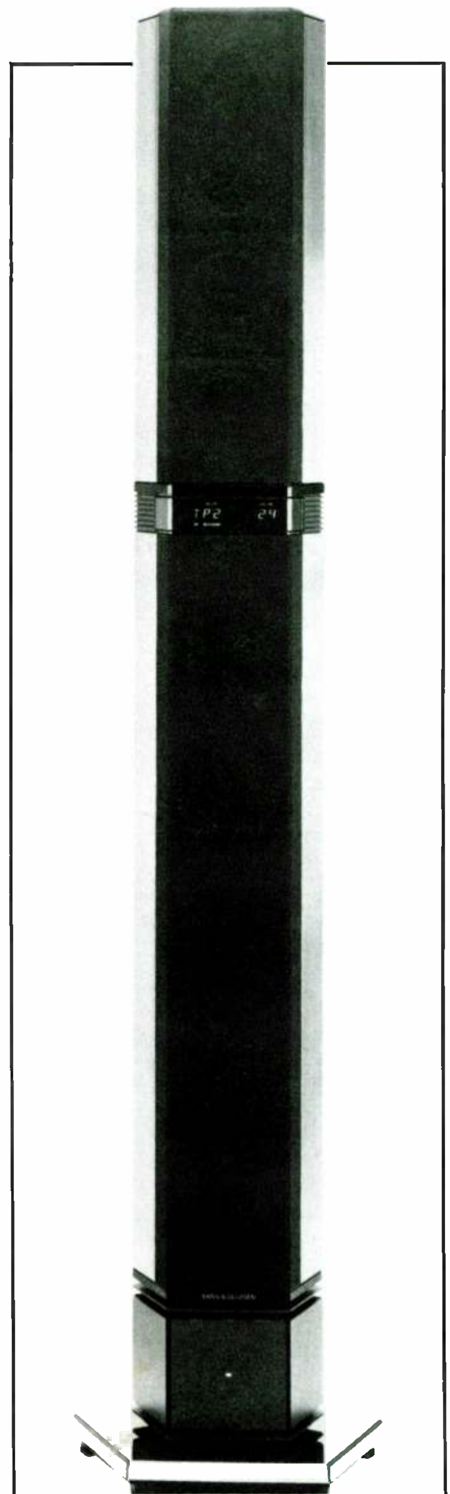
DCL COMPANY has released the LFD/E Program, a new CAD (computer-aided design) program for speaker designers and builders. The manufacturer claims the program features high-level user friendliness while meeting the professional designer's needs for technical sophistication.

LFD/E provides complete instruction to builders, while an accompanying manual shows various functional design sequences.

Currently designed for Apple computers, LFD/E has a suggested retail price of $29.95.

For complete details, contact DCL Company, 6340 SW 145th St., Miami, FL 33158, or call (305) 232-2199.

Fast Reply #IK138



ALTEC LANSING CONSUMER PRODUCTS has released two new speaker systems.

The company's top-of-the-line "Voice of The Highway" speaker is the ALS-693, a 6x9" three-way system. The woofer is a 6x9" carbon fiber cloth cone with double damper, 20.8 oz. magnet, and 1½" ribbon wire voice coil. The midrange has a 20mm polymide dome, while the tweeter utilizes a 14mm polymide dome, both of which are immersed in magnetic fluid for damping and heat dissipation.

The ALS-693 delivers a frequency response of 50Hz to 22kHz ±3dB, and sensitivity (SPL) of 93dB/W/meter. At 1W, the THD is 0.6% over a frequency range of 70Hz to 20kHz. Power-handling capacity is 60W nominal to 120W maximum, and impedance is 4Ω.

Suggested retail price: $250/pair.

Also among Altec's latest releases is the top-of-the-line home speaker system, the Model 501. The 46" high tower system features two 10" carbon fiber woofers, a 2" polymide vacuum-deposition titanium midrange, and a 1" polymide tweeter.

Frequency response is 28Hz to 22kHz ±3dB, while THD from 100Hz to 22kHz is less than 1%. Crossover frequency is 550Hz/3.5kHz, power-handling is 150W nominal to 300W maximum, sensitivity is 93dB 1W/1M, and impedance is 4Ω.

Suggested retail price: $1,500/pair.

For more information on these products, write Altec Lansing, 1515 S. Manchester Ave., Anaheim, CA 92803.

Fast Reply #IK328

Two new spectrum analyzers have been released by **RAPID SYSTEMS, INC.** The R411 is an FFT spectrum analyzer peripheral for IBM PC, XT, AT and compatible computers. The peripheral requires no programming. Once you plug in the hardware and insert a disk, you can begin to analyze frequency versus amplitude spectrums on your screen.

Its many features include: FFT sizes from 16 to 1024 points, execution and display of 1024 point FFTs every four seconds, log and linear scaling for frequency and amplitude, programmable input ranges from 1.6–320V peak to peak, and spectrum averaging.

Suggested retail price: $888.

Also from Rapid Systems is the R360, a spectrum analyzer and digital signal processing peripheral. The manufacturer claims the R360 is the only PC-based instrument featuring the TI TMS32010, and offering four-channel real-time spectrum analysis.

Applications include multiple channel spectrum analysis to 250kHz, vibration analysis, impact testing, power line monitoring, and audio and speech analysis.

Suggested retail price: $2699. The R360 is also available with a more economical data acquisition module for $1499, and alone it retails for $999.

For complete details, contact Rapid Systems, Inc., 755 N. Northlake Way, Seattle, WA 98103.

Fast Reply #IK948

A Designer Series Manual has been published by **AUDIO CONCEPTS, INC.** The complete do-it-yourself manual is geared to encourage beginning as well as advanced builders to try new projects.

Audio Concepts says the manual answers many of the speaker building questions the company receives each day.

Suggested retail price $15 (overseas, add $2).

To order, write Audio Concepts, Inc., 1631 Caledonia St., La Crosse, WI 54603.

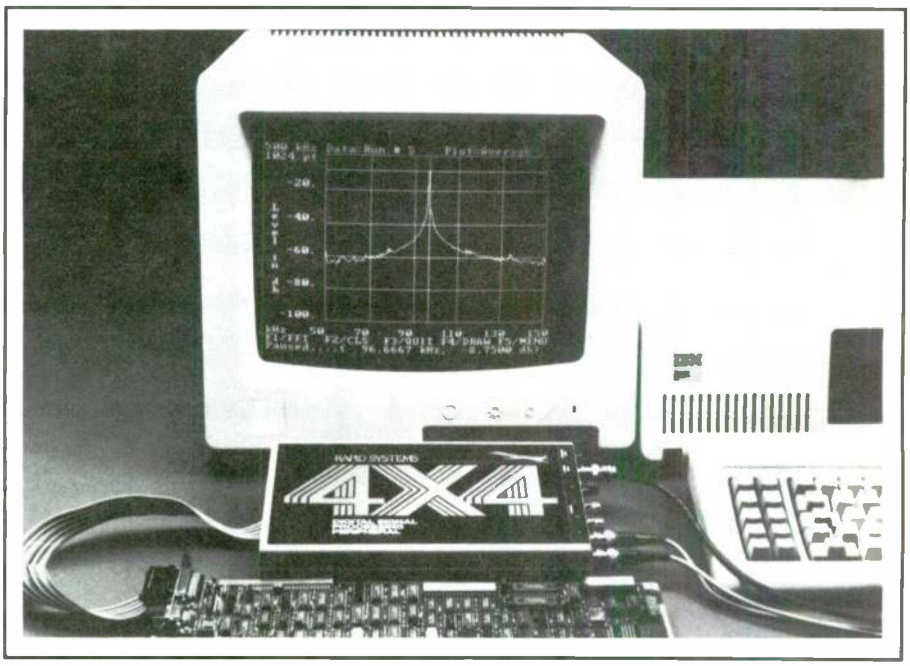
Fast Reply #IK45

Computer-using audiophiles and music lovers who want information, specifications, prices, and the latest news about high-end stereo components can now use their modem to call a new databoard system, **THE AUDIOPHILE NETWORK**.

The network operates from California 24 hours a day, seven days a week, and can be reached by dialing (818) 988-0452. Software is by Russ Systems, and operates on an Apple II with a hard disk and Apple drives.

For further information, call (818) 782-1676, or write to The Audiophile Network, 14155 Kittridge St., Van Nuys, CA 91405.

Fast Reply #IK175



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About This Issue

This jam-packed fourth issue concludes our seventh year of publication and brings another rich selection of work by writers who span the continent from Washington state to California to New York to Louisiana to Ohio. **Max Knittel** has written a brilliant piece of software for exploring the FFT response of drivers which he describes starting on page 7.

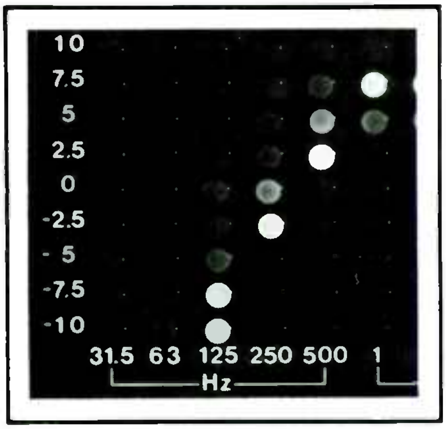
In Louisiana, **Duke LeJeune** put together that rarest of creations, a sonically satisfying ribbon system which passes muster esthetically with his spouse (p. 14). Nearly two years ago, **Bruce Edgar** and I spent a day with the nice people at Acoustic Research interviewing and taking pictures. Even though the prime respondent left the company shortly afterward, editor Edgar pursued his quarry, re-interviewed and re-photographed his subject: Ken Kantor. See page 20 for the results.

Brian Smith was not only lured away from bookkeeping to engineering school at Ohio State by his audio avocation, he is intrigued by horn speaker formats, especially the tractrix. His BASIC program (p. 29) cuts design time by days. The second part of **G. R. Koonce's** exploration of crossover components starts on page 32. On our cover, we look over Ken Kantor's shoulder at the instrumentation for AR's anechoic test bench.

Speaker Builder has grown in pages during 1986 and will probably continue to do so in 1987. The subscription price will rise on January 1, 1987 to $15 per year for four quarterly issues.

SPEAKER BUILDER

VOLUME 7 NUMBER 4 OCTOBER 1986



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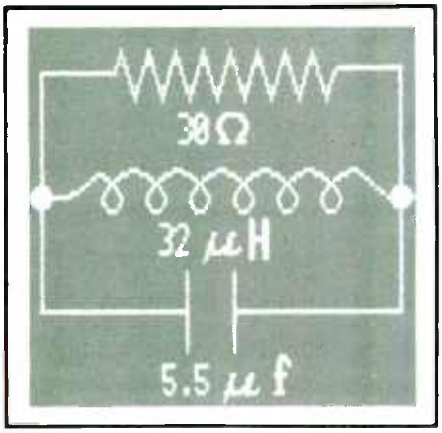

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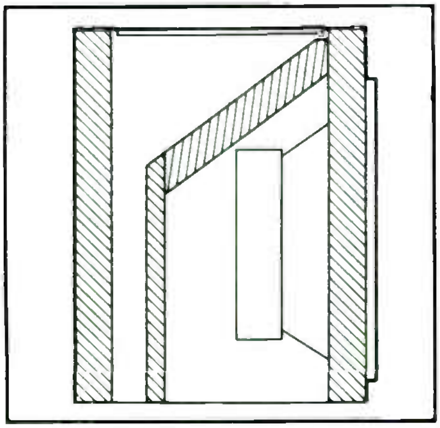

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FEATURES

DEPARTMENTS

Editorial

Mistake Identity

If we all had total recall of our first three months of life, I suspect the earliest words we could remember would probably be "Watch out!" I believe that builds into each of us a powerful programmed response not only to legitimate dangers, which are plentiful enough, but also to any risky enterprise. Accompanying that cautionary atmosphere, we have the added element of shame which goes with failure. The second message, after "Watch out," is probably "You're doing it wrong."

One of the reasons we have avocations, I think, is the chance to undertake projects where the failure rate is not so guilt-laden. If we could assemble statistics on hobby popularity, I suspect the failure-risk-quotient would correlate very well with the quantity of participants. Who can fail at collecting baseball cards or Mickey Mouse watches?

But some hobbies are risky. Racing cars, for example. Climbing sheer rock faces in winter, sky diving. Building loudspeakers.

Building loudspeakers? Indeed. The fear-of-failure syndrome is a tragically effective contraceptive where loudspeaker projects are concerned. Hundreds more die a dried up, overage death on back burners.

The first step toward escape from the clutches of infantile programming is the realization that only the deranged and angels are fearless. Everyone suffers, the differences are only in degree. We are all in that bind together. Others get out of those clutches, so can you.

The second aid to action is the loudspeaker project's limited liability character. Most errors do not blow the main breaker in your electrical distribution box. The *frisson* factor at turn-on is definitely lower than those accompanying flipping the bat handle of a 1kW/channel power amp.

The speaker builder's art is not abstract, either. Its output matches, or does not, the original phenomenon. Honest work draws us to the concert hall and brings us back to our handiwork to listen again, and to compare. Reality is potent and judges unmercifully if we are candid enough to hear the judgment.

But that is also the glory of speaker building. It is not trivial combat. The adversary is sonic reality itself and our attempt to reconstruct that reality is a magnificent challenge. Nature is implacable. The game consists of discovering her game's rules.

I rejoice in the analog nature of sound for the very reason of its variety and unique characteristics. It is an innately challenging continuum. The opportunities for getting closer to the original in all those details is part of the excitement that accompanies the quest.

We are still bound to make mistakes, and some major ones. The beauty of the mistake, however, is the learning that accompanies it. One of my favorite authors, Lewis Thomas (*Late Night Thoughts on Listening to Mahler's Ninth Symphony* is his most recent set of essays), was asked a few seasons ago whether he thought computers could be "intelligent." He said he did not believe they could. He continued by saying that, so far, no computer has made either a joke or a mistake. Now before you rush to your word processors to tell me about misbillings, erroneous checks for millions, or several thousand copies of *Life* magazine mailed to a one room cabin in Oklahoma, remember that those are either badly entered data or an equipment malfunction—not the same thing as a mistake.

Every advance in human development is based on a mistake. Of course, mistakes come in all sizes. But even large ones, like Chernobyl, are massive learning doses. And a loudspeaker project that falls short is still valuable. I deeply admire our Contributing Editors because their articles for this magazine often trace a learning curve dotted with mistakes. We do not pretend to be the flawless, last-word, infallible, "journal" type of publication. Every endeavor is an honest work in progress. Those who wait for perfection are permanently immobilized.

As mistake makers, we are in our most human mode. To paraphrase, better to have built a system and failed, than never to have built one at all. Since my earliest intention to publish do-it-yourself magazines, I have encouraged authors to freely share their mistakes. And the best do. The courage to try, to publish and to build on your own perceptions and the feedback of others is the essence of the adventure. Nothing in life is harder, or better.

—E.T.D.

STEP RESPONSE OF LOUDSPEAKERS

BY MAX KNITTEL

It is no easy task to measure the frequency response of a loudspeaker. You can measure the low-frequency anechoic response of a woofer fairly accurately in a normal room by using swept sine waves and placing the measuring microphone in the near field.[1] Unfortunately, it is difficult to use swept sine waves to measure the full frequency response of a speaker within a normal listening room because of reflections and standing waves created by the room walls, floor and ceiling; hence, the use of anechoic chambers. Since anechoic chambers suitable for audio frequencies are large and expensive, you can use pink noise and octave- or third-octave band analyzers to minimize the creation of standing waves in the test room.

Examining the time response is perhaps just as interesting and as important as examining a loudspeaker's frequency response. Many listeners claim loudspeakers which accurately reproduce the waveform of the input signal are more musical. They produce more immediate and alive transient sounds, and images with proper spatial relationships and dimensions. Perhaps preservation of the relative phase between frequencies (the waveform) is important to the stability and precision of stereo images. Full-range, flat-panel speakers, such as electrostatics, inherently have this characteristic since all frequencies emerge from the same plane in space, producing the acclaimed electrostatic detail and imaging characteristics.

Unfortunately, neither the swept sine wave nor pink noise techniques reveal any information concerning a loudspeaker's time response. Another technique, however, simultaneously examines both the time and frequency domains. With this technique, an impulse or step waveform is sent to the speaker, and a computer is used

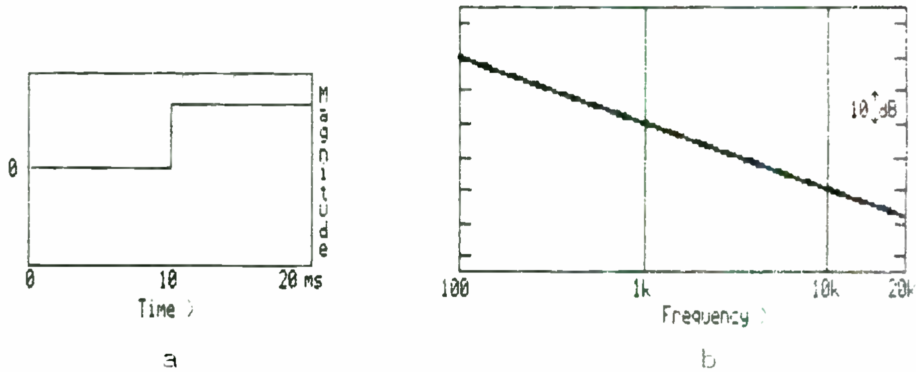


FIGURE 1: Time and frequency responses of a step.

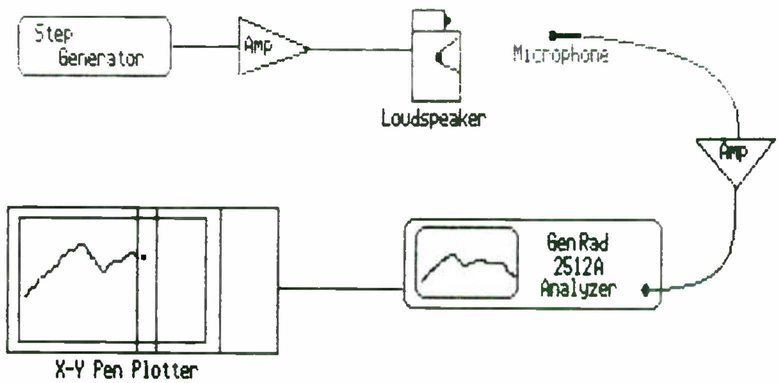


FIGURE 2: Model time responses to a step input.

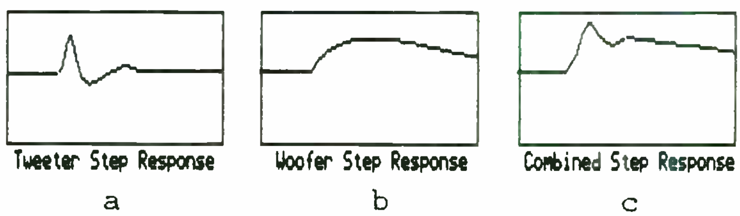


FIGURE 3: Block diagram of step test apparatus.

to analyze a short time-slice of the speaker output. The advantage? These test signals simultaneously send all frequencies to the speaker. The analyzer captures the first direct sounds from the speaker before any reflected ones reach the microphone.

Measuring the time response of a loudspeaker is not a new idea. Heyser[2,3] using impulse signals, has done much work on loudspeaker phase and time distortions. To align drivers with respect to time in a loudspeaker system, Wittenbreder[4] presented plans for constructing a pulse generator and directions for using it. Spangler and McKenzie[5] used an impulse signal and an Apple microcomputer to examine both the time and frequency domains of their modified Strathearn ribbon speaker.

The most interesting speaker testing methods, however, may have been developed by Peter Moncrieff and presented in various issues of his magazine, *International Audio Review*.[6,7,8] Moncrieff describes a general test method that uses the step function as the input signal to the device under test (DUT). His main test instrument is a GenRad 2512A real-time, narrowband spectrum analyzer. Like Spangler and McKenzie's Apple microcomputer and its associated program, it uses a Fast Fourier Transformation (FFT) algorithm to extract the frequency response from the time response of the signal produced by the DUT. Moncrieff is dedicated to the development of testing procedures that quantify and explain differences we hear in audio components. This is in stark contrast to many audio press reviewers, who still perform the same measurements developed more than 50 years ago to test today's equipment.

STEP FUNCTION ANALYSIS. To begin a loudspeaker step response analysis, you must first look at the difference between an impulse and a step, and how the FFT analyzer works with each of them. An impulse is a spike in time. In theory, an impulse contains all frequencies with equal amplitude, but it would have to be infinitely high and infinitely narrow for this to be true. In practice, the impulse duration is set narrow enough to be less than one-half the period of the highest frequency of interest. For example, to measure to 100kHz, the impulse must be less than 5 microseconds in width. Such a brief impulse poses difficulties for the loudspeaker and measuring equipment. The im-

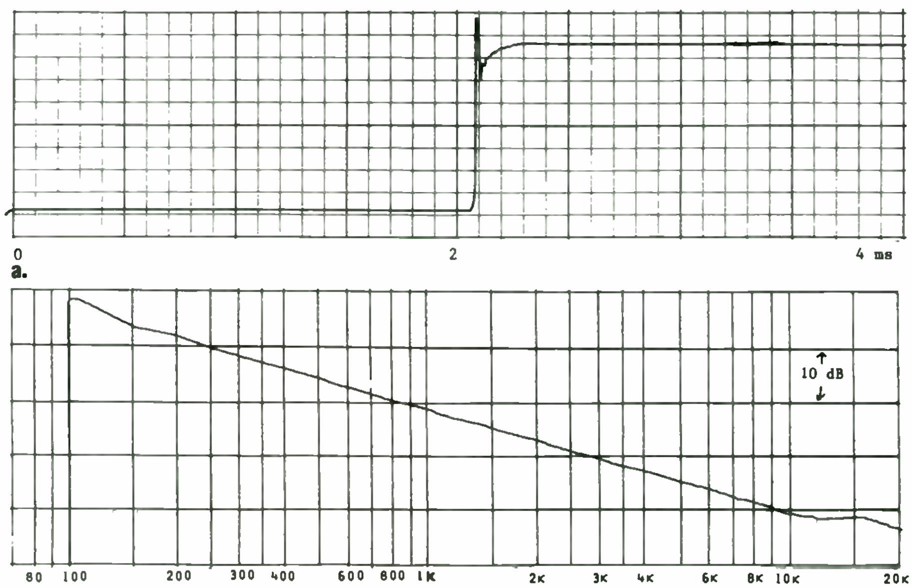


FIGURE 4: Time (a) and frequency (b) responses of the input step.

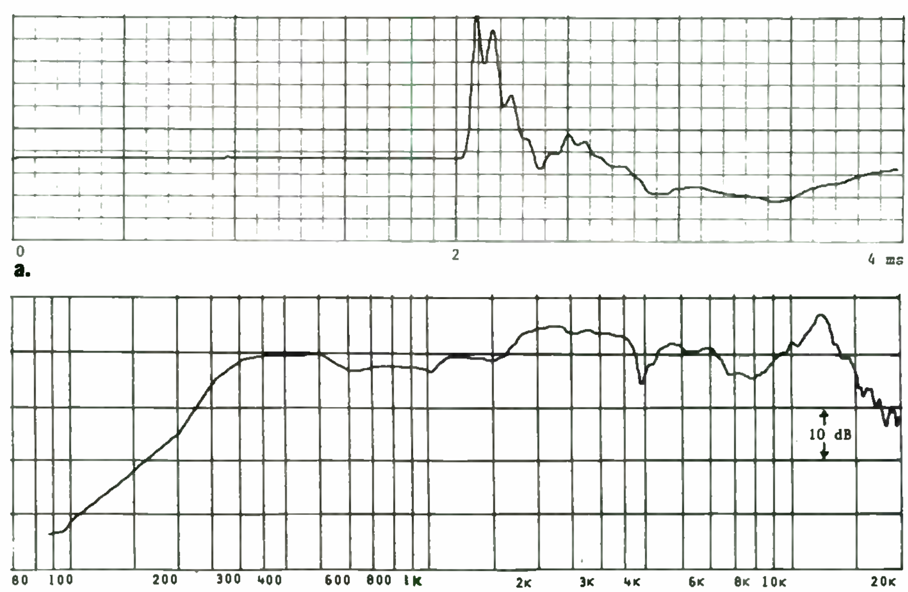


FIGURE 5: Time (a) and frequency (b) responses of Jordan Module.

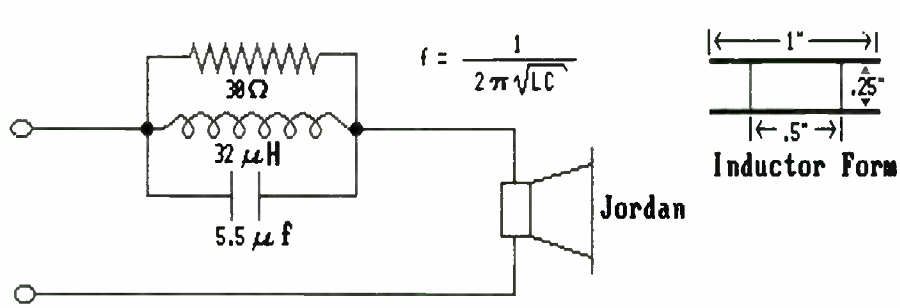


FIGURE 6: Resonance trap attached to Jordan.

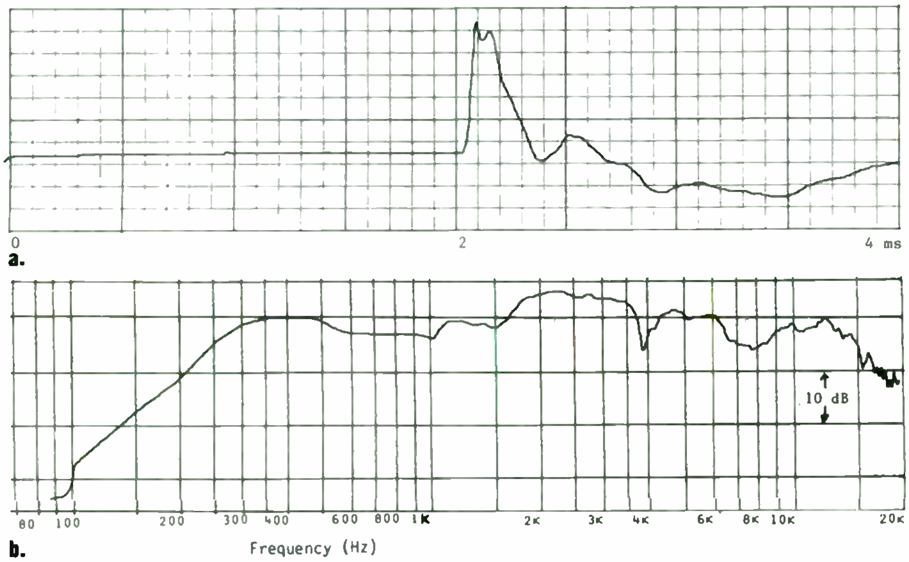


FIGURE 7: Time (a) and frequency (b) responses of Jordan Module with resonance trap.

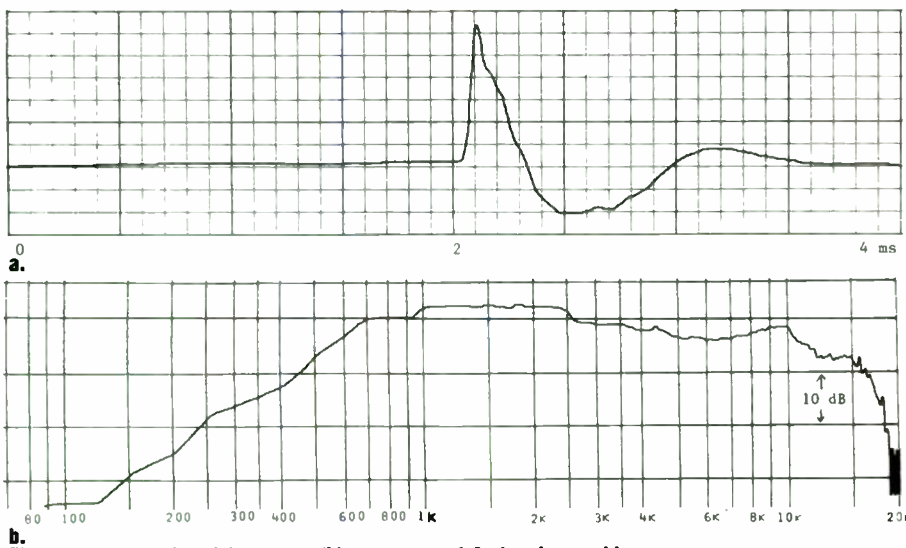


FIGURE 8: Time (a) and frequency (b) responses of Audax dome midrange.

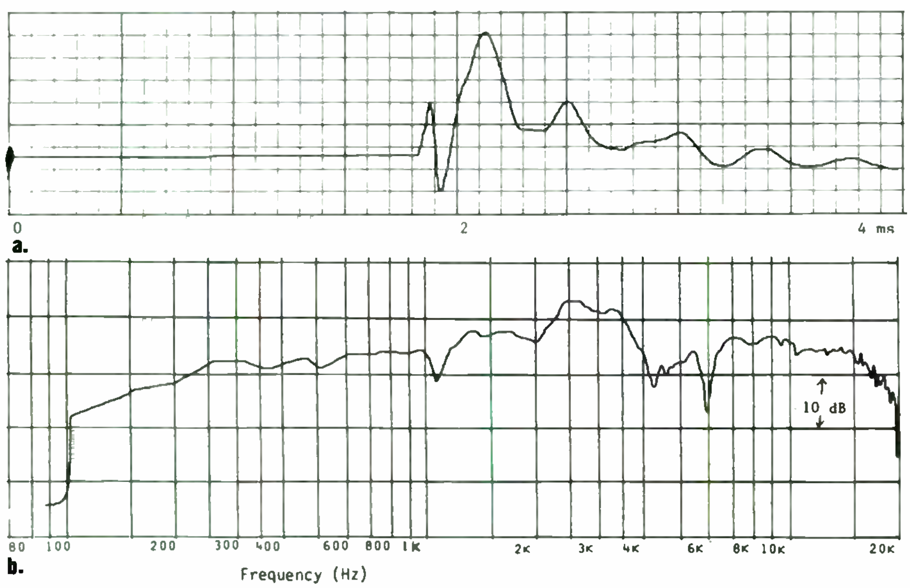


FIGURE 9: Time and frequency responses of the Minimus 7

pulse amplitude, to overcome background noise in the measuring room and equipment, must be quite large, and this can overload the speaker, microphone or electronics.

On the other hand, you can think of the step function as an impulse that is infinitely long in time instead of infinitely short. *Figure 1* shows the time and frequency responses of a step. Because the step contains more energy than an impulse—the energy is the amplitude multiplied by the time; i.e., the area under the time curve—it need not be as large in amplitude to produce a satisfactory signal-to-noise ratio in the measuring equipment. Just as with the impulse, the step contains all frequencies, but they are not of equal amplitude. Instead, the spectral content of a step falls as 1/f; that is, a plot of frequency versus amplitude on a logarithmic scale would be a straight line that falls at 6dB/octave. *Figure 1* can be compared to the impulse time and frequency responses shown in *Figs. 5* and *6* of Ref. 5.

Because the frequency content (or spectral analysis) of the input step function is not a constant, the analyzer output will not be the true loudspeaker frequency response. If a step function is used as the input, you must divide the loudspeaker output spectral response you are measuring with the microphone, by the spectral response you send into the loudspeaker.

The GenRad analyzer offers two features that simplify this analysis: the ability to memorize a spectral response, and to divide a second spectral response by the memorized spectral response. You can store the input spectral response, automatically divide each output spectral response by this stored response, and display the true frequency response on the analyzer's display screen.

STEP RESPONSE OUTPUT. What should the acoustic output from an ideal loudspeaker look like if the input is a step function? When the input voltage is zero, the speaker cone displacement is zero and the acoustic pressure is zero. When the voltage jumps up on the step to a value of +V, the cone accelerates into motion. If it were a nearly massless cone with no suspension stiffness, it would accelerate rapidly while the voltage increases and then continue at some maximum velocity. The ideal speaker should produce an ideal step.

Since speaker cones are not ideal but have significant mass and suspension stiffness, cannot change velocity instantaneously, and continue to move indefinitely, the speaker output is not an ideal step. The cone takes time to overcome its inertia, so the start of the step is not a vertical line but one that slopes upward more gradually. The cone suspension resists the cone movement, eventually stopping it, with some back and forth oscillation. When the cone stops moving, the acoustic pressure must be back to zero, even though the voltage is still at +V. Depending on the particular driver, the acoustic output may look more like a damped resonance than a step. The leading edge of the step depends on the high-frequency response of the speaker, while the extension of the step depends on the low-frequency response of the speaker.

A tweeter tested by itself may produce a sharp spike (such as in *Fig. 2a*), while a woofer by itself may produce a better looking step but with a rounded leading edge (*Fig. 2b*). Put together properly into a loudspeaker, they should produce a sharp step upward and a gentle slope downward (*Fig. 2c*).

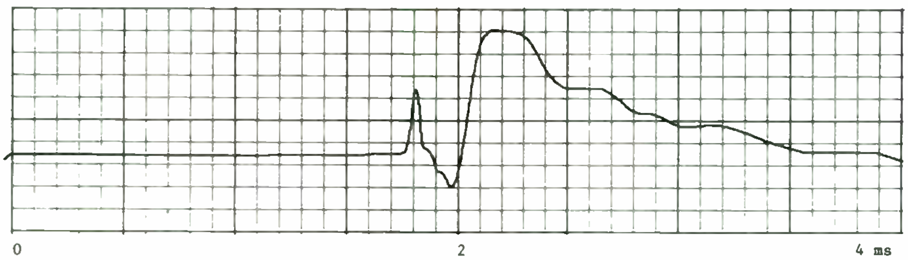
TEST EQUIPMENT SETUP. *Figure 3* is a block diagram of test equipment used to measure and plot the time and frequency responses of a variety of drivers and loudspeaker systems. I used a pulse generator to produce a step longer than 50mS. The time window for analysis was set to 4 or 20mS, with the beginning of the step established in the center of the time window—the window produced a frequency analysis range of 500Hz to 100kHz, and the 20mS window a frequency analysis range of 100Hz to 20kHz. I used the narrow time window to examine the loudspeaker output in the time domain since this spreads the step's leading edge as far as possible. When looking at the frequency domain, I used both the 4 and 20mS windows, but I am presenting plots from the 20mS window since they show more useful information.

I sent the output of the step generator to a power amplifier and then to the loudspeaker. A Shure ES615 microphone with M615 preamplifier detected the loudspeaker output. The microphone output fell off above 17kHz, so loudspeaker measurements above this frequency are unreliable.

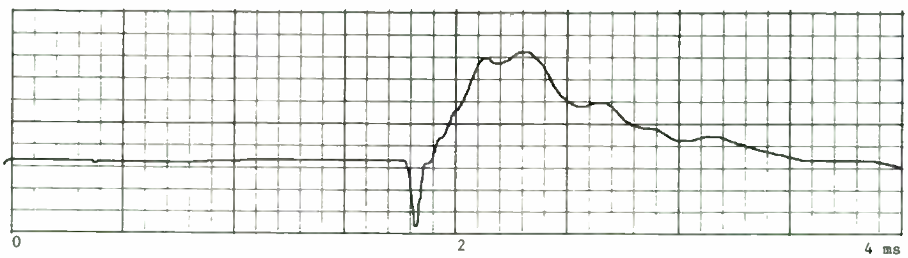
I placed the microphone on the horizontal axis of each speaker one meter away. I spent some time locating the microphone vertically to find a height which produced a smooth frequency response. You can, undoubtedly, find better microphone positions.

To begin making measurements, I fed the speaker input into the GenRad analyzer. This produced the time and frequency plots shown in *Fig. 4*. The frequency spectral response was stored in the analyzer's memory. I reconnected the input to the loudspeaker, and the microphone preamplifier output to the analyzer. The analyzer was set to divide the new spectral response (the speaker output) by the memorized spectral response (the speaker input). *Figure 4a* shows the step generator does not produce a perfect step, but instead has a slight amount of overshoot as it reaches the top of the step. Because you divide by the actual spectral response of this less-than-perfect input step, the displayed frequency responses will be correct.

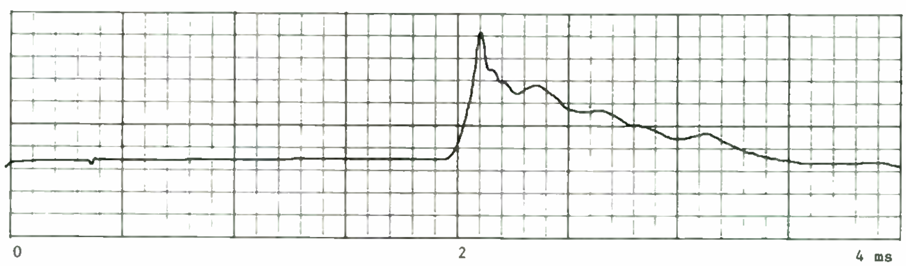
I set the analyzer to begin capturing a waveform when the leading edge of the step is detected. Then, a few milliseconds before the step and a few milliseconds after were captured and the Fourier transform calculated. This gave me both the time (transient) and frequency responses of the speaker.



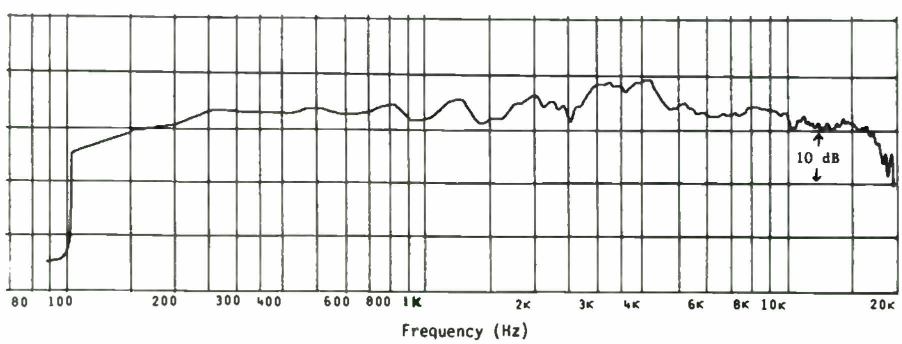
a. Tweeter flush, normal polarity.



b. Tweeter flush, inverted polarity.

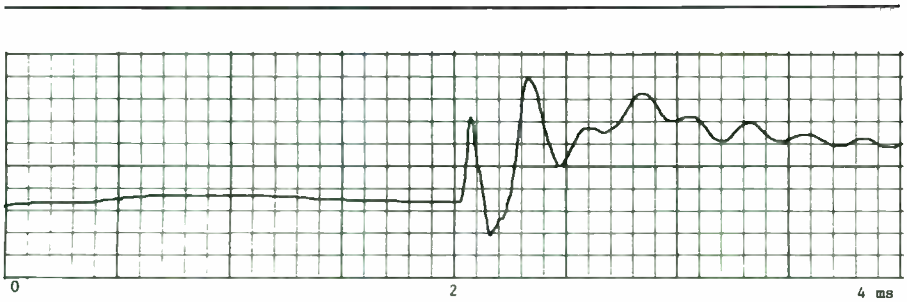


c. Tweeter 10cm back, normal polarity.

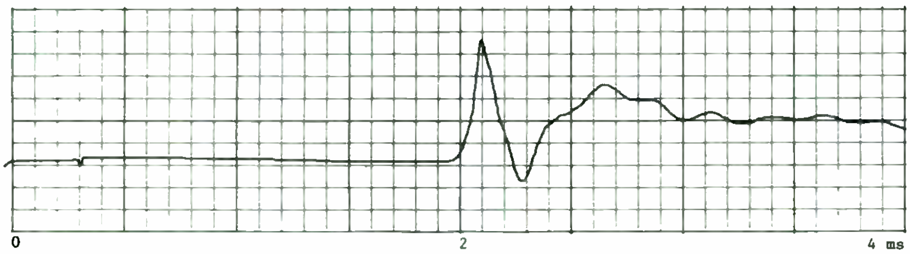


d. Frequency response for final configuration.

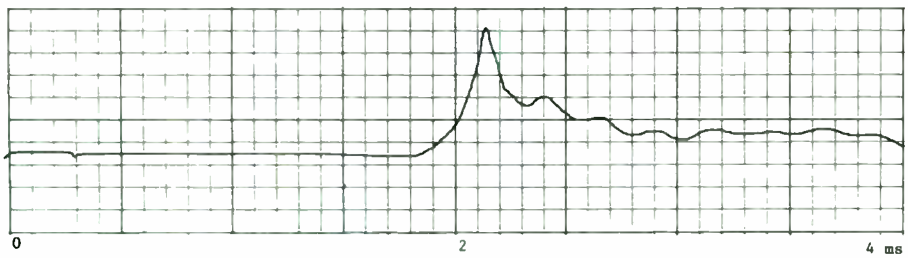
FIGURE 10: Time and frequency responses of the YT1.



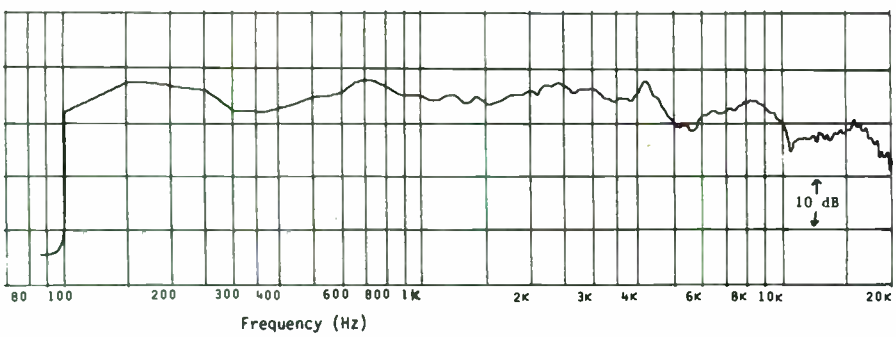
a. Tweeter flush, midrange flush.



b. Tweeterr 8cm back, midrange flush.



c. Tweeter 8cm back, midrange 19cm back.



d. Frequency response for final configuration.

FIGURE 11: Time and frequency responses for the YT2.

TEST RESULTS. For the first tests, I looked at the simplest loudspeakers I could find—two single drivers: a Jordan module and an Audax 1½″ dome midrange. *Figure 5* shows the time and frequency responses of the Jordan. The double peak at the top of the step occurred for two different samples and was also reported by Lampton.[9] The period of this resonance was approximately 0.083mS, which corresponded to 12kHz. Since this was the frequency range where a peak shows up in *Fig. 5b*, there is definitely a high-frequency resonance in the Jordan. Perhaps there was a flexural resonance in the cone.

To reduce this high-frequency resonance, I constructed Lampton's trap circuit. Since my resonance occurred at 12kHz rather than at Lampton's 13kHz, I used a 5.5μF capacitor with the 32mH inductor (using *Table I* from Ref. 10, I wound 49 turns of #22 wire on a coil form whose center is ¼″ high by ½″ in diameter with a 1″ flange diameter). In addition, I found the resonance for my driver evidently needed more damping than Lampton's, so I added a 30Ω resistor in parallel with the capacitor and inductor. *Figure 6* is a circuit schematic, and *Fig. 7* shows the time and frequency responses of the Jordan with the circuit attached. *Figure 6* also gives the formula for finding the trap L (in henries) and C (in Farads) for a particular resonance frequency.

Figure 8 shows the time and frequency responses of the Audax midrange dome. As you would expect, the Jordan had more low- and high-frequency output but was not as smooth. The comparative lack of low-frequency response in the Audax dome showed itself in the time response: the sound pressure dropped to below the zero level more quickly than it did for the Jordan.

Next, I tested two, two-way loudspeakers (both having 5¼″ woofers and 1″ tweeters): a Radio Shack Realistic Minimus 7, and a prototype I built with a movable tweeter, Yours Truly (YT1). *Figure 9* shows the time and frequency responses of the Minimus 7.

I examined the time response of the YT1 two-way system (an Audax woofer and tweeter) for two different tweeter locations. For the first position, *Figs. 10a* and *10b*, I had the tweeter located flush with the woofer, the normal mounting method. In *Fig. 10b*, I reversed the tweeter polarity. The tweeter obviously "speaks" 0.3mS ahead of the woofer, so the loudspeaker produced two pulses for the single input step. In the second position, *Fig. 10c*, I moved the tweeter back until the tweeter step just merged into the woofer step. This turned out to be 10cm (which is as it should be since sound travels a little over 10cm in 0.3 mSec). *Figure 10d* shows the frequency response of this final configuration.

Although the frequency responses of *Figs. 9b* and *10d* are not dissimilar, the time responses of *Figs. 9a* and *10c* certainly are: the YT1 time response looks something like the input step while the Minimus 7 response looks more like a damped sine wave. In listening tests, the Minimus 7 speakers displayed neither the time-aligned YT1's depth and stability of imaging nor the detail and delicacy of high-frequency transients.

For an analysis of a more complex system, *Fig. 11* shows time responses for a Yours Truly three-way loudspeaker (YT2) with an 8″ Audax woofer, 5¼″ Audax midrange, and a ¾″ KEF tweeter. With the three drivers all flush mounted, the time response is displayed in *Fig. 11a*. Moving the tweeter back 8cm relative to the woofer and midrange produced the time response of *Fig. 11b*. Finally, moving the midrange back 19cm relative to the woofer (the tweeter then is 8cm back from the midrange and 27cm back from the woofer) produced the time response of *Fig. 11c*.

As I moved the drivers back so the apparent sound sources for each driver were in a vertical line, the time response looked more and more like the input step. *Figure 11d* shows the YT2 frequency response for this final configuration. The peak just above 4kHz was created by a resonance in the paper-cone midrange.

For historical interest, I also tested a venerable Acoustic Research AR 2ax. This three-way system has a 10″ woofer, and a midrange and a tweeter located adjacent to each other. In all the previously-tested speakers, the drivers were in a vertical line. *Figure 12* shows the time and frequency responses of the AR 2ax. The tweeter was nearly a millisecond ahead of the slow woofer, and the frequency response showed a valley around 1.4kHz, probably due to cancellation between the outputs of two drivers. Shifting the measuring microphone position did not greatly change this frequency response.

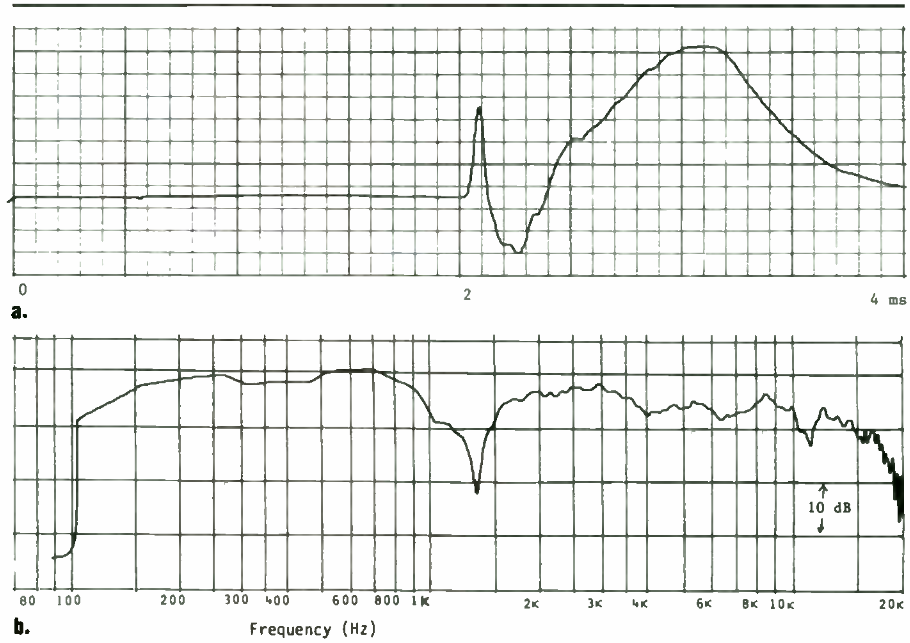


Figure 12: Time and frequency responses of AR 2ax.

ANOTHER TEST METHOD. Using the GenRad real-time analyzer will allow you to fine-tune your design both in the time domain (by locating the drivers in such positions as to produce the most faithful step) and in the frequency domain (by choosing more appropriate crossover frequencies and driver attenuations which will produce the smoothest and most extended response). I have developed and built speakers for nearly ten years without it, however, and I have been doing fairly well with an oscillator, an oscilloscope and two microphones.

Figure 13 shows the test setup of a simple way to find how far back to move the tweeter. The two microphones are located in a line perpendicular to the axis of the speaker. One microphone receives the signal from the tweeter and the other receives the signal from the woofer. Send the microphone outputs to a two-channel oscilloscope which measures the phase difference between the signals the microphones detect. Tune the sine wave oscillator to the crossover frequency, and move the tweeter back until there is no phase difference between the outputs of the two microphones.

For first order 6dB/octave crossovers, the phase shift the crossover introduces will be independent of frequency, but the drivers also have frequency dependent phase shifts. So, to establish a compromise tweeter position, shift the oscillator frequency up and down for an octave on either side of the crossover frequency. Finally, set the tweeter attenuation, mainly by listening, to achieve the smoothest frequency response.[11] You must set the tweeter level last since moving the tweeter back changes the way the outputs of the woofer and tweeter combine around the crossover frequency. The tweeter/woofer orientation (or tweeter/midrange and midrange/woofer) arrived at through this simpler alignment by time method agrees very closely with that suggested by examining the waveform with the real-time analyzer.

CONCLUSIONS. The ultimate questions then become: 1. When designing a loudspeaker, is it important to consider both the frequency and time domains? and 2. If the time domain is important, can I optimize for both domains in a simple and inexpensive manner (i.e., without a $20,000 GenRad analyzer)?

From my own listening experience, the answer to both questions is "Yes." Although expensive and complex equipment can provide enhanced speaker design capabilities, simpler and morereadily available equipment can work nearly as well when coupled

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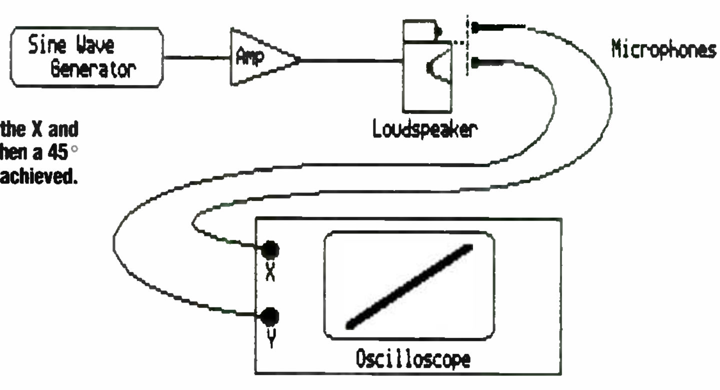

The signals sent into the X and Y axes are in phase when a 45° slope straight line is achieved.

Figure 13: Time alignment test apparatus.

A GOLD RIBBON SYSTEM

BY DUKE LeJEUNE

Home-brewed speakers that satisfy a smitten builder's sonic obsessions are seldom endearing to his wife. Such need not be the case if you accept the challenge of incorporating your wife's preferences into your design criteria. In this article, I will describe a system I built that has pleased my wife and me.

When the time came for me to build a new pair of speakers, there had just been an exciting development in the world of drivers. An American(!) company, Gold Ribbon Concepts, announced a superb ribbon driver. The ribbons were not cheap, but I sold some equipment to help offset the cost. As I began mulling over possible designs and weighed the inevitable tradeoffs, I decided to try a novel approach: I asked my wife what she wanted the speakers to look like. After all, she had to live with them as much as I did. Armed with my pile of *Speaker Builder* back issues, I looked forward to the challenge of getting the best possible performance, given her design preferences.

I showed my wife some of my sketches as well as pictures in magazines. We also looked at speakers in stores. She preferred the configuration you see in the finished speakers in *Photos 1* and *2*, which was based on one of my sketches. I envisioned them being satellites that would (I hoped) serve well as full-range speakers, with a subwoofer as an optional extra rather than a necessity. I figured two 7″ woofers per side would give me a good combination of bass extension, transient accuracy, and dynamic range.

DRIVER MOUNTING. At first glance, the woofer placement on my speakers looks crazy (*Photo 3*). Instinct tells us it is a serious mistake to have the voice coils so obviously misaligned.

Let's look at how far, acoustically, the woofers are behind the ribbons.

For a seated listener in a normal listening position (off-axis to the inside of the

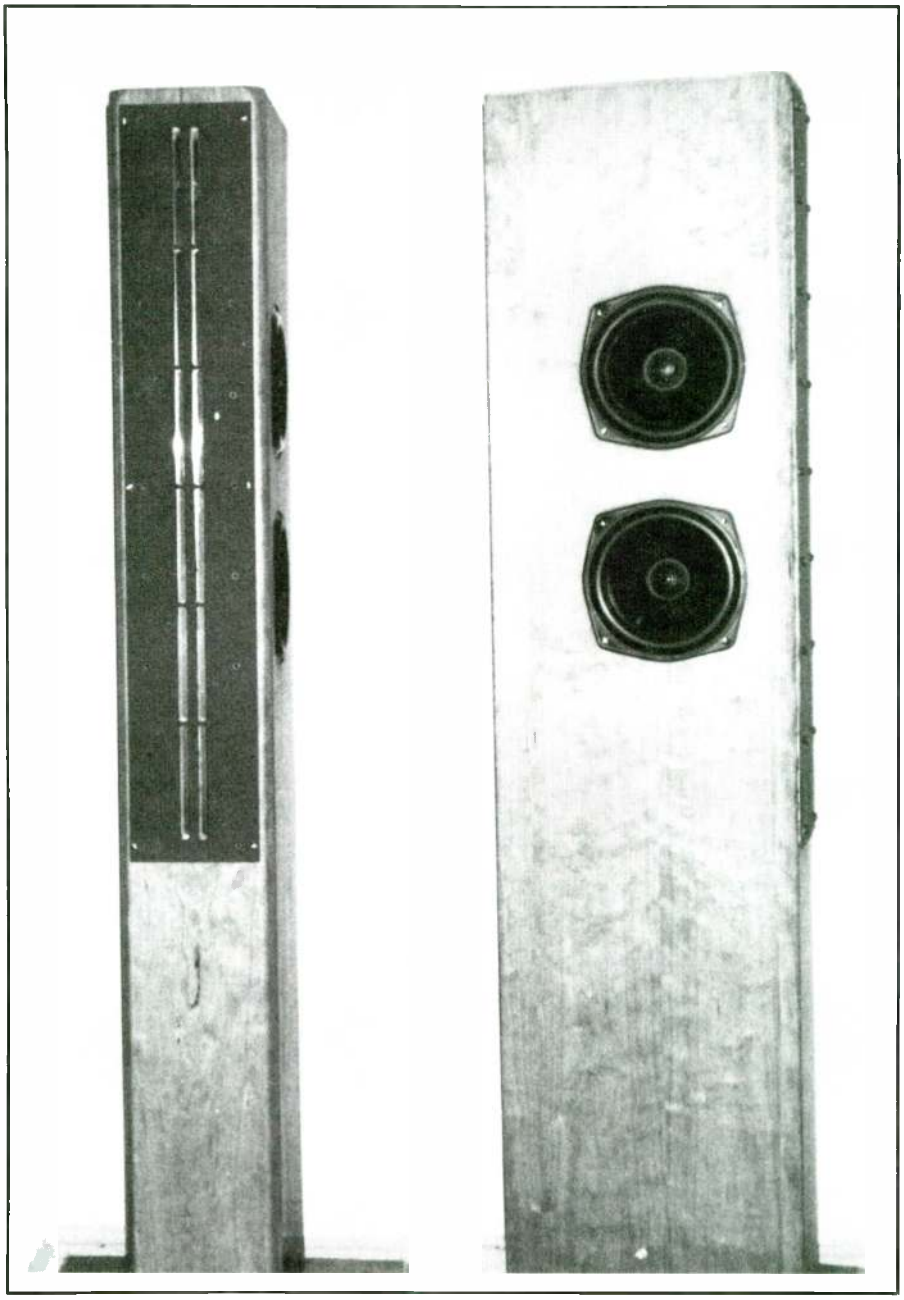

PHOTO 1: Front and side views of the Gold Ribbon system.

speakers), the woofer voice coils are about 3½″ farther away than the ribbon diaphragm. This corresponds to about 1/10 wavelength at the crossover frequency of 400Hz, or a 36° phase lag, which is probably barely audible.

From a normal listening position, the woofer voice coils on my system are about 4″ to the inside of the ribbons, which once again may be barely audible. The path length around the box from the woofer voice coils to the ribbon diaphragm is about 8″, which is less than ¼ wavelength at the crossover frequency. I don't think there is any significant lobing, since lobing occurs where two sources are about one wavelength apart. The small frontal area helps cut down on ribbon output diffraction.

Let's look at other possible driver mounting configurations. *Figure 1a* seems to be the most logical: ribbon over woofers. I tried this with Strathearns, and in my rather small living room, I was aware of the lows coming from below the highs. This caused a slight but distracting lack of coherence. Allen Hulsebus, an engineer at Gold Ribbon Concepts, told me this configuration imaged the best, but he acknowledged the possibility of audible vertical frequency imbalance if you weren't far enough away from the speakers.

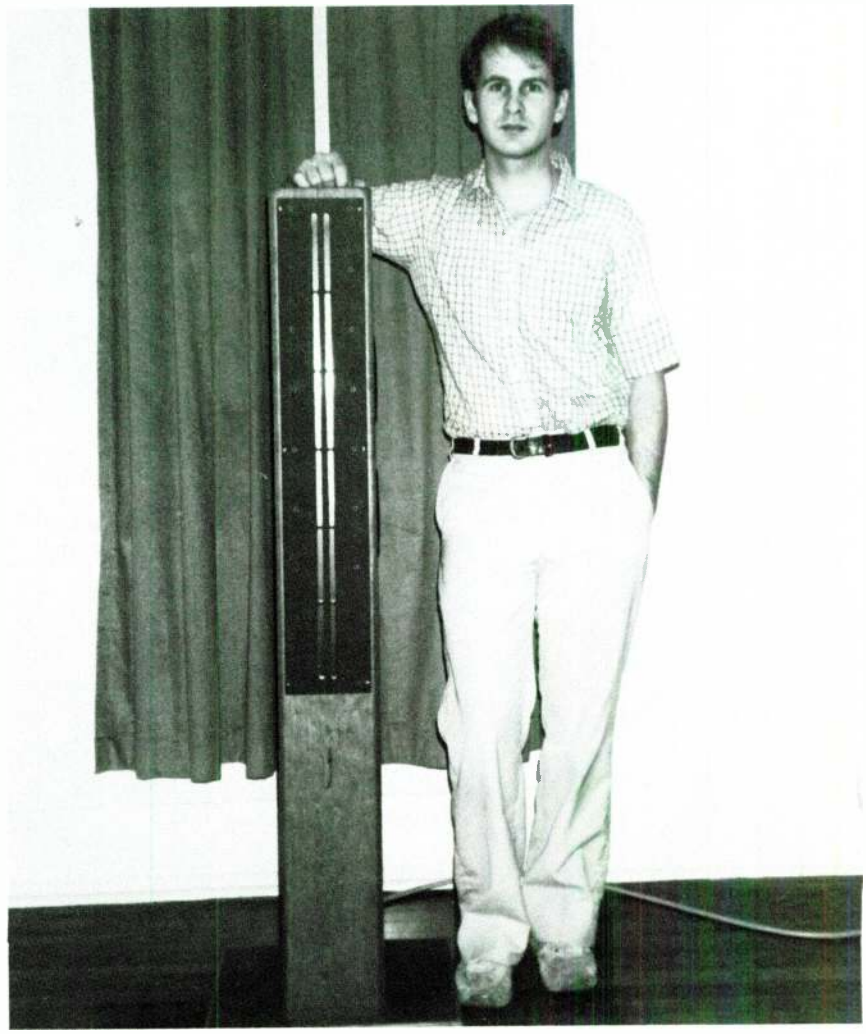

PHOTO 2: Front view of the system with 5′ 8″ author shown for scale.

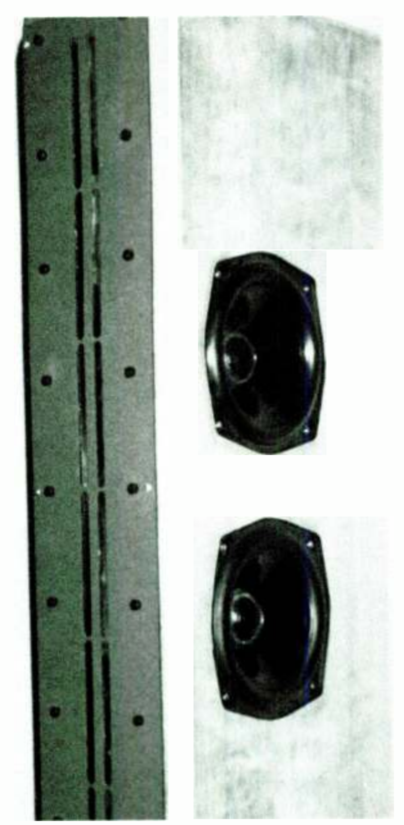

PHOTO 3: Close-up of driver mounting configuration.

Figure 1b shows a configuration I once used, again with Strathearns. It was pretty lousy. There was a peak in the lower midrange (probably attributable to lobing), and weak bass response.

Figure 1c is a configuration I haven't tried yet, but Allen told me it doesn't work well, probably because the inner portion of the ribbon's output is excessively diffracted by the woofer cones.

By swapping these speakers, you'll get a more promising configuration as shown in *Fig. 1d*. The diffraction here is to the outside, which is much less critical. Allen had very good results with this one, and thought the imaging was very close to the ribbon-over-woofers configuration. From a normal listening position, the woofer voice coils would be about the same distance behind the ribbon as in my configuration, and about the same distance to the side.

By now, you can see that tradeoffs are involved when you select a configuration. I chose to trade off a little imaging for a little coherence and better looks. If you wish to build a similar ribbon system, I believe mounting the woofers on the side of the enclosure is a viable option. For me, it is the best choice because my wife likes it, and because it sounds better to me than other configurations I've tried.

One beneficial (and unplanned) side effect of mounting the woofers on the side of the cabinet is it minimizes unwanted midrange contributions from the woofers. The crossovers are first order, and apparently the woofers add a small but audible amount of coloration up to a few octaves above the crossover point. The woofers tend to beam at these upper frequencies, so mounting them on the side of the cabinet directs this unwanted output to the side. From a normal listening position, the combined crossover and beaming rolloff on the woofer is first order in the octave above the crossover, then it accelerates up to roughly fourth order by the time you reach the third octave above the

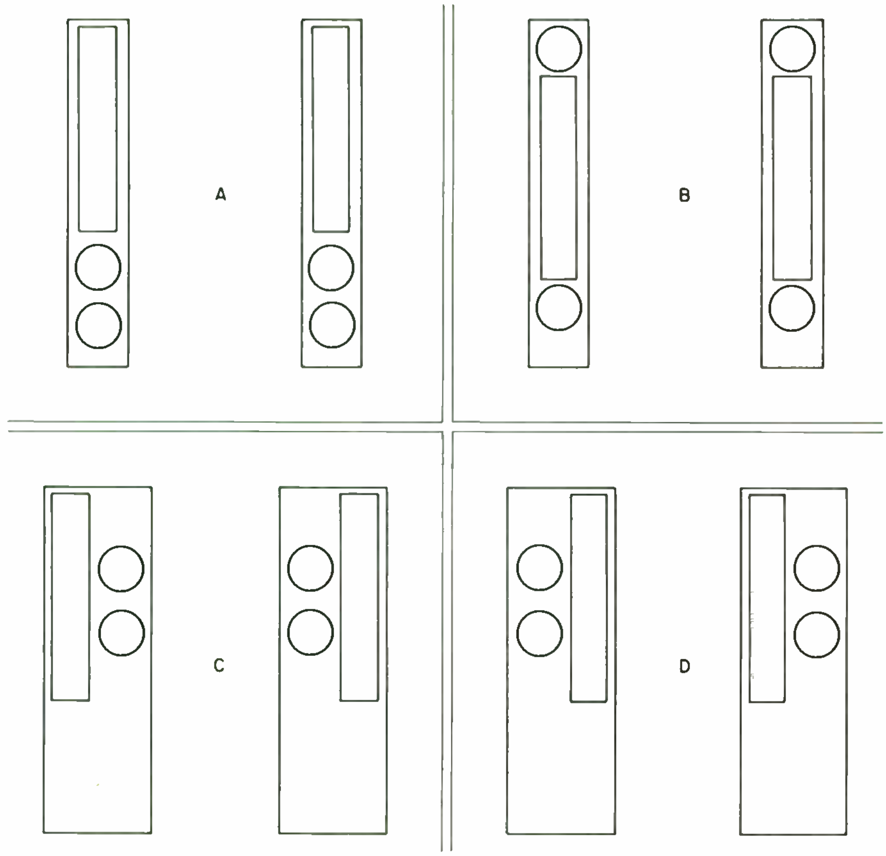

FIGURE 1a-1d: Possible alternative driver mounting configurations.

crossover (see pp. 11-14 of Martin Collom's *High Performance Loudspeakers*, Pentech Press, second edition).

It was a challenge for me to design a satisfactory bass system for the slim, low internal volume enclosures. You see, I am a transmission line devotee, which implies I'm more familiar with large boxes. Transmission line rules of thumb (line lengths that work out to about 7 to 10', and cross-sectional areas greater than or equal to woofer cone area—see Gary Galo's article on p. 7 of *SB* 1/82) indicated I'd need a box four times the size available if I wanted to use two 7″ woofers. Obviously, I would have to break some traditional rules.

SHRINKING THE LINE. Not everyone sticks strictly to the book. B.J. Webb's transmission line (*The Audio Amateur*, 1/75) was tapered to an area much smaller than tradition would call for. Judging by his response curve, I think he traded off some low bass, but he took roughly ⅓ off the box volume, saving about 2 cubic feet. I planned a similar approach, and since I was calling my speakers "satellites," I could add a subwoofer later without too much embarrassment.

In his article, "The Use of Fibrous Materials in Loudspeaker Enclosures" (The *Journal* of the Audio Engineering Society, 4/76), Bradbury indicated another possible place to fudge. It seems that the speed of sound is reduced by a fibrous tangle, which means your stuffed transmission line doesn't need to be as long as tradition would call for. Although this effect is frequently mentioned in connection with transmission lines, few home designers seem to trust it much, probably because the real world doesn't consistently conform to the equations. Apparently, the big boys (such as IMF) somehow take advantage of the "Bradbury effect" in their designs in order to get away with much shorter lines than the rest of us are accustomed to (see Richard Painter's letter on p. 37 of *SB* 3/84). Bradbury's equation gives the speed of low frequency sound in a stuffed enclosure as being equal to

$$\frac{1{,}130'/\text{sec}}{\alpha}$$

where $\alpha = 1 + P/P_a$, with P being the packing density of fiber in kilograms/cubic meter, and P_a being the density of air (1.29kg/cubic meter). For calculating convenience, 1kg/cubic meter is equal to approximately 1 oz./cubic foot.

I called Gold Ribbon Concepts for help because they were using short systems. Allen told me he'd found a 16 oz./cubic foot density of "Polyfill" by Fairfield Industries (sold at K-Mart) would slow down the speed of sound to close to 308'/sec., which is what Bradbury's equation predicted. I was told the area immediately behind the woofers should be stuffed with only 12 oz./cubic foot. Allen said the minimum line length is about 45″, roughly ¾ wavelength at 60Hz. He added there would be a very minor bump here, attributable to the ¾ wavelength resonance, and the −3dB point would be about 40Hz. Although transmission line theory (explained in Mr. Galo's previously cited article) says to use a ¼ wavelength line, Allen found that using a ¾ wavelength line, along with the proper stuffing density, gave the best results. Lines much shorter than 45″ produce major peaks above 100Hz, while longer lines produce deeper bass. Longer lines also need less stuffing, and you can decrease the stuffing density roughly in proportion to the increase in line length.

By using the information Allen gave me, along with the Webb article, I ended up with a 45″ line having a cross-sectional area that averages slightly more than half the woofer cone area. This should be seen as an absolute minimum, and it breaks most of the rules.

My woofers are Focal 7N501s. I had planned to use the Dynaudio 17W75 woofers, but Allen said he'd had excellent results with the Focals. By the time you read this, Gold Ribbon Concepts should have received shipment of custom-made 7N501s featuring twice the linear excursion of the stock model and improved dust caps.

The little Focals in their transmission lines are sonically superior to a slightly larger Isobarik woofer system I had on hand that measured −3dB at 40Hz. I wondered what sort of bass response other enclosure types would yield from a box the size of my small transmission lines (32 liters devoted to the woofer section). I calculated the −3dB points for several pairs of 7″ woofers (including the ones I used) in both sealed and vented enclosures, and few approached 40Hz. I had similar results when I repeated the calculations for single 8″ woofers. In other words, my system goes at least as low as most other enclosures of its size, yet retains the transmission line's transient accuracy and low inherent coloration. I believe the small transmission line is a viable alternative to other enclosure types where size is limited.

To give you an idea of the bass response I traded off by shrinking my transmission lines, Allen recommended a 60–65″ line with a cross-sectional area equal to the woofer cone area which would go down to 35Hz. He has built a no-compromise 90″ line which has a cross-sectional area of 1½ times the woofer cone area, and is –1dB at 30Hz. These systems use the same woofers I used, and their enclosures would be about three and six times larger than mine, respectively.

I believe the transmission line's most significant benefits are in the midrange instead of the bass, so I used a transmission line to handle the ribbons' rear wave. I had used this sort of enclosure for some of my Strathearns, and preferred it to open-backed or sealed (2′ deep triangular) enclosures. I used a ±2′ line length and a stuffing density similar to that in the woofer section. I tapered this line much more than the woofer line because, by the time the rear wave of the ribbon approaches the end of the line, it has been greatly attenuated by the Polyfill.

CONSTRUCTION. To give you a reference point for your own designs, I will briefly describe my speaker construction.

Figure 2 is a simplified rear view of the internal baffling. The woofer line goes down the woofer side of the enclosure and then about ⅓ up the far side. The thin vertical partitions are ⅜″ plywood, the front baffle is solid ¾″ cherry, and all other boards shown are ¾″ particle board. I used about 15 internal cross-braces and a couple of cleats, all of which I have omitted from the drawing.

Figure 3 is a top view of the same thing, showing some important dimensions. Note the 30° angled board has its forward corner trimmed to make room for the ribbon's magnet structure. In the woofer chamber, the angled board and thin partition are both lined with 1″ open-cell foam, with appropriate slices removed to accommodate the woofer magnets. In the ribbon chamber, all surfaces in the forward (triangular) part are lined with ⅜″ felt. (Foam and felt are not shown.)

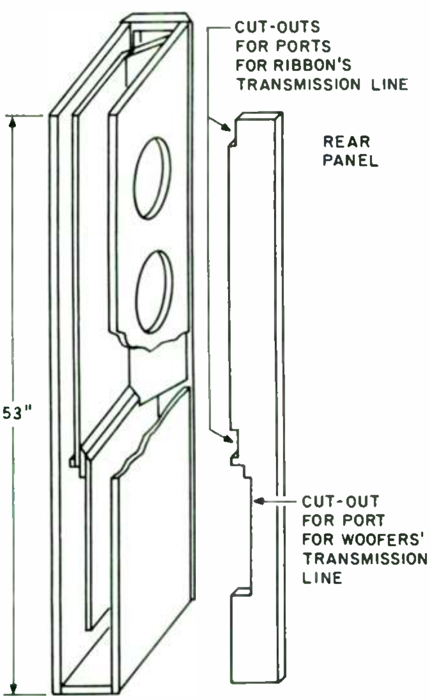

FIGURE 2: Simplified rear view of the cabinet's internal structure. (For clarity, the braces, drivers, ribbon chamber partitions, and top have been omitted.)

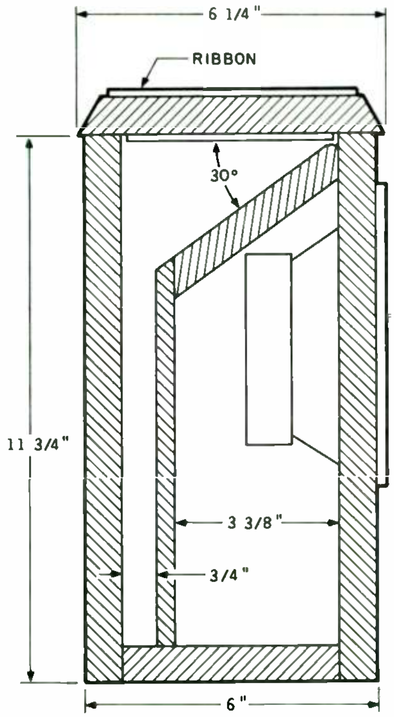

FIGURE 3: Simplified top view of the internal cabinet structure.

Figure 4 shows the internal partitions in the ribbon's transmission line. Since the partitions are all ¾″ thick, they are easy to cut from any ¾″ board. You can vary the geometry as you see fit.

Using wood glue and screws, I first built the enclosures as you see them in *Fig. 2*, except that the top board was in place. I glued the foam in place before gluing on the side panel containing the woofer cutouts. The assembled and tested crossovers went on the bottom of the enclosure. The capacitors are pretty big, so I had to be careful to leave enough room so that the line wouldn't be constricted. I put the input terminal on the rear panel below the port for the woofer's line.

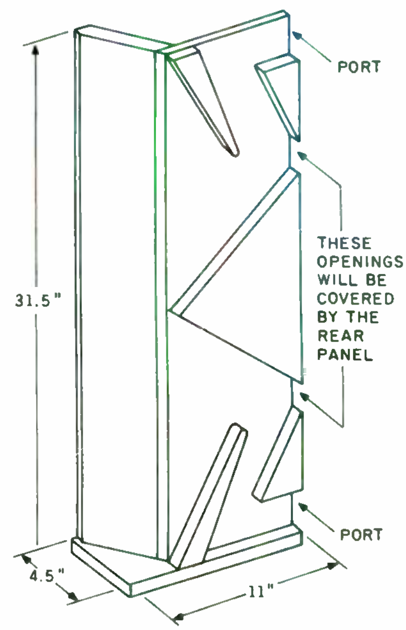

FIGURE 4: Ribbon chamber internal partitions.

The rear panel fit snugly between the sides, as shown in *Fig. 3*. Don't put on the rear panel until you have completed the wiring and stuffed the enclosures. Stuffing is critical and took me several hours. Note that no provisions are made in my design for getting back into the cabinet and restuffing it. I do not recommend this unless you are very confident about your ability to properly stuff a transmission line. The Polyfill will not settle. In *Fig. 4*, there are a couple of little openings that will be covered by the rear panel. These openings facilitate stuffing.

Once I had the rear panels in place, I was ready to apply ⅛″ cherry plywood as a "veneer." Note in *Fig. 3* the front baffle is ⅛″ wider on either side than the rest of the box. This is so it will match up well when you apply the cherry plywood to the sides. The cherry plywood is fairly soft and easily marred, so proceed carefully. In an effort to approximate flush mounting of the woofers, I cut holes in the cherry plywood corresponding to the woofers' shape, as shown in *Photos 1* and 3. I covered the top of the enclosure with a solid piece of ¼″ cherry.

Since the speakers are top-heavy, you'll need a base. For durability, I used ¾″ plywood with ¼″ solid cherry around the edges, and covered the tops of the base with ⅛″ cherry plywood.

I used fine steel wool and natural Danish oil to bring out the beauty of the cherry. Once the oil is thoroughly dried and the finish looks good, you can mount the drivers.

CROSSOVERS. I had everything necessary to assemble a biamplified (or even triamplified) system, but I sold all of this equipment to help finance the ribbons. As a result, I used passive crossovers. I calculated component values and called Allen for confirmation. Since he had spent a great deal of time perfecting a crossover for these drivers, he helped me refine the crossover design. The filters are first order types, with impedance compensation on the woofers and high quality components throughout. A biamplified system could include a 2.5msec delay in the high pass section to compensate for the woofers being behind the ribbons.

Refer to *Fig. 5*. You form C1 by paralleling two 80μf Chateauroux, one 40μf Chateauroux, and one 1μf Chateauroux. I substituted some leftover WonderCaps. Form R1 by paralleling multiple small resistors to get at least 40W power handling. I used eight 4Ω, 10W resistors for an 80W rating. The purpose of R1 is to match the level of the woofers to the ribbons. Do not use a high DC resistance inductor for L1 in an effort to include R1 in the internal resistance of L1. Form R2 and R3 by paralleling two 15Ω, 10W resistors.

LISTENING TESTS. The remainder of my stereo equipment consists of an Apt/Holman Preamplifier Two, an Electron Kinetics Eagle 2 amp, a Luxman D-405 disc player, a Nakamichi cassette deck, and a Mitsubishi turntable. I need a high current amp with these speakers since the ribbons are a 2Ω load.

My wife and I were impressed with the musicality of the Golds. With a good recording, they sound convincing and even startling. You can follow individual instruments through complex passages, and there is a most welcome freedom from those little aberrations that can ruin it all.

I was curious about how they would compare with a really fine reference system. The best speakers in town were the Theil CS3s, which sound better than any other commercial system I've heard for less than $4,000. After extensive listening, I thought the Theils were a bit hot at the top end, but otherwise were very convincing. In my opinion, the Golds are easier to forget, and they convey more of the feeling present in live music.

Realizing my critique of my own speakers was, at best, open to question, I called in audiophile Stanley Morris. He liked the Golds better than any other speakers he'd heard, except for the Theils. He was impressed with my speakers' tight bass response and clean highs, but thought the sound was slightly veiled when compared to the highly-analytical Theils.

I also called in musician Steven Henry. Steven is not only an accomplished clarinetist, he is also blind, and has incredibly acute hearing. He listened at length to both systems and thought they were both excellent. He quickly picked out the slightly more prominent highs of the Theils, but was not sure which system had the more natural highs. He told me the Gold system had better clarity, definition, and more precise instrument location. He commented on how easy it was to either listen to the music as a whole or listen to only one instrument, and how individual voices were easily discernible on good choral recordings. Steven also thought my speakers had more natural-sounding bass response than the Theils.

None of those who listened to my speakers thought I needed a subwoofer, but the ultra low bass does lack the depth and power a fine subwoofer could add. Let me stress that your subwoofer must be of very high quality so as not to detract from the system's transient speed.

CONCLUSION. I hope this article gives you some new ideas as you contemplate your next speaker project. Mounting the woofers on the side of the enclosure is appropriate in certain applications, and the shrinking I have described extends the transmission line option into the area of fairly small enclosures. It was challenging and rewarding to include my wife's preferences in my design criteria, and she and I are pleased with the results.

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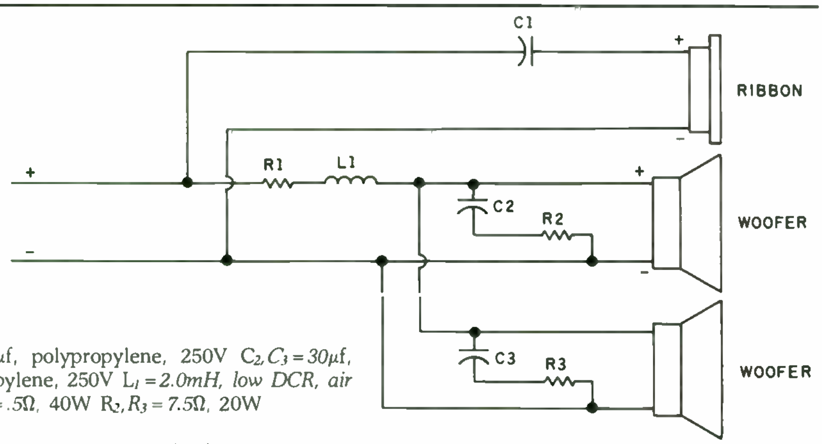

$C_1 = 200\mu f$, polypropylene, 250V $C_2, C_3 = 30\mu f$, polypropylene, 250V $L_1 = 2.0mH$, *low DCR, air core* $R_1 = .5\Omega$, 40W $R_2, R_3 = 7.5\Omega$, 20W

FIGURE 5: Passive crossover for Gold Ribbons and Focals.

ABOUT THE AUTHOR

Francis "Duke" LeJeune works as a landman for the O'Niell Oil Company in Louisiana. He has been an audio buff for nearly ten years, and has been building loudspeakers since 1979. The speakers described in this article are his seventeenth project.

ACKNOWLEDGMENTS

I am deeply indebted to Allen Hulsebus of Gold Ribbon Concepts. His technical assistance was essential to the success of my project, and he kindly allowed me to include his information in this article. I would like to thank Stanley Morris, Steven Henry and, most of all, my wife Shelle.

The hafler new XL 280 evolution

As an evolutionary development with revolutionary implications for amplifier evaluation standards, this latest David Hafler circuit may be the most important power amplifier event in recent years. It is certainly the most thought provoking. Described as **EXCELINEAR**, it demonstrates previously unattainable levels of amplitude, phase and transfer characteristic linearity. For the first time, an amplifier is so accurate, it can be directly compared with the classic zero-distortion reference: a straight wire.

EXCELINEARITY reduces *all* distortions to inaudibility—even those which have not yet been identified or quantified. An audacious statement? Here's the proof: In a differential comparison test with a straight wire the XL-280 is the first amplifier to achieve a null (on the order of 60dB) which taxes your perception. In a differential arrangement the straight wire and the amplifier are fed the same signal, with the input signal to the amplifier adjusted for an overall gain of unity.

The speaker reproduces only signals which differ—*only the distortion*—and you hear *all* of it, too. This test surpasses conventional measurements in its sensitivity and precision. When it has been tried in the past, amplifier flaws were so apparent that input signals had to be specially corrected. No longer. This test is definitive with any program content. In justifying some unquantifiable 'golden ear' assessments of differing amplifier sounds, it gives new meaning to the saying "Silence is golden."

As delivered, the XL-280 has conventional harmonic and intermodulation distortion specifications comparable to the current Hafler DH-220 amplifier. It includes a phase correcting circuit which enables individual adjustment for the differential null. This has been set at the factory for optimum interfacing with a 'standard' loudspeaker load. Different speaker designs present varying loads to the amplifier which can introduce minute phase and amplitude related variations. This is one reason why all amplifiers do not sound the same, and why the sound of any one may vary on different speakers.

For the purist, the XL-280 includes the capability for 'tweaking' each channel to optimally match (null with) your speaker. This range of adjustment is entirely within the XL-280's printed specifications, and is of significance only to 'golden ears'. But it can be performed without test instruments, with patience, in quiet surroundings. It does require very precise adjustment of the amplifier's input level. The procedure is described in the XL-280's manual, and we will supply details to those who request them. Comparing 'nulls' of other amplifier designs is sure to provide grist for erudite discussion.

The XL-280 appears similar to a DH-220, but is an inch wider. It retains the traditional Hafler full complementary circuit driving MOSFET outputs. It has J-FET input stages for lower distortion, lower noise, and improved thermal stability. We've eliminated the conventional input capacitor to reduce low frequency phase shift. The output choke has vanished to improve high frequency phase characteristics, yet excellent stability has been retained. Overall feedback has been reduced about 30dB to lower TIM distortion. And you can switch to bridged mono operation when you need a real powerhouse.

But the biggest changes are to separate power supplies for each channel, and in the output stage. Only the transformer is common to the two channels. Power supply capacitance is up 56%. There are separate high and low voltage supplies on each channel. And we've added another pair of the same rugged MOSFETs on each side. The results are improved channel separation and greatly increased power into low impedance loads. Evolutionary—yes. But of revolutionary significance. And the price goes up only $100! Check with your dealer the end of October.

COMPARING THE XL-280 AND DH-220

	DH-220	XL-280
Rated Power into 8Ω each channel at specified distortion levels with both channels driven. (FTC rating)	115 watts	145 watts
Power into 4Ω—per channel continuous	220 watts	280 watts
Power into 2Ω—per channel short term	200 watts	360 watts
Power into 1Ω—per channel short term	100 watts	325 watts
Maximum RMS output current—per channel	10 amps	20 amps
Number of output devices per channel	4	6
Number of power supplies	2	4
Power supply capacitance	20,000μF	31,200μF
Cost per watt @ 8Ω rated power	$2.17	$2.07
Cost per watt @ 4Ω rated power	$1.25	$1.07

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A VISIT WITH KEN KANTOR

BY BRUCE EDGAR
Contributing Editor

I first met Ken Kantor during a side visit to Acoustic Research while attending a scientific meeting in Boston in 1985. AR had just introduced the Magic Speaker, and Ken gave me a short demonstration of its capabilities. It was a radical departure from the bookcase speaker design AR had traditionally produced.

On a subsequent visit to AR last July, Ed Dell and I interviewed Ken, and Alex deKoster (AR's Director of Engineering). We also toured the lab and production facilities. Ed and I were very impressed with Ken and Alex's openness on technical matters.

As you will see, Ken Kantor represents a new breed of loudspeaker designers who are not tied to any one company for an entire career. Ken links up with loudspeaker production companies who want new and innovative products. His keen observations shed new light on such topics as loudspeaker design and testing, and the future of loudspeakers. My knowledge design was enlarged by the interview, and I'm sure your horizons will also be similarly broadened.

FIGURE 1: Ken Kantor, loudspeaker designer.

Speaker Builder (SB): *Ken, how did you get started in the audio business?*

Ken Kantor (KK): Well, like many others, I grew up with a dual fascination for equipment and music. With a constant interplay of the two, I began to experiment with "improving" commercially available equipment.

SB: *For example?*

KK: I started with a mono system but became very interested in the idea of stereo. Walking home from school one afternoon, I found an old record player with a tiny mono amp. It had been left for the trash collector. I plugged it in and got a wicked shock. I repaired and fitted it with a $4 crystal stereo cartridge and used two different speakers—whatever I could find. Suddenly, I experienced stereo; the sound didn't come from the speakers, it came from somewhere else.

As soon as I managed to save enough money, I purchased some old Dynaco equipment and a used AR table and had the classic system for its day. I began to play with the equipment to see what changes I could make. Eventually, I designed my own equipment from scratch.

SB: *How did you decide to go into engineering?*

KK: I was an amateur musician at the time, doing some recording and experimenting with the early synthesizers. I decided whether I really wanted to pursue that course, I needed to learn electronics, not only in terms of making electronic music but also in terms of sound reproduction. I looked around at the colleges that would give me access to both music and engineering. MIT was my choice, and I received a B.S. in E.E., with a minor in music.

SB: *While you were at MIT, did you work with audio equipment?*

KK: Yes, I began to combine theory with my practical knowledge and designed equipment for local companies. I did circuitry and the like to earn money for school.

As an undergraduate, I was privileged, (somewhat accidentally) to be assigned to a lab at MIT which was one of the real audio hotbeds in the US. Dr. Bose did his original research in that lab, as did many other researchers in psychoacoustics and perception. I began doing less and less of my assigned project and became involved with graduate students who were doing their work in audio.

My first contact with AR, as a matter of fact, was in 1975 in the form of a very shy letter to Bob Berkovitz (who was then Director of Research at AR) proposing an idea for a new kind of loudspeaker. I was thinking about doing my undergraduate thesis on it, and I explained my concept of controlling the radiation pattern and delaying the ambient field to more closely approximate what I thought were the characteristics of an ideal loudspeaker.

SB: *What influence did the ideas of Bose have on you?*

KK: I guess there were several streams of thought, one was Bose's, that formed the genesis of my ideal loudspeaker concept. You have to give him credit for identifying some of the fallacies of what people thought about loudspeakers.

SB: *For example?*

KK: You can't lump the radiation of a loudspeaker into a single parameter, plot it versus frequency, and say it characterizes the system. It doesn't work that way because the brain doesn't work that way. Bose was one of the first people to commercially explore the concept of designing a loudspeaker that explicitly dealt with sound going in different directions to achieve a certain goal.

He sat down and said, "Look, there is a qualitative difference between direct and reflected sound, and this difference contributes to what you hear." The limitation in his thinking, in my opinion, was he didn't fully explore what makes reflected sound special. Sound waves do not change properties simply by reflecting off a wall.

I think the key question is: "In a sonic environment, what makes the reflected sound different from the direct sound?" One difference is the direction from which it arrives, another is its relative frequency spectrum, and a third is its relative arrival time. You can't mix 1 oz. of direct sound with 8 oz. of reflected sound and expect perfect reproduction unless all three issues are adequately addressed.

SB: *To backtrack somewhat, did you own any AR speakers in your student days?*

KK: I grew up on the AR sound. AR was science, and since it was science, it had to be right. Most companies were just making boxes and putting drivers in them. AR seemed to be thinking about sound reproduction in a logical and intelligent way. They seemed trustable, and you always want to trust in something. I owned secondhand 2axs and, later, secondhand LSTs, along with various other brands.

Then, one day, I ventured into a "high-end" shop. I was intrigued by all the exotic designs they sold. I started listening, and I heard details I missed before. Suddenly, I could really tell where all the instruments were. I began to think about what speaker design attributes allowed you to hear all that detail.

SB: *What aspects made the high-end speaker sound different?*

KK: Well, I went home and did some listening with filters and equalizers, and they pretty much eliminated bandwith and frequency response as the cause. I decided it was high directivity, with minimal side reflections and diffraction. Unfortunately, these directional speakers, while detailed, did not seem like real music to my ears. Something was still missing in the approach.

SB: *So, you want your speakers to sound musical?*

KK: Yes, I want the sense of a physical entity in the room. You listen, for example, to some of the finer electrostatics, and you get an extremely high resolution picture of the music. But it is a picture and not a physical entity in the room. On the other hand, when you go back to listen to the old AR-3As, you get this wonderful sense of fullness, but without the detail.

This is what led me to investigate radiation patterns. I began to read psychoacoustics literature. I concluded that reflections are important, yet everyone was sweeping them under the rug. Designers either try to avoid reflections or to integrate them into the measurements in some way that will make them disappear. It should be an issue of optimizing the characteristics of the ambient field in the listening room.

SB: *You felt it was the treatment of ambient sound that was the most important item?*

KK: No, it is the ambient sound and first arrival sound. The important issue was how to get one without sacrificing the other. What would allow my brain to handle the localization details and still get a sense of room acoustics? That's when I began to think about delaying rather than eliminating reflections.

SB: *You were doing this at MIT. What did you do after graduation?*

KK: I finished my undergraduate thesis[1] and produced some prototypes that displayed the capabilities of the idea. I demonstrated them for AR. They said my idea was interesting, but not appropriate for their current product development. The prototypes wound up in my home stereo system. I had to stop working on them, but the idea stayed in my mind. (See *Fig. 2.*)

I did consulting work for six months, then returned to MIT for a Master's degree at the Center for Advanced Visual Studies. It consisted of a group of people involved in technology as it relates to artistic and creative pursuits. It was a very nice place for me. People were working with sound and music, computer graphics, holography, video art, and the like.

SB: *Was it an interdisciplinary program?*

KK: Yes. People, including many innovative artists, would come in on fellowships with credentials in very diverse fields.

SB: *Did you work on speakers then?*

KK: Not explicitly. I did more work in psychoacoustics of localization, and helped develop the Sound Concepts ambience restoration unit, a system for elec-

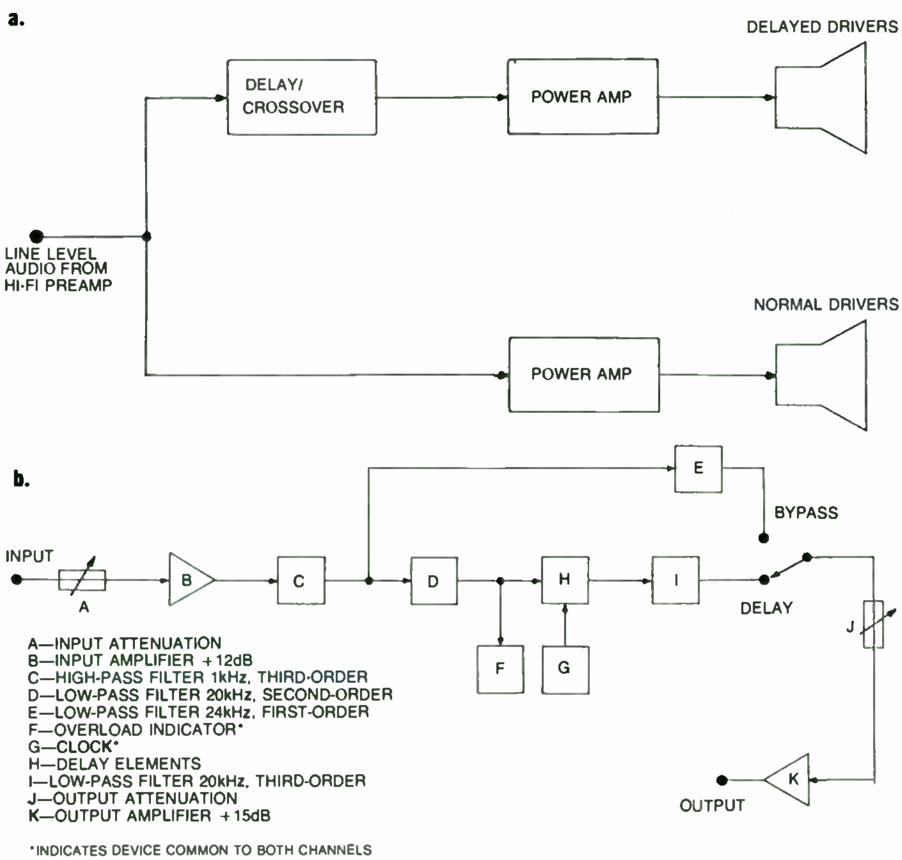
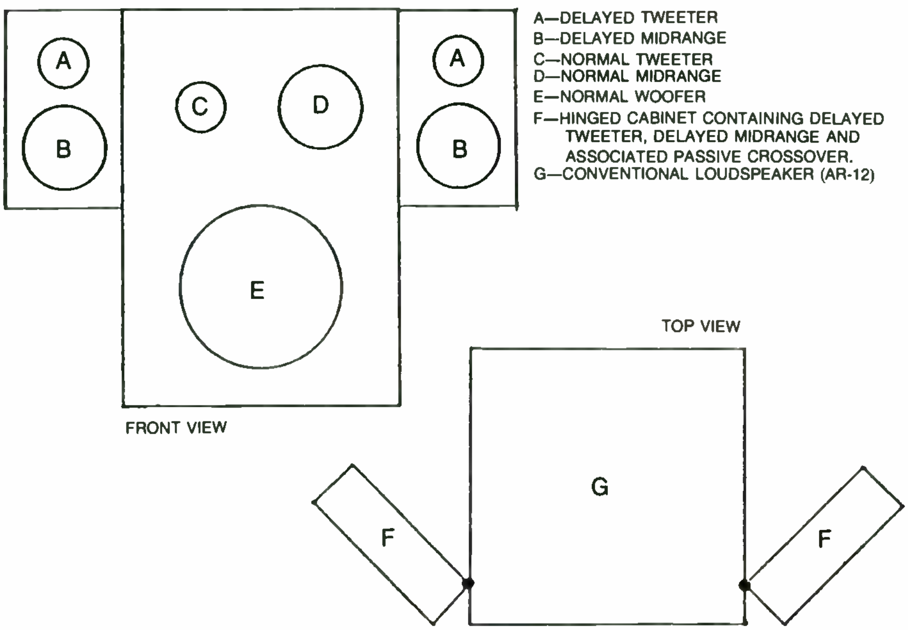


FIGURE 2: Some original drawings from Kantor's MIT undergraduate thesis which formed the genesis of the Magic Speaker concept: (a) block diagram of loudspeaker system; (b) block diagram of delay unit.

a.



b.

FIGURE 3: Original driver layout in Kantor's MIT thesis (a) and a prototype array developed at AR (b).

tronically reducing interaural cross-correlation. I also did hardware work for the AR adaptive digital signal processor. I was most interested in applying advanced audio technology to creative ends, such as music compositions that used three-dimensional space. After leaving MIT, I went to work for the NAD research group in London.

SB: *Did AR come after you?*

KK: We never lost touch. NAD grew out of AR anyway, so there was communication. I eventually returned to AR as a consultant to generate some saleable products from the digital signal processor technology. This led to the SRC remote control, which was my first product for AR.

After the SRC, I was invited to join the company as director of research and development. My mandate was to restructure the R&D effort to better complement product development. After I took over R&D, the president of AR came to me and asked, "What can we do in terms of new speakers?" I just happened to have my thesis project in my back pocket. I began to refine the concept into a product and helped fit it into the company's marketing structure.

SB: *I remember seeing a picture in* Audio *with a "sweet sixteen" type array of drivers. Was that an early effort? (See Fig. 3.)*

KK: That was one of my attempts to control directivity. It had 32 drivers per channel and many problems, but I was able to get management to listen to it and say, "Yes, let's continue."

SB: *How did the array evolve into the present form of the Magic speaker? (See Fig. 4.)*

KK: Basically, there are three ways to control the directivity of a loudspeaker. One is to alter the size of the radiating surfaces, including the use of horns. The second is to use multiple radiators in an array, controlling the amplitude and phase of each. The third way is to use absorptive materials. We tried each method.

We dismissed using a single radiator early on because of the large change in polar pattern over the wide range of frequencies in question. There are some clever ways around this problem, but none was considered viable for production. Arrays are very useful for experiments, since it is relatively easy to modify their directional characteristics.

SB: *But isn't an array very difficult to control over many octaves?*

KK: That's right. Drivers, large enough to work well at low frequencies, cannot be placed close enough together to avoid lobing at high frequencies. We could not get a practical array to work over a decade of frequency. Using the absorptive material resulted in relatively constant directivity without lobing. Some radiated energy was wasted, but it was not excessive.

SB: *Did you use foam before to control patterns?*

KK: AR had used foam for other acoustical purposes, but not pattern control. My first prototypes used thick acoustical felt.

SB: *You ended up with a vertical array. What does that do?*

KK: One of the most sonically detrimental speaker/room interactions is the floor reflection. The typical floor reflection occurs at about 2mS, perfect for causing midrange coloration. The foam we used did not adequately reduce this reflection at lower midrange frequencies, since longer wavelengths are more difficult to absorb. For this reason, a special midrange placement is used. This placement creates a null in the vertical radiation pattern in such a way as to complement the foam's characteristics. At any rate, vertical arrays make sense. They tend to keep the stereo image more stable with frequency changes, and they reduce wave interference problems.

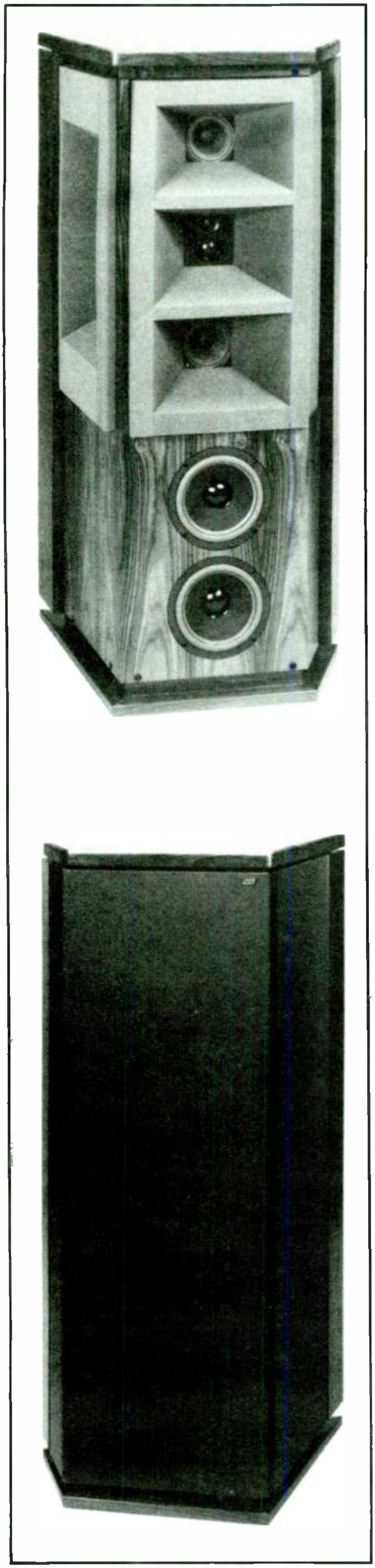


FIGURE 4: The AR MGC-1 "Magic Speaker."[2]

SB: *Over what range is the floor reflection critical?*

KK: It will manifest itself from, say, 300Hz up to several kilohertz.

SB: *What were the design factors of the side-firing drivers?*

KK: Over most frequencies, the side system should exactly complement the front system so you have a hemispherical radiator if you ignore the time delay. Because of the long wavelength of low frequencies, there is no way to control their pattern with a practical size enclosure. You know you don't want the side speakers to radiate at low frequencies. You only want to delay frequencies you can control.

At the upper end, it's tough to determine how high to take the side radiation. Very high frequencies do not contribute much to a sense of ambience, and they restrict the maximum delay time you can use before you hear a discrete "slap" echo. On the other hand, too little high frequency information can upset the ambient tonal balance and restrict listening position. The compromise we chose for the MGC-1 is 5kHz, though I'm not sure it's optimum.

SB: *What is the lower frequency limit?*

KK : About 400Hz.

SB: *Does this frequency span correspond to what psychoacoustics says is important for reflections?*

KK: I think 400Hz to 5kHz is in agreement with what is considered the most important region.

SB: *What is your notion of an ideal loudspeaker?*

KK: First and foremost, you need a flat first arrival signal. This relates to a good anechoic chamber response, but you can't stop there. The second thing is a speaker design that preserves the flat response in a real room to the greatest extent possible—that is, a speaker that minimizes room interactions. It's fine to say you have a flat anechoic first arrival, but if it is polluted by room or cabinet reflections before the brain can perceive it as such, it won't do you any good.

SB: *Your brain/ear system does get distracted by these extraneous signals?*

KK: Absolutely. Early reflections change the tonal perception of a loudspeaker. They also change the localization characteristics and are quite important for final sound quality.

Third, we need a sensible way of dealing with the long-term reverberant field in the listening room. By sensible, I mean right in time and right in frequency. I think with just about every loudspeaker on the market today, the reverberant field is an afterthought. A correct on-axis response in no way guarantees a correct reverberant field. It is necessary for the designer to decide what is wanted, and to find a way to achieve it.

SB: *The way you have done it with the Magic speaker is you have divided it into different regimes of radiation patterns and time domains.*

KK: There are really two speaker systems in there. One system's job is to get an accurate first arrival with no sacrifices. I don't want a floor reflection, delay distortion, vertical interference pattern, and so on. I want a flat, clean arrival, and I want it to be maintained in any room, in any position. The second speaker system means I want a correct reverberant field.

SB: *Some people have referred to this as "soundstage." Is that correct?*

KK: I've heard the term soundstage used in several different ways. Sometimes, it refers to where the source positions are localized, and other times to the spread of the ambient field. I think soundstage is a product of both first arrival and ambient field.

In essence, I'm talking about a very strict approach. We have something existing in the recording environment. We want to reproduce it in the listening environment. I don't want to ignore anything or to sweep anything under the carpet. It's too easy to make excuses, to say, "Well, the listening room is smaller, so the mean free path is shorter, so the reverb can't be the same." I'd rather say, "Here's an original. Let's do the best job possible at reproducing each facet of it."

It's a subtle point, but this approach differs from the philosophy of accurately reproducing the recording microphone signal. I believe the latter approach is non-operative in the real physical system we must confront. Spatial information is lost in the recording process—any recording process—and that information must be restored in the best way possible.

SB: *Do you believe you acheived most of those goals with the Magic speaker?*

KK: Do I think it is a perfect speaker? Of course not. Do I think it opens up a new area of understanding about what hi-fi could create? Yes, I do. Listening beyond whatever limitations there may be, I believe the speaker demonstrates the merit of thinking this way about the problem. I hope it is not the final embodiment of the concept.

SB: *You've done much original thinking about how speakers should be tested. What, in your opinion, are the most pertinent tests that need to be performed on loudspeakers?*

KK: We have the means to test speakers more appropriately. It's a matter of thinking carefully about how to apply the tools we have available. You can take a power amp and test it into an 8Ω load using a frequency sweep. Or, you can test it into a difficult reactive load with the same sweep. It's a matter of finding the experimental conditions that relate to the real world.

It's the same with a loudspeaker. If you have an anechoic chamber, a third-octave analyzer and an FFT analyzer, you pretty much have what you need. I think the test equipment should be fully exploited, however. I guess it comes down to steering the measurements with an understanding of the ear. For example, I talk to engineers who believe the purpose of an FFT analyzer is to make anechoic-type measurements without having to build a chamber. They still try to ignore room reflections, even though they now have the power to measure and tailor them.

With a good FFT analyzer, it is possible to include room reflections in a manner similar to the human ear—use a time-weighting function. Or, you can apply critical-band filtering to judge, with the same resolution as the human ear, the influence of frequency peaks and dips. It is possible to look at the speaker's transient response and judge imaging capabilities. These are not pie-in-the-sky techniques. They can be used routinely. (See *Fig. 5*.)

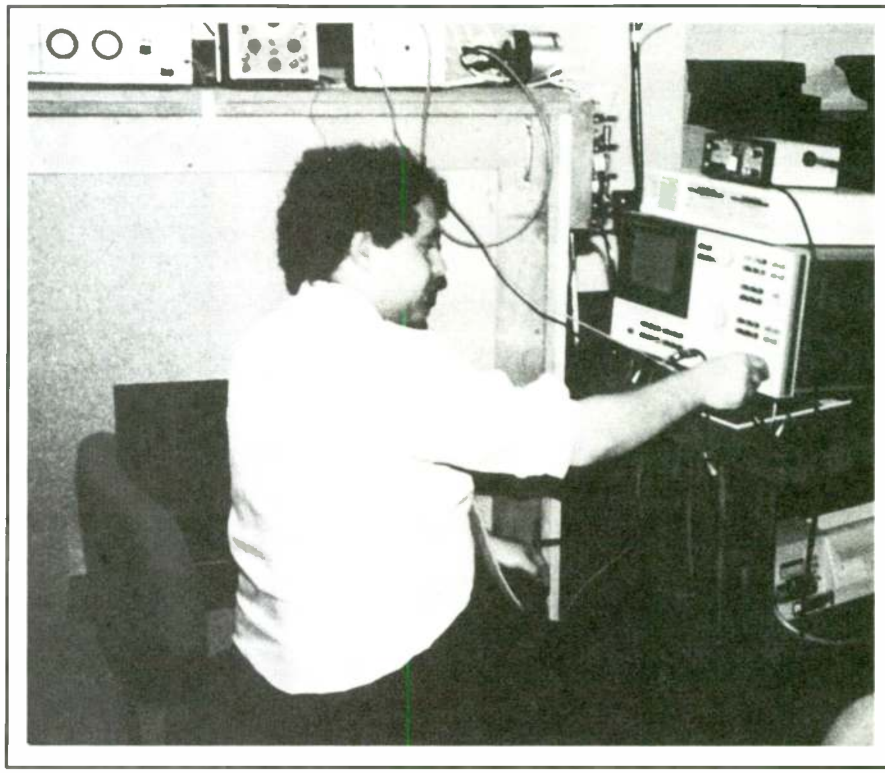


FIGURE 5: Ken Kantor's loudspeaker test facility, which includes a test chamber for small drivers, an HP3562A Dynamic Signal Analyzer, and a laboratory reference microphone.

SB: *How does your approach to measurements differ from others?*

KK: There certainly have been intelligent attempts recently to get subjectively-correlatable speaker measurements. Everybody in the business knows you can't measure a speaker in a chamber and get more than a first-order handle on the sound. Most manufacturers use a combination of listening tests, anechoic sweeps and third-octave room response. This method, especially with a concentration on listening, can produce superb products, but it won't break any conceptual barriers about system design.

Some researchers, notably Canada's Floyd Toole,[3] have defined better test approaches. Dr. Toole's work concentrates on single-speaker frequency domain testing. He reports excellent validation by listeners. I look at the impulse response first, and I am very concerned with stereo imaging. Many methods, however, can enable you to arrive at the same goal.

SB: *Do you have a favorite type of crossover?*

KK: I wish I didn't need to use them. I think they represent a weak link in system design. You can optimize them for this or that, but you can't ever make them do everything you wish. Computers have really improved things, but I'm still waiting for the ideal crossover. I'd rather see improved full-range drivers.

SB: *What do you do differently from published designs?*

KK: I don't have any better ideas. I draw on what has been done by others. I'm working on a design that uses a novel positioning of drivers to ease the job of the crossover.

SB: *What do you mean by novel positioning?*

KK: What makes crossover design difficult is the fact that drivers are physically separated, and so form interference patterns. I'm working on an approach, using reflection and phantom sources, to try to get the effective acoustic centers of the midrange and tweeter coincident. Lower crossover points are less of a problem because of the longer wavelengths.

SB: *What bearing has the phase response of the loudspeaker?*

KK: For all my concern with time domain response, I am fairly convinced the phase response of a single speaker is irrelevant, provided the group delay remains under 2mS in the midrange. Many good studies support this view. 2mS equals about 2′ of offset between drivers, so you can see where I stand on so-called "time-aligned" systems. On the other hand, two speakers in a stereo pair must be well matched in phase to assure proper imaging.

SB: *What about the phase transition from driver to driver?* *Continued on page 26*

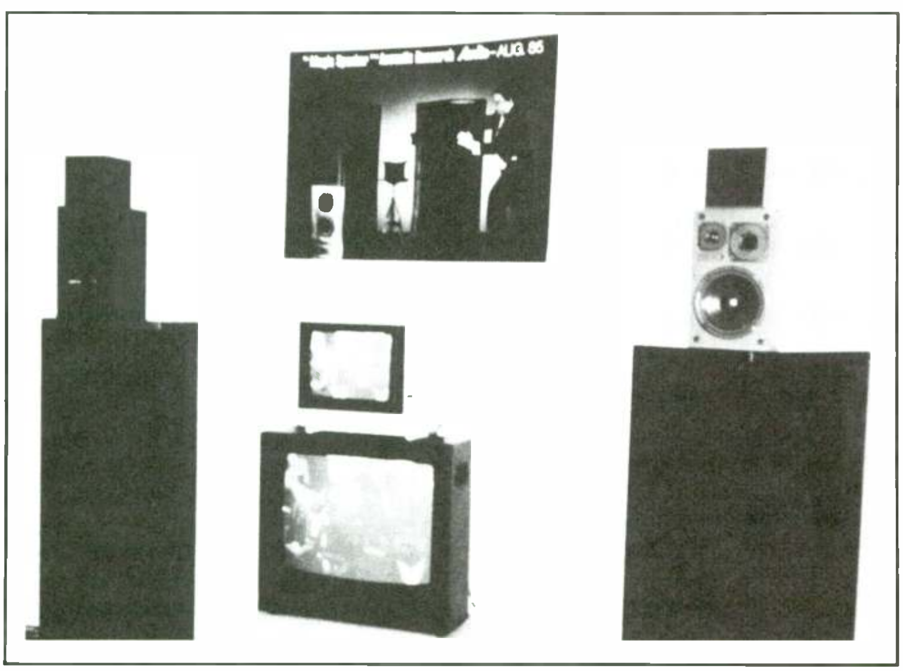


FIGURE 6: Ken Kantor's audio-video setup with video monitors and Magic Speakers.

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THE BEAUTY OF GOLD

GOLD RIBBON CONCEPTS

Continued from page 24

KK: That is critical and is intimately related to crossover design. You can't get correct axial or reverberant response without careful attention to the phase transition.

SB: *When you do a system design, do you worry about matching the phase to achieve the seamless phase response?*

KK: If my axial frequency response is correct, and remains correct throughout the listening area, I know my phase response must be correct. So, it is an implicit part of my design approach.

SB: *What is the current state of audio measurement?*

KK: We designers seem to swing back and forth between periods when we trust specs and when we don't. I think we are in a transition period. On one hand, you see much vocal opposition to a quantitative approach to equipment evaluation. This is a reaction to a commercialized overemphasis on unimportant specs by the mass market brands, and an unwillingness on the part of the audio establishment to question and improve their test methods. On the other hand, the consumer wants guidance about what to buy. Most consumers have never learned how to make listening evaluations and strenuously avoid doing so. Just try to listen to a hi-fi in a chain store or department store, where most people buy. Forget it. Also, real product innovation cannot proceed without appropriate measurement technology. I think we will see the growth of better techniques in the near future. I hope they relate more to product quality. Better techniques are the only way to meet everyone's needs.

SB: *In your opinion, how does the ear really perceive sound?*

KK: That's a pretty misty subject, but some rules of thumb are relevant to audio reproduction. I tend to divide sound arriving at the ear into three general time groups: less than 1mS, 1–20mS, and greater than 20mS. The initial information dominates the perception of source location and size. The second time group establishes tonal balance and gives cues about the location of walls and other reflecting objects. The last period indicates room size and reverb time.

These numbers are very rough and depend on frequency. It's a tough issue for designers because very little recent psychoacoustics research has been directed specifically towards loudspeakers.

SB: *What do you say to people who tell you they have very good ears and can detect differences between loudspeakers by listening to them at different times, shows, and so on?*

KK: Well, I say that myself, so I can't be too critical of that attitude. But, honestly speaking, you must be realistic. Auditory memory is notoriously short term, and that is easily proved. The physiology of the ear is baroque at best. Without getting into theology, the engineering design of the human ear is not what I would call state-of-the-art. For example, have a drink at your audio club meeting. Your blood alcohol level goes up and affects the reflex time of the muscles in the inner ear, which, in turn, alters the frequency response of your ear. It's not like an analyzer that recalibrates itself every five minutes. You listen to some speakers and announce, "Great," or "Terrible," and you have no real reference. You may be right, and you may be wrong. In a scientific situation, where you put it on the line, you find your ears change more than you think they do—from day to day and minute to minute.

Another pet peeve is the idea of using arbitrary recordings as some kind of reference for judging speakers. I think it is unfair to judge a product based on four minutes of listening, using one or two recordings you and your friends think sound good on your home system. To subtly influence purchase decisions, ask any honest audio salesperson how they choose demo material.

I am not saying ears aren't wonderfully sensitive, they just are not perfectly consistent. Audiophiles sometimes condemn very thoughtful and careful experimental evidence rather than admit ears can deceive. On the other hand, an untrained listener can quickly hear things a $40,000 FFT analyzer can't find. I don't think any approach is infallible. I personally trust my ears as the final arbiter of my designs.

SB: *What are the differences between what the ear perceives and what a microphone measures?*

KK: One of the differences is the head constantly moves in space, in an intelligent way. For example, with the aid of involuntary head movement, you detect whether a source is in front of you or behind you. Microphones can't do that, so information is lost. They can't do spatial averaging like the ear does, and they can't discriminate an interference pattern from a resonance. These processes tell you what is and isn't a relection; whether someone is walking behind you. A microphone can't do that.

In some ways, the microphone is *too* sensitive. If you measure a speaker in a room, you can see 20dB peaks your ear seems to ignore.

SB: *At AR, you dealt exclusively with acoustic suspension speakers. Are you still a believer in that type of design?*

KK: Yes, I am. I feel obligated to keep up with the current work in low frequency theory, and I think recent work in reflex and horn design has been interesting. For consumer applications, however, where the total acoustic power output is not extreme, I remain a believer in acoustic suspension.

SB: *For what reasons?*

KK: One is that it works. That's a fundamental reason. Nobody has proven to me any shortcomings with the approach except efficiency. The notions of fast and slow woofers are extremely unconvincing. Acoustic suspension is a simple and elegant system, a logical system. The parameters are well-controlled, and designs behave very much according to theory. Also, vented systems run into problems below resonance. In commercial sound situations, though, it a different story.

SB: *Where do you see loudspeaker material technology going?*

KK: I can't answer that without discussing where system design is going. Looking at drivers from a materials point of view is a mistake unless we identify the end application. You can easily begin chasing your own tail unless you address the limitations of present drivers.

In the long range, today's drivers are terribly inadequate for the task at hand. The fundamental properties of dynamic drivers, or any kind of mass/compliance energized surfaces, are not very satisfactory.

SB: *You don't think the moving coil loudspeaker has much of a future?*

KK: It depends on what people can come up with. Right now, it remains competitive with the alternative technologies. Ribbons, horns, electrostatics, and so on all suffer, in one way or another, from wacky radiation patterns and other problems I don't like.

SB: *What do you see replacing them?*

KK: I don't know. It's all fantasy at this point. It seems to me there must be a better way to move air and correctly control directivity. Someday, I suppose people will make massless drivers and control their polar patterns in the same way radio antenna arrays are designed—some kind of electrode array.

SB: *In the near future, where do you see the loudspeaker business going?*

KK: Well, the two non-economic factors that have impacted the audio business the most over the last few years have been video and digital. Digital makes very specific and quantifiable demands on a loudspeaker—dynamic range and power handling at the frequency extremes, requirements consumer product makers have never fully dealt with before. It is true recording studio monitors had to deal with high levels above 10kHz, but manufacturers took for granted home systems would never encounter that problem. That's not true anymore. Many inexpensive power amps can now put out several hundred watts peak, and more and more recorded sources demand it.

Continued on page 28

Continued from page 26

To me, the idea of what video will mean to loudspeaker design is much more interesting. Up to now, when people have talked of video speakers, all they meant is the magnets are shielded. That's not so interesting. But, suddenly, when you have loudspeakers interacting with video images, localization and ambience takes on a whole new aspect. You now have a logical place where both the images and the ambience should be. I'm actively involved in getting images correctly up front and center, while wrapping the ambience around the listener. (See *Fig. 6*.)

SB: *How do you do that?*

KK: You do that by understanding how the ear determines what is ambience and what is a direct, focused source. I deal with the two differently in the speaker, and I'm working on electronic circuits that recognize the differences.

SB: *You are saying you have to go to different loudspeakers and also do processing in the signal to determine the ambient from the direct?*

KK: That's one way. It depends on the nature of the source material. Ideally, I could make a recording that didn't require any processing. To work with existing software, you must do electronic processing.

When two-channel technology was first introduced, a fair amount of debate went on as to whether you should record left and right mikes, or front and ambient mikes. I'm perpetually amazed that stereo works as well as it does. You can get a decent stereo image from a session that used 50 mikes and five weeks of overdubs. You can get a stereo image from a dummy head, or a spaced pair or a coincident pair or whatever. Because stereo is so tolerant, very little effort has been made to standardize recording techniques. As a result, things are harder for speaker designers. It is easy to get consistently good reproduction, and almost impossible to get consistently excellent reproduction.

SB: *Where do you see US and foreign speaker manufacturers going?*

KK: Everybody is scrambling to make everything. I think it spells trouble when every other electronics manufacturer has "secret" plans to introduce a speaker, and every other speaker manufacturer plans electronics. Audio companies are trying video, pro companies are trying consumer; everyone seems desperate to increase sales. I don't think the market can possibly support this kind of expansion.

More and more audio is being sold as rack systems. You can cringe and shudder, but it is a commercial reality. The days of the giant loudspeaker companies are over. Most people want turn-key home entertainment. I don't and you don't, but most people, who might have been hobbiests 10 years ago, do. I see it very likely that in ten years, 75% of the audio sold will be integrated in a video system.

People don't shop for audio the same way anymore. They don't go out and listen to many speakers, or try to match their cartridge to their turntable. It is polarizing the business into small companies and big companies: the mass market versus the purist approach. The companies in- between are mostly struggling.

SB: *How does this trend affect the US speaker manufacturers? We see them using drivers made in the Far East.*

KK: Because driver manufacturing is both labor intensive and critical tolerance work means you either use a large labor force, or you spend capital to automate production. You can make a better product with automation, because the tolerances are so close. But both approaches seem prohibitively expensive to beleaguered American manufacturers.

The Japanese, too, are discovering the same problem. Speaker production is moving to Taiwan and Korea, sometimes as a Japanese joint venture, and sometimes not. I suspect some production will move back to the States as economic forces shift. Europe should benefit as well. European companies have been struggling, pretty much unsuccessfully, to maintain any kind of presence in the consumer market. You look at the cost of comparable quality goods from Europe and Asia and it is hard to imagine what is keeping the Europeans in business. You can buy a decent three-way 12″ system in Taiwan for the price of a pair of some of the premium European tweeters.

SB: *You left AR in the Fall of '85. What are you doing now?*

FIGURE 7: Kantor's stock of drivers used in designs for loudspeaker clients.

KK: Mostly freelance product design in the consumer electronics field; some in audio, some in video. Also, I'm doing technical marketing work, helping companies explain and sell their ideas and philosophies. Loudspeakers remain my love and are a substantial portion of my business. I have a number of speaker clients on a contract basis. In addition, I like to write and have done several magazine articles this year.

SB: *What differences do you see in the Far East's drivers versus those from Europe?*

KK: I suppose it is like cars. When you can't any longer compete on price, you do things with your product to make them distinctive. These may improve performance, or they may just improve your image. I think the Europeans have concentrated on high-end specialty drivers, with plenty of craftsmanship and detail, and charged a premium for them. They have a niche market. As with the expensive European cars, some perform better and some don't. The Asian drivers are getting better and better. The Japanese might eventually decide to compete at the top end, but it is a small market.

I dream of producing my own products. I know the realities of doing it on any significant scale, and know it takes loads of money and time. But I have the ideas, and I am heading in that direction.

SB: *What sorts of jobs do you do for your clients?*

KK: I am doing straight-up loudspeaker design, from a driver level to a system level. At the other extreme, I am doing long-term strategic planning for companies; how best to estimate and meet the future needs of the industry. My work is pretty broad-based. (See *Fig. 7*.)

SB: *Any more thoughts?*

KK: Just that I love audio because it is such a weird and wonderful subject. You can put 3% distortion in a power amp and barely hear it. Or you can put a match stick 6″ in front of a speaker and ruin the imaging. It remains very magical. Whatever happens in the mass market, I never worry that audiophiles will disappear, or tire in the quest for better sound.

SB: *Thank you for your cooperation and candor.*

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TRACTRIX HORN DESIGN PROGRAM

BY BRIAN D. SMITH

If you are planning to design a tractrix horn, one of the biggest problems you'll face is the equation for the tractrix curve, where the "x" coordinate is a complex function of the "y" coordinate. This relationship makes it impossible for you to pick a desired point along the horn and to find the dimensions of your horn at that point. You can, however, easily solve this problem, by employing a computer to do the drudgery of picking points and plotting results.

The program I am about to describe is only 64 lines long, but in just a few minutes it will accurately perform many thousands of calculations that might take days to perform by hand. If the results are not satisfactory, you may re-run the program with different data until you are satisfied. With the computer, this process may take only minutes instead of days or weeks.

SOLVING THE PROBLEM. To get the program to solve the tractrix problem, first use Bruce Edgar's equations in his *SB* 2/83 article to compute the the horn's mouth and throat radii. Use the letter "a" to refer to the mouth radius in the tractrix equation. Subtract a small number "r" from "a." From the tractrix equation, compute the length, "x", of the horn at this radius.

You must subdivide the horn into a pre-determined unit for giving output. In the version for this article, the unit is 1″. Therefore, the program will find the horn dimension at 1″ intervals until it reaches the throat. It then compares the computed "x" to the first 1″ interval. If "x" is less than the interval, the program will loop back and subtract a little more from "r." This process will continue until "x" is just slightly more than the first interval. The program will then print out the value of the first interval, along with

```
10 PRINT "THIS PROGRAM CALCULATES DIMENSIONS FOR A TRACTRIC HORN"
20 LPRINT "THIS PROGRAM CALCULATES DIMENSIONS FOR A TRACTRIC HORN"
30 PRINT "All lengths are in inches. Areas are in square inches."
40 LPRINT "All lengths are in inches. Areas are in square inches."
50 PRINT " "
60 INPUT "Key in driver resonance frequency (Fs) in Hertz",FS
70 INPUT "key in desired tractrix cutoff frequency",F
80 INPUT "Key in driver Q (Qes) ",Q
90 INPUT "Key in driver compliance (Vas) in cubic inches",VAS
100 TOTDIST!=0
110 XTWO=0
120 AREA1!=(6.2831853#*FS*Q*VAS)/13500
130 THROAT!=SQR(AREA1/3.1415927#)
140 A!=13500/(6.2831853#*F)
150 B!=SQR(3.1415927#*(A^2))
160 C!=(VAS/((F/(Q*FS))-1))
170 LPRINT "Driver Vas equals ",VAS;" cubic inches"
180 LPRINT " "
190 LPRINT "Driver Qes equals ",Q
200 LPRINT " "
210 LPRINT "Driver resonance frequency equals ",FS;" Hertz"
220 LPRINT " "
230 LPRINT "Horn cutoff frequency equals ",F;" Hertz"
240 LPRINT " "
250 LPRINT "Area of enclosure behind driver should be ";C!;" cubic inches"
260 LPRINT " "
270 LPRINT "Mouth radius is",A!;" Inches"
280 LPRINT " "
290 LPRINT "NON CORNER MOUTH AREA IS ",(A!^2)*3.1416;" Square Inches"
300 LPRINT " "
310 LPRINT "Corner mouth area is ",((A!^2)*3.1416)/8;" Square Inches"
320 LPRINT " "
330 LPRINT "Throat radius is ",THROAT!;" Inches"
340 LPRINT " "
350 LPRINT "Throat area is ",AREA1!;" Square Inches"
360 LPRINT " "
370 LPRINT "Mouth square dimension IS ",B!;" Inches per side"
380 LPRINT " "
390 LPRINT " "
400 LPRINT "Length from","Curve Length","Throat radius";" Sq. dim.","Horn area"
410 LPRINT "mouth"," at X"," at X"," at X"," at X"
420 LPRINT " "
430 R!=A!
440 YONE=SQR((R!^2)*3.14159)*.5
450 LENGTH=LENGTH+1
460 R!=R!-.005
470 YTWO=SQR((R!^2)*3.14159)*.5
480 COUNTER=COUNTER+1
490 PRODUCT=SQR((A!^2)-(R!^2))
500 XONE=XTWO
510 X!=A!*(LOG((A!+PRODUCT)/R!))-PRODUCT
520 XTWO=X!
530 DIST=SQR(((XONE-XTWO)^2)+((YONE-YTWO)^2))
540 TOTDIST!=TOTDIST!+DIST
550 IF R!<THROAT THEN GOTO 590
560 IF X!>LENGTH THEN GOTO 590
570 YONE=YTWO
580 GOTO 460
590 AREA=(R!^2)*3.1416
600 S!=SQR(AREA)
610 LPRINT X!,TOTDIST!,R!,S!,AREA
620 IF R!<THROAT! THEN GOTO 640
630 GOTO 450
640 STOP
```

FIGURE 1. The complete Basic code for the author's Tractrix horn design program.

the dimensions ("x") of the horn at that point.

Finally it loops back and adds one inch to the previous interval, and proceeds to loop until the calculated "x" value is just slightly more than the next interval, or 2″. The program will stop when the horn radius is just a little more than the calculated radius.

The whole process will take a few seconds or several minutes, depending upon the length of your horn. Be prepared to wait a short while, especially at first, since the program will perform more calculations for the first intervals than for the last.

HOW IT WORKS. Let's look at an example to see how all this happens. In this example, the driver's resonant frequency is 49.5Hz. The driver's Q_{es} is .31. The driver's compliance or V_{as} is 864″^{3}, and the desired low frequency cutoff is 70Hz. Start the program by typing "run" and pressing the "enter" key.

The program will ask for the driver resonant frequency. Type in 49.5 and press the "enter" key. You will then be asked for the tractrix L.F. cutoff. Type in 70 and press "enter." When the program asks for Q_{es}, type .31 and press "enter." Finally, the program will ask for driver compliance (V_{as}) in cubic inches. Type 864 and press "enter." The printer will immediately produce all general horn data, such as mouth area, throat area, and so on. After a minute or so, it will print the first line. The next line will print after another minute. Time intervals between printings will decrease until the program finishes. I have included a sample printout (*Fig. 2*) at the end of the article.

This program offers one special twist: it calculates the curve length at each point "x." I included this piece of data because of a past experience. Long before I wrote this program, I decided to build a horn. I calculated everything and went to work, constructing it out of four separate pieces of thin plywood. I drew a line down the length of each piece of wood and perpendicular lines to the length at 1″ intervals. I then plotted all the points on either side of the length, connected them, and cut out each piece. When I joined all the pieces together, the flare at the end of my horn caused the overall length to be shorter than I originally planned. When I wrote this program, I knew I had to solve this problem so I determined the length (L) of the curve at any point "x" along the horn as given to be:

$$L = a\ (\log(r/a))$$

where "a" is the mouth radius and "r" is the radius of the horn at "x." You'll find the proof for this equation at the end of the article. This seems fine except all the radius dimensions have been converted to square dimensions. So, to give the square curve length for a square horn, I used the Pythagorean theorem. I calculated the distance between each two adjacent points on the curve and added it to a running sum of the curve length.

The curve length is used to draw perpendicular lines at the length intervals printed under the "curve length at 'x'" column. This ensures the overall length will be "x" when you put the horn together.

CORNER HORNS. If you are building a corner horn, your starting point will not be zero, but rather at that point "x" where the horn area equals the corner horn mouth area as calculated by the program. In the example, the program calculated the corner mouth area to be 370″^{2}. This area is somewhere between 23 and 24″ from

```
THIS PROGRAM CALCULATES DIMENSIONS FOR A TRACTRIC HORN
All lengths are in inches. Areas are in square inches.
Driver Vas equals                 864   cubic inches

Driver Qes equals                 .31

Driver resonance frequency equals              49.5  Hertz

Horn cutoff frequency equals                   70  Hertz

Area of enclosure behind driver should be  242.5776  cubic inches

Mouth radius is                   30.69417  Inches

NON CORNER MOUTH AREA IS          2959.802  Square Inches

Corner mouth area is              369.9752  Square Inches

Throat radius is                  1.401485  Inches

Throat area is                    6.170591  Square Inches

Mouth square dimension IS         54.404  Inches per side
```

Length from mouth	Curve Length at X	Throat radius at X	Sq. dim. at X	Horn area at X
1.001476	2.991469	27.5385	48.81077	2382.491
2.002092	4.846433	25.78312	45.69944	2088.439
3.001206	6.454411	24.36782	43.19089	1865.452
4.003916	7.936473	23.14256	41.01916	1682.571
5.001423	9.325939	22.05732	39.09564	1528.469
6.004799	10.66258	21.06711	37.34053	1394.315
7.003557	11.94711	20.16192	35.73611	1277.07
8.003975	13.19757	19.32174	34.24693	1172.852
9.006374	14.42107	18.53657	32.85526	1079.468
10.0005	15.61038	17.80642	31.56109	996.1021
11.00582	16.79249	17.11127	30.32897	919.8462
12.00124	17.94558	16.46113	29.17663	851.2754
13.0035	19.09145	15.84103	28.07752	788.3472
14.0081	20.22673	15.25101	27.03175	730.7153
15.00138	21.3378	14.696	26.04801	678.4989
16.00607	22.45133	14.16099	25.09973	629.9962
17.00976	23.5546	13.65098	24.19575	585.4345
18.00861	24.64443	13.16597	23.33609	544.5732
19.00955	25.72923	12.70096	22.51188	506.7848
20.00975	26.80665	12.25594	21.72312	471.894
21.00619	27.8741	11.83094	20.96981	439.7329
22.00813	28.94203	11.42093	20.24309	409.7825
23.00073	29.99515	11.03092	19.55181	382.2734
24.00743	31.05869	10.65091	18.87826	356.3889
25.01361	32.11753	10.2859	18.2313	332.3804
26.00233	33.15435	9.940891	17.61979	310.457
27.01497	34.21267	9.600884	17.01714	289.5831
28.00504	35.24434	9.280876	16.44994	270.6007
29.00068	36.2789	8.970869	15.90047	252.825

FIGURE 2. A sample run of the author's Tractrix horn program using a resolution of 0.005 and an interval of 1″.

the mouth of a free-standing horn. The horn length at the throat is printed as 85.35″. For a corner horn, you would use the point "x" at 24″ making the horn 85.35″ − 24″ = 61.35″. The point at "x" 24″ then becomes the new mouth, or "x" = 0 point. At each point, you must subtract the curve length at "x" = 24″ to find the new curve length at each point. If you are building a corner horn, however, you probably won't care about curve length.

I offer some final pointers should you wish to modify this program for a special need. The 1″ intervals are controlled in statement 450. If you wish to print data at ½″ intervals, replace = 1 with = .5. For 2″ intervals, replace = 1 with = 2. Data accuracy is controlled by statement 460. In this version, I used .005. You will get more accurate measurements by using .0005, however, the program slows geometrically as this number decreases. Conversely, the program will run much faster if you replace .005 with .05. The measurements using .05 will be less accurate than at .005. Try a variety of measurements and choose one that suits you best. If you wish to specify your own mouth radius and throat, remove statements 60, 70, 90, and 120 through 390. Replace them with the following two statements:

```
60 A = XXX
70 THROAT = ZZZ
```

Where XXX is your desired mouth radius and ZZZ is your desired throat radius.

If you have access to a personal computer with Basic programming language capability, this program will allow you to design a tractrix horn with minimal effort. My modification suggestions should allow you to design any type of horn using any type of driver. Have fun, and if you have any problems or comments, write to me in care of *SB*.

30.00068	37.3153	8.670862	15.36872	236.1976
31.00376	38.35239	8.380856	14.8547	220.662
32.00849	39.38886	8.10085	14.3584	206.1636
33.01333	40.42331	7.830843	13.87982	192.6495
34.01662	41.45418	7.570837	13.41898	180.0689
35.01654	42.47977	7.320832	12.97585	168.3727
36.01113	43.49822	7.080826	12.55045	157.5138
37.02011	44.5298	6.845821	12.13392	147.2319
38.02092	45.55155	6.620816	11.7351	137.7127
39.01116	46.56119	6.405811	11.35402	128.9137
40.0125	47.58088	6.195806	10.98179	120.5998
41.02472	48.61045	5.990801	10.61843	112.7511
42.02157	49.6233	5.795797	10.27279	105.5303
43.02688	50.64371	5.605792	9.936019	98.72448
44.01235	51.64306	5.425788	9.61697	92.48611
45.00334	52.64714	5.250784	9.306782	86.61621
46.02896	53.68542	5.07578	8.996596	80.93875
47.02985	54.69789	4.910776	8.704134	75.76195
48.00201	55.68064	4.755773	8.429397	71.05473
49.0072	56.69607	4.600769	8.15466	66.49848
50.01351	57.712	4.450766	7.888786	62.23294
51.01977	58.72728	4.305763	7.631774	58.24397
52.02468	59.74063	4.165759	7.383624	54.51791
53.02683	60.7507	4.030756	7.144337	51.04156
54.02468	61.75595	3.900753	6.913913	47.80219
55.01655	62.75473	3.77575	6.692351	44.78756
56.00064	63.74526	3.655748	6.479651	41.98588
57.01806	64.7689	3.535745	6.266952	39.27468
58.02648	65.78311	3.420742	6.063115	36.76136
59.0237	66.78573	3.31074	5.86814	34.43506
60.00736	67.77441	3.205737	5.682028	32.28544
61.02416	68.79606	3.100735	5.495916	30.20509
62.02541	69.80181	3.000733	5.318666	28.28821
63.00831	70.78889	2.90573	5.150279	26.52537
64.02419	71.80881	2.810728	4.981892	24.81924
65.01907	72.80741	2.720726	4.822367	23.25522
66.04767	73.83963	2.630724	4.662843	21.7421
67.05223	74.84751	2.545722	4.51218	20.35977
68.02904	75.82738	2.46572	4.370381	19.10022
69.03827	76.83956	2.385718	4.228581	17.88089
70.01586	77.81987	2.310717	4.095644	16.77429
71.02586	78.8325	2.235715	3.962706	15.70304
72.07056	79.87973	2.160713	3.829769	14.66713
73.00586	80.81723	2.095712	3.714557	13.79793
74.04626	81.85986	2.02571	3.590482	12.89156
75.04521	82.86088	1.96071	3.475271	12.07751
76.07798	83.89561	1.89571	3.360062	11.29001
77.06334	84.88278	1.83571	3.253714	10.58666
78.08156	85.90272	1.77571	3.147367	9.905919
79.04574	86.86849	1.72071	3.049882	9.301781
80.04134	87.8656	1.66571	2.952397	8.716648
81.07046	88.89622	1.61071	2.854912	8.150523
82.03709	89.86415	1.56071	2.766289	7.652357
83.03528	90.86363	1.51071	2.677667	7.169898
84.06714	91.89673	1.46071	2.589044	6.703149
85.02655	92.85724	1.41571	2.509284	6.296504
85.35316	93.1844	1.40071	2.482697	6.163783

ABOUT THE AUTHOR

Brian D. Smith is a certified public accountant who, after five years with a major oil firm, decided to return to school to study electrical engineering. An electronic hobbyist for eleven years, his special interest is loudspeaker theory supported by construction projects utilizing many driver types. He is enrolled as a full time EE candidate at Ohio State.

Readers interested in a demonstration copy of Mr. Smith's Tractrix Horn Program should use Fast Reply #IK1088 for full details. The program runs on the IBM PC/XT machines (and compatibles) and will be distributed on 5¼″ DS/DD, 40-track floppy disks.

CROSSOVER COMPONENT CAPABILITIES and REQUIREMENTS

CROSSOVER VOLTAGE & CURRENT FACTORS

BY G. R. KOONCE
Contributing Editor

In this issue, I will examine how the common parallel crossover (CO) configuartions (first-order through third-order) for two-way and three-way speaker systems affect their component's voltage and current requirements. CO sections are filter sections which exhibit circuit Qs based on CO type, circuit topology and the selected CO frequences. As a result, the voltage across and current through components will vary with frequency, and at certain frequencies may exceed the system input voltage and load current. I will show you how to determine how much excessive voltage or current each component must tolerate.

APPROACH. I programmed my computer to solve a ladder network that could represent any CO section I wanted to investigate. This program solved for the voltage and current through each network component at 41 frequencies over a two decade spread. Processing all the COs of interest produced several hundred pages of voltage/current versus frequency data. The remainder of this article summarizes my findings and describes how you can apply them.

I analyzed all CO sections with an 8Ω load. For two-way systems, the CO frequency (f_C) was always 1kHz. This does not limit the generality because the results scale directly and are independent of the computational values. Three-way systems had the lower CO frequency (f_L) at 1kHz and three different upper CO frequencies (f_H) to show frequency spread effects on the midrange section.

FACTOR DEFINITIONS. I have presented the results in terms of "factors," which you can multiply by known quantities to get results specific to your system. The voltage factor for a component is defined as:

$$\mathrm{VF} = \frac{\text{max. RMS voltage occurring at some frequency}}{\text{RMS input voltage (1V in computations)}}$$

So if you multiply the VF by the maximum RMS input voltage you expect

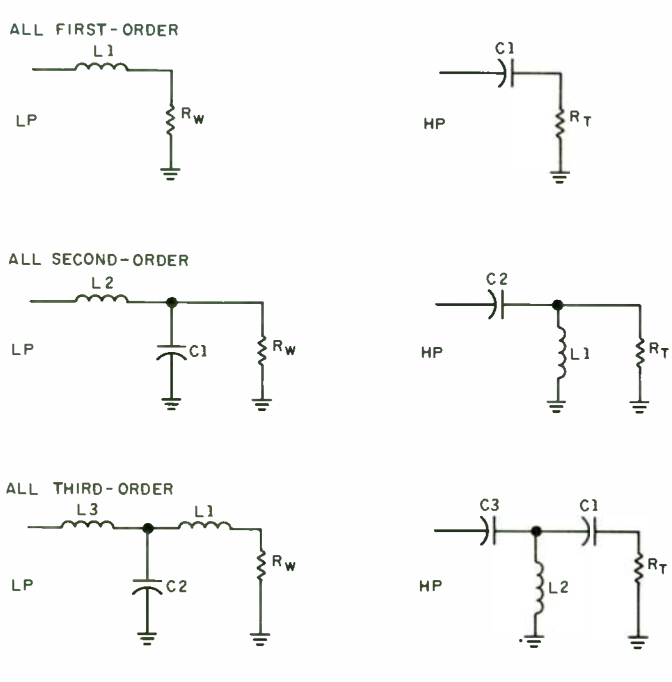


FIGURE 1: Schematics for two-way crossovers.

TABLE 1: ALL FACTORS for TWO-WAY CROSSOVERS

First-Order

	L1 VF at f/f_c		L1 IF at f/f_c		L1 CF	C1 VF at f/f_c		C1 IF at f/f_c	
	1.00	HF	1.00	LF	1.13	1.00	LF	1.00	HF

Second-Order: Low-Pass

	L2 VF at f/f_c		L2 IF at f/f_c		L2 CF	C1 VF at f/f_c		C1 IF at f/f_c	
BW	1.27	1.25	1.03	0.5	1.38	1.00	LF	0.5	1.00
AP	1.16	1.40	1.00	LF	1.59	1.00	LF	0.25	1.00
B1lck EC	1.21	1.40	1.00	LF	1.46	1.00	LF	0.35	1.00

Second-Order: High-Pass

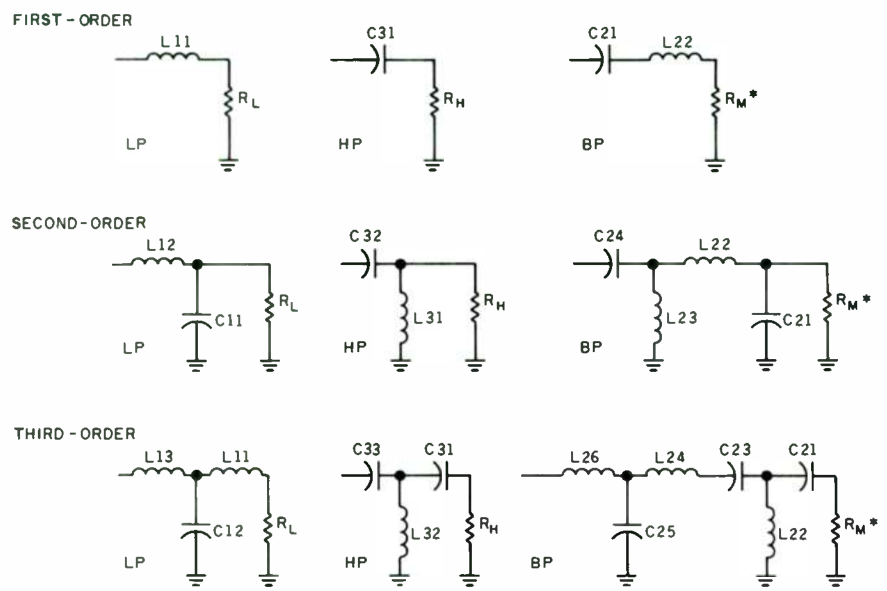
	L1 VF at f/f_c		L1 IF at f/f_c		L1 CF	C2 VF at f/f_c		C2 IF at f/f_c	
BW	1.00	HF	0.5	1.00	0.67	1.27	0.8	1.02	2.00
AP	1.00	HF	0.25	1.00	0.40	1.16	0.7	1.00	HF
B1lck EC	1.00	HF	0.35	1.00	0.52	1.20	0.75	1.00	HF

Third-Order: Butterworth--All Factors

	VF at f/f_c		IF at f/f_c		CF		VF at f/f_c		IF at f/f_c	
L1	0.36	0.9	1.00	LF	0.80	C1	0.36	1.10	1.00	HF
L2	1.02	2.0	1.06	1.1	1.04	C2	1.02	0.5	1.06	0.9
L3	1.47	1.1	1.09	0.7	1.50	C3	1.46	0.9	1.09	1.4

Table 2: ALL FACTORS for FIRST-ORDER THREE-WAY CROSSOVERS

f_H/f_L		Butterworth VF - f/f_c	Butterworth IF - f/f_c	Butterworth CF	All-Pass VF - f/f_c	All-Pass IF - f/f_c	All-Pass CF
=2	L11	1.00--HF	1.00--LF	1.13	1.00--HF	1.00--LF	1.13
	C31	1.00--LF	1.00--HF		1.00--LF	1.00--HF	
	L22	1.51-1.17	1.00-1.00	1.11	1.00--HF	1.00-1.00	0.65
	C21	1.50-0.81	1.00-1.00		1.00--LF	1.00-1.00	
=4	L11	1.00--HF	1.00--LF	1.13	1.00--HF	1.00--LF	1.13
	C31	1.00--LF	1.00--HF		1.00--LF	1.00--HF	
	L22	1.00--HF	1.00-1.00	0.65	1.00--HF	1.00-1.00	0.50
	C21	1.00--LF	1.00-1.00		1.00--LF	1.00-1.00	
=6	L11	1.00--HF	1.00--LF	1.13	1.00--HF	1.00--LF	1.13
	C31	1.00--LF	1.00--HF		1.00--LF	1.00--HF	
	L22	1.00--HF	1.00-1.00	0.50	1.00- HF	1.00-1.00	0.43
	C21	1.00--LF	1.00-1.00		1.00- LF	1.00-1.00	



R_M is total load resistance. If R_A is used then, R_M = R_A + midrange resistance.

FIGURE 2: Schematics for three-way crossovers (cascaded MR topology).

to feed the system, you will get the maximum RMS voltage that will occur across that component.

Current factor is defined as:

$$\text{IF} = \frac{\text{maximum RMS current occurring at some frequency}}{\text{expected load RMS current (1V/8}\Omega\text{ in comp.)}}$$

If you multiply IF by the maximum RMS current you expect to deliver to the driver on that CO section, you will find the maximum RMS current through that component. Maximum expected RMS load current is equal to your maximum RMS input voltage, divided by the load resistance. For all this work, I assumed impedance correctors were used on all drivers, thereby approximating resistive loads. (See *SB* 1/83 p. 11, and 1/85 p. 19.)

These voltage and current multiplying factors do not apply throughout the entire audio band, but rather at usually one or sometimes two frequency ranges within the band. Since CO circuit Qs are generally low, these frequency bands tend to be wide, so don't ignore them.

To help you know where these frequency bands are located, I have indicated the band centers on the data charts. For two-way COs, it is given as the ratio f/f_C for the lowest frequency band to occur. Multiply your CO frequency by this f/f_C factor to find the location of the voltage or current maximum (they don't always occur at the same frequency). For three-way COs, the frequency is shown as f/f_M, where f_M is the geometric mean frequency between your lower and upper CO frequencies (i.e. $f_M = (f_L \times f_H)^{1/2}$. See *SB* 2/85 p. 26 for further clarification on f_M.

How severe is the CO design on the required coils? Saturation of a cored inductor is proportional to the coil current and the number of turns on the coil for a given core. Since inductance is proportional to number of turns squared, the following core factor indicates how "hard" the coil is on the core.

$$\text{CF} = \text{Current Factor} \times (\text{inductance in mHy})^{1/2}$$

The lower the CF, the less core area is needed to wind the coil so that it will sustain a given power to the load. This factor will help you judge the effects of CO type, circuit topology and CO frequency on coil implementation difficulty. Reducing CO frequency raises inductance, making the coil more difficult. Changing driver resistance at a given power level will have no effect on core requirement because current increase (decrease) is directly offset by turns decrease (increase).

TWO-WAY CO RESULTS. *Figure 1* shows the schematic diagram for all two-way COs covered in this work. Component labeling is in agreement with Bullock (*SB* 1/85 p. 13). *Table 1* shows the results for all two-way COs. The first-order has almost all unity factors and, as expected, low stress on the parts. For second-order COs, the Butterworth (BW) appears to be toughest on its components, while the All-Pass (AP) is the mildest. The Bullock Equal Compromise is, as you might expect, in between. The low-pass (LP) inductor sees about the same current as the load, while the high-pass (HP) inductor sees a maximum of about one-half

Table 3: VOLTAGE FACTORS for SECOND-ORDER THREE-WAY CROSSOVERS

Format: VF-f/f_m

	Butterworth Cascaded Midrange	Butterworth Special Cascaded Midrange	Butterworth Transp. Midrange	Constant-Power Cascaded Midrange	Constant-Power Transp. Midrange	All-Pass Cascaded Midrange	All-Pass Transp. Midrange
	For f_H/f_L = 2						
L12	1.27-0.9	1.27-0.9	1.27-0.9	1.27-0.9	1.27-0.9		
C11	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF	No Design for	
L23	2.31-1.2	2.56-1.3	2.53-1.2	1.59-1.1	1.52-1.6	for these cases.	
C24	2.54-0.7	2.32-0.8	2.56-0.8	1.53-0.6	1.53-0.6		
L22	3.71-1.3	3.00-1.0	1.00-1.0	1.64-1.6	1.00-1.0		
C21	3.00-1.1	3.68-0.8	1.00-1.0	1.40-1.0	1.00-1.0		
L31	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF		
C32	1.27-0.6	1.27-0.6	1.27-0.6	1.27-0.6	1.27-0.6		
	For f_H/f_L = 4						
L12	1.27-0.6	1.27-0.6	1.27-0.6	1.27-0.6	1.27-0.6	1.14-0.7	1.14-0.7
C11	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF
L23	1.42-1.6	1.63-2.0	1.64-2.0	1.32-1.1	1.38-2.5	1.30-1.3	1.26-2.8
C24	1.63-0.5	1.42-0.6	1.63-0.5	1.38-0.4	1.38-0.4	1.26-0.4	1.26-0.4
L22	1.84-2.0	1.44-1.0	1.00-1.0	1.46-2.2	1.00-1.0	1.29-2.8	1.00-1.0
C21	1.44-1.0	1.84-0.5	1.00-1.0	1.24-1.0	1.00-1.0	1.15-1.0	1.00-1.0
L31	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF
C32	1.27-0.4	1.27-0.4	1.27-0.4	1.27-0.4	1.27-0.4	1.14-1.4	1.14-1.4
	For f_H/f_L = 6						
L12	1.27-0.5	1.27-0.5	1.27-0.5	1.27-0.5	1.27-0.5	1.15-0.6	1.15-0.6
C11	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF	1.00--LF
L23	1.25-1.8	1.47-2.4	1.47-2.5	1.22-1.6	1.34-2.9	1.28-1.3	1.24-3.2
C24	1.47-0.4	1.25-0.5	1.47-0.4	1.34-0.4	1.32-0.3	1.24-0.3	1.24-0.3
L22	1.57-2.6	1.24-1.0	1.00-1.0	1.39-2.8	1.00-1.0	1.26-3.1	1.00-1.0
C21	1.24-1.0	1.57-0.4	1.00-1.0	1.16-1.0	1.00-1.0	1.12-1.0	1.00-1.0
L31	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF	1.00--HF
C32	1.27-0.3	1.27-0.3	1.27-0.3	1.27-0.3	1.27-0.3	1.15-1.7	1.15-1.7

load current. The HP capacitor can see about 30% overvoltage compared to the input, and should be so rated. The BW was the only third-order CO I examined. Coil current factors are near unity, but capacitor C3 can see about 50% overvoltage in the region of CO frequency.

THREE-WAY CO RESULTS. *Figure 2* shows schematics for the first-order through third-order COs, using cascaded midrange (MR) topology. Component numbering and schematic layouts are in agreement with Bullock (*SB* 2/85 p. 26). *Figure 3* shows schematics for revised MR topologies. See reference 1 for design of the transposed MR topology, and reference 2 for the special cascaded MR topology. For consistency I have maintained component labeling in agreement with *Fig. 2*.

Table 2 shows the results for BW and AP first-order COs. Capacitor VF only becomes a problem for C21 at low f_H/f_L ratios with the BW CO. The coil CFs are unity throughout.

Table 3 shows VF for a variety of second-order CO types and topologies. At low f_H/f_L ratios, capacitors can see as much as three times the input voltage with some BW CO topologies. *Table 4* shows IF results for the same second-order COs. At low f_H/f_L ratios, some BW topologies are quite hard on coils, with L23 seeing up to nine times the load current. This drops quickly as the CO frequency-spread increases, and is not a problem at all with the transposed MR topology.

Table 4: CURRENT FACTORS for SECOND-ORDER THREE-WAY CROSSOVERS

Format: IF-f/f_m

	Butterworth			Constant-Power		All-Pass	
	Cascaded Midrange	Special Cascaded Midrange	Transp. Midrange	Cascaded Midrange	Transp. Midrange	Cascaded Midrange	Transp. Midrange
	For f_H/f_L = 2						
L12	1.03-0.4	1.03-0.4	1.03-0.4	1.03-0.4	1.03-0.4	No Design for	
C11	0.50-0.7	0.50-0.7	0.50-0.7	0.50-0.7	0.50-0.7	for these cases.	
L23	8.14-0.8	9.25-1.2	1.02-0.9	0.83-0.7	1.00-1.0		
C24	9.25-1.2	8.06-1.3	1.02-0.9	1.96-1.0	1.00-1.0		
L22	4.53-1.2	3.49-0.8	1.14-0.8	1.47-1.0	0.40-0.7		
C21	3.44-1.2	4.49-0.8	1.16-1.2	0.57-1.6	0.40-1.6		
L31	0.50-1.4	0.50-1.4	0.50-1.4	0.50-1.4	0.50-1.4		
C32	1.02-2.8	1.02-2.8	1.02-2.8	1.02-2.8	1.02-2.8		
	For f_H/f_L = 4						
L12	1.03-0.3	1.03-0.3	1.03-0.3	1.03-0.3	1.03-0.3	1.00--LF	1.00--LF
C11	0.50-0.5	0.50-0.5	0.50-0.5	0.50-0.5	0.50-0.5	0.22-0.5	0.22-0.5
L23	1.42-0.6	2.14-0.7	1.03-1.0	0.65-0.5	1.00-1.0	0.38-0.5	1.00-1.0
C24	2.15-1.4	1.42-1.8	1.03-1.0	1.53-1.0	1.00-1.0	1.32-1.0	1.00-1.0
L22	1.69-1.4	1.01-0.5	0.70-0.6	1.29-1.1	0.42-0.5	1.67-1.0	0.28-0.5
C21	1.01-1.8	1.70-0.7	0.70-1.8	0.52-2.0	0.42-2.0	0.32-2.0	0.28-2.2
L31	0.50-2.0	0.50-2.0	0.50-2.0	0.50-2.0	0.50-2.0	0.22-2.0	0.22-2.0
C32	1.02-4.0	1.02-4.0	1.02-4.0	1.02-4.0	1.02-4.0	1.00--HF	1.00--HF
	For f_H/f_L = 6						
L12	1.03-0.2	1.03-0.2	1.03-0.2	1.03-0.2	1.03-0.2	1.00--LF	1.00--LF
C11	0.50-0.4	0.50-0.4	0.50-0.4	0.50-0.4	0.50-0.4	0.24-0.4	0.24-0.4
L23	0.94-0.5	1.58-0.6	1.03-1.6	0.60-0.4	1.00-1.0	0.35-0.4	1.00-1.0
C24	1.58-1.6	0.94-2.2	1.03-1.6	1.35-1.0	1.00-1.0	1.25-1.0	1.00-1.0
L22	1.38-1.6	0.76-0.4	0.61-0.5	1.20-1.3	0.44-0.4	1.14-1.0	0.28-0.4
C21	0.76-2.2	1.38-0.6	0.61-2.3	0.51-2.5	0.44-2.5	0.31-2.5	0.28-2.5
L31	0.50-2.5	0.50-2.5	0.50-2.5	0.50-2.5	0.50-2.5	0.24-2.5	0.24-2.5
C32	1.02-4.9	1.02-4.9	1.02-4.9	1.02-4.9	1.02-4.9	1.00--HF	1.00--HF

Table 5: CORE FACTORS for SECOND-ORDER THREE-WAY CROSSOVERS

	Butterworth			Constant-Power		All-Pass	
	Cascaded Midrange	Special Cascaded Midrange	Transp. Midrange	Cascaded Midrange	Transp. Midrange	Cascaded Midrange	Transp. Midrange
	For f_H/f_L = 2						
L12	1.38	1.38	1.38	1.38	1.38		
L23	4.46	4.14	1.37	1.15	1.04	No design for	
L22	3.51	7.65	1.08	1.29	0.66	these cases.	
L31	0.47	0.47	0.47	0.47	0.47		
	For f_H/f_L = 4						
L12	1.38	1.38	1.38	1.38	1.38	1.64	1.64
L23	1.37	1.15	0.80	0.88	0.73	0.65	0.71
L22	1.09	1.17	0.81	0.84	0.70	0.77	0.51
L31	0.34	0.34	0.34	0.34	0.34	0.34	0.34
	For f_H/f_L = 6						
L12	1.38	1.38	1.38	1.38	1.38	1.61	1.61
L23	1.03	0.76	0.62	0.81	0.58	0.58	0.60
L22	0.74	0.95	0.75	0.65	0.64	0.65	0.49
L31	0.27	0.27	0.27	0.27	0.27	0.16	0.16

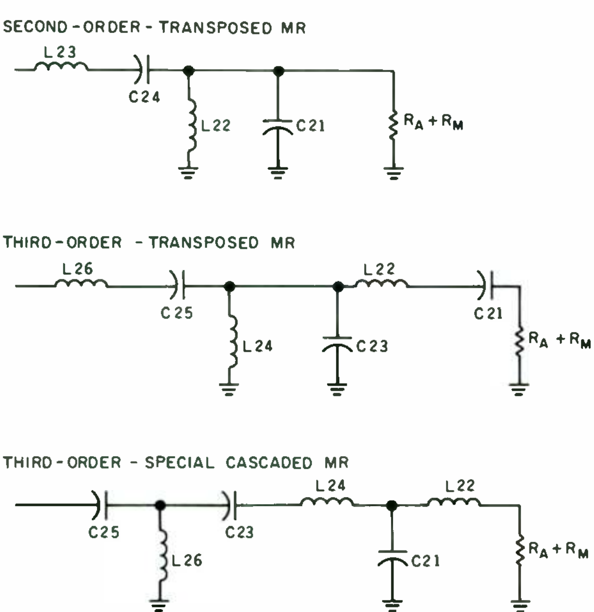


FIGURE 3: Special midrange topologies.

Table 6: VF & IF for THIRD-ORDER THREE-WAY HP & LP SECTIONS

Format: VF-f/f_m IF-f/f_m

	Voltage Factor		Current Factors	
	Butterworth Constant-Power	All-Pass	Butterworth Constant-Power	All-Pass
	For f_H/f_L = 2			
L13	1.47-0.8	1.85-0.8	1.09-0.5	1.62-0.7
C12	1.02-0.4	1.38-0.7	1.06-0.6	1.98-0.8
L11	0.36-0.6	0.77-0.7	1.00--LF	1.18-0.6
C33	1.46-1.3	1.86-1.2	1.09-2.0	1.62-1.4
L32	1.02-2.8	1.38-1.6	1.06-1.6	1.96-1.4
C31	0.36-1.6	0.77-1.4	1.00--HF	1.18-1.6
	For f_H/f_L = 4			
L13	1.47-0.6	1.63-0.6	1.09-0.4	1.26-0.5
C12	1.02-0.3	1.14-0.4	1.06-0.5	1.44-0.5
L11	0.36-0.5	0.54-0.5	1.00--LF	1.04-0.4
C33	1.46-1.8	1.63-1.8	1.09-2.8	1.26-2.2
L32	1.02-4.0	1.14-2.5	1.06-2.2	1.44-2.0
C31	0.36-2.2	0.54-2.0	1.00--HF	1.04-2.8
	For f_H/f_L = 6			
L13	1.47-0.5	1.57-0.5	1.09-0.3	1.19-0.4
C12	1.02-0.2	1.09-0.3	1.06-0.4	1.29-0.4
L11	0.36-0.4	0.47-0.4	1.00--LF	1.01-0.3
C33	1.46-2.2	1.56-2.2	1.09-3.4	1.18-2.7
L32	1.02-4.9	1.09-3.4	1.06-2.7	1.29-2.5
C31	0.36-2.7	0.47-2.5	1.00--HP	1.01-3.9

Note: All-Pass CO is the positive-polarity bandpass type.

Table 7: MR VOLTAGE FACTORS for THIRD-ORDER THREE-WAY CROSSOVERS
Format: VF-f/f_m
Butterworth

	Special Cascaded Midrange	Constant-Power Cascaded Midrange	Constant-Power Transp. Midrange	All-Pass Cascaded Midrange	All-Pass Transp. Midrange
	For f_M/f_L = 2				
C25	2.97-0.8	1.95-0.9	1.97-0.7	2.34-0.9	2.49-0.8
L26	2.51-0.8	1.97-1.4	1.97-1.4	2.48-1.2	2.48-1.2
C23	3.86-1.2	2.57-0.7	1.00-1.0	3.87-0.8	1.00-1.0
L24	6.03-1.2	0.66-1.2	1.00-1.0	1.20-1.2	1.00-1.0
C21	1.78-1.2	0.53-0.8	0.27-0.8	0.81-0.9	0.32-0.9
L22	1.83-1.2	2.02-1.0	0.27-1.2	2.68-1.0	0.32-1.2
	For f_M/f_L = 4				
C25	1.94-0.5	1.41-0.7	1.66-0.5	1.43-1.4	1.81-0.5
L26	1.49-0.6	1.66-2.0	1.66-2.0	1.82-2.0	1.82-2.0
C23	1.00-0.6	1.97-0.5	1.00-1.0	2.23-0.7	1.02-1.4
L24	2.54-1.8	0.48-1.8	1.00-1.0	0.62-2.5	1.02-1.4
C21	1.60-1.4	0.42-0.6	0.30-0.6	0.52-0.8	0.36-0.6
L22	0.76-1.8	1.46-0.9	0.30-1.8	1.53-0.7	0.36-1.8
	For f_M/f_L = 6				
C25	1.75-0.4	1.26-0.6	1.58-0.4	1.29-1.8	1.69-0.4
L26	1.30-0.6	1.58-2.5	1.58-2.5	1.69-2.5	1.69-2.5
C23	0.68-0.5	1.78-0.4	1.00-1.0	1.93-0.4	1.03-0.6
L24	2.05-2.3	0.44-2.2	1.00-1.0	0.52-2.2	1.03-0.6
C21	1.36-1.6	0.39-0.9	0.31-0.5	0.46-0.5	0.36-0.5
L22	0.58-2.0	1.29-0.8	0.31-2.3	1.34-0.6	0.35-2.2

Note: All-Pass CO is the positive-polarity bandpass type

Table 8: MR CURRENT FACTORS for THIRD-ORDER THREE-WAY CROSSOVERS
Format: IF-f/f_m
Butterworth

	Special Cascaded Midrange	Constant-Power Cascaded Midrange	Constant-Power Transp. Midrange	All-Pass Cascaded Midrange	All-Pass Transp. Midrange
	For f_M/f_L = 2				
C25	5.00-0.8	2.85-1.2	1.00-1.0	5.92-1.2	1.00-1.0
L26	5.59-0.8	3.84-1.0	1.00-1.0	6.65-1.0	1.00-1.0
C23	2.11-1.2	2.12-1.0	0.92-1.2	3.01-0.9	1.31-1.2
L24	2.11-1.2	2.12-1.0	0.93-0.8	3.01-0.9	1.29-0.8
C21	2.80-1.2	1.96-1.0	1.00-1.0	2.58-1.0	1.00-1.0
L22	2.15-1.0	1.58-0.8	1.00-1.0	2.63-0.8	1.00-1.0
	For f_M/f_L = 4				
C25	2.31-0.6	1.71-1.8	1.00-1.1	2.27-1.8	1.10-0.6
L26	2.53-0.6	2.06-1.1	1.00-1.1	2.32-0.6	1.10-0.6
C23	1.64-1.6	1.53-0.8	0.97-1.8	1.74-0.6	1.27-1.8
L24	1.64-1.6	1.53-0.8	0.97-0.6	1.74-0.6	1.27-0.6
C21	1.82-1.8	1.43-1.0	1.00-1.0	1.47-1.4	1.01-1.4
L22	1.46-1.0	1.27-0.6	1.00-1.0	1.64-0.6	1.01-1.4
	For f_M/f_L = 6				
C25	1.77-0.5	1.44-2.2	1.01-1.6	1.77-2.2	1.13-2.0
L26	1.88-0.5	1.63-1.6	1.01-1.6	1.79-2.0	1.13-2.0
C23	1.44-1.8	1.35-0.6	1.00-2.3	1.50-0.6	1.24-2.2
L24	1.44-1.8	1.35-0.6	1.00-0.5	1.50-0.6	1.23-0.4
C21	1.51-2.3	1.27-1.0	1.00-1.0	1.28-1.6	1.02-1.6
L22	1.28-1.0	1.18-0.4	1.00-1.0	1.44-0.4	1.02-1.6

Note: All-Pass CO is the positive-polarity bandpass type

The Constant-Power (CP) and AP configurations show much less current multiplication and are thus the preferred types to use. *Table 5* shows CF for the second-order COs. Again, avoid the BW at low f_H/f_L ratios to prevent severe MR coil requirements.

Table 6 shows VF and IF for the LP and HP sections of third-order COs. These results are not too sensitive to f_H/f_L ratio, with the AP CO always the worst case. *Table 7* shows third-order MR VF results. The AP CO covered in this work is the positive-polarity bandpass type. Low f_H/f_L ratios show large capacitor voltage multipliers.

Table 8 shows third-order MR IFs. At low f_H/f_H ratios, IFs enlarged for the cascaded topologies, showing the transposed topology clearly easier on the MR coils. *Table 9* shows CF for all portions of the three-way CO, verifying the transposed topology is clearly superior and the CP CO slightly superior in terms of coil requirements.

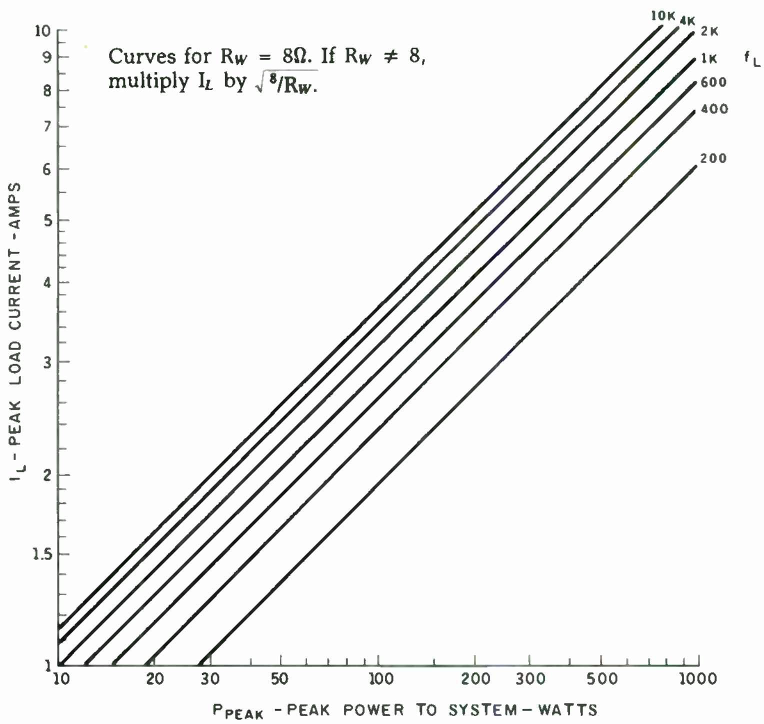


FIGURE 4: Woofer load current versus peak power.

OVERALL FACTORS. It would be nice to arrive at one overall factor for

Table 9: CORE FACTORS for THIRD-ORDER THREE-WAY CROSSOVERS
Butterworth

	Special Cascaded Midrange	Constant-Power Cascaded Midrange	Constant-Power Transp. Midrange	All-Pass Cascaded Midrange	All-Pass Transp. Midrange
	For f_M/f_L = 2				
L13	1.50	1.50	1.50	1.87	1.87
L11	0.80	0.80	0.80	1.09	1.09
L26	4.03	2.39	1.22	3.54	1.37
L24	3.04	1.06	0.98	1.77	1.13
L22	1.72	1.82	0.46	2.67	0.51
L32	0.73	0.73	0.73	1.31	1.31
	For f_M/f_L = 4				
L13	1.50	1.50	1.50	1.59	1.59
L11	0.80	0.80	0.80	0.90	0.90
L26	2.02	1.13	0.78	1.22	0.84
L24	1.31	0.58	0.97	0.70	1.16
L22	1.17	1.35	0.36	1.62	0.38
L32	0.52	0.52	0.52	0.70	0.70
	For f_M/f_L = 6				
L13	1.50	1.50	1.50	1.55	1.55
L11	0.80	0.80	0.80	0.85	0.85
L26	1.60	0.79	0.62	C.84	0.67
L24	0.85	0.42	0.99	0.49	1.14
L22	0.46	1.22	0.30	1.41	0.32
L32	0.42	0.42	0.42	0.51	0.51

Note: All-Pass CO is the positive-polarity bandpass type

Table 10: OVERALL FACTOR and AVERAGE CORE FACTOR--ALL COs

Format OAF-Avg CF

	TWO-WAY	THREE-WAY f_M/f_L=2	THREE-WAY f_M/f_L=4	THREE-WAY f_M/f_L=6
FIRST-ORDER:				
BW	1.00-1.13	1.13-1.12	1.00-0.89	1.00-0.82
AP		1.00-0.89	1.00-0.82	1.00-0.78
SECOND-ORDER:				
BW - cas MR	0.95-1.03	2.72-2.46	1.25-1.05	1.10-0.86
spcl cas MR		2.78-3.41	1.28-1.01	1.12-0.84
tran MR		1.16-1.08	1.02-0.83	0.99-0.76
CP - cas MR		1.10-1.07	1.05-0.86	1.01-0.78
tran MR		0.93-0.89	0.95-0.79	0.95-0.72
AP - cas MR	0.85-1.00		0.98-0.85	0.91-0.75
tran MR			0.86-0.80	0.86-0.72
Bullock EC	0.89-0.99			
THIRD-ORDER:				
BW spc cas	1.00-1.11	2.02-1.97	1.29-1.22	1.15-0.94
CP - cas MR		1.55-1.38	1.17-0.98	1.08-0.86
tran MR		1.01-0.95	0.94-0.82	0.94-0.77
AP - cas MR		2.34-2.04	1.41-1.12	1.25-0.94
tran MR		1.32-1.21	1.14-0.93	1.09-0.84

$$\text{OAF} = \text{Overall Factor} = \frac{\text{Sum of Capacitor VFs + Sum of Coil IFs}}{\text{Total Number of Components}}$$

$$\text{Ave CF} = \text{Average Core Factor} = \frac{\text{Sum of Coil Core Factors}}{\text{Number of Coils}}$$

a CO type and topology. I have done this, believing current is the key factor on coils, and voltage on capacitors.

$$\text{OAF} = \frac{\text{(sum of coil IFs)} + \text{(sum of capacitor VFs)}}{\text{number of components in CO}}$$

Table 10 shows OAF for all CO types, along with average core factor:

$$\text{ACF} = \frac{\text{sum of core factors}}{\text{number of coils}}$$

This table shows there is not much difference between two-way COs of various types and orders. For three-way COs above first-order, things become worse as f_H/f_L gets smaller. A CP or AP CO is preferred over the BW, and the transposed topology is somewhat better than either of the cascaded topologies. The three-way first-order CO is not too sensitive to frequency spread or CO type. These results agree with those previously stated.

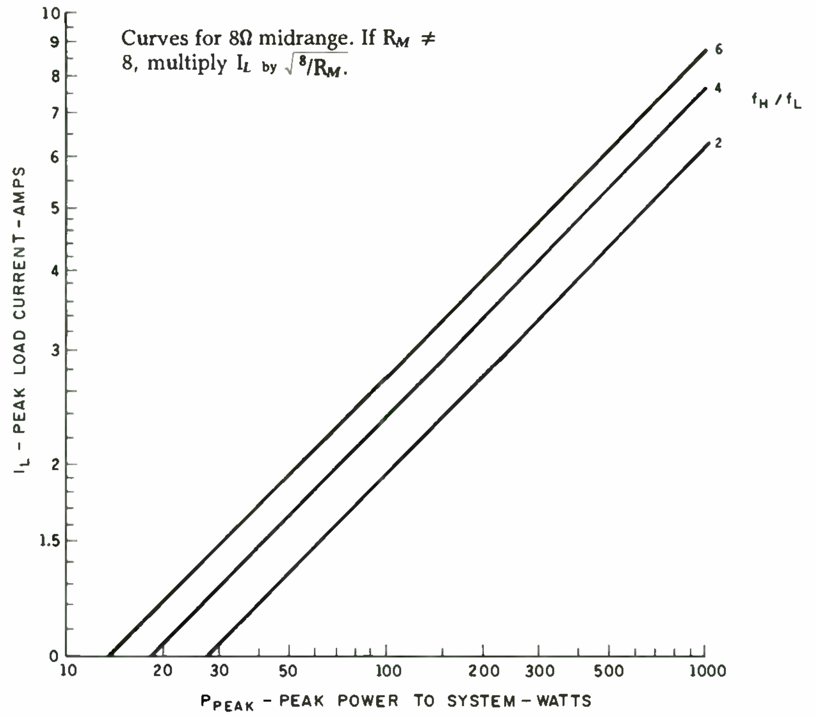


FIGURE 5: Midrange load current versus peak power.

DATA APPLICATION. If simple equations could use the above data to produce accurate CO component requirements for your application, life would be easier. Unfortunately too many variables are involved. For example, the factors I presented were computed for steady-state conditions, so I question their use for peak power (transient) conditions. When combined with trying to obtain how peak music power splits for your music and your CO shape, you can see why a rigorous mathematical solution is beyond the scope of this work. I will present my approach and you can make any changes you believe to be appropriate.

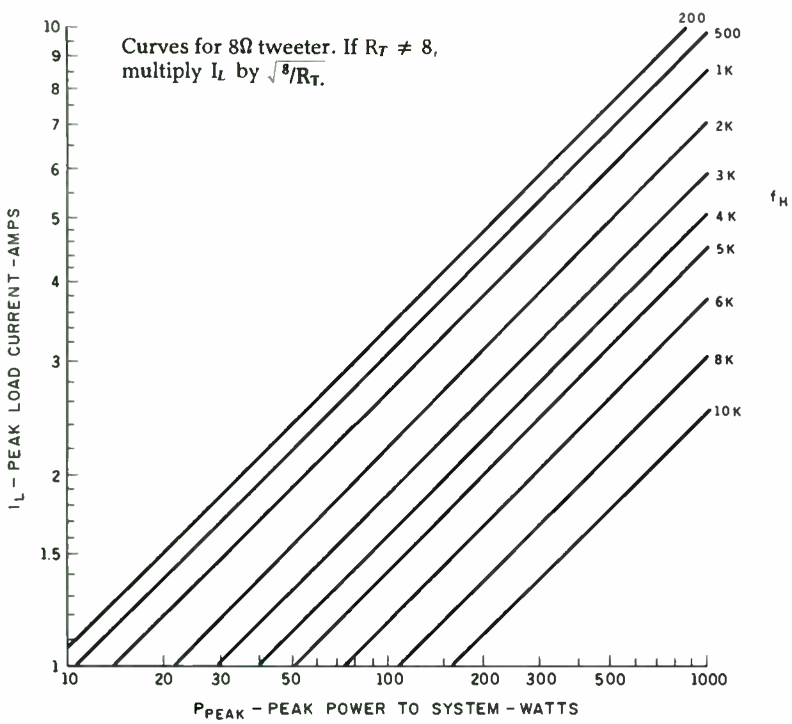


FIGURE 6: Tweeter load current versus peak power.

Based on tests of coils used in woofer COs where distortion problems existed, my approach is conservative for woofers. I believe it is also conservative in the MR and tweeter areas, but I have no data. If you use an equalizer or tone controls to boost portions of the audio band, you must make the appropriate corrections.

Start with the peak power (P_{peak}) to be delivered to the speaker system. As Part I indicated, ten times the maximum average power is a good P_{peak} value. If the driving amplifier is known, then power supply rail voltage will establish an upper limit on P_{peak} for amplifiers with no output transformer. You can approximate maximum P_{peak} as 2x RMS power rating, corrected for dynamic headroom. Comparing the two vertical axis scales (*Fig. 6* [Part 1]) will convert dB dynamic headroom into the correct power multiplying factor. Don't make P_{peak} one kilowatt just because the amplifier can deliver it. Be realistic, or you will make the CO components difficult to realize.

Next, find peak input voltage (V_{peak}) as:

$$V_{peak} = (P_{peak} \times R_{nom})^{1/2}$$

where R_{nom} is the nominal speaker system resistance. If the drivers are not all the same resistance, use an average or take the highest value for a worst case V_{peak}. With parallel COs,

Continued on page 38

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A brand new series of test recordings for Long-Playing equipment, studio environments, and for Compact Disk players. CBS recordings are designed to assist both the audio engineer and audiophile in the evaluation of audio components, studio equipment and reproduction systems. Each record contains a series of tests designed for a particular application, thus eliminating the need for numerous test records.

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CONTENTS: Tracks 1. Reference, L & R, 0dB, 1kHz; 2. Left separation: 0dB, 1k, 125, 4k, 10k, 16kHz; 3. Right, separation similar to track two; 4. Output noise, L & R, Digital zero w/o emphasis; 5. Dynamic range, L & R, 1kHz, −60dB, 4, 8, 17, 31Hz; 7. 61, 127, 251, 499Hz; 8. 997, 1999, 4001, 7993Hz; 9. 10,007, 12,503, 16,001, 17,989Hz; 10. 19,997Hz (Also used for pitch error); 11. Sweep frequency response, 0dB, 5Hz—22.05kHz; 12. De-emphasis Error, L & R, 1k, 125, 4k, 10k, 16kHz; 13. Intermodulation Distortion (SMPTE, twin tone) L & R, 60Hz + 7kHz, 11kHz + 12kHz; 14. Linearity, 997Hz, L & R, 0dB, −1, −3, −6, −10, −30, −39.99, −49.97, −59.94, −70.31, −80.77, −90.31dB; 15. Wow & flutter, L & R, 0dB, 3150Hz; 16. Access Time, L & R, 0dB, 317Hz; 17. Square Wave, L & R, 0dh, 1002.27Hz; 18. Linearity with Dither, 997Hz, L & R, −70.31, −80.77, −90.31, −100dB; 19. Impulse & Polarity Test, 0dB, L & R; 20. Fade to Noise, L & R, −60dB, 500Hz; 21. Monotonicity, L & R, 1105.5Hz, 10LSB.

STR 101: SEVEN STEPS TO BETTER LISTENING. This high precision test record enables you to make sure your equipment functions properly to tune your system to your ears and your room acoustics. Included is a detailed 16-page booklet by *Audio's* Edward Tatnall Canby explaining how to use the record to improve your system's performance.

CONTENTS: Side A. STEREO TESTS. Bands: 1. Left-Right identification; 2. Phasing Test; 3. Loudspeaker Balance; 4. Tone Control Setting (⅓-octave noise bands); 5. Alternate Phasing Test.

SIDE B. STEREO-MONAURAL TESTS. Bands: 1. Tone Control setting (⅓-octave noise tone); 2. Buzz and Rattle Elimination (high level glide tone); 3. Lateral Tracking Test; 4. Vertical Tracking Test.

CTC 300: PHONOGRAPH CARTRIDGE TEST RECORD. Used for measuring the frequency response, crosstalk, low frequency resonance, polarity, compliance, and tracking ability of phonograph cartridges. It contains swept frequency left and right channel test bands, in both audio and infrasonic ranges, whose logarithmic frequency sweeps are compatible with the chart speeds of many graphic level recorders.

CONTENTS: Side A. Bands: 1. Left channel sweep, 2. Right frequency sweep, 3. Left separation, 4. Right separation, 5. Lateral polarity test, 6. Vertical polarity test, 7. Left 1kHz tone, 8. Right 1kHz tone.

SIDE B. Bands: 1. Lateral sweep: tone arm resonance, 2. Vertical sweep: tone arm resonance, 3. Lateral Compliance: 100Hz, 4. Vertical compliance: 100Hz, 5. Lateral tracking: 300Hz, 6. Vertical tracking: 300Hz.

CTC 310: DISTORTION TEST RECORD. Designed for measuring the nonlinear distortion of phonograph pickup cartridges. Distortions are often the result of a non-linear relationship between the stylus velocity (or displacement) and the electrical output voltage from the cartridge. Part of this nonlinearity may originate in the voltage generating system of the cartridge, and in part from the coupling between the stylus and the voltage generator which may be magnetic, piezo-electric, or some other element. Other distortions are fundamental to the the shape of the cutting and playback stylii, their effective vertical tracking angles, or the coupling of the stylus to the record groove. The total distortion usually increases with recorded level throughout the normal operating range of the cartridge, becoming very large as the mechanical limits of the cartridge components are approached or exceeded.

CONTENTS: Side A. Bands: 1. Left Two-tone sweep: F2 = 1.0-50kHz; F1 = 0.685 to 49.685; (F2-F1 = 315Hz; 2. Left: 1kHz Square wave; 3. Right: 1kHz Square wave; 4. Left spots: 20kHz-20Hz; 5. Intermodulation: 4kHz + 400Hz in ascending level steps, phase locked; 6. Right, Intermodulation: 4kHz + 400Hz in Ascending level steps, phase locked; 7. Left 1kHz reference tone; 8. Right 1kHz reference tone

SIDE B. Bands: 1. Right, Two-tone sweep: F2 = 1kHz to 50kHz; F1 = 0.685kHz to 49.685kHz(F2-F1 = 315Hz); 2. Lateral, 1kHz square wave; 3. Vertical, 1kHz square wave; 4. Right spots: 20kHz—20Hz; 5. Right spots: 20kHz-20Hz; 5. Lateral, Intermodulation: 4kHz + 400Hz in ascending level steps, phase locked; 6. Vertical, intermodulation: 4kHz + 400Hz in ascending level steps, phase locked; 7. Lateral, Noise band: 16kHz.

CTC 330: STUDIO TEST RECORD. This was developed to assist broadcast engineers and technicians, advanced audiophiles, and other professionals in evaluating and adjusting the performance of audio disk playback equipment. It provides the range of test frequencies and levels required to measure phono cartridge sensitivity, frequency response and stereo channel separation as well as the sensitivity and frequency response of monophonic cartridges. Standard level signals can be used to set the gain in other parts of the reproducing system and to identify the left and right stereo channels. Other test bands are used to check system phase and measure or adjust the rotational speed of a turntable.

CONTENTS: Side A. Bands: 1A. Frequency response measurement (l. chan); 2A. Frequency response measurement (r. chan.); 3A. Lateral frequency sweep; 4A. Spot frequencies (l. chan.); 5A. High level sweep (l. chan.); 6A. High level sweep (r. chan.); 7A. Left channel reference tone; 8A. Right channel reference tone.

SIDE B. Bands: 1B. Lateral spot frequencies; 2B. Right channel spot frequencies; 3B. Lateral reference tone; 4B. Vertical reference tone; 5A (B) Stroboscope (120Hz); 5B (B) Stroboscope (100Hz).

CTC 340: ACOUSTICAL TEST RECORD. Intended to be used for measuring the performance of an entire reproducing system, including the loudspeakers. The program uses random noise suitable for measurement with instruments or, in some cases, interpretation by ear. The signals are characterized as "pink" noise, which is random in nature and has equal average energy in each octave of the audio frequency range, from 20Hz to 20kHz. A spectrum analysis of pink noise using a proportional bandwidth analyzer (such as the "real time" spectrum analyzers often used for acoustical measurements) produces a linear (or "flat") frequency response. In contrast, a constant bandwidth analysis of pink noise yields a response that decreases at 3dB per octave with increasing frequency.

CONTENTS: Side A. Bands: 1. Left sweep: ⅓-octave pink noise, 20Hz—20kHz; 2. Right sweep: ⅓-octave pink noise, 20Hz—20kHz; 3. Lateral sweep: ⅓-octave pink noise, 20Hz—20kHz; 4. Left spots: ⅓-octave pink noise, 20Hz—20kHz; 5. Left channel: wide band pink noise, 20Hz—20kHz; 6. Right channel: wide band pink noise, 20Hz—20kHz; 7. Left: 1kHz Reference tone; 8. Right: 1kHz Reference tone.

SIDE B. Bands: 1. Lateral spots: ⅓-octave pink noise, 20Hz—20kHz; 2. Right spots: ⅓-octave pink noise, 20Hz—20kHz; 3. Left and right wide band noise (in phase), 20Hz—20kHz; 4. Left and right wide band random noise (random phase); 5. Lateral: 1kHz Reference tone.

CTC 350: TURNTABLE AND TONE ARM TEST RECORD. The three basic components of a phonograph record player are the turntable, which rotates the record at (ideally) a constant angular speed, the tone arm, which supports the cartridge in a correct geometrical relationship to the record groove, and to the cartridge, whose stylus traces the record groove and which generates voltage analogs of each groove wall modulation. Because the components of a record player system interact to some degree, it is difficult to measure or specify the performance of one without considering the properties of the other two. In most cases it is acceptable to specify the makes and model numbers of related parts of the system (e.g. the cartridge or tone arm), and their operating conditions where applicable, when measuring the performance of another part.

CONTENTS: Side A. Bands: 1. Left: 1kHz reference tone; 2. Right: 1kHz reference tone; 3. Wow and flutter; 4. Lateral sweep: 2—100Hz; 5. Vertical sweep: 2—100Hz.

SIDE B. Bands: 1. Rumble: Reference tone; 2. Rumble: Quiet grooves; 3. Lateral: 1kHz Reference tone; 4. Vertical: 1kHz Reference tone; 5A. Vertical strobe: 120Hz; 5B. Vertical strobe: 100Hz.

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Table 11: CROSSOVER COMPONENT REQUIREMENTS: An Example

System: All Drivers 8Ω f_L = 800Hz, f_H = 4kHz
Peak power input to system = 200W

A. 2nd-Order All-Pass--Transposed MR (See *Figs. 2* and *3*)

Coil

Coil	IF	IL	I_{pk}-A	L-mHy	I_{pk}x(L)*	AD*
L12	1.00	3.9A	3.9	3.29	7.1	W
L23	1.00	3.6A	3.6	0.53	2.6	M
L22	0.28	3.6A	1.0	3.95	2.0	M
L31	0.22	2.1A	0.5	0.66	0.4	T

Cap

Cap	VF	Vin	V_{pk}-V	C-μFd	IF	I_{pk}-A	AD*
C11	1.00	40V	40	12.0	0.22	0.9	W
C24	1.26	40V	50	15.0	1.32	4.8	M
C21	1.00	40V	40	2.0	0.32	1.2	M
C32	1.14	40V	46	2.4	1.00	2.1	T

B. 3rd Order Butterworth--Spec. Casc MR (See *Figs. 2* & *3*)

Coil

Coil	IF	IL	I_{pk}-A	L-mHy	I_{pk}x(L)*	AD*
L13	1.09	3.9A	4.3	2.39	6.6	W
L11	1.00	3.9A	3.9	0.80	3.5	W
L26	2.53	3.6A	9.1	0.86	8.5	M
L24	1.64	3.6A	5.9	0.54	4.3	M
L22	1.46	3.6A	5.3	0.20	2.4	M
L32	1.06	2.1A	2.2	0.24	1.1	T

Cap

Cap	VF	Vin	V_{pk}-V	C-μFd	IF	I_{pk}-A	AD*
C12	1.02	40V	41	33.1	1.06	4.1	W
C25	1.94	40V	78	24.0	2.31	8.3	M
C23	1.00	40V	40	32.9	1.64	5.9	M
C21	1.40	40V	56	6.9	1.82	6.6	M
C33	1.46	40V	58	3.3	1.09	3.9	T
C31	0.36	40V	15	9.9	1.00	2.1	T

*AD=Associated Driver: W=Woofer M=Midrange T=Tweeter

Continued from page 36

as this work covers, each section sees V_{peak} as input. Therefore, find the capacitor's voltage requirements by multiplying V_{peak} by the VF for each capacitor in the CO you plan to use.

To use the IFs, you must establish the expected load current (I_L). I do this by apportioning the peak power to the various drivers, based on the work I presented in Part I, plus some added safety factors. *Figure 4* relates woofer I_L to system P_{peak} for various f_L values. If R_W is not 8Ω, be sure to correct I_L as indicated. The MR is a bit messy since you have two CO frequencies that can vary. I examined the available data and found, for a given f_H/f_L ratio, the percent peak power did not vary greatly with CO frequency. *Figure 5* thus uses the worst case results to relate I_L to system P_{peak} for the MR driver. *Figure 6* relates I_L to system P_{peak} at various f_H values for the tweeter. Once you know I_L, the IF can be applied to yeild coil peak current.

Look also at the peak capacitor currents. Examine the capacitors you plan to use before you put these currents through them. Test the capacitors at average current (peak current divided by 3.16) with an amplifier, to see whether they heat up or fail. I have seen too many COs with tiny capacitors in them trying to pass surprisingly large currents. Capacitors can cause as many distortion problems as coils.

AN EXAMPLE. Lets look at the case of a three-way system using all 8Ω drivers, with f_L = 800Hz and f_H = 4kHz. The unit will be driven by a 60W RMS amplifier to produce a peak power of 200W. This power corresponds to about 20W average music power and seems realistic for the system. System impedance is a nominal 8Ω, so V_{peak} = 40V.

We want to consider two basic CO types: the second-order AP with transposed MR, and the third-order BW with special cascaded MR. *Table 11* shows the results. I used *Fig. 4—6* to establish I_L for each CO section. Then I applied the appropriate VF and IF values for each CO type. In the example, f_H/f_L = 5:1, but we only have factors for 4:1 and 6:1. I used the 4:1 factors to stay conservative. *Table 11* shows peak coil current and peak capacitor voltage and current. I also show the component values and a factor (coil peak current times square root of inductance) which indicates how difficult it is to implement the cored coil.

The second-order CO offers no surprises on capacitor voltage, but C24 could see as high as 5A peak. Coils for the MR and tweeter are not difficult, but as expected, the woofer coil L12 is by far the most difficult to realize. I used the third-order BW CO with special cascaded MR in the example because it has severe component requirements. With only 20W average input, of six capacitors used, three should be greater than 50V units and five could see peak currents of 4A and above. The woofer coils are of expected difficulty, but remember, both coils are in series with the woofer, making DC resistance an important factor.

It may come as a surprise to you that the MR coils are more difficult to implement than the woofer coils. This is common with any cascaded MR topology due to the response peaking, and it gets increasingly severe as f_H/f_L is reduced.

The two cases I have examined in this example clearly indicate that, for a given set of drivers, fixed CO frequencies and a given system input power, the CO type and topology can make a major difference in the component stresses. It is just one more factor on your already long list to determine which CO type to use.

CONCLUSIONS. In Part II I have presented voltage and current multiplying factors for many of the common parallel CO configurations covered in R.M. Bullock's design articles (*SB* 1/85 and 2/85). I have provided the additional curves and approach needed to arrive at a conservative set of CO component stresses, along with a two-part example.

REFERENCES

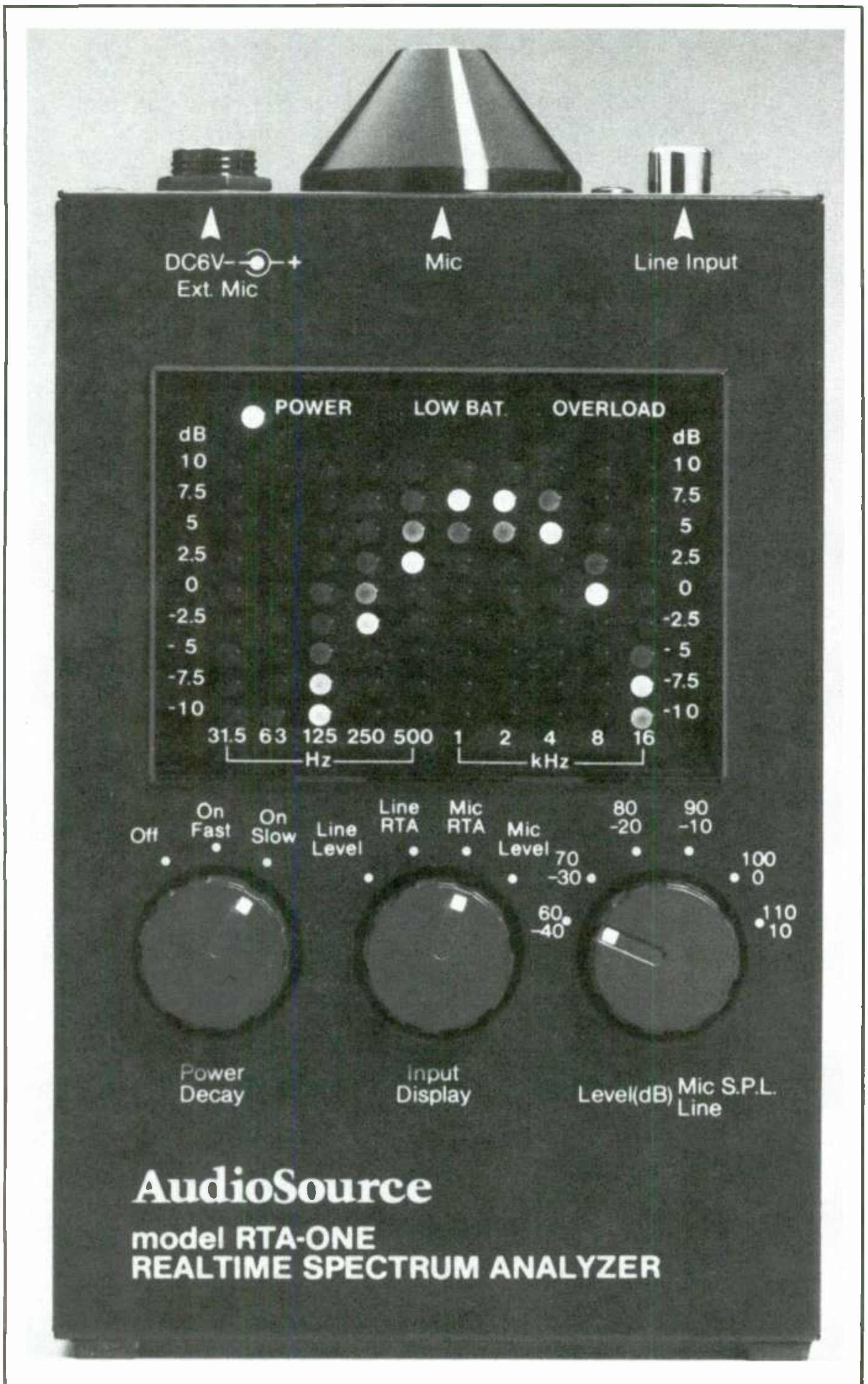
1. Bullock, III, R.M., "The Analysis and Synthesis of Passive Constant Power and All-Pass Three-Way Crossovers," AES Preprint 1995 (F-1).
2. Allie, M., and R.A. Greiner, "Synthesized Three-Way Loudspeaker Crossovers," AES Preprint 2007 (F-3).

SOURCES

Cored crossover and ferrite bobbin coils available from:
Dynamic Acoustics
PO Box 646
San Ramon, CA 94583
Fast Reply #IK4

Solenoid wound ferrite core coils are available from:
McGee Radio
1901 McGee Street
Kansas City, MO 64108
Fast Reply #IK44

Kit Report
AudioSource RTA-ONE



If you read *SB* 1/86, you may have noticed the AudioSource RTA-ONE octave band analyzer which displayed the response of my midrange horn. I have used the RTA-ONE since mid-1985 and have found it to be a valuable diagnostic tool. I also recommend the optional pink noise generator, the PNG-ONE. Both are manufactured and distributed by AudioSource, 1185 Chess Drive, Foster City, CA 94404, (415) 574-7585.

Suggested retail price for the RTA-ONE is $249.95, and for the PNG-ONE is $54.95. A kit, which includes the RTA-ONE, the PNG, an AC adapter, and a carrying case retails for $385.95.

Description

The AudioSource RTA-ONE is a small but rugged battery-operated, 10 octave band sound analyzer calibrated in sound pressure level (SPL) or line level. A built-in electret condenser microphone measures the incident sound level, while an RCA jack accepts the line level inputs.

The 20dB range (−10, 0, +10dB) LED display is segmented in 2.5dB steps. For easy reading, a separately colored LED indicates the 0dB reference level. A level switch selects the absolute SPL or line dB level for the relative 0dB display line. The level select ranges from 60dB SPL (or −40dB line level) to 110dB SPL (or +10dB line level) in 10dB steps, for a total dynamic range of 70dB. This range allows you to monitor sound levels in a room (60-100dB), or at a rock concert (up to 120dB). An overload LED indicates whether the 20dB range is being exceeded in one or more of the octave bands.

The other two control switches select decay characteristics and input selection. You can set the display decay for fast or slow. When testing speakers for steady state pink noise, either decay characteristic will work well. For real-time transient monitoring, however, the fast decay mode is especially valuable. The input selection switch picks the display mode, i.e., mike and line real-time analyzer (RTA) or mike and line level. The RTA display modes have been described in previous sections, but the mike and line level display modes deserve some explanation. All of the octave band levels go to the same level, which apparently corresponds to an "A" weighted SPL measurement used by acoustic researchers. I do not find this feature to be particularly useful for speaker builders.

Applications

For many years, I struggled unsuccessfully to match driver levels at the crossover point. I became so enthralled with the sound of a driver enclosure combination (Beranek's Law) that I simply ignored the fact that the woofer horn was 6dB more efficient than the midrange. Later, my wife would let me know the sound, although great, was somewhat bass-heavy. Then, a technician co-worker showed me his RTA-ONE and let me borrow it. After using it for a short while, I was hooked.

I found I could hold it close to each driver for a near field test of driver level. If the levels weren't the same, the meter told me, within a couple of dB, what attenuation I needed. By ear, I could then use L-pads to fine-tune the level equalization. When I later checked the total response in the lab with an FFT, I discovered the level matchups were very good indeed.

There is, however, a pitfall in using the analyzer: you can get readings that can fool you. You will get reliable readings if your drivers are reasonably flat, but if there are response irregularities, such as a strong resonance (some of my prototype horns have had this problem), the readings may be anomalously high. If you continually encounter this situation, a ⅓ octave analyzer, such as the more expensive Heathkit (reviewed in *Audio Amateur* 3/85), may be a better choice.

The optional Pink Noise Generator (PNG-ONE) is also a good buy. I tested the generator on an integrating FFT and found the generator maintained the 1/f characteristic down to 15Hz before the response rolled over. Since octave band bandwidth increases linearly with frequency, the pink noise source needs the inverse frequency characteristic for an accurate transducer (i.e., loudspeaker, amp, and so on) response measurement.

The marketing people at AudioSource, however, seem to have confused white noise with pink noise in their ad copy—they say pink noise is "flat" (sic). It is obvious to me the design engineers knew what they were doing, because the RTA-ONE and the PNG-ONE work quite well together.

Summary

The AudioSource RTA-ONE will make a valuable addition to your test gear collection. Its small size and flexibility make it a very good buy.

Bruce C. Edgar
Contributing Editor

STEP RESPONSE

continued from page 12

with your knowledge, intuition, patience and listening experiences.

If "time alignment" is so necessary for "accurate" sound reproduction, why are so few commercial loudspeakers built with it? First, it costs more to produce a speaker with the drivers stepped back; more pieces of wood cabinetry must be cut and assembled. The loudspeaker also looks peculiar unless it is covered with a large cap of foam (itself a detriment to high-quality sound). An alternative to physical alignment of the drivers is to use an active crossover with the appropriate time delays built into it. It is not easy, however, to construct a totally acoustically transparent active crossover, and the extra power amplifiers would multiply the loudspeaker cost several times over.

Finally, most loudspeaker manufacturers, including many with full-time research staffs and many times more sophisticated equipment than I used, have yet to be convinced of the need or commercial success of time-aligned loudspeakers. Amateur speaker builders need not worry about these problems.

ACKNOWLEDGEMENTS

My thanks to Rich Atneosen and Arnold Amenda for their great contributions and criticisms in developing the techniques presented here.

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Craftsman's Corner

Strathearn System

PHOTO 1: The Strathearn design exposed. My design is the result of time-delay spectrometry testing.

I have spent the better part of the past two years building a complete speaker system around the Strathearn driver, which is rather difficult to use and extremely difficult to match with other drivers. My design, shown in *Photo 1* and *Photo 2*, is the result of time-delay spectrometry testing in WED Enterprises laboratories, and more than 200 hours of listening tests.

At present, I use electronic crossover between the Dynaudio sub-bass units and the mid-bass upward units. I use a passive crossover (*Fig. 1*) between the KEF, Strathearn and Panasonic units. I have bought and tried a couple of enclosure kits, with less than satisfactory results. The enclosure I am using is a monopole with pressure relief, so the Strathearns' backwave is totally absorbed without loading the diaphragms.

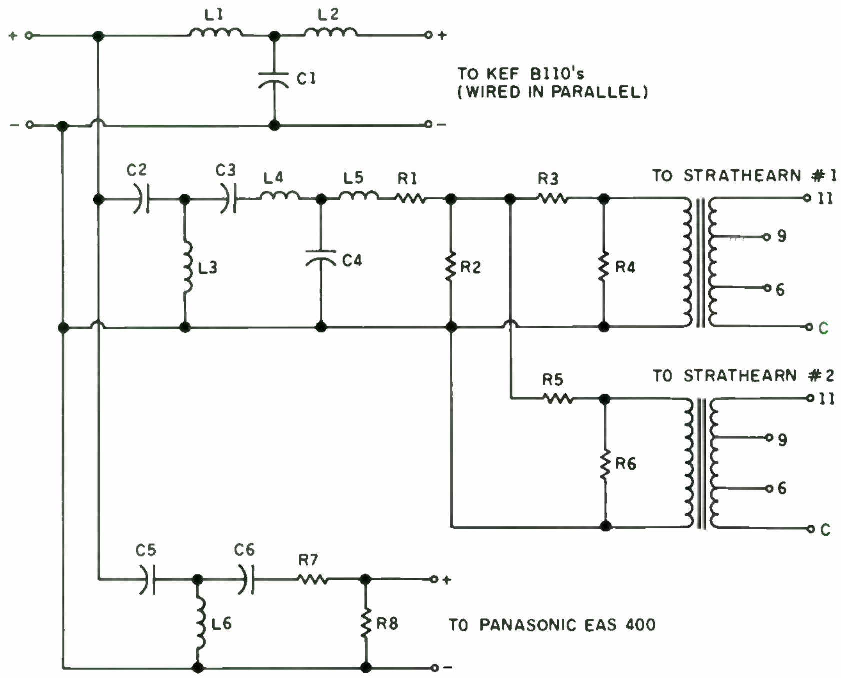
The design does not adhere strictly to acoustic theory, but rather evolved from modifications suggested by extensive listening sessions. In all, six people were involved in these sessions, and all thought a better-designed system would cost several thousand dollars.

Many Strathearn users have tried to get the response down to 150-200Hz by using large baffle surfaces. I have tried this method, and it works. The problem is that regardless of the baffle surface, the Strathearns still have a very irregular response at high levels (90-110dB) and can catch on the edge of the magnet structure with the amount of excursion they must produce, thereby ripping the foil conductor.

My answer to this problem was to use a small (5 to 6-inch) driver to fill in from 100-500Hz. The KEF B110 works very nicely, although using it lowers the overall system efficiency. One advantage of using the B110 is it is very difficult to determine where the DEFs cross over to the Strathearns. Another advantage is with the narrower enclosure and rounded edges, the system's imaging is outstanding.

The response curves shown in *Figs. 2-4* were run without the Hafler equalizer

Continued on page 43



L1 = 2.2mH
L2 = 740μH
L3 = 1.2mH
L4 = 158μH
L5 = 60μH
L6 = 95μH

C1 = 115μF (Composite)
C2 = 60μF (Composite)
C3 = 185μF (Composite)
C4 = 5μF (Polypropolene)
C5 = 4μF (Polypropolene)
C6 = 10μF (Polypropolene)

Resistors R1, R2, R7, R8*
Resistors R3 & R5: 0.5Ω (matched) 25W non-ind metal clad
Resisters R4 & R6: 20.0Ω (matched) 100W non-ind metal clad

Note: The author did not provide values. For information write to: Don E. Prock, 590 Maverick Lane, PO Box 3698, Rubidoux, CA 92509—Ed.

FIGURE 1: My passive crossover design. Note: all coils are custom wound with 16 gauge magnet wire, and matched to within 1% tolerance of each other and values listed. All caps are matched to within 1% tolerance.

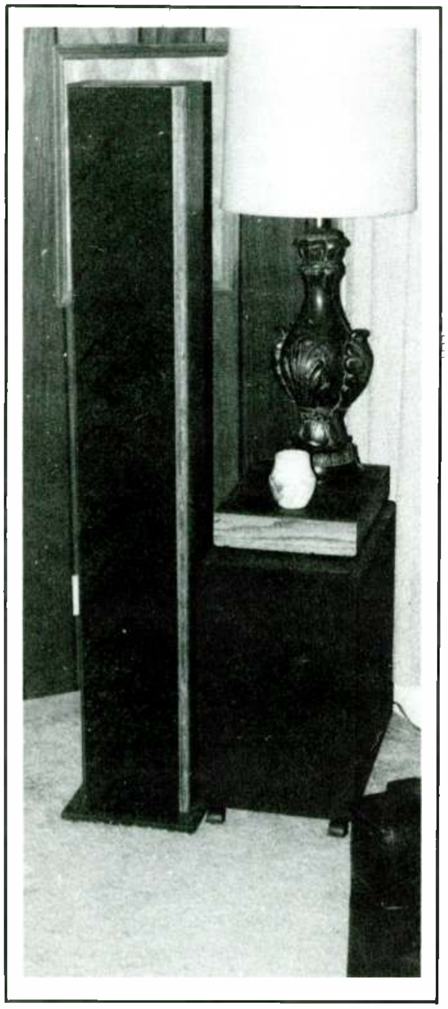


PHOTO 2: The finished Strathearn in my living room.

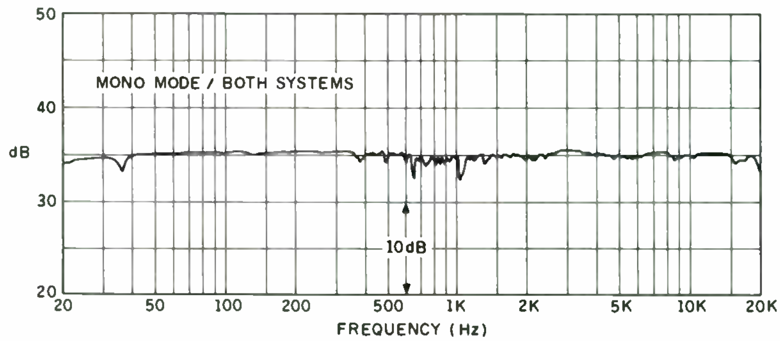


FIGURE 2: The dynamic/ribbon line source response curve at 60 seconds with both systems on mono mode.

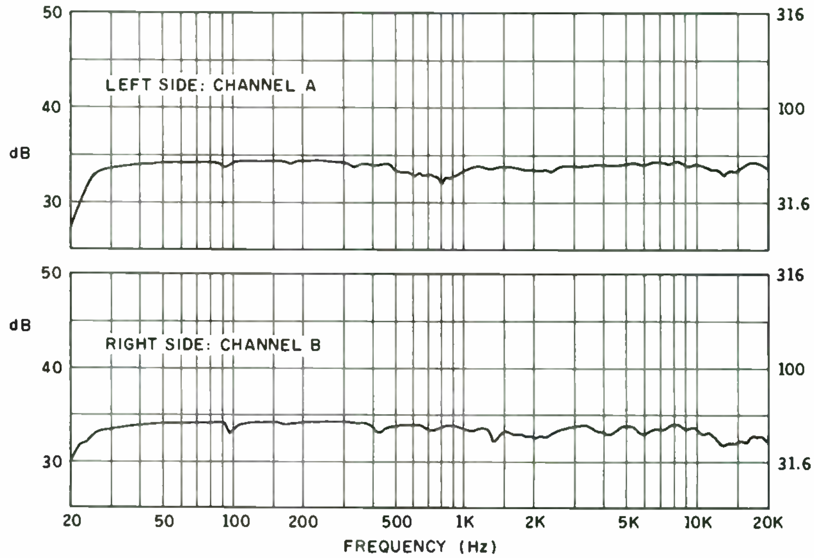


FIGURE 3: The response curve with a sweep speed of 15 seconds.

Continued from page 41

in place. I used the equalizer because the system is so flat that most listeners do not hear what they are used to hearing. In my living room, which is heavily furnished with sound-absorbing materials, the system measures ±1.5dB from 25Hz–18kHz. It is very fast and clean, and I have been able to produce 115dB peaks on an Ivie spectrum analyzer with approximately 25W of power. My living room is about 4,200 cubic feet, and I made the measurement at 10 feet from the speakers.

I have listened to Beveridge, IRS and Magneplanar units, and I would not trade my system for any of them. When I feed the Strathearns with my modified Leach amps, it produces sound that raises the hair on the back of my neck. For some reason, the Strathearns seem to make the Leach amps sound almost as sweet as tube amps from the better manufacturers.

Don E. Prock
Rubidoux, CA 92509

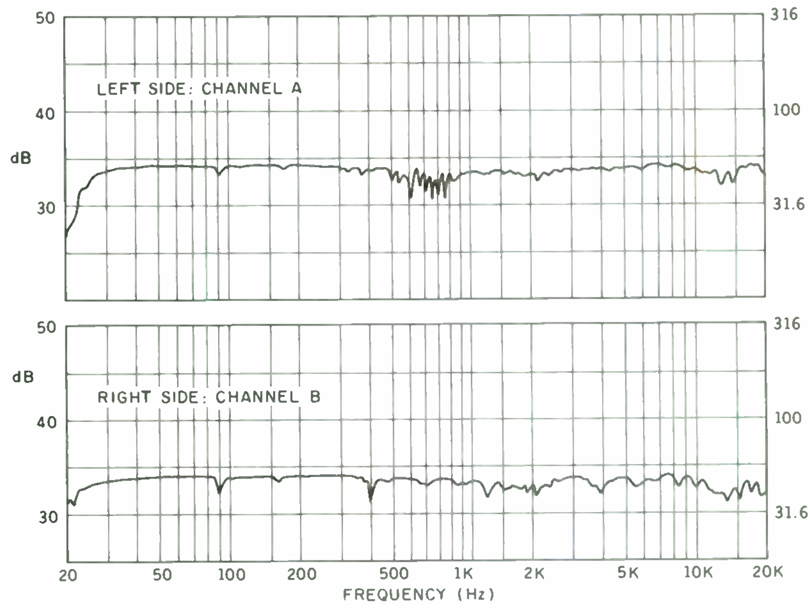


FIGURE 4: The response curve with a sweep speed of 30 seconds.

SB Mailbox

ESL GREMLINS

A couple of gremlins crept into my letter (*SB* 2/86), "Examining the ESL." The first error occurred in the fourth paragraph. The diaphragm to plate spacing on the Malme speaker was ¼ (0.25) of an inch, not the ¾″ indicated in the article.

In the fifth paragraph, the cited book should have read "Fundamentals of Acoustics" by Harry Olson, not "electrical engineering."

Paragraph 12 indicated "other designers have maintained the width at a value which is equal to or less than 1.3 of the wavelength of the speakers' uppermost frequency." The 1.3 should have been ⅓ (one-third).

Ronald Wagner
Fremont, CA 94539

THE VOICE OF EXPERIENCE

I am an amateur singer (tenor) and an opera lover, so voice reproduction is extremely important to me.

Because I don't have the room, money, or knowledge to deal with test equipment, I must rely on my own ears or someone else's experienced ears to come up with suitable solutions to my questions.

For several years, I have been using Philips drivers in a triangulated TL configuration, built from an article in *TAA*. My midrange is a 5″ sloped cone with textile surround and a plastic rear chamber. So far, I am pleased. It's fairly smooth and its low resonance and low crossover point (±400Hz) makes it handle the vocal range almost in its entirety.

I want to improve my speaker system, but I have plenty of doubts. With so many choices in different catalogs, there is no way I can listen to them all, and it is too expensive to try them on my own system.

I am attracted to the fiberglass-coned Siares, but I have never heard them. Has Gary Galo had any experience with them? How about the MB Electronics drivers (they make the PHT and PMT dome tweeters and midranges, sold under the Peerless name)? They have made a titanium dome midrange that apparently goes way down in range (McGee catalog claims 340Hz crossover!). They cost less than Dynaudios (±35% less), and although they do not handle Dynaudio's prodigious amounts, they do much better than Audax and Philips.

My fear of purchasing something extremely revealing is compounded by two things. First, although I have and listen to compact discs, I also own many old records and tapes of old recordings, most of which are now unavailable. As you know, they are plagued by noise, scratches, and so on, but the performances by great artists are irreplaceable.

Also, most of the higher quality amplifying equipment available is short on filters, mono switches, and so on. My indecision alternates between keeping what I have as-is, or having maybe two sets of speakers—something up-to-date to deal with good records and compact discs, and maybe a smaller system (not too sophisticated) to enjoy my older opera records. Most of the old records do not go very high, and absolutely not very low!

Regarding the possibility of two alternate systems for my listening sessions, I have read some articles which condemn simultaneously using more than one system in a room. Maybe I could use the second set as an ambience or rear speakers. Any suggestions?

As a point of reference, most of the music I listen to is operatic, but I also listen to a lot of classical—from Baroque to Twentieth Century. I listen to only a little pop and jazz.

Rafael Lopez
Miami, FL 33125

Mr. Galo replies:

For many years, the Philips driver was considered to be the best of their cone midrange types. You are certainly correct about its accuracy in the vocal range. By today's standards, however, this driver lacks upper midrange detail and overall power-handling.

The Peerless PMT 51 midrange looks good on paper, although I have not used it. The data says it is good down to 500Hz, but given its high F_s (380Hz), operating it that low might be asking for trouble. I do not know whether Peerless has used any special damping techniques (such as Dynaudio used in the D-76) to allow safe crossover at 500Hz.

I have not used the Focal or Siare drivers, but hope to be able to try them in the future. Like yourself, I must purchase most of the drivers I test, although I occasionally get a dealer or manufacturer loaner. I hope more dealers and manufacturers will be willing to submit drivers to me for testing. Whenever they do, I will report the results in SB.

Regarding old recordings playback, you raise some worthwhile questions. If you subscribe to Audio Amateur, *I refer you to my 1/85 article, "A Preamp for Vintage 78s." I believe the key to successful playback of old records lies in the proper turntable, arm, stylii, and preamp equalization. I own nearly 2500 78 rpm discs, including numerous vocal records which I play frequently. I use the same playback system for them that I use for my best LPs, with the exception of the above-mentioned pieces of equipment which must be tailored specifically for 78s.*

My friends are continually amazed at the exceptional sound quality contained in the grooves of these recordings. I would go so far as to say that well-recorded 78s from the mid-1930s sound more impressive using the playback methods I use than modern LPs and tapes played on the cheap gear used by the average American record buyer (let us remind ourselves that most record buyers are not audiophiles, and use pretty poor playback gear). This often makes me wonder just how much progress has really been made in sound reproduction for the average consumer.

For the playback of extremely noisy discs, a parametric equalizer (such as the Phoenix Systems unit I reviewed in TAA *2/86), and perhaps a dynamic click and pop eliminator will help (they probably won't eliminate them completely, but they will reduce the annoyance level). A "wet playing" with distilled water will reduce audible scratches and other noise resulting from wear on old LPs. I don't recommend this for everyday use, but it will allow you to make a low-noise tape recording of your worn records.*

Therefore, I do not see the need for a different amplifier and speaker system for old *records. I prefer to treat the recordings properly at the source, and play the processed signal through my main loudspeakers.*

SOLVING SOME TL MYSTERIES

I had the good fortune of meeting the only other (that we know of) *SB* subscriber in the New Orleans area, Francis "Duke" LeJeune. He's built a system using two woofers in a transmission line (TL) with a Gold Ribbon mid/tweeter. I was very impressed by the smoothness of the bass response. Two of my friends are about to commission me to build them new speakers: one set a main (35Hz F_3), and the other an extension pair (possibly as satellite units) with a cutoff of about 100Hz. I'd like to use a TL loading in both of these, but I have a few questions about TLs I cannot see addressed in current writing.

Has Gary Galo studied electrical TL theory and, if so, what analogies does he see with acoustic TLs? Specifically, I am concerned with addressing TL design not simply from wavelength versus tube length, but from driver impedance (generator impedance) versus line impedance.

In his original article (*SB* 2/86), Mr. Galo stated that if the ¼ length of the line is unacceptably long, take the ¼ length of some higher frequency and base the line on that. What would be the effect on ultimate F_3 and ripple from using, say, a driver of 45Hz F_s on a 100Hz line? What would the effect be if the crossover contains a 100Hz high-pass filter anyway?

Finally, Mr. Galo says it is good practice to include a 45° angle piece in the first section of the line to reduce early reflections to the woofer(s). In his experience, at what distance into the line from the woofer can we stop worrying about early (?) reflections? This question arises from my desire to make a large-scale (25Hz) TL for my CVA that would have a spiral cross section and, therefore, no parallel walls at all.

One thing I like most about *SB* is that heretofore mystery-shrouded areas (TLs, horns) are being illuminated. I think Mr. Galo's contributions have done much toward that, and I look forward to whatever input he can provide.

Scott Ellis
New Orleans, LA 70116

Mr. Galo replies:

There are anlogies between electrical transmission lines and undamped quarter wavelength acoustic lines (i.e., acoustic labrynths). The electrical equivalent to the ¼ wave labrynth, which is a pipe open at one end and closed at the other, is a short-circuited electrical TL. In the shorted electrical line, if it is ¼ wavelength long, the incident current (outward-going current) is cancelled by the reflected currents. The current at the point ¼ wavelength from the short will, therefore, be 0. This point is called the current node. This is analogous to the ¼ wave acoustic pipe, where the pressure node at resonance opposes the acoustic motion of the source (i.e., the woofer), and inhibits its vibration. For a more detailed explanation of electrical TLs, I recommend Chapter 20 in the pre-1977 Radio Amateur's Handbook.

The above analogies apply to undamped pipes. When the line is filled with damping material, such as wool, its characteristics are no longer the same. Harry Olson, in Music, Physics, and Engineering, *pp. 86-88, gives formulas for determining the impedance of undamped pipes, but I do not know how to compute the impedances for wool-filled lines.*

If you are building a satellite system which would be crossed over to the subwoofers at 100Hz, you do not want F_3 to occur at 100Hz, since this will alter the crossover roll-off characteristics. It is better to put F_3 at least one-half octave lower than the crossover frequency, at around 75Hz. If you use a very low F_s driver in a line that is ¼ wavelength at a much higher frequency, the 3dB down point in the frequency response will probably be lower than expected. In my

SB *Mailbox*

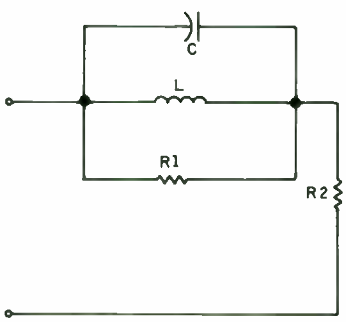
TL-5 system, using an Audax 5" driver with an F_s of 35Hz, the 3dB down point is actually 39Hz, even though the line is ¼ wave at 60Hz.

I don't think it is necessary to include right angle pieces farther than half-way down the line's length. Without parallel walls, reflection problems are minimized even further.

A WELCOME SUGGESTION

I was interested to see the empirical difference between notch filters using NP electrolytics and those using Mylar capacitors, as revealed in David Weems' *SB* 2/86 article.

Has he investigated the effect of the impedance load, R_2, below?



For a given R_1, the notch's cut will go to 0 if R_2 is increased significantly, whereas for "small" R_2, a very large cut may result. Perhaps his formulas reflected a value for R_2 different than the one he gave it. This might partially account for the difference between his measurements and the theoretical values of Q.

From my calculations, he could choose:

$R_1 = R_2Q - R_2$, then
$C = Q/2\pi fR_1$, and
$L = R_1/2\pi fQ$

As Mr. Weems mentioned, he is still left with the problems of imperfect capacitors, complex (non-resistive) driver loads, and interactions with the crossover.

In reference to his *Fig. 6*, while it would be beneficial to place a driver impedance compensation circuit between the notch filter and the driver (so the notch will have a more resistive load), I believe he must insert a different compensator between the crossover and the notch (Fig.

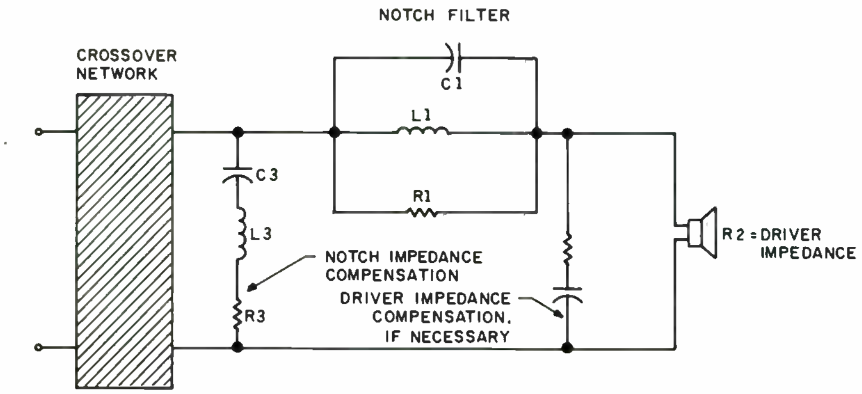


FIGURE 1

1) so the crossover won't see a high impedance at the notch frequency (when the notch frequency is close to the crossover frequency). Perhaps this one would be helpful:

$C_3 = L_1/R_2^2$
$L_3 = R_2^2C_1$
$R_3 = R_2(1 + R_2/R_1$

I've enjoyed Mr. Weems' articles and books. I'd like to know if he finds any of these calculations useful.

Ralph Gonzalez
Philadelphia, PA 19143

Mr. Weems replies:

You are right about the effect of the load impedance (R_2 in your diagram) on the performance of a notch filter. But I measured the impedance and Q of each filter with no load (i.e., $R_2 = 0$), and yet the values I obtained were below those predicted.

Your "notch impedance compensation" filter suggestion is a good one. It is precise, removes the impedance variation produced by the notch filter, and increases the effectiveness of the filter. Using it with an impedance equalizer on the driver should produce a more uniform impedance load for the crossover network.

As to the value of the impedance equalizer after the notch filter to provide a more resistive load for the filter, it is good in theory, but in a practical situation may not be an important virtue.

For example, you are likely to find the Q of your notch filter is lower than predicted. In that case, you increase the value of the shunt resistance, R_1. That move, combined with the impedance equalizer on the driver, can produce a deeper than desirable cut at the notch frequency.

The simplest solution for such a problem is to use the circuit I showed in Fig. 6. *A more elegant solution would be to make up a new notch filter with a different LC ratio and a higher Q. Your choice might depend on how much pleasure or aggravation you get from making up notch filters.*

The only other objection I can see with your circuit is it violates the adage "what you don't add on won't cause you any trouble." But so does the notch filter.

LOWERING THE BOOM

I received my first issue (1/86) of *SB* last week, and was interested in Bruce Edgar's answer to Frederick Weber regarding the Speakerlab SK.

I built a pair of SKs from scratch in 1977 and have never been really happy with the sound of the upper bass range. They've always sounded "boomy." I added polyfill behind the woofer, which smoothed the sound somewhat.

I would, however, like to learn what modifications I can make to the horn(s) and/or crossovers to improve the sound.

Dave Wharran
Clearville, PA 15535

Contributing Editor Edgar replies:

I went back and looked again at my calculations for the EVM15L and discovered a mistake. Instead of an optimum throat size of 46 square inches, it should be 61 square inches.

The SK horn has a throat size of 3x13", or 39 square inches. I have not found any other 15" woofers that want to work into that small a throat area.

If you are willing to try an experiment, put in a JBL2220H ($F_s = 37Hz$, optimum throat size = 55 square inches). An EVM12L driver will work into 39 square inches, but the 13" wide throat of the SK does not match the 12" diameter very well.

My radical suggestion for modifying the SK horn is as follows: remove the 15" driver and set it aside, select a suitable sledgehammer and demolish it into small pieces, and tie up the remains for the trash pick-up.

In all seriousness, I've found that is the only way to deal with that "turkey." I have had to do the same with some of my experimental horns that didn't quite work out.

The SK woofer ($F_s = 24.7Hz$, $Q_{ts} = .38$, and $V_{as} = 23.3$ cubic ft.) is better suited for a sealed *box enclosure. For a* Q_c *of 0.7 and an* F_c *of 47Hz, you will have to build a 9.7 cubic ft. box. A smaller 4.7 cubic ft. box will give a* Q_c *of 1.0 and an* F_c *of 68Hz.*

I regret none of these solutions are quite optimum, but maybe they will spur you to try your hand at bass horn design.

WONDERFUL WARBLER

I find the Warbler oscillator [Dick Crawford, "The Warbler," *TAA* 1/79, pp. 22-27, 43. Available from Old Colony Sound, Box 243, Peterborough, NH 03458. Kit No. KK-3.] to be a very useful item, especially in a horribly difficult room. I use it carefully since it is exceptionally sensitive to object placement, including one's own body.

I found it accelerated the side-to-side balancing process, terminating with a ⅓ octave white noise band, and providing greater and more stable accuracy. To me, global side-to-side balance at every ⅓ octave is far more important to total system performance than global flatness response for the audible, subaudible and superaudible spectrum, providing it is done with the highest possible precision.

Regarding Isobarik (*SB* 3/85), I have placed two JBL2235s (15"), glued front-to-front, in a transmission line system. I don't give a hoot if the parameters are equivalent to a single driver, or if I lose 6dB. In a small room (15x18'), I have ample volume for overloading, even at 16Hz.

I do know I have 24,000 gauss to control the diaphragm and, with a proper amplifier, the results are the purest lows I have heard during my 34 years as an audiophile.

By the way, I use a Kenwood M2, 200W amplifier (enhanced per driver) with electronic crossover at 36dB/octave, and use the Sigma drive option. It far surpasses other amplifiers of far greater intrinsic value and cost.

You should never underestimate the power required for an efficient driver at normal listening levels. Dafos, for example, require nearly 50W per driver on the Japanese drum.

Michel Deneault
Ste. Foy, Quebec G1W 2V3

CVA IN DIXIE

I have finally finished my second Curvilinear Vertical Array (*SB* 2/85), and I would like to report the results.

My pair is located in my apartment, which is a remodeled attic and, therefore, a pentagonal cross section. The speakers are placed about 14′ apart, and due to furniture placement, they are about 2′ from the angled ceiling, as measured from the top of the arrays.

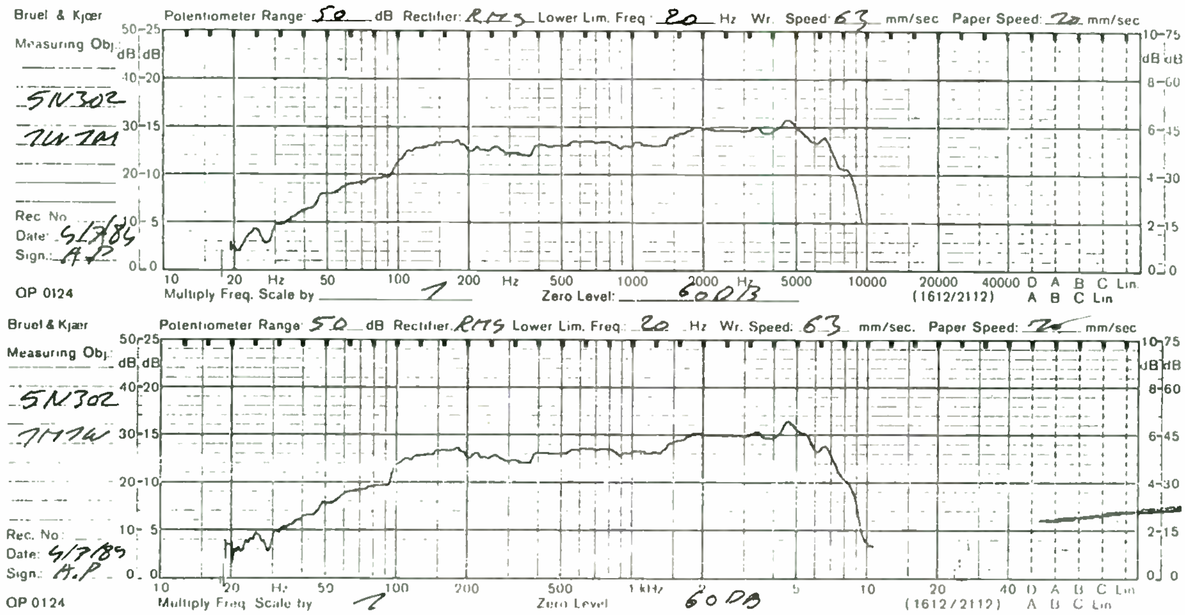
Initially, I had the CVAs rotated out at the traditional 45° angle. This produced a rather diffuse image, without a stable center, even on monophonic material. I then rotated the pair inward, with only about a 15° outward cant. The image stabilized and offered a great deal of "air" and internal space.

I have been fortunate to listen to live broadcasts (via microwave to the transmitter) of the New Orleans Symphony. The imaging and sense of "air" I get at home is quite similar to what I have experienced in the concert hall. With well-mixed LP or CD classical recordings, the imaging is superb, if not the "pin-point" imaging some lesser speakers offer due to upper midrange beaming. With rock material, featuring artificially-added echo, the results range from the sublime to the ridiculous. In many cases, the image seems to expand beyond the walls.

If the CVA configuration has a disadvantage, the very wide (120° or so) dispersion gets the sidewall reflection into the act more than necessary. In a listening room such as mine, with close, hard-surfaced walls, this tends to dilute the image. As with many large speakers with numerous drivers, the CVAs demand some distance between themselves and the listener, lest the listener hear the individual drivers rather than the array itself. An interesting phenomenon occurs when I stand midway

ProvenPair PRECISION DRIVE UNITS

Madisound Speakers in cooperation with France Filieres, manufacturers of Focal Loudspeakers, is proud to present Focal ProvenPair drive units.



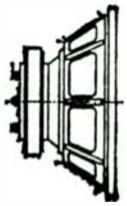
Example of 5N 302 ProvenPair Response documents:

ProvenPair units for the first time offer drive units that are carefully matched and shipped with individual response information. This can insure that the serious speaker constructor can turn his design into reality. ProvenPair drivers offer the following:

1. Frequency response matched to within 1dB throughout the range of desired use (typically one octave above resonance to the frequency where the nominal impedance is doubled).
2. Units sold only in pairs.
3. Each unit is accompanied by a signed and dated individual response curve from France Filieres. Measurement system is by Bruel & Kjaer, including a BK 4133 microphone and a BK 2307 chart recorder.

Madisound is offering the ProvenPair option on several of Focal's midranges and woofers. It is our feeling that the high frequency units do not show significant unit to unit variation, but bass and midrange speaker construction involves many hand operations and parts from several subcontractors. The quality control advantages of ProvenPair design results in a very good value for the purchaser.

MODEL		PRICE per pair
5 N 302	**Neoflex midrange ProvenPair**	**$ 65**
5 N 402dB	**dual voice-coil 5″ woofer ProvenPair**	**68**
7 N 402dB	**dual voice-coil 6.5″ Neoflex ProvenPair**	**88**
7 N 303	**Lower mid 40mm flat wound coil, 93dB ProvenPair**	**130**
7 N 501	**Neoflex 6.5″ woofer; 40mm flat wound voice-coil ProvenPair**	**101**
8 N 511	**Neoflex 40mm flat wound voice-coil, cast frame, ProvenPair**	**106**
8 P 511	**Polypropylene 40mm flat wound voice-coil, cast frame, ProvenPair**	**110**
10 N 501	**Neoflex 10″ woofer, 40mm, cast frame, ProvenPair**	**122**



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FAST REPLY #IK20

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between my canted-in speakers. The effect is much like putting on a pair of headphones—some of the images appear to originate within my head!

I changed the pods' suspension method from the pylon and threaded rod arrangement to a spine which holds the pods with clamps from the rear. Lack of reflections from the former pylons, to my surprise, is indeed audible and welcome.

I thank Angel Rivera of Brooklyn, NY for pointing out that C_z of the crossover should be 156.8μF, not the 15.68 I had (mis)calculated earlier.

Scott Ellis
New Orleans, LA 70116

DRIVER QUERY

Thanks to Bob Bullock for his series of articles about Thiele-Small alignments, and his part in the preparation of the BOX-RESPONSE program. While the information in these articles is interesting and useful, I have a number of questions.

In Mr. Bullock's reply to Carmen Gitto (*SB* 4/86), he said when two drivers are paralleled and assumed to be one, the only Thiele-Small parameter altered is V_{AS}. Please explain why. I have outlined my thoughts, such as they are, below:

$$Q_T = \frac{Q_E\,Q_M}{Q_E + Q_M}$$

Therefore, any change in Q_E or Q_M alters Q_T.

From Martin Collom's "High Performance Loudspeakers" (p.78):

$$Q_E = \frac{2\pi\, f_o\, M_T R_C}{B^2 l^2}$$

for two drivers in parallel, R_C will be half its single driver value, therefore, for Q_E to remain the same, either M_T or B^2l^2 alters.

What is M_T? In the formula concerning sealed box systems, M_{AC} was the total mass, including the diaphragm and adjacent air. Is M_T perhaps a version of total mass including port mass in a vented system?

Mr. Colloms uses 5.5 x $10^{-2}m^{-2}$ for M_T, the area of driver diaphragm. Assuming this is somehow correct, two drivers in parallel would have an M_T value twice that of a single driver. The changes in M_T and R_C cancel each other out. So far, so good. B^2l^2 doesn't change, which seems reasonable.

What happens with four drivers in series/parallel? R_C remains the same as one driver, but does M_T become four times as large, leading to a new Q_E four times the original?

Looking at:

$$Q_M = \frac{2\pi f_o M_T}{R_{MS}}$$

two drivers will produce twice the original M_T, but what of R_{MS} (suspension resistance)? I'm not sure what R_{MS} is, but my first inclination is to guess that two equal values in parallel will produce half the original value. Do two drivers produce twice the value of R_{MS}?

At this stage, I am completely confused. Maybe changes occur in both Q_E and Q_M which somehow cancel each other out, leading to the same Q_T no matter how many drivers are combined in whatever manner. Could Mr. Bullock clear this up for me?

My second question concerns adding series resistance to alter the Q value of a driver. For example, using a Focal 10N501 and the QB_3 alignments (*SB* 3/81, p. 21):
Q_T=.231, f_3=47.2Hz, V_B=37.9 1,
but Q_T=.31, f_3=33Hz, V_B=81.5 1.

To alter Q_T from .231 to .31 requires a series resistance of 2Ω:

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$$\left(R_s = \frac{Q_{Tnew}}{Q_{Told}} R_E - R_E\right)$$

I know many manufacturers use the resistance of series coils to obtain the Q_T they desire, and I wonder whether there is any reason, other than power loss in the resistor, not to do this? Is there a maximum series resistance that would be unwise to surpass? Obviously, doing so will reduce the output from the driver for a given applied voltage, but I presume if I have two drivers in parallel or four in series-parallel, the gains will offset the loss due to the added resistor.

I am also interested in the relationship between the reference efficiency of a speaker, the sensitivity measured by the manufacturer, and the maximum loudness capability as predicted by BOXRESPONSE. I made this table to compare reference efficiency and 2.83V sensitivity of various speakers:

Vifa M30WO	1.270	94dB
Dalesford D300	1.083	90dB
Vifa P25WO	.847	90dB
Focal 10N501	.837	92dB
Peerless TA305F	.825	89.5dB
Dynaudio 30W54	.660	96dB
Dalesford D100/250	.630	88dB

Except for the Dynaudio 30W54, the order of reference efficiencies and sensitivities make some sense. Can Mr. Bullock give a brief and reasonably simple explanation of why a driver, such as the Focal 10N501, won't produce the full output implied by its sensitivity rating (i.e., 1W=92dB, 10W=102dB, 100W=112dB)? For a number of drivers, BOXRESPONSE shows the reference output to be 3-6dB lower than expected. Again, the effect of creating one driver from two or four similar ones is interesting:

$$N_o = \frac{2\pi^2 f_o{}^3 V_{AS}}{C^3\, Q_{ES}}.$$

If Q_{ES} is unaffected by this process, then only doubling or quadrupling V_{AS} will affect N_o.

Finally, when using BOXRESPONSE for a driver consisting of two 8Ω drivers, is W multiplied by 2 or 4 (two drivers and halved resistance = 4 times as many watts)?

Michael Anderson
Lorne, Victoria Australia

Contributing Editor Bob Bullock replies:

I have never been satisfied with the bass sound quality in my systems when the driver Q is raised by a significant amount of series resistance (more than 1Ω), whether it be by adding a series resistor between filter and driver, or by using a high resistance coil in a filter series arm. Of the two techniques, I think the added resistor route is safer for the home builder, although it might be worthwhile for you to try both and decide whether you hear a difference.

In theory, putting resistance in a filter coil changes the filter's response shape over what it would be without resistance, and the extent of the change increases with the amount of resistance added. A maximum tolerable amount is necessarily a subjective decision. Manufacturers can measure its consequences for overall system response and limit the added resistance to keep within given design specifications. Most home builders must rely on their ears to decide when there is too much.

I do not see how you get BOXRESPONSE to predict maximum SPLs which are 3–6dB less than maximum SPL = nominal sensitivity + 10 log (power). I ran the Vifa P25WO, Focal, Peerless, and Dynaudio drivers through BOXRESPONSE, and the greatest disagreement I found was 1.9dB for the Peerless. It is not clear from my spec sheet at what voltage they measured sensitivity. This could account for the entire discrepancy. I do not have specs for the other drivers, so I could not check them. My Dynaudio data, by the way, shows a 92dB sensitivity. Send me some samples of your calculations, and maybe I'll be able to reconcile our different findings. It is not always clear what sensitivity means. I highly recommend you read Joe D'Appolito's letter on the difference between power and voltage sensitivity in SB 2/85, p. 48.

For a given driver, I use voltage sensitivity defined by sensitivity@ 1m:

$$e_v = 94 + 20 \times \log\left(\frac{r_{ho} O S_d B_{le}}{(2 pi M_{ms} R_e)}\right)$$

with $e = (R_e)$, which gives 1W across $R_e \Omega$. I then interpret this as @1m, 1W. Manufacturers' specs generally set $e = 2.83V$ and call it @1m, 1W (i.e., they use $R_e = 8$ for all nominal 8Ω drivers). My values usually agree with theirs.

When comparing compound driver sensitivities, I keep e the same in all cases. Then, two drivers in series have the same sensitivity as one, but in parallel they have 6dB greater sensitivity. Note these are no longer necessarily comparable as 1m, 1W sensitivities.

I don't usually think in terms of power, so I rarely use efficiency eta0. BOXRESPONSE calculations are all done in terms of voltage and converted to power values by $P = e \times e/R_e$. The maximum loudness capability in BOXRESPONSE is calculated from the above sensitivity formula, but with e being the maximum voltage that can be applied to the system without exceeding its thermal or excursion limits.

To answer your question about Mr. Collom's notation, M_T on p. 78 is exactly the same as M_{AC} on p. 71. In the example on p. 78, $M_T = M_D$ + Airload = .045 + .01 = .055, which happens to equal S_D only by accident.

Now, about multiple driver configurations. Take the following statements as axioms when two identical drivers are replaced with one hypothetical equivalent driver:

1. Voice coil resistance R_e doubles (halves) for series (parallel) connection.
2. Force factor Bl doubles (is unchanged) for series (parallel) connection.
3. Mechanical mass M_{ms} (compliance C_{ms}, resistance R_{ms}) doubles (halves, doubles).
4. Piston area S_d doubles.
5. Linear excursion limit x_{max} does not change.
6. Thermal power limit P_e doubles (answering your last question).

With these, you can verify my assertion to Mr. Gitto. For example, driver radian resonant frequency w_s ($= 2 \times p_i \times f_s$) is found from:

$$w_s = \frac{1}{\sqrt{(M_{ms} \times C_{ms})}}$$

$$w_s^2 = \frac{1}{\sqrt{(M_{ms}^2 \times C_{ms}^2)}}$$

$$= \frac{1}{\sqrt{\left(2 \times M_{ms} \times \frac{(C_{ms})}{2}\right)}}$$

$$= \frac{1}{\sqrt{(M_{ms} \times C_{ms})}} = w_s$$

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where the two subscripts refer to the equivalent driver parameters.

You can also use these axioms to deal with four drivers. For example, the single equivalent driver would have four times the M_{ms} and R_{ms}, and one-fourth the C_{ms} of the single driver. In series/parallel (two each in series, the resulting pairs in parallel?), Bl would double, R_e would remain the same.

IN PRAISE OF ACTIVE CROSSOVERS

The crossover network for multielement speakers is at least as important as the drivers in producing realistic quality sound. Thus, the crossover network should occupy a proportionate amount of the speaker builder's time in planning, designing, applying, building, testing and experimenting with all types of crossover networks.

Admittedly, I am prejudiced (with good reason) in favor of active (electronic) crossover networks. My reasons for favoring active crossovers are as follows:

- The changing impedance of the drivers does not affect the performance of active crossovers.
- The direct coupling of amplifiers results in improved damping and better transient response.
- The sound-pressure level (SPL) matching of the drivers is easily and accurately accomplished.
- Contouring voltage levels to compensate for deviations in driver frequency response is easily accomplished within the active crossover network.
- Amplifier intermodulation distortion is reduced.

The active crossover network is a high-impedance, low-power device inserted between the preamplifier, as shown in *Fig. 1.* The power amplifiers are buffers between the active crossover network and the drivers for impedance matching. The changes in the driver coil's reactive nature and total impedance caused by changes in frequency have no effect on the crossover voltage output and phase angle or on the power amplifier output.

For comparison, a passive crossover network is shown in *Fig. 2.* When properly designed, a passive crossover includes the real and imaginary parts of driver impedance as part of the crossover network circuit. A simple first-order, high-pass (HP) filter (often used as a tweeter crossover) is shown without any load in *Fig. 3,* along with phase and voltage relations. The same HP filter is shown in *Fig. 4* with a connected load (R_L) to simulate the tweeter coil.

Driver Coil Parameters

You can see that the driver coil parameters must be included as part of the crossover circuit, since no-load and load conditions change drastically. In the real world, the actual speaker impedance will not be pure resistance as represented by R_L. It will be a complex impedance (Z) represented by the following equation:

$$Z = R + jX_C - jX_L$$

It will be represented in magnitude by the formula

$$Z = \sqrt{R^2 + X_C^2 + X_L^2}$$

where Z is the total impedance, X_C is the capacitive reactance, X_L is the inductive reactance, and R is the resistive component. Note that R is not the ohmic resistance as measured statically by an ohmmeter.

When excited by an AC voltage, the driver coil is in constant motion, vibrating at the AC frequency rate. This continuous cyclic motion of the driver coil

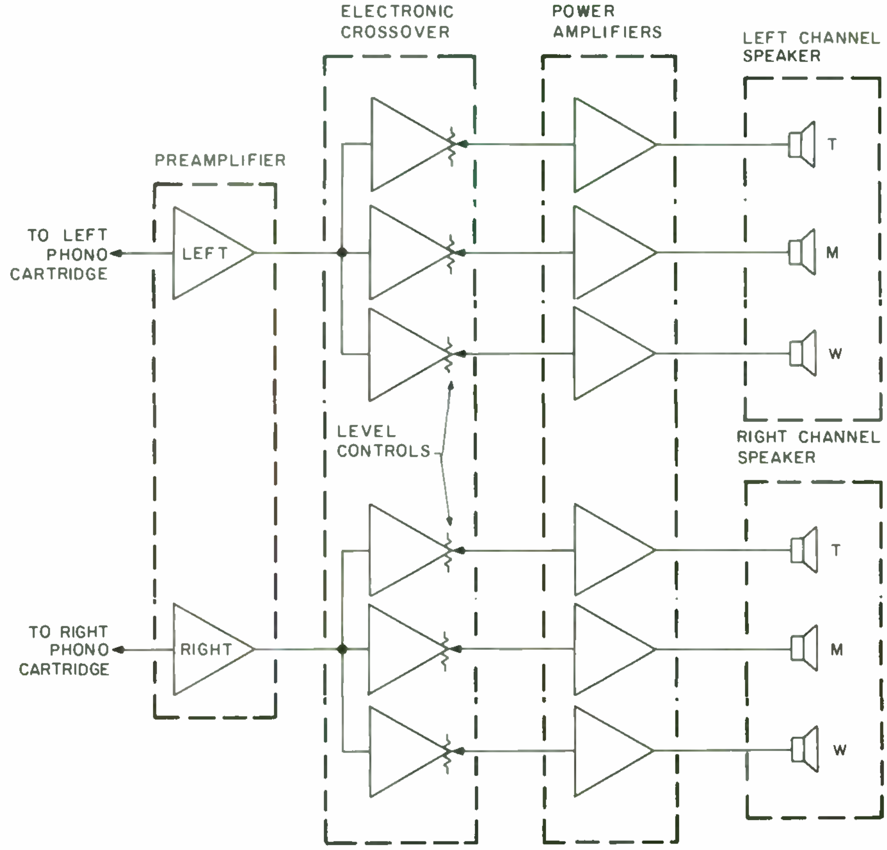

FIGURE 1: Complete stereo system with an active (electronic) crossover.

within its magnetic field generates a separate voltage called a counter emf (electromotive force), which is in opposition to the applied voltage. This counter emf alters the static values of R, L and C. Thus, R, L and C of the driver coil are not only complex variables, they are also dynamic variables changing with applied frequency. The values of R, X_C and X_L for the driver coil used in the crossover network calculations are those at the crossover frequency, f_C. As you can see, the passive crossover is designed to satisfy only one frequency for the driver parameters R, L and C.

Designing a proper passive crossover network is further complicated by the design of the midrange portion of the crossover. The midrange crossover is made up of HP and LP (low-pass) filter sections connected in series. These HP and LP sections cannot be designed as independent filters because they react upon each other. The input impedance of the last section becomes the load impedance of the first section.

It should now be obvious that the driver coil impedance has a definite impact on the design and performance of a passive crossover network. This is not the case with an active crossover network.

Active crossovers allow the direct coupling of power amplifiers to driver coils, which creates the optimum condition of a very-low-impedance connection between amplifier and driver (assuming, of course, that something better than #18 lamp cord is used). The very low impedance maintains a high damping factor (DF) for the driver coil. Most solid-state power amplifiers have high DFs of 20 or more, with some as high as 200. (I believe the importance of extremely high DFs is overstated. The driver coil provides its own damping effect by its internal DC coil resistance, which for a normal 8Ω driver might be 5 to 7Ω. The difference between controlling cone motion with a DF of 20 and one of 200 is inconsequential.)

In passive crossovers, however, a DC resistance of 1 to 2Ω is common in inductance coils. When these are in series with the driver coil in a LP filter section, a deterioration in driver damping can result. All the connection points and extra wiring in a passive crossover network contribute to a higher DC resistance and a lower DF.

Even though an extremely high DF is less important in damping the driver cone's motion, a high DF goes hand in hand with excellent AC voltage regulation. This is very important in transient response. The tight direct coupling of amplifier to driver in an active crossover system will maintain this excellent AC voltage regulation right up to the driver terminals for superb transient response. In contrast, inserting a passive crossover between the amplifier and drivers introduces additional resistance and impedance and results in a degradation of transient response.

One of the requirements for good stereo effect is driver matching. ("Matching" means the matching of SPL at all useful frequencies.) Matching between the left and right channels and between the drivers in each channel is required. Most high-volume manufacturers of speaker drivers do not achieve a consistent SPL between drivers of the same type, even though they appear to be identical physically. The cost of reasonable consistency could be exorbitant. Most commercially produced loudspeakers use passive crossovers, and producers of high-quality speaker systems match drivers by "test and select." These manufacturers purchase or produce drivers in sufficient quantity that the test-and-select procedure is economically practical.

The lonely speaker builder who wants to build one high-quality pair of speakers must either find a supplier who will provide honestly matched pairs of drivers, modify the passive crossover with great difficulty and dubious results, or use an active crossover network.

You can manipulate the passive crossover circuitry to achieve a contoured response. Adding padded level controls also helps you obtain matched SPLs. All this manipulation adds to the complexity of the passive crossover network, however, and causes further deterioration of DF and transient response.

With active crossover networks, you can achieve matching drivers by using level controls and circuit contouring for frequency response. This complements and matches the SPLs between driver pairs with no effect on DF or transient response.

Intermodulation Distortion

Intermodulation (IM) distortion is the mixing of two different frequencies to produce two more frequencies (sum and difference), which are not harmonically related (a multiple) to the fundamental. Since the distortion products are not harmonically related, they can be several times more irritating than simple harmonic distortion.

Active crossovers have multiple power amplifiers—one for each driver. In a typical three-way system, the woofer and its amplifier operate at 200Hz and below, the midrange from 200 to 3,600Hz and the tweeter at 3,600Hz and above. Since the woofer amplifier and driver do not see a frequency of 1,000Hz, there can be no IM products from 1,000 and 100Hz. In the same way, IM products cannot be formed from frequencies that are not present in the woofer, midrange or tweeter amplifier. Of course, IM products in the woofer amplifier can result from frequencies such as 50 and 200Hz. The scope of IM distortion products has, however, been cut to about one-third in a three-way system. From listening, it also seems that the IM distortion is less irritating when the span between frequencies creating IM is shorter.

A passive crossover has only one amplifier, so all audio frequencies are present and capable of producing IM distortion products. It is, therefore, apparent that systems using passive crossovers have more inherent problems with IM distortion products.

If you agree that an active crossover network is the only way to go, you might be wondering how many speaker builders can afford three big 200W power amplifiers. The rule of thumb has been that a biamped or triamped system should use identical amplifiers. Since the woofer is the most power hungry, the minimum power rating required for it becomes the norm for the other two.

That rule of thumb is hogwash. It came about for two reasons. First, it allows manufacturers and retailers to sell more big amplifiers. Second, it stems from a real concern about differences in phase, linearity and power sensitivity between

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dissimilar amplifiers. We need concern ourselves only with the second reason.

All three amplifiers must track on phase angle through the audible frequency range. They must be linear in regard to voltage input and frequency. Most quality amplifiers will comply reasonably well with these requirements. Occasionally, you will find an amplifier that has been designed with a reversal in the circuit so that the output is 180 degrees out of phase with the input. In this case, you need only reverse the driver leads.

Amplifier power sensitivity is watts output per volts input. With dissimilar amplifiers, it is unlikely that you will find equal power sensitivity. You can, however, adjust the level controls in the active crossover network's output to produce equal SPLs for the same voltage input to crossover.

The curve in *Fig. 5* represents peak music power versus frequency. Using this curve, you can choose the woofer amplifier rating based on either the largest amplifier you have or plan to buy. If you buy one, its rating should be based on the maximum peak watts rating of the woofer. You can then select the lowest

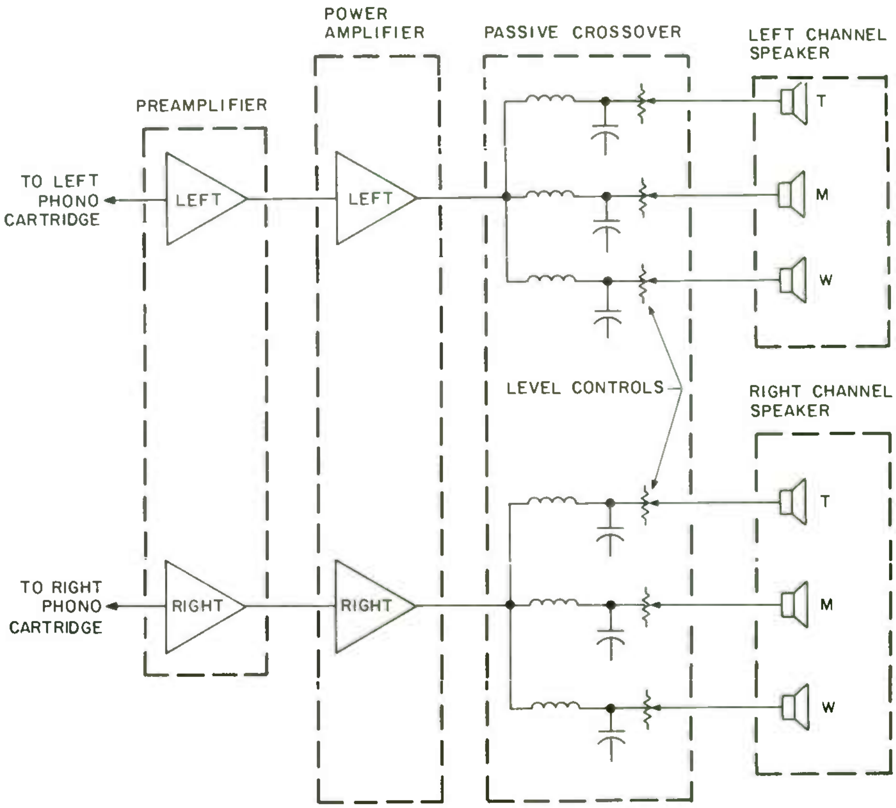

FIGURE 2: Complete stereo system with a passive crossover.

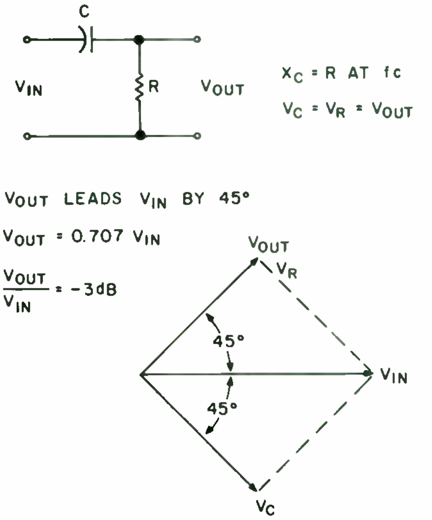

FIGURE 3: First-order, high-pass filter with no load.

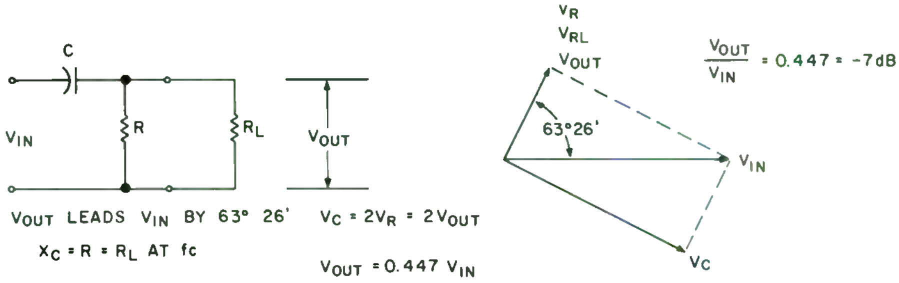

FIGURE 4: First-order, high-pass filter with load R_L.

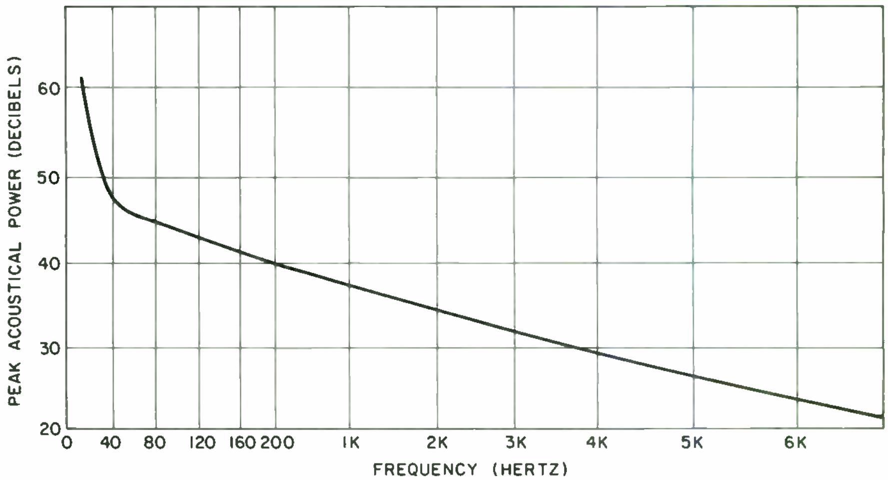

FIGURE 5: Typical peak acoustical power distribution in music.

operating frequency for the woofer. If you chose 30Hz, this would be 50dB on the curve *(Fig. 5)*, and your woofer amplifier could be selected as 200W. The crossover to midrange would be at 200Hz. Peak acoustic power would then be down to 40dB, which is ten times less. The midrange amplifier rating should be a minimum of 20W. The crossover to the tweeter would be at 3,600Hz, where the

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curve is at 30dB, which is again ten times less than the midrange or 2W minimum for the tweeter. Note that these ratings are minimum values.

In my system, I use 200W, 40W and 15W amplifiers for the woofer, midrange and tweeter. The midrange and tweeter amplifiers never clip. The woofer clips occasionally, but my only resort would be to use something like 1,000W per channel.

In spite of all the problems with passive crossover networks, some manufacturers and audiophile speaker builders can design and build excellent-sounding speaker systems using passive crossovers. Still, active electronic crossovers do the job easier and better than the best passive crossovers.

I would like to hear from all you speaker builders and audiophiles. Can you refute what I have said and remove a few nails from the coffin of the passive crossover? Or do you have a few more nails to add?

Bob Ballard
Johnson City, TN 37601

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Allows adjusting the input capacitance of the phono input of every preamp and receiver with low loss Polystyrene Capacitors

DBP-6MC RESISTIVE LOADING KIT $36.95
Allows adjusting load resistance from 5 to 200 Ohms for moving coil cartridges Gold plated phono plugs in both kits

DBP-9AU BANANA PLUGS Eight gold plated, solderless ..$16.50
DBP-9H BANANA HANDLES Four red, four black $5.50
DBP-9P GOLD PLATED DUAL BANANA PLUGS 2 pk .. $17.95
DBP-9J GOLD PLATED DUAL BANANA JACKS 2 pk . $17.95
DBP-10 PHONO ALIGNMENT PROTRACTOR 24.95
Allows adjusting the lateral tracking error of a mounted cartridge to within ¼ of one degree Non-technical instructions & case included

DBP-12 AUDIO CABLE 10 meter (33 ft.) $70.00
Low capacitance (400pF) stereo interconnect cable, terminated with rugged gold plated phono connectors Custom lengths available

DBP-12X AUDIO CABLE 10 meter (33 ft.) $95.00
As above but terminated with premium DBP-13PX plugs

DBP-13J GOLD PLATED PHONO JACKS (¼") 8 pk $16.50
DBP-13JR GOLD PLATED PHONO JACKS (3/8") 8 pk . $21.95
DBP-13P GOLD PLATED PHONO PLUGS 8 pk $9.95
DBP-13PM GOLD PLATED PHONO PLUGS, 4 red, 4 blk ..$21.95
DBP-13PX GOLD PLATED PHONO PLUGS 2 pk $16.50
DBP-14 GOLD PLATED SPADE LUGS 8 pk $5.95
DBP-15 GOLD PLATED "Y" ADAPTORS 2 pk $12.95
DBP-16 12dB INPUT ATTENUATORS $12.95
DBP-CK CRAMOLIN AUDIO KIT contact treatment ...$17.50
DBP-SC SOUTHER RECORD CLAMP $10.00
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THE VANCOUVER AUDIO SOCIETY publishes a bimonthly newsletter with technical information, humor and items of interest to those who share our disease. We have 40 members and meet monthly. Six newsletters per year. Call (604) 299-4623 or write Dan Fraser, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be on your mailing list.

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues include monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-Bing equipment. New members welcome. Contact: Bill Donnally (201) 334-9412; or Bob Young, 116 Cleveland Ave., Colonia, NJ 07067, (201) 381-6269.

THE COLORADO AUDIO SOCIETY is a group of audio enthusiasts dedicated to the pursuit of music and audiophile arts in the Rocky Mountain region. We offer a comprehensive annual journal, five bimonthly newsletters, plus participation in meetings and lectures. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212, or call Art Tedeschi, (303) 477-5223.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, AB listening tests, equipment clinics, recording studio visits, annual picnic and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 1464, Berkley, MI 48072-0464.

Classified Advertising

AUDIOPHILES IN CENTRAL PENNSYLVANIA (also eastern Pennsylvania and Delaware): Interested in forming a serious audio organization? Contact Steve Gray, 625F Willow St., Highspire, PA 17034 or phone (717) 939-4815.

FOR SALE

Beautiful Linkwitz three box system, ¾″ mahogany with black walnut inlay, satellite stands, crossover as is, $300; Audio Concepts unstained oak Quartz J speakers with bone grilles, about one year new, $350; Hafler DH-200, some mods, $175. You ship. David Tannenbaum, 2206 W. Porter Ave., Fullerton, CA 92633, (714) 879-4269 evenings.

Mission 7000R CD Player, $500 or best offer; Linn Ittok, $250; Proton electronic crossover for car, $75; EV Sentry SEQ, $75; Al CST-80 moving-coil transformer, $65; DeCoursey 2kHz, 18dB crossover, $140; Interlink phono cables, $25; RS dB meter, $25; RH 1200 horns, $50; University coax speaker, $35; EV Aristocrat box, free pick up. Steve (203) 397-4965.

Grace 747 tone arm with Grace F9E Ruby cartridge. Arm excellent condition. Cartridge needs stylus (will accept Ruby or regular stylus). Original cost $300 each. Both for $175 or best offer. Gary Galo, 72 Waverly St., Potsdam, NY 13676, (315) 265-4268.

Two Macintosh moving-coil 75 mono tube amplifiers each with an extra set of new KT88 output tubes. All for $650 plus shipping. Louis Dyer, 1584 W. Holland, Fresno CA 93705, (209) 221-8592.

ReVox A-77 tape recorder, less than 2 hours use, $650. Barney Rigney, 412 West Broadway, Glendale, CA 91210, (818) 956-3397: 9-4:30 PDT, after 6 PM: (213) 226-0410.

Citation II,III, IV all in excellent, checked out condition. Extra output tubes for II. Best offer. Call F.H. Blanding (201) 276-5132.

Hewlett-Packard oscilloscope 120B, excellent condition, $150; Audio Research tube crossover EC2, $150; Audio Research D100 amplifier, $575. (415) 851-2779.

Precision Fidelity C-8 tube preamp, $395; Luxman MQ68C 30W tube amp, extra tubes, mint, $450; Symmetry ACS-2A crossover (factory sealed, list $1200), $450; Six 1m pairs Monster Interlink Reference, $40 pair; 40 new Telefunken ECC 8035, legendary "Premium" substitute for 12AX7A, $18 each, $32 MP. L.P. Wilkins, 550 Baker #6, San Francisco, CA 91117, (415) 931-2783.

Audio Concepts Model M (compact monitor) kit: MW160 high-power 6″ woofers, Dynaudio D28af tweeters, high quality crossover, enclosure plans, 2 years old, light use, $135. Ralph Gonzalez, 4612 Chester Ave., Philadelphia, PA 19143, (215) 387-0860.

DeCoursey modules 2kHz, 18dB, $40 per pair; 855-SSB, $22 per pair; EV SEQ bass equalizer, $75; RH 1200Hz horns $50 per pair; Proton 270 auto crossover $65; Linn Ittok, $275; Monster phono DIN cables, $25. All mint. Call Steve (203) 397-4965.

IQS 401-S FFT response analyzer. For use with Apple II computer. Latest software versions. Will measure many speaker response parameters, $600; Tektronix 2215 60mHz oscilloscope, calibrated to NBS traceability, $900. Have Apple II for sale also. Chris Lovett, 100 Northill Dr. #30, Brisbane, CA 94005, (415) 468-2990 (office). (415) 223-5183 (home).

ARC SP3-1A regulator boards, fully stuffed, $20 each or $60 for four; Moscode 300, stock, $550; Sherwood 6040 CP amp., dual mono, MOSFET, $200; Hitachi HCA8500 MKII preamp, $200, MFSL "Dark Side," sealed, $25. I pay shipping. Roger Artman, 18265 Lasso Loop, Penn Valley, CA 95946, (916) 432-4124.

Heil air motion transformers, private collection, great Heils, disco horns, power rings, performance classics, tempests, more. Some in original factory sealed boxes. Spare diaphragms available, $25 - $175 each. Rick (916) 988-0673.

Warbler oscillator, for checking speaker performance (*Audio Amateur*, 1/79, p.22), Old Colony KK-33 kit with reprint (unassembled), $30. Free shipping in USA. Robert Agee, 241 E. Fiesta Green, Port Hueneme, CA 93041, (805) 984-6859 after 5.

Onkyo T-9090 tuner, $375; Technics RS-M253X 3-HD cassette deck, B-C-DBX, under 200 hours, list $450, sell $220; ADC SS-315X 10-band equalizer, list $380, sell $140; Akai GX-747 DBX reel-to-reel, new with warranty, half-price. John (904) 641-5830 evenings.

Jordan 50 MM with ferro fluid, $125 per pair, Dynaudio, Scan-Speak, Peerless, modified Panasonic EA200. All at exceptional prices. Call or write for more information: Bob Pickard, 1720 Lilac Dr., Walnut Creek, CA 94595, (415) 930-9104.

Pair Altec 1570B amps, 165W, 811A tubes, Peerless transformers, $200 or best offer; 2 mono Williamson type power amps, Acrosound TO-300 transformers, $100 or best offer; Scott 299 tube integrated amp, $40; Fisher X100B tube integrated amp rel-caps, $90; 100B FM tuner $40, both with original packing; Heath mono tube FM tuners AR-11, $20; AR-30, $25; pair of AR2 speakers, $45; Thorens TD166 MKII, 3 months old, $200; pair SEAS 332BX-DD best 13″ woofers, little use, $250; with teak cabinets, $350. All plus shipping. Bob, (201) 968-7783, evenings (201) 647-0194.

WANTED

Riders Perpetual Troubleshooters Manual for pre WWII schematics and parts for radios, jukeboxes, PA systems. Beginning of radio to 1946. Brian Dettling, 120 South College St., #2, Akron, OH 44304, (216) 434-1037.

Experienced craftsman to repair Strathearn ribbon diaphragms. Call (212) 242-1996 and leave message.

MC-225 tube cages. S. Hluchan, 21 Westward Rd., Woodbridge, CT 06525, (203) 397-4965.

LUX/NEC 50C-AIO and 6240G tubes for LUX MQ 68C, new or used; Superphon Revelation dual mono preamp; Luxman L-11 integrated amp; new or used KT88 and KT77 tubes. L.P. Wilkins, 550 Baker #6, San Francisco, CA 94117, (415) 931-2783.

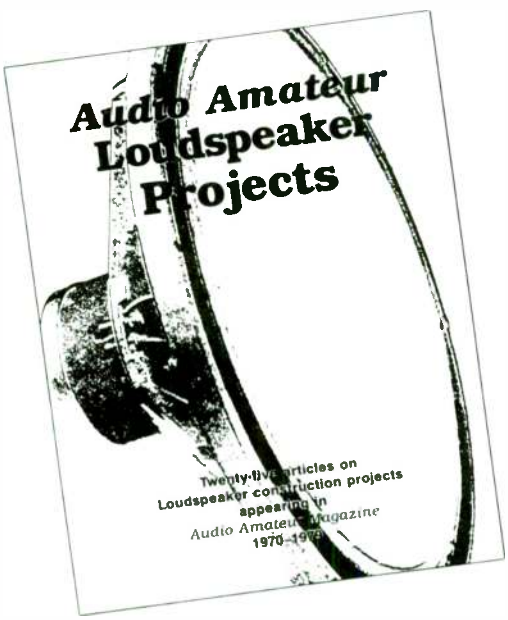
Audio/Acoustical instrumentation B & K, HP, GR, etc. Particularly want B & K ½″ condenser mike. G. Whelpley, 4305 W. Saturn Way, Chandler, AZ 85226.

Schematic and/or repair advice for KLH 9s, fee offered. Jack Rumora, 626 Tall Trees Dr., Ashtabula, OH 44004.

Advertising Index

Audio Amateur Loudspeaker Projects

Twenty-five articles on Loudspeaker construction projects appearing in *Audio Amateur Magazine* 1970–1979



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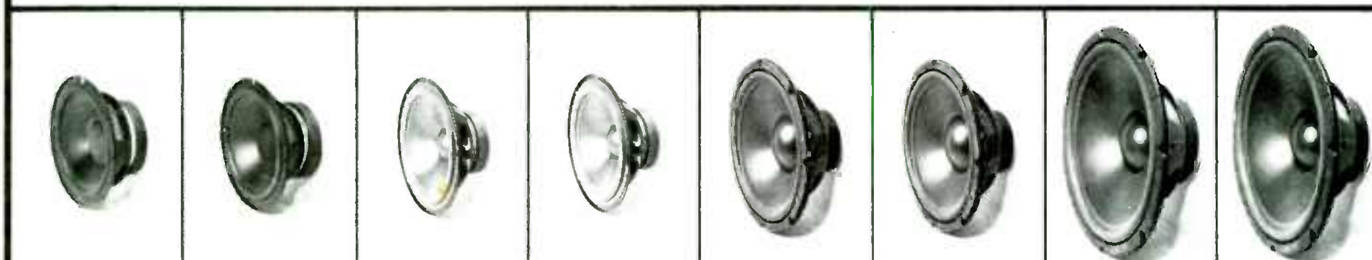

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