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THE LOUDSPEAKER JOURNAL

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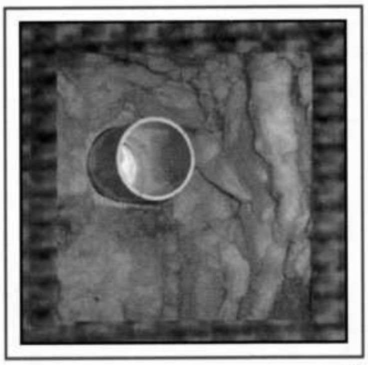

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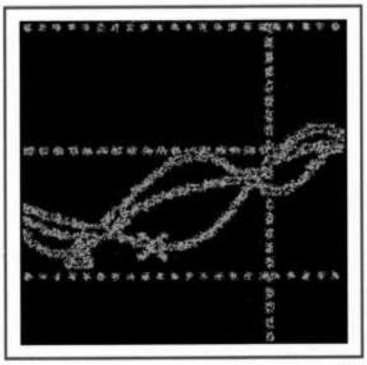

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A SIMPLE PHASE TESTER

By Andy Lewis

Unfortunately, in order to reproduce the entire audio spectrum with moving-coil drivers, it is generally necessary to use several drive units of different sizes. The human ear simply responds to a wider range of frequencies than any single driver is capable of reproducing. Consequently, you are forced to rely on multidriver systems, using woofers, tweeters, and sometimes one or more midranges as well.

PHASE COHERENCE: A LITTLE BACKGROUND

As early as 1935, listening revealed that when the drivers in a system were separated by a distance in space, they tended to be perceived as separate sources of sound, rather than as parts of a whole.[1] Experimentation then demonstrated that when the drivers were aligned properly, the perception changed to one of a single, full-range source.

Ever since, designers have disagreed about both the threshold of audibility of this problem and the best way to overcome its ill effects in their designs. Most agree, however, that phase distortion is audible and that some degree of phase linearity is an important design goal in any multiway speaker system. If you accept this notion, a design question is how to align your drivers to most effectively compensate for their differences in phase.

Some manufacturers have concentrated on their systems' capacity to be phase coherent, or in phase with themselves. They have also taken different approaches to solving the phase-response problem. You may remember the systems from Rectilinear Research in the '60s and early '70s. Notable was a bookshelf-sized 10″ three-way system that Rectilinear claimed was able to pass a square wave virtually undistorted.

It accomplished its touted phase linearity in a very clever way: the design avoided midband-crossover phase irregularities by simply avoiding a midband crossover! The crossover frequency from the woofer to the midrange was very low, and from the midrange to the tweeter very high. Interestingly, a sample I examined achieved the low crossover from woofer to midrange by using an enormous coil in series with the woofer.

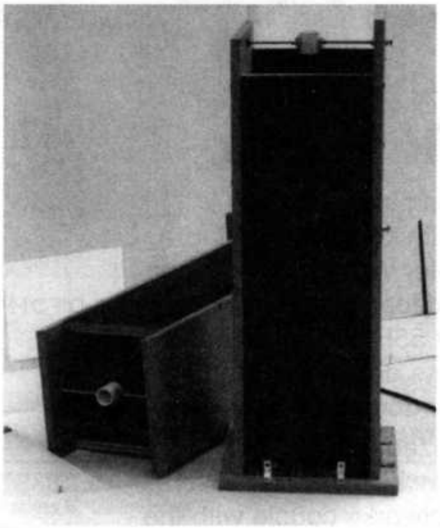

PHOTO 1: Test jigs.

Unfortunately, the DC resistance of this coil undermined the outstanding damping characteristics of the Rectilinear woofer, adversely affecting the bass. Although it wasn't rocket science, it had a fundamental correctness of approach. The design was important because of its early attention to phase coherence in a home-speaker system.

PHASE-LINK SYSTEMS

In the '70s, phase alignment really came into its own. Another interesting approach was the "phase link" system introduced by Bang & Olufsen. The designers, recognizing that precise second-order filters would result in an out-of-phase condition at the crossover point, inserted a driver with a very narrow pass band to "fill the hole" exactly at that point.

Some of these systems had, to my ears, a remarkably smooth and uncolored midrange. Such a system would necessarily be a little top-heavy in terms of crossover cost, and although this concept never sparked widespread imitation, the design was significant in that it acknowledged the unruly nature of crossovers with respect to phase.

Also introduced in the early '70s was the celebrated Dahlquist DQ-10. This important design accomplished phase coherence through the use of "staggered" drivers. Rather than mounting all the drivers on a common baffle, as was then customary, Dahlquist chose to displace the drive units with respect to each other on the horizontal axis.

The idea was to physically compensate for the drivers' inherent time differences by altering the positions of the individual sources. When carefully positioned, the drivers would become "time-aligned®" and behave more as a unit.

This concept has withstood the test of time. Not only are a great many DQ-10 systems still around 20-odd years later, but legions of hobbyists and professionals alike have endorsed the concept of staggered drivers through their imitation of Dahlquist's principle.

THE NEED FOR PHASE TESTING

Everyone has seen the "stairstep" enclosures used to physically "time-align" the speakers in a multidriver system (*Fig. 1*). Most of these, of course, owe a debt to Dahlquist.

While they might look strange at first, the concept is simple. Each driver, when considered as a source of sound, has an apparent exact location in space, the point on the speaker's axis where the sound wave is exactly in phase with the input voltage. This apparent location is sometimes referred to as the drive unit's "acoustic center."

In this illustration, assume that the

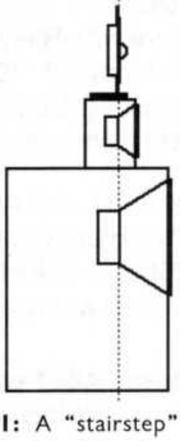

FIGURE 1: A "stairstep" time-aligned system.

acoustic centers of the three drivers are in the vicinities of their voice coils. The enclosure then is built in such a way that the drivers are physically offset by the precise amounts necessary to "time align" the system.

The assumption that a driver's acoustic center is located approximately where the voice coil meets the cone or dome is sometimes considered the default rule of thumb among designers when empirical data is unavailable.[2] Without arguing the merits of this assumption, the inherent correctness of staggering the drivers seems obvious. But wouldn't it be nice if you could perform a simple test on your drive units that would allow you to stagger your enclosures, if necessary, with confidence that it was doing more good than harm?

Little has been offered to help the low-budget designer in this area, and designing systems to be in-phase with themselves has generally been a matter of guesswork and reliance on assumptions. While sophisticated equipment may be available to provide phase-alignment information, hobbyist builders have been left in the cold, aware that phase alignment is a design consideration worthy of attention, but ill equipped to measure the effects and precisely compensate for them.

I have devised a simple and inexpensive gadget, the phase tester, to measure phase differences between drivers. I will describe my method of testing for phase alignment to show how you can make your own phase measurements as you design your systems.

REINFORCEMENT AND CANCELLATION

All of you have probably had the experience of accidentally connecting your speakers out of phase with each other. In most cases, the resulting loss of bass is dramatic. This is the result of wave cancellation. When two sources a short distance from one another produce the same signal at the same amplitude, but 180° out-of-phase, they work against each other. One source produces a positive wavefront, while the other produces a negative one. As a result, the sound pressure is reduced to zero in an ideal case, and often to near zero in the real world.

This is the nature of wave activity—to add and cancel with respect to phase. It is a source of phase problems and a reason you must consider phase accuracy in the first place. Ironically, however, this very property can become a tool for measuring the apparent distance between two drivers mounted on the same plane.

It also makes possible my method of phase-testing. The cancellation and reinforcement of in-and-out-of-phase sine waves provides a "sonic ruler" that you can use to simply and precisely measure phase differences between drivers you wish to use together.

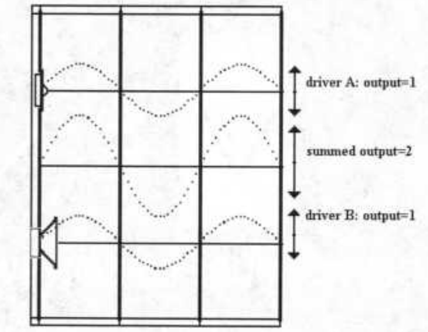

FIGURE 2: Wave reinforcement when drivers are in phase.

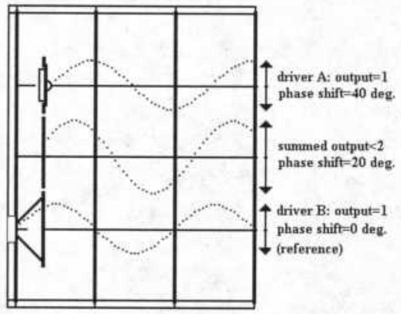

FIGURE 3: Incomplete cancellation and resulting phase shift.

HOW IT WORKS

To see how this is done, first consider the case of two drivers mounted on a common plane and electrically in phase with each other (*Fig. 2*). In this diagram, assume that the acoustic centers of the two drivers are aligned. The two separate sources are working together. The summed amplitude, represented by the middle sine curve, is double that of either individual driver's output, and the phase of the summed outputs is the same.

Now consider a second situation (*Fig. 3*) where the two drivers are mounted on a more-or-less common baffle. They are not time aligned, and the acoustic centers of the drivers are separated by a distance that is less than half a wavelength. Each driver produces the same sine-wave signal, individual outputs are identical, and the output of each is represented by its own sine curve.

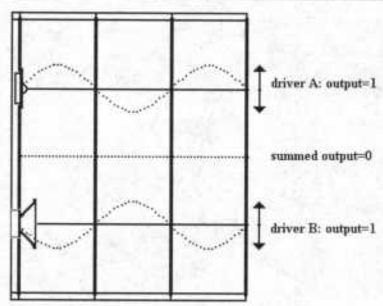

FIGURE 4: Complete cancellation using out-of-phase drivers.

In this case, the waves tend to cancel, but not completely, and the result is the summed waveform shown, which has a smaller amplitude than the summed output in *Fig. 2*. Note that when the outputs of the two drivers are added, the resulting curve is shifted in phase with respect to those of the individual drives. As effective displacement between the two sources becomes one-half wavelength, the summed output drops to zero (*Fig. 4*).

LONGITUDINAL VS. TRANSVERSE WAVES

I have represented sound waves in *Figs. 2–4* with mathematically generated sine curves. This is a common practice, and in fact is a good graphical representation of the varying amplitudes of the compressions and rarefactions that comprise a sound wave. But to represent a sound wave with a "wavy line" sometimes gives the false impression that there is some kind of side-to-side activity as a sound wave propagates. I have seen at least one commercial speaker-design how-to book "explain" that after leaving the vicinity of the driver, the in-and-out motion of the cone is somehow "transformed" into the side-to-side motion implied by the wavy lines often used to represent waves.

Transverse waves are those in which particle motion is as wavy lines would imply: in a direction perpendicular to the direction of propagation. A guitar string, for example, exhibits transverse-wave activity. As the wave travels up and down the length of the string, each individual tiny section of the string itself moves back and forth in a direction perpendicular to the length of the string. Electromagnetic waves are also transverse waves.

Sound, however, is an example of a longitudinal wave. The particle (air molecule) motion is parallel to the direction of propagation. So again, while the picture of the sine wave is representative of the mathematical model of the sound wave, it does not look anything like a sound wave would if it were visible.

CANCELLATION

Consider the same two drivers mounted in a time-aligned configuration, but wired out of phase with each other (*Fig. 4*). Under this obviously adverse condition, the two drivers work in perfect opposition to each other. In this example, when the summed outputs are represented by a curve, the result is a constant zero, indicative of complete cancellation.

This leads to the observation that when

outputs are equal, phase is reversed, and drivers are time-aligned, cancellation is complete. When you electrically reverse phase to one driver with the goal of complete cancellation in a sine-wave test, there are only two variables involved: relative output level and time alignment. This is the key to simple at-home phase testing.

But how can you arrange these special circumstances? Where would you put a microphone? How could you possibly move drivers with respect to each other, correct output levels to achieve cancellation, and accurately measure time-alignment data? And what about room reflections?

These questions tend to make the default method of aligning the voice coils seem attractive by comparison. But if you could adjust the physical positioning of drivers to duplicate these special conditions, it might be possible to measure time-alignment data accurately.

SIMPLE SOLUTIONS

Several adjustments occurred to me. First, by acoustically separating the speakers from each other, I could avoid interference and room reflections. Second, that changing the precise position of a microphone is much easier than moving a speaker. And, finally, that using a separate microphone on each speaker might enable me to control outputs precisely enough to imitate the ideal case represented in *Fig. 4*. Starting to make sense? The idea is really quite simple.

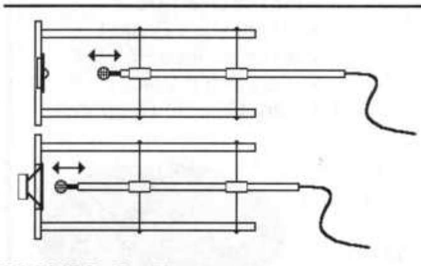

FIGURE 5: Phase-testing jigs. Note the PVC-tube guides supported by the eight carriage bolts.

TABLE 1

PARTS LIST—JIGS

QTY	ITEM
4	plywood or particleboard, 27″ × 9 1/8″ × ¾″ (short sides)
4	plywood or particleboard, 30″ × 10 5/8″ × ¾″ (long sides)
2	plywood or particleboard, 15″ × 15″ × ¾″ (end pieces)
8	4½″ × ¼″ carriage bolts
8	¼″ nuts
8	¼″ washers
8	¼″ T-nuts
16	1″ L-brackets
32	#8 × 5/8″ wood screws
2	lengths 1″ steel tube 36″ long "EZ Pull 1″ EMT"
4	1½″ lengths 1″ PVC tube (guides), 1″ PVC 1120 125 PSI irrigation pipe
2	matching microphones (Radio Shack #270-092 or similar)

The phase tester is also very simple. It consists of two large wooden jigs designed to separate a speaker and a microphone by a precise distance, and a flexible electronic system that connects everything together to make it work. There is nothing complicated or expensive, assuming you own an accurate sine-wave generator, a stereo, and a voltmeter.

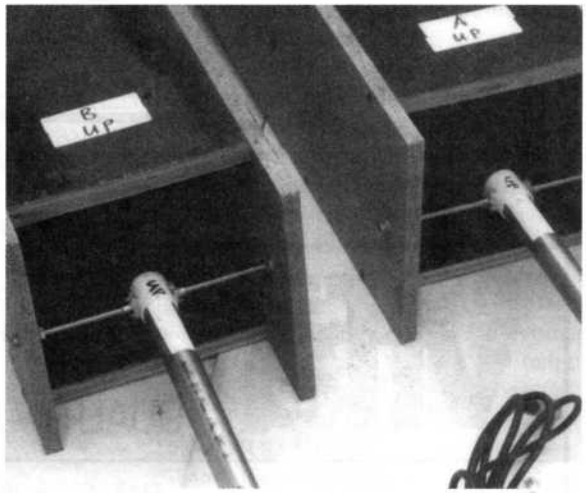

PHOTO 2: Steel pipes in PVC guide assembly in wooden tunnels.

Figure 5 is a schematic representation of the two jigs that hold the speakers and microphones. A drive unit is mounted at one end of each wooden tunnel. Microphones are installed in such a way that you can move them closer to or farther away from the speakers, thus adjusting the distances as desired. In theory, if you can achieve cancellation and the microphone positions are accurately measured, the device will provide phase-alignment data at any chosen frequency.

The microphones you use must match, but the demands on them are very slight, so the cheapest ones imaginable will give excellent results. I have used the Radio Shack electret capsule for applications much more demanding than phase testing, and this cheapie will do nicely.

CONSTRUCTION

After gathering the necessary parts (*Table 1*), construction of the test jigs is simple. First, assemble two wooden tunnels from the eight long pieces of wood. For ease of construction, I used glue and screws to assemble mine. The tunnels don't necessarily have to be airtight, as in the case of a speaker enclosure, so do whatever's easiest for you.

Notice that the pieces are of unequal length (*Photo 1*). This is to provide access to a good reference point for making measurements, as described later. One end of each tunnel will then have extended "lips" and the other end will be flush.

At the flush end of each tunnel, use #8 × 5/8″ wood screws to mount eight "L-brackets" for easy installation and removal of the end pieces that will hold the speakers you're testing. After installing the brackets on the tunnels, stand each tunnel upright on its end piece and use the remaining wood screws to attach the end pieces to the main assemblies (*Photo 1*).

It will be necessary to remove and reinstall the end pieces to calibrate the phase tester, so don't glue the end pieces in place. Do, however, use a felt pen to mark which way the end pieces are oriented so you can easily line them up when reinstalling them.

Now mount the two microphones in the ends of the 36″ steel pipes that slide back and forth in the PVC guides. The guides are supported by the carriage bolts that run through opposite sides of the wooden tunnels (*Photo 2 and Fig. 5*).

OBLIGING STORE

When I built my prototypes, I was fortunate to find a lumber store with a wide variety of diameters of stock to choose from, and the patience to let me experiment for a while. The steel pipe and PVC tube I selected work perfectly, without any play, but with enough space between the inner pipe and outer tube to make it possible to stick computer labels on the steel pipe for calibration marks (*Photo 2*). In the parts lists, I have included, verbatim, the manufacturers' labeling of the pipe and tube. If you can find exactly these same materials, it will save you some trouble.

When cutting the four 1½″ PVC guides, be careful to saw the ends as straight and cleanly as possible. The ends of these guides provide reference points for making measurements, and a little early care will pay dividends later. Also, it's a good idea to cut a few extras. I ruined several before I mastered the art of gluing T-nuts to them.

For each PVC guide, use a ruler to mark the center, ¾″ from the each end. Drill a 1/16″ hole in that spot. Use the "eyeball" method to locate the spot diametrically across the tube from the first hole, and drill a second 1/16″ hole. Using the ruler, check the locations of the holes for uniformity in distance from the ends of the guide.

When you're satisfied that you've located two spots on the PVC guide directly oppo-

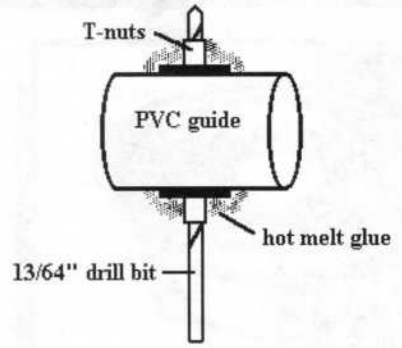

FIGURE 6: Drill bit being used to line up T-nuts.

TABLE 2

PARTS LIST—ELECTRICAL

QTY	ITEM
2	microphones matched (included in *Table 1* as well)
1	sine-wave generator
1	frequency counter (optional)
1	stereo amplifier
1	double-throw switch
1	L-pad (optional)
1	microphone mixer (or stereo preamp)
1	voltmeter
1	load resistor, 8Ω or similar

site one another, increase the size of the holes to 13/64″. A drill bit of this diameter fits nicely through a ¼″ T-nut. Remove the 13/64″ bit from your drill and insert it through both holes in the tube. This provides a straight axis to precisely line up two T-nuts as you glue them to the tube (*Fig. 6*).

To a T-nut, apply a small amount of hot-melt glue on the outer edge of the flange (away from the threads), and affix the T-nut using moderate pressure. Be careful you don't press so hard that you distort the shape of the tube. Mount all eight T-nuts in this fashion.

GLUE LIBERALLY

After the glue has had several minutes to set, you can remove the drill bit and apply more glue. Although hot-melt glue adheres tenaciously to PVC, you must apply enough so it extends over the flange of the T-nut, or the metal nut can easily fall off, particularly under stress.

After you've glued the T-nuts to the PVC guides, you can mount the guides in the tunnel assemblies using the 4½″ carriage bolts (*Photo 2*). Drill a 5/16″ hole 2″ in from the end of each long side, centered with respect to the short dimension. Drill a second hole 12″ back from the first. It is important to measure carefully when drilling these holes. The holes on opposite sides must be lined up exactly, so the PVC guides will be straight.

After you've drilled the holes, mount the carriage bolts from one side only. Use the washers and nuts to hold them in place, and tighten them finger-tight. Then, carefully mount the PVC guides, using the T-nuts, on the ends of the carriage bolts. Don't overtighten them, or you will break the glue holding the T-nuts to the guides. Now insert the remaining carriage bolts from the opposite side, using the washers and nuts, and thread the bolts into the waiting T-nuts. Again, be careful not to tighten them too much.

Next, tighten the first hex nut against the internal wall of the tunnel to hold the first carriage bolt. Then readjust the opposite bolt and insert one of the 36″ steel pipes to see if everything lines up. If it does, tighten the remaining hex nuts and washers holding the bolts in place. When carefully constructed, the assembly is fairly rugged.

MICROPHONE INSTALLATION

A simple sine wave presents a very easy sound for a microphone to transduce, and virtually any matched mikes will do nicely. Again, if you don't own any, it's not much of a problem. Cheap high-impedance models are readily available for portable tape recorders, and should work fine.

I recommend using a microphone that has a smaller diameter than the PVC guides. If it's too big to fit through the guides, the entire metal pipe must be inserted from the speaker end before you install the end piece. A mike of smaller diameter can be inserted and removed from either end, making the process easier.

Once you have selected your microphones, you mount them in one end of each steel pipe, with the cords coming out the other end (*Photo 3*). How you choose to fasten the mikes depends on their shape and whether or not you will want to remove them when you're done. I chose to use duct tape. Be creative.

CALIBRATION

With the microphones installed and the steel tubes sliding freely, you must calibrate the phase tester. First, define a reference point along the axis of the speaker, with respect to which you can make measurements. Since any differences in apparent locations will ultimately be expressed as a distance between the drivers' mounting surfaces, I chose to use those surfaces as the references. This makes calibration very simple.

So, with the microphones installed, I attached the end pieces to the speaker ends of the testing jigs before cutting the holes for the drivers. Then I simply moved both microphones toward the speakers until they came into contact with the end boards.

The PVC guides at the open ends provide excellent reference points (*Photo 2*). I simply stuck a 1″ × 3″ computer label on each steel tube, and indicated the calibration points on these labels by marking the ends of the guides with a fine-tipped pen. When both testing jigs have their reference lines even with their PVC guides, both microphones are at exactly the same point with respect to where the drivers will be mounted, and calibration is complete. You can then measure any differences in position between the two microphones against these marks, thus accurately determining each microphone's position.

ELECTRONIC SETUP

Whereas the jigs are used to vary the location of the microphones relative to the speakers, the ancillary electronic setup is designed to control power to the speakers, compensate for levels present at the microphones if necessary, and measure relevant voltages. *Figure 7* shows a block diagram of the electrical setup. The parts list is in *Table 2*.

A sine-wave generator drives Channel A of a stereo amplifier. If you have access to a frequency counter more accurate than the indicator on your source, by all means use it. You use the amplified sine wave to power the drivers for testing, and the amplifier to drive both speakers. Don't forget that the two drivers must be connected electrically out of phase with each other.

Between the amplifier and the two drive

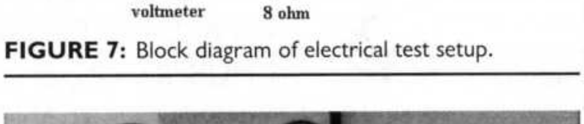

FIGURE 7: Block diagram of electrical test setup.

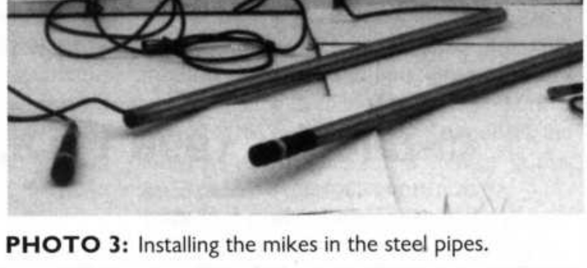

PHOTO 3: Installing the mikes in the steel pipes.

units is a three-position, double-throw switch. When you make a measurement of phase difference, it is necessary to equalize the test signal for levels before you can achieve complete cancellation. The easy way to do this is by switching from one speaker to the other. When levels are roughly equal, both speakers must be active simultaneously while you adjust the microphone positions. The switch makes it easy to go from A to B or to both.

When you're testing, one speaker will almost certainly be inherently louder than the other, and it can help to insert an L-pad attenuator in the path of the louder of the two drivers. You generally must attenuate tweeters with respect to midranges and woofers.

An opposing effect is the decrease in SPL as you move a microphone farther from a source, and the L-pad is useful in compensating for this as well. It may take a little experimentation with the drivers to determine which speaker to attenuate. As an alternative, you could use an attenuator on each speaker, sometimes attenuating one driver, and sometimes the other.

Each combination of drivers presents its own compensation problems, depending on their output levels and the microphone positions at the point of maximum cancellation, so you might have to experiment. As the L-pad is purely resistive, it has no effect on phase measurements in these tests.

THE MIXER FUNCTION

The two speakers, inside the two tunnels, provide their respective microphones with audible sine waves. The microphones are connected to a mixer, which serves as another point of level compensation between the two signals. I used an old Shure M68 for mixing. Again, the demands on the equipment are slight, and even a low-quality mixer can provide good results. Or you could use a stereo preamp, with the balance control adjusting relative microphone levels to a summed output.

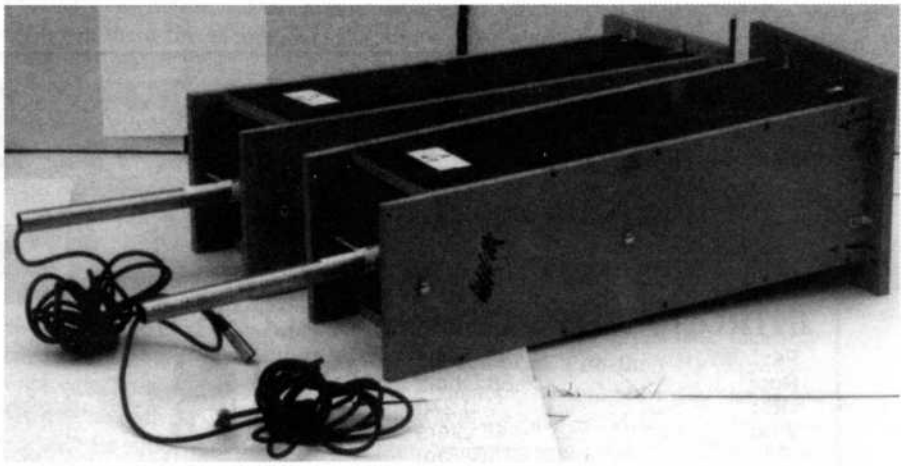

PHOTO 5: The phase tester ready for testing.

Theoretically, if you use the L-pad carefully enough, this point of adjustment isn't necessary. I found it to be quite useful in my testing, however, and in any event, you must amplify the microphones' low-level signals to line level before they will be adequate to drive the second power amplifier.

This second power amplifier is the next stage. The second channel of the stereo amplifier (designated "Amplifier B" in *Fig. 7*) has one simple purpose: to boost the signals from the mixer to an adequate voltage for easy testing. This amplifier receives the summed signals from the microphones, and drives a simple resistor. The voltage as measured across this resistor indicates reinforcement or cancellation.

Figure 7 shows an analog-style voltmeter, but in my tests I chose to use both an analog and a digital voltmeter at the same time. When searching for peaks and valleys in a changing voltage, an analog unit is nice, because a change in direction of the needle is very easy to see. A digital meter, on the other hand, is somewhat more precise when determining absolute minima. When switching between drivers to equalize levels, you achieve greater accuracy by using the digital unit as well. For fast, accurate testing, it's best to use both.

USING THE PHASE TESTER

There follows a step-by-step description of the process of testing for phase difference between a ScanSpeak 13M/8621 cone midrange and a Dynaudio D28 tweeter. *Photo 4* shows these two units installed in the wooden tunnels, ready for testing.

After setting everything up, it makes the measuring and recording of data easier if you designate one test jig the "control" unit, and perform all the adjustments on the other one. Also, if you do all of the work on one unit, you can place the other as far away as the length of your cords will allow. (At one point, I actually moved my control unit to another room.) Although interference between the separate test jigs was at no time apparent, I felt that any step to avoid trouble was positive.

Following the default assumption—mounting the high-frequency unit behind the one with lower frequency—it should be easier to perform the tests with the lower-frequency driver as the control. You can pul the microphones far away from the drivers but there is obviously a limit to how clos

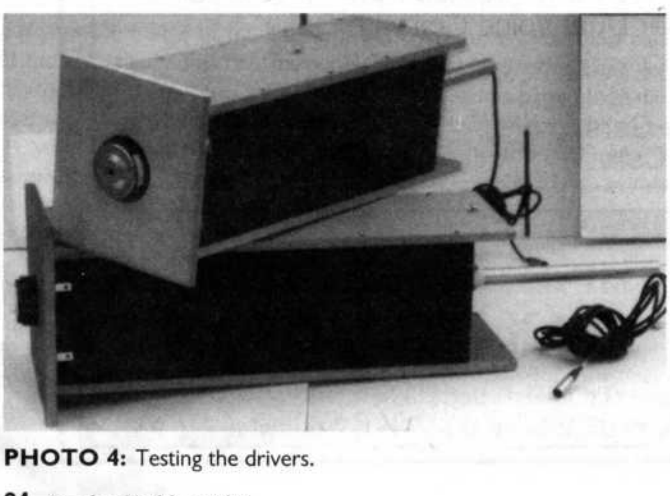

PHOTO 4: Testing the drivers.

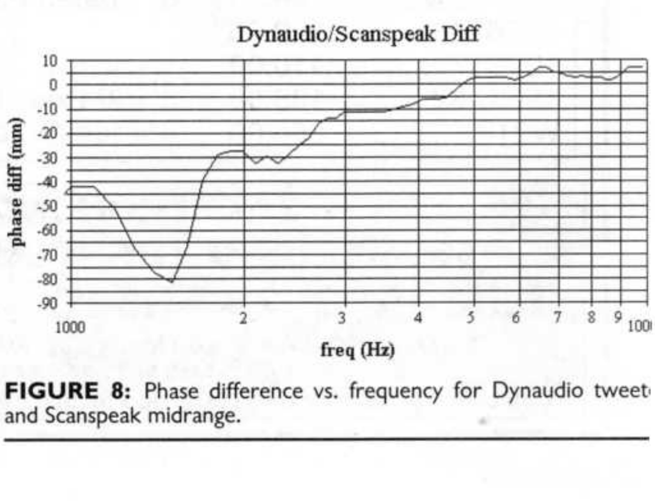

FIGURE 8: Phase difference vs. frequency for Dynaudio tweet and Scanspeak midrange.

you can get them. With this midrange and tweeter, I found that testing was easier with the lower-frequency driver as the control unit. But this will not necessarily be the case with your drivers.

Once you've decided on a control unit (in my case the Scanspeak midrange), you must determine an initial position for the control microphone. Use extreme caution here. Inadvertently placing either microphone too close to its driver could easily damage the delicate cone or diaphragm. Be aware of the calibration mark made earlier, and stay away from it!

For safety's sake, a good place to start is with the microphone on the control unit pulled 1″ or 2″ back from the calibration mark. When you pull the microphone back to this resting point, carefully measure the distance between the calibration mark and the PVC guide, and make another mark at this point. Then you can pull the microphone on the "test" jig back to exactly the same point, and mark its position as well.

WIDE FREQUENCY RANGE

I decided that phase-alignment data would be most useful if taken over a wide range of frequencies. At low frequencies, where the wavelength is quite long, you needn't worry about being off by a complete wavelength, but above, say, 10kHz, where the wavelength can be less than an inch, it's conceivable that you could be off by that much when you start. Your readings would seem correct, but the results would be flawed.

I therefore decided to start at the lowest frequency I could without damaging the tweeter, and work my way up. As the Dynaudio unit is fairly robust, I took a chance, and started at 400Hz. The wavelength at this frequency is over 2′, and it was unlikely I would miss the mark by that much.

With the control unit selected and the adjusted reference marks in place, testing can begin. For each frequency to be tested, the procedure is the same:

1. Switch the output of amplifier A from one driver to the other, using the double-throw switch, and correct levels as necessary so voltage on the load resistor stays the same as you switch back and forth.

2. Position the switch to connect both speakers at once.

3. Move the test microphone backward (away from its drive unit), and locate a minimum in the resistor voltage as clearly as possible. If you must move the microphone closer to the driver for the voltage to drop, swap control and test units.

4. Correct levels again, as in step 1.

5. Locate the position of maximum cancellation again, as in step 3.

6. Measure and record the microphone's position at this point.

COMPLETE CANCELLATION

As you can see, this is a process of closing in on the exact point of maximum cancellation (minimum voltage) as you adjust changing levels. I found that after going through steps 1–6 twice, I was generally able to achieve near-absolute cancellation. For example, voltage on the resistor would start at about 4V for either driver individually, and decrease to 0.02V–0.04V in many cases.

The actual cancellation is so complete that the microphone position necessary for maximum cancellation is easy to determine with great precision, often well within a millimeter. You'll find that once you've done it a few times, it becomes quite easy.

When I first started the phase measurements, I made multiple marks on the computer labels affixed to the steel tubes. I found, though, that it was difficult to make sense of the many marks after the tests were done. It was much easier simply to measure the microphone displacement with a ruler as I proceeded, and to record the data separately. With a good ruler having fine gradations, in good lighting conditions, it is possible to make quite precise measurements—within .25 mm or so.

I made phase measurements on these two drivers at more than 70 different frequencies, from 400Hz to over 12kHz. *Figure 8* shows the results of the tests on these two speakers between 1kHz and 10kHz.

The effective difference in displacement of the acoustic centers of the drivers when mounted on a common plane is shown on the y-axis in millimeters. A positive number indicates how far in front of the midrange's effective location the tweeter's center lies. Over most of this range, the number is negative, meaning that the tweeter's acoustic center is located behind that of the midrange driver. This would, qualitatively at least, tend to bolster the assumption that lining up the voice coils can be helpful.

THE IDEAL SITUATION

The ideal curve for such a graph would be a flat, straight line, indicating two speakers whose acoustic centers you could align at all frequencies simply by staggering their mounting surfaces by a distance equal to the displacement indicated by the line. With hundreds of woofers, midranges, and tweeters to choose from, it might in fact be possible to find combinations that would yield such a well-mannered curve. But alas, in the case of these two drivers, the curve is anything but flat!

Any information, however, is better than none. A close look at this curve can still be helpful. Consider these points of interest: at about 4.7kHz, the curve crosses the line of zero displacement, indicating that the two drivers are exactly in phase with each other at this frequency when they're mounted on a common baffle.

From about 2.3–6.7kHz, the overall trend is quite uniform in direction, i.e., increasing with respect to frequency. Near the middle of this region, from around 3–3.5kHz, you find the elusive flat line—not as long as you might desire, but it's a "given" in this situation.

I would be inclined then, when designing a system using these drivers, to cross them over squarely in the center of that flat line, at about 3.25kHz. The displacement at this frequency is approximately 11mm, so the tweeter's mounting surface would be behind that of the midrange by this amount. This would place the drivers precisely in phase through the region closest to the crossover frequency. By using fairly sharp filters of 18 or 24dB per octave, you would minimize in importance the phase differences above and below the crossover point.

MAKING THE BEST OF IT

Whether such an interface is "phase aligned" is a question of degree. Many design decisions are a matter of making the best of what you have to work with, and with these two drivers, I believe phase-test data has made it possible to use the units more effectively than might otherwise be the case.

Consider that using a common baffle would represent an offset of 11mm at 3.25kHz. This would be a phase shift of about 38°. Coincidentally, because this tweeter is horn-loaded, I estimate that this mounting would very nearly align the voice coils. In this case, it would seem that you can improve somewhat upon the configuration that your default assumption would yield.

There are, of course, other design considerations involved, such as the power handling and frequency response of the drivers, which might make such a straightforward, if hasty, approach impossible. What if, for example, the only flat spot on your curve was below the tweeter's resonant frequency? It's conceivable that two drivers you had considered for use together might not be a good match in terms of phase characteristics.

With some combinations of drivers, it's entirely possible that you could achieve excellent phase-alignment using a common baffle. Others could work best with the voice coils aligned, or at some other point. To find a combination of drivers yielding a curve closer to the desired flat line would make design easier and capable of more correct phase alignment. Testing large numbers of speakers in this way can be time-consuming, though, and a perfect combination is an ambitious goal.

I never said that phase-alignment data would make crossover design any easier, only that it is necessary before you can accomplish any truly corrected design. In terms of expediency, it's much easier to keep the blinders on and rely on the assumptions we've used for years!

CAVEATS

I must also emphasize that filter design is just as important to a system's phase behavior as anything offered here. While this article is intended to provide a means of measuring physical time-alignment data, both active and passive crossovers introduce phase effects unrelated to anything I've discussed herein.

To make things worse, if the (default)

resistor model of a moving coil's loudspeaker's impedance is used for passive filter design, any carefully measured alignment data is compromised in applicability anyway. Anyone having doubts about the inadequacy of this resistor model would do well to review Victor Staggs' excellent "Exploring Speaker Impedance" (*SB 5/94*).

Another area of concern is the question of sine-wave vs. transient reproduction. It's all well and good to define a phase difference when simple sine waves are tested, but what does it tell us about the music reproduction? My best response to this legitimate concern is that you are at the mercy of your drivers.

Some manufacturers publish relevant information, such as tone-burst pictures that show a remarkable ability to track a signal accurately under transient conditions. It stands to reason that the better your driver can accurately follow transient signals, the more valid the phase information from sine-wave tests will be. Even in worst cases, however, any information is better than none at all.

DAMPING EFFECTS

What about installing insulation or other damping material inside the phase-testing jigs? When I first used my prototypes, I lined the tunnels with 1″ fiberglass insulation, partly because using damping materials tends to be second nature to a speaker builder. But I had concerns about internal reflections blurring the effect of cancellation at the point of minimum measured voltage, making the tester harder to read.

Later on, however, I tried making the same measurements in both an unlined setup and with the tunnels stuffed. Interestingly, the measured results were the same regardless of damping material. So don't worry about it. Also, significantly, because the speed of sound decreases in a stuffed enclosure, this demonstrates that the phase differences measured are characteristics of the drivers themselves, and are unrelated to the speed of sound in the medium into which they radiate.

The wooden tunnels illustrated in *Photo 5* are obviously not big enough to be used with large drivers. An 8″ speaker is the largest that will fit into these test jigs. In order to test a larger driver, it would be necessary to build some larger jigs. Although I haven't had a need to try this, it should work, with a simple recalibration.

REFERENCES

1. Alexis Badmaieff and Don Davis, *How to Build Speaker Enclosures*, Howard W. Sams and Company, 1966, p. 27–28.

2. Hank Zumbahlen, "Zobels and All That," *Audio*, June 1995, p. 36.

OTHER THINGS TO CONSIDER

If you assemble your own phase tester and want to spend some time playing with it, there are some other areas to consider. I will offer these without elaboration, as ways in which you can investigate wave activity on your own.

1. What if you were to connect the two drivers in-phase electrically instead of out-of-phase as in the tests I've shown? Could you still make accurate phase measurements? Under this different condition, of what would the point of maximum cancellation be indicative, and could its location be predicted at a given frequency from the measurements already made in the out-of-phase configuration? And how about the point of maximum reinforcement?
2. In addition to measuring displacement differences between drivers, could the phase tester be configured to determine the actual acoustic center of a driver, and if so, how?
3. Could the phase tester be used to measure the speed of sound?
4. These test results would seem to indicate a change in acoustic center of one or both of the drivers as frequency changes. What would be the cause(s) of the fluctuations in acoustic center of a drive unit with respect to frequency?